eMG80/100& eMG800& UCP& vUCP

Feature Description & Operation Manual

iPECS is an Ericsson-LG Brand

Please read this manual carefully before operating System.
Retain it for future reference.

Revision History

ISSUE	DATE	DESCRIPTION OF CHANGES
1.0	July, 2013	SW Version 1.0.x - Initial Release
1.1	Dec., 2013	- Changed Ericsson-LG to Ericsson-LG Enterprise
1.2	Feb, 2015	SW Version 1.1.x - Updated contents according to S/W integration for both eMG80 & eMG800
1.3	June, 2015	SW Version 1.2.x - Emergency page - Message Cascade. - LCR (Least Call Routing). - Remote Control from mobile Phone. - ARS (Alternative Route Selection) for Digit conversion table. - Message Wait Reminder Tone. - Call park for internal call. - Unified Loop button.
1.4	Jan., 2016	SW Version 2.0.x - System announcement is changed from 70 to 200. - LIP-9071 NFC for Call forward & Hot desk. - Added Activating/Deactivating/Joining conference group. - Searching Directory in LDAP & System DB, refer to Dial by name. - Moving from Mobile phone to Desktop phone, refer to Mobile Extension. - Added the Message retrieval for hearing or skipping date/time prompt while hearing. - Deleted Urgent Text Message Service. - Added Mobile Native Voice Support for UCS Mobile Client. - Updated Message cascade. - Added Station Flexible button default table. - Added Multiple Mailbox Greeting. - eMG80 Expansion. - eMG80/800 can’t support Conference recording. - Added Lift Handset for Page in Paging. - Added Privacy function in Barge in and Intrusion. - Agent statistics for multiple groups in ACD group statistics report.
1.5	Mar., 2016	SW Version 2.0.x - Unified SMB Manual: Integrate eMG and UCP manual
1.6	Aug, 2016	SW Version 2.1.x - The condition is updated in Dial-by name. - LED flash rate is updated according to adding Message wait (call back) in Flexible LED Flash rates.

ISSUE	DATE	DESCRIPTION OF CHANGES
		- The Alarm Ring Coverage button is added for covering the alarm from all to the specific alarm by using Alarm code type in System Alarm Notification control. - Speed numbering is updated according to adding more types in Flexible speed.
1.7	August, 2017	SW Version 2.2.x - Preset Call Forward is updated about Call Forward Type by adding Internal Unconditional and External Unconditional. - Dial-by-Name is updated about LDAP and System DB for LIP/LDP with 3 soft button and LIP-9071. - Executive/Secretary Forward is updated: assign the flexible button into Forward button on Executive or Secretary phone. - Multiple Mailbox Greeting is updated about playing the prompt 601~604 and prompt 80 if there is no User greeting. - Station Speed Dial is updated about Net station number. - System Speed Dial is updated about Net station number. - Call Log Display is updated about Call Log Usage, Call Log Menu, Answered by Group Member, and Internal Call Log. - Call Profile Routing is updated about adding Call Profile Apply Option. - Station Flexible Button is updated; Telephone number including CO access code is available.
1.8	Jan, 2018	SW Version 3.0.x - Company Name Service - Dial-by-Name (Directory Search & Dial) - System Alarm Announcement - Triggering of External Contact Relay Port for Alarm - Message Wait/Call Back - Security - User base license &Single log-in vs Multi log-in - Push notification for UCS Mobile
1.9	Jun, 2018	SW Version 3.1.x - Updated the conference room - Updated the Message Cascade - Updated the LNR Operation of LDP/LIP Phone
2.0	Mar, 2019	SW Version 3.5.x - Updated the Internal Call Support of Personal CCR/User Greeting DISA section - Added the Flexible Page Feature section - Updated the Barge In section - Added the Phone Lock/Unlock section - Updated the “Condition” of Two-Way Record section”
2.1	Nov., 2019	SW Version 3.6.x

ISSUE	DATE	DESCRIPTION OF CHANGES
		- Added eMG100 System
2.2	Dec.03,2019	S/W version 4.0.x. - Updated Condition section - Added Condition section - Updated the Attendant Operation. - Added Condition about the Call Group Number - Updated the Attendant Operation. - Added Condition about the Call Group Number - Updated the Station Call Coverage Operation. - Updated the Auto Called Number Redial (ACNR) Operation. - Updated the Executive/Secretary Forward Operation.
	Dec.18,2019	- Updated the Station Call Coverage Operation. - Updated the Auto Called Number Redial (ACNR) Operation. - Updated the Executive/Secretary Forward Operation.
	Feb.13,2020	- Added 1000i Series IP Phone
2.3	Sep., 2020	S/W version 4.1.x. - Added the Multi-Barge In Room - Updated Description (Enhanced VSF MOH 4~30) - Added the Programmable MOH Based On Station User & Call - Added the SMDR Enhanced Options for Internal Call - Added the Flexible Page from Mobile Extension at the Flexible Page Codes - Updated BGM (Enhanced VSF MOH 4~30) - Added the 1048ilss,1048idss models
2.4	Mar., 2021	S/W version 5.0.x. - Added the “ Hybrid redundancy” at the Redundant System Processor for UCP

Table of Contents

1 Introduction

1.1 Manual Application

This document provides detailed information covering description and operation of thenumerous features available in iPECS Release system software. The documentis written assuming the system employs the default-numbering plan 2.

Virtual machine UCP provides most features except the following features:

- Redundant System processor (Local Redundancy).
- Door Open, LBC (Loud Bell Control), and External Control Device.
- Internal/External & All Page.

1.2 Manual Organization

Features are arranged alphabetically in nine (9) different major groupings as follows:

- Section 1Introduction
- Section 2 System Features
- Section 3 Intercom Features
- Section 4 CO/IP lines
- Section 5 iPECS IP & LDP Phones
- Section 6 Attendant
- Section 7 SLT(Single Line Telephone)
- Section 8 SIP Extension
- Section 9 iPECS UCS Client

1.3 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 Ericsson-LG Enterprise phones (LDP digital and LIP/IP phones)and SLT(Single Line Telephone)s. In some cases a Flexible button on Ericsson-LG Enterprise phones can be assigned to activate the feature. In this case, the user entries to assign the button are provided.
- **Conditions**: explains known feature interactions and constraints related to the feature.
- **Programming**: lists database entries that may be required for proper feature operation. This lists are related to Station Admin programming on Administration and Programming manual. Regarding WEB admin, you directly access to the web admin and check the programming.
- **Related Features**: lists related topical information to aid in understanding the feature.
- **Hardware**: lists hardware required for proper feature operation.

1.4 Buttons &Term Variations

The eMG supports proprietary digital and IP phones. The proprietary models include the LDP- 7000 series, LDP-9000 series, LDP-9200 series, LIP- 7000 series, LIP-8000/8000E series, and LIP-9000 Series, and LIP-907x series, and 1000i series. Each series employs similar but slightly different designations for the buttons. Details of the buttons on your phone are provided in the User Guide. For the purposes of this manual, several of the designations may be used. For example, the button to handle messages is called either **[MSG]**, **[CALL BK]**, or **[MESSAGE/CALLBACK]**. Note the fixed button names are enclosed in brackets '**[ ]**' and Flex buttons are enclosed in braces '**{ }**'. In addition, some buttons have dual functions; the **[HOLD/SAVE]** button places your active call on hold and stores entries you make when assigning Flex buttons.

The description, operation and conditions attempt to use common terms such as "Voice Mail" or "outside Line" so that a novice installer can understand the contents. The Programming and Hardware sections however employ terms specific to the iPECS systemas used in the respective manual such as "VSF" (used for voice mail) and "CO/IP lines" for outside Lines.

Note the feature code is based on **Flexible numbering plan 2** in Operation part. First check your numbering plan 'System ID (100)' and for the detailed feature code about the numbering plan, refer to Appendix on Administration & Programming manual.

1.5 eMG80/100 Expansion

1.5.1 eMG80 Expansion

The maximum number of TDM port can be expanded in BKSU-IE/AE. For more detailed information about KUSIE & AE, refer to Hardware Description and Installation Manual.

To install

1) Connect Expansion cable(HDMI) between BKSU-IE/AE of master and slave.
2) Set TNET program.

Conditions

- ✓ IP device should be registered to Master system.
- ✓ A WTIB can be installed in Master system, not in Slave system.
- ✓ ISDN clock is synchronized only in Master system, so stable clock should be provided by ISDN line of Master system, not in Slave system.
- ✓ eMG80N-EXPM should be installed in Master system.
- ✓ eMG80N-EXPS should be installed in Slave system.

1.5.2 eMG100 Expansion

The maximum number of TDM port can be expanded.

To install

1) Connect Expansion cable(LAN) between master and slave.
2) Set master/slave DIP switch
3) Set TNET program.

Conditions

- ✓ IP device should be registered to Master system.
- ✓ ISDN clock is synchronized only in Master system, so stable clock should be provided by ISDN line of Master system, not in Slave system.
- ✓ eMG100N-EXPM should be installed in Master system.
- ✓ eMG100N-EXPS should be installed in Slave system.

2 System

2.1 8-Digit Station Numbering Plan

Description

The system supports Station Numbering Plans from two to eight digits. The Station number consists of a Prefix digit string and a suffix called “Add Digit”. A Prefix digit string can be up tosix (6) digits that do not conflict with the system Flexible Numbering Plan used for feature codes. The Add Digit defines the number of digits from the Station Number that are added as the suffix to the Prefix digits to form the Station Number. The Add Digits can be 2 digits in lengthusing the right most digits of the Station Number and must uniquely identify the station.

Operation

System

Operation of the 8-digit Station Numbering Plan is automatic when configured.

Conditions

- ✓ The system employs the complete Station number, 2 to 8 digits, when sending messages to an external device such as an adjunct Voice Mail and TAPI device such as ez-Attendant. It is the responsibility of the 3rd party software to support the defined Station Numbering Plan.
- ✓ When using Keyset Admin to configure the 8-Digit Table, Prefix Usage must be enabled in the System ID PGM 100 (Flex button 5).
- ✓ The 8-digit numbering is configured in the 8-Digit Table (PGM 238).

Programming

Keyset Admin.

System ID

- Prefix Usage (PGM 100 – Flex button 5)
- 8-Digit Table (PGM 238)

Web Admin.

SYSTEM ID & NUMBERING PLANS

- 8Digit Extension Table➤8 Digit String
- 8Digit Extension Table➤Add Digit

Related Features

- ■ Flexible Station Number

Hardware

2.2 Account Code

Description

You may allow tracking of specific calls by entering a non-verified variable length (upto 12 digits) identifier for a call.

The identifier or “Account Code” is output as part of the StationMessage Detail Recording (SMDR) for the call as discussed in Section 2.80.This is often used to allocate call costs to a specific account or client.

Operation

iPECS IP & LDP Phones

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

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “84” + **[HOLD/SAVE]**

*To assign a Flex button for **{ONE-TOUCH ACCOUNT CODE}** operation*

- **[TRANS/PGM]** + **{FLEX}** + **[TRANS/PGM]** + “84” + Account code (up to 12 digits) + **[HOLD/SAVE]**

*To enter an Account Code using an **{ACCOUNT CODE}** button prior to placing a call*

1) Lift the handset.
2) Press the {ACCOUNT CODE} button.
3) Dial the Account Code (up to 12 digits).
4) Press “*”, Intercom dial tone is heard.
5) Place the outside call as normal.

Or,

1) Lift the handset or press the [SPEAKER] button.
2) Press the {ONE-TOUCH ACCOUNT CODE} button.
3) Place the outside call as normal.

*To enter an Account Code using an **{ACCOUNT CODE}** button during a call*

1) Press the {ACCOUNT CODE} button.
2) Dial the Account Code (up to 12 digits).
3) Press “*”

Or,

1) Press the {ONE-TOUCH ACCOUNT CODE} button.

SLT(Single Line Telephone)(Single Line Telephone)

To enter an Account Code prior to placing a call

1) Lift the handset.
2) Dial the Flex Numbering Plan code “*550”.
3) Dial the Account Code (up to 12 digits).
4) Press “*”.
5) Place the outside call as normal.

To enter an Account Code during a call

1) Momentarily press the Hook-switch.
2) Dial the Flex Numbering Plan code "*550".
3) Dial the Account Code (up to 12 digits).
4) Press "*".

Conditions

- ✓ When entering an Account Code during a call, DTMF digits are transmitted to the connected party.
- ✓ A maximum of 1000 {ONE-TOUCH ACCOUNT CODE} buttons may be assigned in the system.
- ✓ If an Authorization Code is entered as the Account Code, the SMDR record will show the station number or the index number for a System Authorization Code rather than the user entered Authorization Code.
- ✓ If the Forced SMDR Account Code attribute of the station is enabled, you must enter an Account Code prior to placing an outgoing call.

Programming

Keyset Admin.

NUMBERING

- Flexible Numbering Plan (PGM 106-Button 8)

STATION

- Forced SMDR Account Code (PGM 113-Button 24-21)

Web Admin.

SYSTEM ID &NUMBERING PLANS

- Flexible Numbering Plan ➤SMDR Account Code Enter

STATION DATA

- Common Attributes ➤Forced SMDR Account Code

Related Features

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

Hardware

2.3 Alarm Signal/Door Bell

Description

The system can be configured to recognize the status of an external contact (normally open orclosed). The system will signal assigned iPECS IP and LDP Phones when the contact activates. Thiscapability is commonly employed to provide remote Alarm or Door Bell signals a user.

Assigned stations receive the Alarm Signal, either a single tone burst repeated at 1-minuteintervals or a continuous tone. The Alarm Signal may be terminated at the phone bydialing the Alarm Reset code or, if assigned, pressing the **{ALARM RESET}** button. To rearm theAlarm function, the alarm condition must be cleared and the Alarm signal terminated.When used as a Door Bell, assigned iPECS IP and LDP Phones receive a single tone burst each time theexternal contact is activated and no reset is required.

Operation

System

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

iPECS IP & LDP Phones

*To assign a Flex button as an **{ALARM RESET}** button to terminate the Alarm Signal*

- **[TRANS/PGM]** + {FLEX} + “*565” + **[HOLD/SAVE]**

To terminate an Alarm Signal, while idle

1) Dial the Flex Numbering Plan code “*565”, confirmation tone is received and the Alarm Signal is terminated. If the alarm condition is cleared, the system will automatically rearm the alarm monitoring or press the **{ALARM RESET}** button.

Conditions

- ✓ The Alarm contacts must be “dry”, no voltage or current source connected.
- ✓ Only iPECS IP and LDP Phones may be assigned to receive Alarm/Door Bell signals.
- ✓ The LCD of phones assigned to receive Alarm/Door Bell signals will show “ALARM” or “DOOR BELL” as appropriate.
- ✓ The Alarm Reset code is also employed to terminate alarm signals from digital and IP trunk alarms as well as Cabinet alarms.

Programming

Keyset Admin.

SYSTEM

- Alarm Enable (PGM 163-Button 1)
- Alarm Contact Type (PGM 163-Button 2)
- Alarm/Doorbell Mode (PGM 163-Button 3)
- Alarm Signal Mode (PGM 163-Button 4)

STATION

- Alarm/Door Bell Attribute (PGM 113-Button 8)

Web Admin.

SYSTEM DATA

- Alarm Attributes ➤Alarm Enable
- Alarm Attributes ➤Alarm Contact Type
- Alarm Attributes ➤Alarm/Doorbell Mode
- Alarm Attributes ➤Alarm Signal Mode

STATION DATA

- Common Attributes ➤Alarm/Door Bell Attribute

Related Features

- Door Open
- Station Flexible Buttons
- System Alarm Notification Control
- Cabinet Alarm for UCP

Hardware

- iPECS IP or LDP Phone
- Alarm or Doorbell input contact, refer to Hardware Description & Installation Manual to install.

2.4 Authorization Codes (Password)

Description

Authorization Codes provide a means to control access to Voice Mail, Off Premise Call Forward, WalkingCOS (Class of Service), or DISA and may be required for outgoing outside Line or LCR access based onconfiguration of the system database.

When you dial an Authorization Code that matches anAuthorization Code stored in the database, the system invokes the Station COS (Class of Service) or the COS (Class of Service)assigned to Authorization code. Each Authorization code has separate Day/Night mode COS (Class of Service)assignments.

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

The Station Authorization Codes includes the associated station number and the assignedcode. The structure of the System Authorization code can be set as either "*", or "*" the Authorization table index and the code digits. The later allows duplicate codes to be employedusing entry of table index to provide a unique identification of the entry.

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

Operation

iPECS IP & LDP Phones

To assign a Station Authorization Code

1) Press the **[TRANS/PGM]** button.
2) Dial "33", the Authorization Program Code.
3) Dial the Station number.
4) Dial the Authorization Code (1 to 12 digits).
5) Press the **[HOLD/SAVE]** button.

To enter an Authorization Code when second dial tone is received

1) Dial the Station number for the Station Authorization code or, for a System Authorization Code, dial "*" or "*" and the Authorization table index.
2) Dial the corresponding Authorization Code.
3) Place your call as normal.

SLT(Single Line Telephone)(Single Line Telephone)

To assign a Station Authorization code

1) Lift the handset.
2) Dial SLT(Single Line Telephone) Program Mode Entry code "*561".
3) Dial Authorization Program Code "33".
4) Dial Authorization Code (1 to 12 digits).
5) Momentarily press the Hook-switch, receive confirmation tone.

To enter an Authorization Code when second dial tone is received

1) Dial the station number for the Station Authorization code or, for a System Authorization Code, dial “*” or “*” and the Authorization table index.
2) Dial the corresponding Authorization Code.
3) Momentarily press the Hook-switch.
4) Place your call as normal.

DISA

To enter an Authorization Code when second dial tone is received

1) Dial the station number for a Station Authorization code or, for a System code, “*” or“*” and the Authorization table index.
2) Dial the corresponding Authorization Code.
3) Place call as normal.

System Attendant

To assign an Authorization Code

1) Press the **[TRANS/PGM]** button.
2) Dial Attendant Station Program code “031”.
3) Dial the Station number of a Station code or, for system code, “*” or “*” and the Authorization table index.
4) Dial the corresponding Authorization Code.
5) Press the **[HOLD/SAVE]** button.

Conditions

- ✓ When an outside DISA Line is marked for Authorization Code entry, the caller will hear fast busy tone and must input a valid Authorization Code to continue. In case of an entry error, the caller may retry entry of the code.
- ✓ A DISA caller must enter a valid code within the number of attempts assigned as the FAC(Forced Account Code) Retry Count. The station, if enabled, is placed to COS (Class of Service) 7 after repeated failure attempts.
- ✓ A user may enter an Authorization Code from any station to place an outside call using Walking COS (Class of Service).
- ✓ The eMG80/100/eMG800/UCP system provides memory for up to 500/2400/5200 Authorization codes, 140/1200/2400for station codes and the remaining 360/1200/2800 are system codes.
- ✓ Outside Line groups may be configured to require entry of a valid Authorization Code. In this case, a second dial tone is provided when the Line group is accessed. If the code entered is invalid, error tone is returned and the user must enter a valid code within the FAC(Forced Account Code) Retry Count.
- ✓ If the Check Password option is enabled in the LCR (Least Costing Routing) database, when dialed digits match the LDT (Leading Digit Table) table digits, the system will send second dial tone to request that you input a valid Authorization code.
- ✓ An Authorization code may include any dial pad digit except “*” and “#”.

- ✓ Duplicate or conflicting System Authorization codes are not allowed when using the"*" and code operation. For example, code "*1234" conflicts with code "*123" and cannot be recognized as a unique code. Since the index operation employs the Table index and the station number forms part of the Station code, codes cannot be duplicated and conflicts will not occur.
- ✓ Use of Authorization codes varies based on the system Nation Code. In some regions, particularly the US and UK, a System Authorization code may be required for DISA access. Entering a Station code on a DISA line will fail in these areas. In other regions, a System Authorization code can be used with Walking COS (Class of Service).
- ✓ The system can be configured to require an End of Entry code or simply "End Code". If "End Code" is enabled, you must enter "#" after entry of the Authorization code.

Programming

Keyset Admin.

STATION

- Station Account (PGM 112-Button 19)
- CO/IP Group Authorization (PGM 141-Button 8)
- DISA Authorization Code (PGM 142-Button 5)

SYSTEM

- DISA Retry Counter (PGM 160-Button 5)
- AUTH Retry Count (PGM 160-Button 16)
- Old Auth Code Usage (PGM 161-Button 16)
- COS (Class of Service) 7 When Auth fails (PGM 161-Button 17)
- End code(#) usage in System Auth Code (PGM 160 – Button 24-2)

TABLES

- LCR Check Password (PGM 221-Button 6)
- Station Authorization Code Table (PGM 227-Button1)
- System Authorization Code Table (PGM 227-Button2)

Web Admin.

STATION DATA

- Common Attributes ➤Station Account

CO LINE DATA

- Common Attributes ➤CO/IP Group Authorization
- Common Attributes ➤DISA Account Code

SYSTEM DATA

- System Attributes ➤DISA Retry Count
- System Attributes ➤AUTH Retry Count
- System Attributes ➤Simple Auth Code Usage
- System Attributes ➤COS 7 When Auth Fail
- System Attributes ➤End Code(#) usage in System Auth Code

TABLES DATA

- LCR LDT (Leading Digit Table)➤Check Password
- LCR LDT (Leading Digit Table)➤Station& System Authorization Code Tables

Related Features

- Account Code
- Auto Service Mode Control
- Direct Inward System Access (DISA)
- Outside Line Groups
- Outside Line Access
- Temporary Station COS (Class of Service)/Lock
- Walking COS (Class of Service)
- Call Forward
- Station User Programming & Codes

Hardware

2.5 Auto Call Recording

Description

Your iPECS IP or LDP Phone can be configured to automatically record calls to a Voice Mailbox or the hard disk drive of UCS Client soft phone or an external call recording server such as iPECS IPCR. The type of calls recorded can be set as None, All calls or outside Line calls only.

When recorded to a mailbox, you manage the recording through your Voice Mailbox. For recordings to UCS Client, recordings are managed directly by UCS Client with the ability to listen to, delete or send the recording to others via e-mail.When using an external server, use the procedures associatedwith the server.

Operation

Recording of calls is automatic when assigned.

To manage the recordings, use the procedures outlined in iPECS UCS Client User Guide, iPECS IPCR User Guide, or third party guide, as appropriate.

Conditions

- ✓ UCS Client can record one call at a time and must be idle. While recording, if UCS Client places or receives a call, recording terminates.
- ✓ When call recording begins, you and the connected party will receive a Call Recording confirmation tone.
- ✓ UCS Client used for recording must be operating in a ‘Local’ mode not remote.
- ✓ A third party SIP Voice Mail or recording server may be used to record calls. In this case, the Recording Destination should be assigned with the Station group of the SIP server.
- ✓ When you activate call recording to an iPECS IPCR, the entire call is recorded as long as the call is active when you activate recording. It would be expected that other SIP recording servers would provide similar On-demand recording.
- ✓ The IPCR Agent Table in the iPECS system must be configured when using the IPCR application or a third-party SIP recording server. Assigning an Agent Id automatically enables Auto Talk Recording.
- ✓ User cannot record individually 1:1 after starting conference in case of add-on conference. Only supervisor can record the conversation on Conference or Add-on conference.
- ✓ Conference recording is not supported in eMG system.

Programming

Keyset Admin.

STATION

- Two-way Record Privilege (PGM 112-Button 11)
- Auto Call Recording (PGM 112-Button 20)
- Call Recording Destination (PGM 112-Button 21)

Web Admin.

STATION DATA

- Station Voice Mail Attributes ➤Two-Way Record
- Station Voice Mail Attributes ➤Auto Talk Recording Option
- Station Voice Mail Attributes ➤Auto Talk Recording Destination

CO LINE DATA

- Common Attributes ➤Auto Call Recording Destination

SYSTEM DATA

- System Attributes➤Record Warning Tone System Timers
- System Attributes➤Record Warning Repeat Timer

TABLES DATA

- IPCR Agent Table

Related Features

- iPECS IP or LDP Phone
- PC with UCS Client
- Recording server

Hardware

✓ iPECS IP or LDP Phone, or PC with UCS Client or Recording server

2.6 Auto Call Release

Description

Outside or intercom calls (except Hands-free Answerback) will be released automatically if you do not complete dialing or, for intercom calls, the called party does not answer after apre-determined time.

Operation

System

Auto Call Release of Intercom calls

If you place an intercom call and the called station does not answer in the IntercomCall Release Time, the call is terminated and you will receive error tone. In addition, if the Inter-digit Timer expires while dialing, the call is terminated and you will receive error tone.

Auto Call Release of outside calls

If you seize an idle Line and do not dial within the AutomaticRelease Time, the call is terminated and you will receive error tone.In addition, if the Inter-digit Timer expires while dialing, the call is terminated and you will receive error tone.

Conditions

- ✓ If the Automatic Call Release Timer is set to “0”, Auto Call Release is disabled.
- ✓ When the handset is used to place a call, you will receive error tone for 30seconds followed by 30 seconds of Howler tone and the station is placed in a fault mode. If on-hook dialing is used, you receive error tone for one (1) second and the phone returns to idle automatically.

Programming

Keyset Admin.

STATION

- Howling Tone to Station (PGM 111-Button 5)

SYSTEM

- CO Call Automatic Release Timer (PGM 180-Button 12)
- STA Call Automatic Release Timer (PGM 182-Button 4)

Web Admin.

STATION DATA

- Terminal Attributes ➤Howling Tone to Station

SYSTEM DATA

- System Timers ➤Automatic CO Release Timer, Station Auto Release Timer, Inter-digit Timer

Related Features

- ■ Howler Tone

Hardware

2.7 Automatic Call Distribution (ACD)

Description

iPECS ACD incorporates flexible incoming call routing, real-time agent monitoring andsupervision, and call record statistics as well as ACD Event messages for managementreporting.

Calls route to an ACD group directly or otherwise including via call transfer, CCR (Customer Call Routing) and ICLID (Incoming Calling Line Identification) routing.

The caller may receive announcements then route to an available agent. When no agent is available, calls queue to the group awaiting a free agent. Should the call Overflow based on number of queued calls or queued duration, the call can be routed to an alternate destination.

2.7.1 ACD Caller Controlled ICLID Routing

Description

With Caller Controlled ICLID enabled for the ACD Group, at any time during or within 5 seconds after a Guaranteed Announcement, the system will monitor for digits dialed by the caller. Received digits are compared to the ICLID Table entries.

If a match is found, the call routes to the destination defined in the ICLID Ring Assignment Table. Available destinations are a System announcement, system speed dial, a station or a station group. If a match is not found, the call remains in the queue for normal ACD group routing.

Operation

System

ICLID Routing of ACD group calls is automatic based on the ACD group assignments.

Conditions

- ✓ The conditions of System Announcements and ICLID Call Routing apply.
- ✓ Only the caller-entered digits are employed for ACD Caller Controlled ICLID Routing, the Caller Id received from the network for the incoming call is not used for this feature.
- ✓ The ICLID Table routing is only followed if the 1st ACD announcement is Guaranteed, ICLID Usage is enabled and the digits are entered during the 1st Announcement.
- ✓ The caller may correct an entry by dialing ‘*’ to erase all digits entered then dial the correct entry.

Programming

Keyset Admin.

STATION GROUPS

- ACD Group (PGM 191)
- ICLID Usage (PGM191-Button 21)

TABLES

- ICLID Route Table (PGM 203)
- ICLID Ring Assignment (204)

Web Admin.

STATION GROUP DATA

- Station Group Assignment➤ACD Group Type
- Station Group Attributes➤Entered Caller ID(ICLID) Usage

ISDN LINE DATA

- ICLID Route Table
- ICLID Ring Assignment

Related Features

- ✓ Automatic Call Distribution
- ✓ ICLID Call Routing
- ✓ Integrated Auto Attendant

Hardware

2.7.2 ACD Statistics Report

Description

ACD reports can be requested by the Supervisor and can be programmed for periodic output over the SMDR port or selected TCP/IP port.

The system will provide reports for the ACD Group and Agent statistics as follows:

ACD Group Statistics Report

- Group Number
- Time stamp
- Total calls
- Number of unanswered calls
- Average queue time
- Longest queue time
- Total number of calls placed in queue
- Number of times calls experience all agents busy
- Total time all agents were busy

Average ring time before answer

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

Operation

Supervisor iPECS IP & LDP Phone

*To assign a **{ACD Supervisor Status}** button at the Supervisor*

- **[TRANS/PGM]** + **{FLEX}** + "*576" + "Group Number (Station Group for ACD)" + **[HOLD/SAVE]**

To output ACD Statistics Report

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the {ACD Supervisor Status} button.

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

3) Dial "#", the ACD Group Statistics Print code.
4) Press the **[MUTE]** button to initialize the ACD database after printing; this eliminates overlap of future reports.

To output the Agent Statistics Report

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the {ACD Supervisor Status} button.

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

3) Dial “3”, the Agent code, the LCD menu will update.

[1] DUTY STATUS
[2] DUTY ON/OFF
[#] DUTY PRINT

4) Dial “#”, the Agent Statistics Print code.
5) Press the **[MUTE]** button to initialize the ACD database after printing; this eliminates overlap of future reports.

Conditions

✓ The ACD status can be printed periodically. The period is assigned in Admin Programming. The ACD records contain information for both ACD agents and ACD group.

✓ The Supervisor must be assigned as a Supervisor for the ACD Group and be logged into the group to access the Group Status display information.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Group Supervisor Status (PGM 107-Button 7)

STATION GROUPS

- ACD Group (PGM 191)
- ACD Group Supervisor (PGM 191-Button 18)

Web Admin.

SYSTEM ID &NUMBERING PLANS

- Flexible Numbering Plan ➤ACD Supervisor Status

SYSTEM DATA

- System Attributes ➤ ACD Print Enable, ACD Print Timer

STATION GROUP DATA

- Station Group Assignment ➤ACD Group Type
- Station Group Attributes ➤Supervisor

Related Features

■ Station Groups

2.7.3 Agents

2.7.3.1 Agent Auto Connect/Zap Tone

Description

Calls from an ACD group can automatically connect to agents using a headset. This feature removes the requirement for the agent to answer ACD callsmanually. The Agent receives a brief tone (Zap Tone), if assigned, and thenthe call is connected to the agent.

Operation

System

When Zap tone is enabled, operation of this feature is automatic.

Conditions

✓ The station must be in the headset mode for this feature to operate.

Programming

Keyset Admin.

STATION GROUPS

- CD Group (PGM 191)
- Zap Tone (PGM 191-Button 24-1)

Web Admin.

STATION GROUP DATA

- Station Group Assignment ➤ACD Group Type
- Station Group Attributes ➤Zap Tone

Related Features

■ Station Groups

Hardware

■ iPECS IP or LDP Phone

2.7.3.2 Agent Automatic Wrap-Up

Description

When an Agent completes an ACD group call, the Agent automatically enters into the Wrap-up state. In this state, an Agent will not receive ACD calls, allowing the Agent to complete paperwork, etc.

The Agent remains in this automatic Wrap-Up state for the duration of the ACD group Wrap-Up Timer. After the Wrap-Up Timer or by using “Wrap-Up-End” feature, the Agent returns to available status and can receive ACD group calls.

Operation

Activation is automatic when the Agent completes an ACD Group call

Agent iPECS IP & LDP Phones

*To assign a **{WRAP-UPEND}** button*

- **[TRANS/PGM]** + **{FLEX}** + “*585” + Group Number (Station Group for ACD) + **[HOLD/SAVE]**

To manually deactivate the Wrap-Up state

✓ Dial “*585” the Wrap-Up End code. Or, Press the **{WRAP-UPEND}** flexible button.

Agent SLT(Single Line Telephone)

To manually deactivate the Wrap-Up state

✓ Dial “*585” the Wrap-Up End code.

Conditions

✓ During Wrap-Up, the assigned {WRAP-UPEND} Flex button flashes. The button LED extinguishes when the Wrap-Up Timer expires or if the button is pressed, both actions return the Agent to available.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Agent Wrap-Up End (PGM 109-Button 9)

STATION GROUPS

- ACD Group (PGM 191)
- ACD Wrap-up Timer (PGM 191-Button 9)

Web Admin.

SYSTEM ID &NUMBERING PLANS

- Flexible Numbering Plan ➤Wrap-Up End

STATION GROUP DATA

- Station Group Assignment ➤ACD Group Type
- Station Group Attributes ➤Wrap-Up Timer

Hardware

2.7.3.3 Agent Help Request

Description

Agents request assistance from a Supervisor using the ACD Help Code, default *574. UsingiPECS IP or LDP Phones, agents with the **{ACD Help Code}** button can request assistance without interrupting anactive conversation.

Operation

Agent iPECS IP & LDP Phone

*To assign **{ACD Help Code}** buttons*

- **[TRANS/PGM]** + **{FLEX}** + “*574” + ACD Group Number (optional) + **[HOLD/SAVE]**

To request Supervisor assistance while on an active ACD call

✓ Press the **{ACD Help Code}** button.

SLT(Single Line Telephone)(Single Line Telephone)

To request Supervisor assistance while on an active ACD call

1) Hook-flash.
2) Dial “*574”, the Agent Help code.
3) Dial the desired ACD Group number.

Conditions

✓ Agents receive error tone to indicate there is no active Supervisor.

✓ Using the **{ACD Help Code}** button during a call, no signals are sent on the active conversation and the connected party is unaware of the request for assistance.

✓ If Supervisor already monitor the agent conversation, Help Request will be ignored

✓ Agent on recoding state cannot use Help Request

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Help Code (PGM 107-Button 5)

STATION GROUPS

- ACD Group (PGM 191)
- ACD Supervisor (PGM 191-Button 18)

Web Admin.

SYSTEM ID &NUMBERING PLAN

- Flexible Numbering Plan ➤ACD Help Code

STATION GROUPS

- Station Group Assignment ➤ACD Group Type
- Station Group Attributes ➤Supervisor

Related Features

- Station Groups
- Supervisor Help Response

2.7.3.4 Agent ID Login/Logout

Description

Stations or 4-digit Agent ID are assigned as members of an ACD group. A station or Agent ID can be assigned to a maximum of two (2) ACD groups. Each Agent has a unique 4-digit Agent ID used to login and logout of active ACD group participation.

Agents can login from any station in the system but only at one station. When an Agent logs in, any active login for the Agent is terminated and the new login is honored. An Agent may login to two (2) groups at one time, logging into and out of the primary and secondary group separately.

Operation

Agent iPECS IP & LDP Phones

*To assign **{LOGIN}** or **{LOGOUT}** buttons*

- **[TRANS/PGM]** + **{FLEX}** + Login or Logout code + ACD Group Number (optional) + **[HOLD/SAVE]**

To Login to a primary group using the Agent ID

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial "*581", the ACD Agent Primary Login code, or press the Login Flex button.
3) Dial the ACD group number.
4) Dial the Agent ID, the agent is logged into the ACD group.

To Logout of the primary ACD Group

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial "*582", the ACD Agent Primary Logout code, or press the Logout Flex button.
3) Dial the ACD group number.
4) Dial the Agent ID, the agent is logged into the ACD group.

To Login to a secondary group using the Agent Id

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial "*583", the ACD Agent Secondary Login code, or press the Login Flex button.
3) Dial the ACD group number.
4) Dial the Agent ID, the agent is logged into the ACD group.

To Logout of the secondary ACD Group

1) Lift the handset or the press **[SPEAKER]** button.
2) Dial "*584", the ACD Agent Secondary Logout code, or press the Logout Flex button.
3) Dial the ACD group number.
4) Dial the Agent ID, the agent is logged out of the ACD Group.

SLT(Single Line Telephone)(Single Line Telephone)

To Login to a primary group using the Agent Id

1) Lift the handset.
2) Dial "*581", the ACD Agent Primary Login code.
3) Dial the ACD group number.
4) Dial the Agent ID, the agent is logged into the ACD group.

To Logout of the primary ACD Group

1) Lift the handset.
2) Dial "*582", the ACD Agent Primary Logout code.
3) Dial the ACD group number.
4) Dial the Agent ID, the agent is logged out of the ACD Group.

To Login to a secondary group using the Agent Id

1) Lift the handset.
2) Dial "*583", the ACD Agent Secondary Login code.
3) Dial the ACD group number.
4) Dial the Agent ID, the agent is logged into the ACD group.

To Logout of the secondary ACD Group

1) Lift the handset.
2) Dial "*584", the ACD Agent Secondary Logout code.
3) Dial the ACD group number.
4) Dial the Agent ID, the agent is logged out of the ACD Group.

Conditions

✓ The system outputs ACD events including Login/Logout messages when Discovery Manager Print (Output ACD Event Message) is enabled.
✓ The Agent ID can be any 4-digit number. iPECS does not verify the Agent ID, other than requiring that four digits be entered.
✓ Any Agent ID may be used to login except at a Hot Desk Station where a Station Authorization code is required.
✓ PC UCS Client can control ACD Login/out using GUI.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Agent Primary Login (PGM 109-Button 5)
- ACD Agent Primary Logout (PGM 109-Button 6)
- ACD Agent Secondary Login (PGM 109-Button 7)
- ACD Agent Secondary Logout (PGM 109-Button 8)

STATION GROUPS

- ACD Group (PGM 191)

SYSTEM

- ACD Manager Print (Output ACD Event Message) (PGM 160-Button 13)
- Unified Message Format (PGM 161-Button 18)

Web Admin.

SYSTEM ID &NUMBERING PLANS

- Flexible Numbering Plan ➤ACD Agent Primary Login, ACD Agent Primary Logout, ACD Agent Secondary Login, ACD Agent Secondary Logout

STATION DATA

- Terminal Attributes ➤UCS ACD USE

STATION GROUP DATA

- Station Group Assignment ➤ACD Group Type

SYSTEM DATA

- System Attributes ➤ACD Manager Print
- System Attributes ➤Unified Message Format

Related Features

- Station Groups
- Hot Desk
- Authorization Codes (Password)

2.7.3.5 Agent On/Off Duty with Reason Code

Description

Agents can control their status, On/Off-duty, and assign a Reason code for an Off-duty state. The system outputs the Reason code as part of the ACD Event message output. The Reason code may be any digit (0 ~ 9, * and #). With the exception of Reason code “0”, when the Agent goes Off-duty manually, the Agent must return to the On-duty state manually.

If assigned as the Auto ACD DND code, using Reason code “0” activates the ACD DND Wrap-up timer. At expiration of the timer, the Agent is placed back On-duty. This provides the flexibility of an Agent activated Wrap-up time.

The Agent may assign an **{ACD ON/OFF DUTY}** button to for easy access. The ACD group number and the Reason code can be assigned in the Flex button for one-button operation. If the button is assigned Reason code “0”, the Agent will return to On-duty status after the ACD DND Wrap-up time.

Operation

Agent iPECS IP & LDP Phones

To assign an {ACD Agent ON/OFF DUTY} button

- **[TRANS/PGM]** + **{FLEX}** + “*571” + (optional ACD group number) + (optional Reason Code) + **[HOLD/SAVE]**

To toggle to Off-duty state from On-Duty,

1) Dial “*571”, Agent On/Off Duty Code.
2) Dial ACD group number.
3) Dial Reason Code (“0” ~ “9”, “*” or “#”).

Or,

1) Press {ACD ON/OFF DUTY} button.
2) Dial group number and Reason Code (“0” ~ “9”, “*” or “#”) unless assigned in the button.

To return to an On-duty state

1) Dial “*571”, Agent On/Off Duty Code.
2) Dial ACD group number.

Or,

1) Press the {ACD Agent ON/OFF DUTY} button.
2) Dial the ACD group number unless assigned in the button.

To assign an {ACD Agent ON/OFF DUTY, ALL GROUPS} button

- **[TRANS/PGM]** + **{FLEX}** + “*58*’ + (optional Reason Code) + **[HOLD/SAVE]**

To toggle to Off-Duty status to both groups at one time

1) Dial “*58*”, the All group On/Off-Duty code.
2) Dial the Reason code (“0” ~ “9”, “*” or “#”).

Or,

1) Press {ACD Agent ON/OFF DUTY ALL GROUPS} button.
2) Dial the Reason code (“0” ~ “9”, “*” or “#”), unless assigned in the button.

To return to an On-duty state for both groups at once

1) Dial “*58*” the code for the All group On/Off-Duty code.
2) Or, Press {ACD Agent ON/OFF DUTY ALL GROUPS} button.

SLT(Single Line Telephone)(Single Line Telephone)

To toggle to Off-duty state from On-Duty

1) Lift handset or press **[SPEAKER]** button.
2) Dial “*571”, Agent On/Off Duty Code.
3) Dial ACD group number.
4) Dial Reason Code (“0” ~ “9”, “*” or “#”).

To return to an On-duty state

1) Lift handset or press **[SPEAKER]** button.
2) Dial “*571”, Agent On/Off Duty Code.
3) Dial ACD group the umber.

To return to an On-duty state for both groups at once

1) Lift handset or press **[SPEAKER]** button.
2) Dial “*58*”, the code for the All group On/Off-Duty code.

To toggle to Off-Duty status to both groups at one time

1) Lift handset or press **[SPEAKER]** button.
2) Dial “*58*”, the code for the All group On/Off-Duty code.
3) Dial Reason Code (“0” ~ “9”, “*” or “#”).

Conditions

- ✓ The system will output an Agent Availability message with the dialed Reason code when an Agent changes availability status. The message will include the button type “Wrap” if the ACD DND timer is activated.
- ✓ The **{ACD Agent ON/OFF-DUTY}** Flex button will flash while the agent is Off-duty for any reason. The button LED extinguishes when the Agent returns to the On-duty status.
- ✓ While Off-duty, the supervisor's Flex button LED assigned for the Agent’s station flashes at the DND rate. The supervisor may call the station overriding the Off-duty status.
- ✓ Only one **{ACD Agent ON/OFF-DUTY}** button with the Group number can be assigned at a station.
- ✓ The default value of the ACD DND Wrap-Up Timer is 1 second.
- ✓ PC UCS Client can operate ACD On/Off duty using GUI.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Agent On/Off Duty (PGM 107-Button 2)
- All group Agent ON/OFF duty (PGM 109-Button 17)

STATION

- Auto ACD DND Reason (PGM 113-Button 15)

STATION GROUPS

- ACD Group (PGM 191)
- ACD DND Wrap-up Timer (PGM 191-Button 20)

Web Admin.

SYSTEM ID&NUMBERING PLANS

- Flexible Numbering Plan ➤ACD Agent On/Off Duty
- Flexible Numbering Plan ➤Agent ON/OFF Duty in ALL GRP

STATION DATA

- Common Attributes ➤Auto ACD DND
- Terminal Attributes ➤UCS ACD USE

STATION GROUP DATA

- Station Group Assignment ➤ACD Group Type
- Station Group Attributes ➤Wrap-up Timer

Related Features

- Agent Automatic Wrap-Up
- Auto ACD DND Unavailable Service
- Event Messages

Hardware

2.7.3.6 Agent Queued Calls Display

Description

An Agent can view the queued call status for an ACD group when not on a call, On-duty orWrap-up status.

In addition, an active Agent can view the queued call status for an ACD groupwhile on a call using a**{ACD Calls in Queue Display}** Flex button.

The Call Queue display indicates theACD group number, the number of calls in queue and the longest queue time.

Operation

Agent iPECS IP & LDP Phone

To assign a {ACD Calls in Queue Display} buttons

- **[TRANS/PGM]** + **{FLEX}** + “*575” + ACD Group Number + **[HOLD/SAVE]**

To display ACD Call Queue status when the Agent is idle and on-duty

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial “*575”, the ACD Call Queue status code.
3) Dial the ACD Group number. The display shows the Queued call status and will update as the queue status changes. If no calls are in queue, the phone returns to idle.
4) Hang-up to return to idle and normal display.

Or,

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the **{ACD Calls in Queue Display}** button. The display shows the Queued call status and will update as the queue status changes. If no calls are in queue, the phone returns to idle.
3) Hang-up to return to idle and normal display.

To display ACD Call Queue status when the Agent is on a call

1) Press the **{ACD Calls in Queue Display}** button, the queue status displays for several seconds then returns to the normal call display.

Conditions

- ✓ The Agent cannot display queued call status while the phone is ringing.
- ✓ If there are no queued calls to display and you activate Queued Calls Display, the phone returns to idle after providing error tone.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Calls In Queue (PGM 107-Button 6)

STATION GROUPS

- ACD Group (PGM 191)

Web Admin.

SYSTEM ID &NUMBERING PLANS

- Flexible Numbering Plan ➤ ACD Calls In Queue Display

STATION GROUP DATA

- Station Group Assignment ➤ACD Group Type

Related Features

- Station Groups

Hardware

- iPECS IP or LDP Phone

2.7.4 Announcements

Description

Each ACD group can provide announcements to incoming callers. Primary and secondaryannouncements are available with control timers. The first announcement is played after 1stcontrol timer.

The 2nd control timer determines the period between the first and secondannouncement. The second announcement can be replayed at defined intervals. If desired, theprimary message can be defined to play in full to all callers as a “guaranteed announcement”or only for calls that queue.When enabled, a CIQ (Calls in Queue) announcement is played to callers that are queued tothe ACD group.

The CIQ announcement (“Your Call is number xx in Queue”) is played toqueued callers after the 1st announcement is played and again after each time the 2ndannouncement is played. Internal callers using an iPECS IP or LDP phone with Display also receive the “Youare xx in queue” display message.

Operation

System

ACD group announcements are played automatically based on the ACD group assignments.

System Attendant

To record a System Announcement for an ACD Group

1) Press the **[TRANS/PGM]** button.
2) Dial “06”, the System Announcement Record code.
3) Dial the sequence number of the voice storage unit.
4) Dial the Announcement number (“001”-“202”).
5) Dial the language type (“1”~”6”), the current announcement is played followed by the “Press # to record” prompt.
6) Dial “#”.
7) After the beep-tone, record message.
8) Press the **[HOLD/SAVE]** button to stop recording and save the announcement.

To delete a recording

1) Press the **[TRANS/PGM]** button.
2) Dial “06”, the System Announcement Record code.
3) Dial the sequence number of the voice storage unit.
4) Dial the Announcement number (“001”-“202”).
5) Dial the language type (“1”~”6”), the current announcement is played followed by the “Press # to record” prompt.
6) Press the **[SPEED]** button during playback to erase the announcement.

Conditions

- ✓ ACD group announcements are recorded in the integrated Auto Attendant/Voice Mail. Up to two hundreds (200)Announcements are available for group announcement recordings. Announcement201 is reserved for MOH and 202 is reserved for Multi Language selection.

- ✓ When equipped to support multiple languages, the Language selection announcement is played prior to a guaranteed announcement, allowing the caller to select between the languages equipped in the system (maximum 3)
- ✓ To define a guaranteed announcement, assign 0 seconds to the 1st announcement control timer.
- ✓ The sequence number for the voice storage device is given in the Administration and Programming Manual 'System Overview Web page'.

Programming

Keyset Admin.

STATION GROUPS

- ACD Group (PGM 191)
- ACD ICLID Usage (PGM 191-Button 21)
- CIQ Mention (PGM191-Button 24-21)

Web Admin.

STATION GROUP DATA

- Station Group Assignment ➤ACD Group Type
- Station Group Attributes ➤Entered Caller ID(ICLID) Usage, CIQ Announcement

Related Features

- Integrated Auto Attendant
- ACD Caller Controlled ICLID Routing
- Multiple Language Support

Hardware

2.7.5 Auto ACD DND Unavailable Service

Description

An Agent who does not answer a call offered from an ACD group within the ACD No-Answertimer is placed in a “No response” unavailable status, called ACD DND. In this state, the Agentis Off-duty and will not receive ACD calls from the group. If the Auto DND Reason code is “0”, after the ACDDND Wrap timer expires, the Agent returns to available status. Otherwise the Agent must return to the On-duty status manually.

Operation

System

Auto ACD DND activates automatically when Reason code “0” is assigned.

Agent iPECS IP & LDP Phones

To return to the On-duty status

1) Dial “*571”, ACD Agent On/Off Duty Code.
2) Dial the ACD group number.

Or,

1) Press the {ACD Agent ON/OFF-DUTY} button.
2) Dial the ACD group number unless assigned in the button.

SLT(Single Line Telephone)

To return to the On-duty status

1) Lift the handset.
2) Dial “*571”, the Agent On/Off Duty Code.
3) Dial the ACD group number.

Conditions

- ✓ The system will output an Agent Available message with a Reason code when an Agent changes availability status.
- ✓ At the Supervisor's phone, the LED of a Flex button assigned for an agent’s Station will flash at the DND rate. The supervisor may call the Station.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Agent On/Off Duty (PGM107-Button 2)

STATION

- Auto ACD DND Reason (PGM 113-Button 15)

STATION GROUPS

- ACD Group (PGM 191)
- ACD No Answer Timer (PGM 191-Button 24-22)

Web Admin.

SYSTEM ID &NUMBERING PLANS

- Flexible Numbering Plan ➤ACD Agent On/Off Duty

STATION DATA

- Common Attributes ➤Auto ACD DND

STATION GROUP DATA

- Station Group Assignment ➤ACD Group Type
- Station Group Attributes ➤No Answer Timer

Related Features

- Agent On/Off Duty with Reason Code

Hardware

2.7.6 Calls-In-Queue routing

Description

When a caller is queued to an ACD Group, various announcements may be played and musicon hold may be sent to the caller. The caller may dial a digit at any time while queued to exitthe queue, except during a Guaranteed Announcement. The dialed digit is compared to digitsdefined in the ACD Group CIQ Route Table. If a match is found, the call is routed to the defineddestination (Station, Station Group, etc.). If a match is not found, external callers receive anerror message and are placed back in queue; internal callers are simply placed back in queue.

Operation

Operation of this feature is automatic based on caller input.

Conditions

- ✓ ACD CIQ routing is not supported during or within 5 seconds of a Guaranteed Announcement. Digits dialed during a Guaranteed Announcement use Caller Controlled ICLID routing to determine routing as discussed in Section 2.7.1.
- ✓ Dialing during ring back tone is not recognized by the system.

Programming

Keyset Admin.

STATION GROUPS

- ACD Group (PGM 191)
- ACD CIQ Route Table (PGM 191-Button 23)

Web Admin.

STATION GROUP DATA

- Station Group Assignment ➤ACD Group Type
- Station Group Attributes ➤CIQ Route Table

Related Features

- Station Groups
- Announcements
- ACD Caller Controlled ICLID Routing

Hardware

2.7.7 Calls-In-Queue Page Alert

Description

The administrator can establish CIQ (Calls-In-Queue) thresholds for each ACD Group so thatan assigned System announcement plays over a defined Page zone, if the number of callsin queue equals or exceeds a CIQ threshold. The message, which is recorded by an Attendant,plays immediately or after a defined delay. The message repeats at assigned intervals until thenumber of Calls in Queue falls below the threshold. The system allows different treatment andannouncements for up to three different Calls-In-Queue thresholds.

Operation

System

When the number of Calls-In-Queue changes

1) Checks CIQ thresholds.
2) Determines if a threshold is reached.
3) Queues announcement for threshold (sets Delay timer).
4) Plays CIQ announcement.
5) Repeats announcement as defined

System Attendant

To record a CIQ Announcement for an ACD Group

1) Press the **[TRANS/PGM]** button.
2) Dial “06”, the System Announcement Record code.
3) Dial the sequence number of the voice storage unit.
4) Dial the System Announcement number (“001”-“200”).
5) Dial language type (“1”~”6”), the current announcement is played followed by the “Press # to record” prompt.
6) Dial “#”.
7) After the beep-tone, record message.
8) Press the **[HOLD/SAVE]** button to stop recording and save the announcement.

To delete a recording

1) Press the **[TRANS/PGM]** button.
2) Dial “06”, the System Announcement Record code.
3) Dial the sequence number of the voice storage unit.
4) Dial the System Announcement number (“001”-“200”).
5) Dial language type (“1”~”6”), the current announcement is played followed by the “Press # to record” prompt.
6) Press the **[SPEED]** button during playback to erase the announcement.

Conditions

- ✓ Thresholds are assigned as total Calls-In-Queue to activate a Page alert and are checked from lowest priority (CIQ #1) to highest priority (CIQ #3). Thus, the highest priority CIQ defined should have the largest threshold and the lowest priority CIQ should have the smallest threshold.
- ✓ The System has two hundreds (200) announcements employed for Auto Attendant, ACD groups, CCR, Page Alerts, etc. Two announcements are reserved for MOH and multi-language support, announcements 201 and 202 respectively.

Programming

Keyset Admin.

STATION GROUPS

- ACD Group (PGM 191)
- CIQ #1 Threshold (PGM 191, Button 24-6)
- CIQ #1 Announcement Location (PGM 191-Button 24-7)
- CIQ #1 Page Zone (PGM 191-Button 24-8)
- CIQ #1 Announcement Delay Timer (PGM 191-Button 24-9)
- CIQ #1 Announcement Repeat Timer (PGM 191-Button 24-10)
- CIQ #2 Threshold (PGM 191-Button 24-11)
- CIQ #2 Announcement Location (PGM 191-Button 24-12)
- CIQ #2 Page Zone (PGM 191-Button 24-13)
- CIQ #2 Announcement Delay Timer (PGM 191-Button 24-14)
- CIQ #2 Announcement Repeat Timer (PGM 191-Button 24-15)
- CIQ #3 Threshold (PGM 191-Button 24-16)
- CIQ #3 Announcement Location (PGM 191-Button 24-17)
- CIQ #3 Page Zone (PGM 191-Button 24-18)
- CIQ #3 Announcement Delay Timer (PGM 191-Button 24-19)
- CIQ #3 Announcement Repeat Timer (PGM 191-Button 24-20)
- VSF Group Number (PGM 190)

Web Admin.

STATION GROUP DATA

- Station Group Assignment ➤ACD Group Type
- Station Group Attributes ➤CIQ #1,#2,#3 Page Alert➤Threshold, Announcement Location, Page Zone, Announcement Delay Time, Announcement Repeat Time
- Station Group Assignment ➤VSF-VM Group Type

Related Features

- ■ Station Groups
- ■ Integrated Auto Attendant/Voice Mail
- ■ Internal/External & All Call Page

2.7.8 Event Messages

Description

Each ACD group sends messages for group and agent events over a defined Serial or TCP/IPport. These event messages are employed by 3rd party applications to provide enhanced CallCenter functionality. Details of the event messages are provided in the iPECS 3rd Party Support Guide.

Operation

System

When enabled, Event messages are sent automatically.

Programming

Keyset Admin.

STATION GROUPS

- ACD Group (PGM 191)

SYSTEM

- ACD Pack (PGM 175- Button 1-9)
- ACD Manager Print (Output ACD Event Message) (PGM 160-Button 13)

Web Admin.

STATION GROUP DATA

- Station Group Assignment ➤ ACD Group Type

SYSTEM DATA

- Serial Port Selections ➤ACD Package Print
- System Attributes ➤ACD Manager Print (Output ACD Event Message)

Related Features

- Station Groups

2.7.9 Group Mail Box

Description

A Voice mailbox can be associated with a Station Group. Calls to the group are assigned to overflowand can be otherwise re-routed to the Station Group Mail box where the caller can leave a voice message.

Messages are retrieved in the same manner as normal voice messages employing the StationGroup number as the station number and the defined Group Mailbox Password. Messagescan be retrieved only if a **{GROUP MAILBOX}** Flex button is assigned to the station.

Operation

Agent iPECS IP & LDP Phone

*To assign a **{GROUP MAILBOX}** Flex button*

- **[TRANS/PGM]** + **{FLEX}** + VSF-VM Station Group + Group Message Wait Station + **[HOLD/SAVE]**

To retrieve Station Group Voice Mail

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the**{GROUP MAILBOX}** button to receive the “Password” prompt.
3) Dial the ACD Group number and Group Mailbox Password. The “Number of Messages” prompt followed by Voice Mail instruction prompts are played.
4) Dial the desired option from the Voice Mail prompt.
5) At completion of the session, hang-up to return to idle.

To leave a voice message

1) After the Mailbox Greeting and beep, leave the message.
2) Hang up to quit recording or dial “*” for further options.

Conditions

- ✓ The System Administrator or Supervisor may change the Group Mailbox Password.
- ✓ An outside caller cannot access the Group Mailbox to retrieve messages.

Programming

Keyset Admin.

STATION GROUPS

- Station Group Attributes for VSF-VM and ACD (PGM 191)
- Mailbox Message Wait Station in ACD Group (PGM 191-Button 24-2)
- Mailbox Password in ACD Group (PGM 191-Button 24-3)

Web Admin.

STATION GROUP DATA

- Station Group Assignment ➤VSF-VM Group Type, ACD Group type
- Station Group Attributes in ACD Group➤Mailbox Message Wait Station, Mailbox Password

Related Features

- External Auto Attendant/Voice Mail
- Station Groups
- Integrated Voice Mail
- Supervisors

Hardware

- iPECS IP or LDP Phones

2.7.10 Group Name

Description

Each Station group is assigned a name of up to 12 characters.

The name is employed as thecalled party display for internal callers and in the output of group statistics.

Operation

System

Operation of this feature is automatic when programmed.

Programming

Keyset Admin.

STATION GROUPS

- Station Group Attributes (PGM 191)
- ACD Group Name (PGM 191-Button 22)

Web Admin.

STATION GROUP DATA

- Station Group Assignment ➤ ACD Group Type
- Station Group Attributes ➤Group Name

Related Features

- Station Groups

2.7.11 Incoming Call Routing

Description

Incoming calls to an ACD group route directly to the station/agent that has been idle the longest(Uniform Call Distribution) or an ACD group announcement recorded. If allagents are busy, the call is queued to the group or routes to the Alternate Destination. Thequeued caller receives ACD group announcements or audio from the defined music source.Callers that remain in queue receive audio from the defined music source or ring-back toneand, after the defined Overflow time, route to an Overflow destination.

Agents can be assigned a priority from“0” ~ “9”, “0” being the lowest priority and “9” the highest.When priorities are assigned, calls are routed to the highest priority Agent that has been idlethe longest.If there are no Agents active in the group when a call arrives, the call will receive the “NoMember” treatment. Intercom calls receive re-order and external callers are routed to anAttendant.

Operation

System

Routing of ACD group calls is automatic based on the ACD group assignments.

Conditions

- ✓ The Alternate ACD Destination can be a group or a station but cannot be a member of the ACD group.
- ✓ The Overflow destination may be any station or group as well as the ACD group Voice Mailbox, but cannot be a member of the overflowing ACD group.
- ✓ If an agent becomes available during an announcement, except a guaranteed announcement, the call is passed immediately to the agent.

Programming

Keyset Admin.

STATION GROUPS

- ACD Group (PGM 191)
- Overflow Destination (PGM191-Button 7)
- Overflow Timer (PGM 191-Button 8)
- Music Source (PGM 191-Button 11)
- Alternate Destination (PGM 191-Button 13)
- Report No Members (PGM 191-Button 10)
- Agent Priority (PGM 191-Button 19)

Web Admin.

STATION GROUP DATA

- Station Group Assignment ➤ ACD Group Type
- Station Group Attributes ➤Overflow Destination, Overflow Timer, Music Source, Alternate Destination, Report No Member, Agent Priority

Related Features

- Automatic Call Distribution

2.7.12 Supervisors

2.7.12.1 Agent Call Monitor

Description

Agent Call Monitor permits an active Supervisor to monitor an Agent's call in progress fortraining purposes or to assist the Agent.

When used, the Supervisor is connected to the callwith the microphone muted, the **[MUTE]** button LED is On. If ACD Warning Tone is enabled, aWarning Tone is provided to the Agent's call indicating the Supervisor has entered the call.

TheSupervisor hears the Agent and connected party and may join the conversation by pressing the**[MUTE]** button to activate the microphone.The Supervisor station must be assigned an **{ACDSupervisor Monitor}** button to activate the AgentCall Monitor feature.

Operation

Supervisor iPECS IP & LDP Phone

To assign an {ACD Supervisor Monitor} button

- **[TRANS/PGM]** + **{FLEX}** + "*577" + "Group Number" + **[HOLD/SAVE]**

To enter an active Agent call from the Supervisor phone

1) Lift the handset or press the **[SPEAKER]** button.
2) Call the desired Agent and receive busy tone.
3) Press the **{ACD Supervisor Monitor}** button, the Supervisor is connected to the active Agent call with the microphone muted, the **[MUTE]** button LED is On.

To converse with the Agent and connected party

✓ Press the illuminated **[MUTE]** button.

Conditions

✓ The Supervisor must be active and logged into the Agent's ACD group to monitor calls in progress.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Supervisor Monitor (PGM 107-Button 8)

STATION GROUPS

- ACD Groups (PGM 191)
- ACD Group Supervisor (PGM 191-Button 18)
- ACD Warning Tone (PGM 191-Button 12)

Web Admin.

SYSTEM ID &NUMBERING PLANS

- Flexible Numbering Plan➤ACD Supervisor Monitor

STATION GROUP DATA

- Station Group Assignment ➤ ACD Group Type
- Station Group Attributes ➤Supervisor (1 to 5),ACD Warning Tone

Related Features

- Station Groups

Hardware

- iPECS IP or LDP Phone

2.7.12.2 Agent Status Control

Description

An ACD Supervisor can view and control the status of Agents in the group with a **{ACD Supervisor Status}** button.

Operation

Supervisor iPECS IP & LDP Phones

*To assign a **{ACD Supervisor Status}** button at the Supervisor*

- **[TRANS/PGM]** + **{FLEX}** + "*576" + "Group Number" + **[HOLD/SAVE]**

To view/control Agent Status

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the {ACD Supervisor Status} button.

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

3) Dial "3", the Agent Duty code, the LCD menu will update:

[1] DUTY STATUS [2] DUTY ON/OFF [#] DUTY PRINT

To view Agent Status

1) Dial "1", the Agent Status Code.

ACD STATUS : XXX TOTAL CALLS : XXX

2) Press [VOL UP] or [VOL DOWN] for:
   - Number of ACD calls served
   - Number of unanswered ACD Calls
   - Average ring time before answer
   - Average ACD call service time after answer.
3) Press the "*" key to view the status of the next Agent.

To control Agent duty status

1) Dial "3", the Agent Duty code.
2) Dial "2" to set the Agent's On/Off Duty status.
3) Dial "0" or "1", ("0": Off Duty, "1": On Duty).

Conditions

✓ The Supervisor must be assigned as a Supervisor for the ACD Group and be logged into the group to access the Group Status display information.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Group Status (PGM 107-Button 7)

STATION GROUPS

- ACD Group (PGM 191)
- ACD Group Supervisor (PGM 191-Button 18)

Web Admin.

SYSTEM ID &NUMBERING PLAN

- Flexible Numbering Plan➤ACD Supervisor Status

STATION GROUP DATA

- Station Group Assignment ➤ ACD Group Type
- Station Group Attributes ➤Supervisor1 ~5

Related Features

- Station Groups

Hardware

- iPECS IP or LDP Phone

2.7.12.3 Group Status Display

Description

An ACD Supervisor may view the status of each of the ACD groups to which the Supervisor isassigned. The display will give the status of the active ACD group at the time of the request.

The display indicates the following statistics:

- Number of calls in Queue
- Wait time for the call in queue longest
- Agents in service and available

When a call is queued to the group, the ACD Supervisor is notified, in real-time, by the LED ofthe **{ACD Supervisor Status}** button. The **{ACD Supervisor Status}** button allows access to the ACDGroup Status display or the Group Status code may be used to display the status information.

Operation

Supervisor iPECS IP & LDP Phone

*To assign a **{ACD Supervisor Status}** button at the Supervisor*

- **[TRANS/PGM]** + **{FLEX}** + "*576" + "Group Number" + **[HOLD/SAVE]**

*To display the Group status when the {**ACD Supervisor Status**} button is flashing*

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial "*576", the Group Status display code.
3) Dial "1" to see the Group Status information.

Or,

1) Press the flashing **{ACD Supervisor Status}** button.
2) Dial "1" to see the Group Status display.

Conditions

✓ The Supervisor must be assigned as a Supervisor for the ACD Group and be logged into the group to access the Group Status display information.

✓ The **{ACD Supervisor Status}** button will flash when the count of Calls-in-Queue reaches or exceeds the programmed 'Supervisor Call Count' after the programmed 'Supervisor Timer'.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Supervisor Status Code (PGM 107-Button 7)

STATION GROUPS

- ACD Group (PGM 191)
- ACD Group Supervisor (PGM 191-Button 18)
- Supervisor Timer (PGM 191-Button 14)
- Supervisor Queued Call Count (PGM191-Button 15)

Web Admin.

SYSTEM ID &NUMBERING PLANS

- Flexible Numbering Plan➤ACD Supervisor Status

STATION GROUP DATA

- Station Group Assignment ➤ ACD Group Type
- Station Group Attributes ➤Supervisor1 ~5,Supervisor Timer, Supervisor Call Count

Related Features

- Station Groups

Hardware

- iPECS IP or LDP Phone

2.7.12.4 Group Parameter Control

Description

An ACD Group Supervisor can adjust several of the ACD Group routing parameters inreal-time.

A **{ACD Supervisor Status}** button is required. Adjustable parameters are:

- Overflow Destination
- Overflow Time
- Wrap-up Time
- ACD Voice Mailbox Password

Operation

Supervisor iPECS IP & LDP Phone

*To assign a **{ACD Supervisor Status}** button at the Supervisor*

- **[TRANS/PGM]** + **{FLEX}** + "*576" + "Group Number" + **[HOLD/SAVE]**

To use the Supervisor Status Menu

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the {ACD Supervisor Status} button.

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

3) Dial ACD Dbase Code "2".
4) Select database item, scrolling with the **[VOL UP]**/**[VOL DOWN]** button;
   - Overflow Destination- station (Net station) (1)/group (2)/system speed dial (3)
   - Overflow Time- xxx seconds
   - Wrap-Up Time- xxx seconds
   - Password- up to 12 digits
5) Enter new data.
6) Press the **[HOLD/SAVE]** button.

Conditions

✓ The Supervisor must be assigned as a Supervisor for the ACD Group and be logged into the group to access the Group Status display information.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Supervisor Status code (PGM 107-Button 7)

STATION GROUPS

- ACD Group (PGM 191)

- ACD Group Supervisor (PGM 191-Button 18)

Web Admin.

SYSTEM ID &NUMBERING PLAN

- Flexible Numbering Plan➤ACD Supervisor Status

STATION GROUP DATA

- Station Group Assignment ➤ ACD Group Type
- Station Group Attributes ➤Supervisor (1 to 5)

Related Features

- Station Groups

Hardware

- iPECS IP or LDP Phone

2.7.12.5 Supervisor Help Response

Description

An Agent may request assistance from a Supervisor using the ACD Help Code, default *574 or **{ACD Help Code}** button. The Supervisor is then notified of the Help request in the Phone display and a flashing **{HELP RESPONSE}** button. The Supervisor can respond using the Help Response button, which is required, and be connected to the Agent's active call with the microphone muted.

The Supervisor hears the Agent and connected party and may join the conversation by pressing the **[MUTE]** button to activate the microphone.

Operation

Agent iPECS IP & LDP Phone

*To assign a **{ACD Help Code}** button*

- **[TRANS/PGM]** + **{FLEX}** + "*574" + "Group Number" + **[HOLD/SAVE]**

To request Supervisor assistance

✓ Press the **{ACD Help Code}** button.

SLT(Single Line Telephone)

1) Hook-flash and dial "*574", the Agent Help code.
2) Dial the desired ACD Group number.

Supervisor iPECS IP & LDP Phones

*To assign a **{HELP RESPONSE}** button*

- **[TRANS/PGM]** + **{FLEX}** + "*574" + "Group Number" + **[HOLD/SAVE]**

To respond to a Help request, at the Supervisor Phone

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the flashing **{HELP RESPONSE}** button, the Supervisor is connected to the active Agent call with the microphone muted, the **[MUTE]** button LED is On.

To converse with the Agent and connected party

1) Press the illuminated **[MUTE]** button.

Conditions

✓ The Agent Help request is sent to the lowest numbered available Supervisor station for the associated ACD Group.

✓ If no Supervisor is available, the Agent receives error tone.

✓ A Warning Tone, if enabled, will be sent to the Agent as the Supervisor is connected to the conversation.

✓ Only one active Help request is allowed at a time, a second help request will cancel any active help request.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Help Code (PGM 107-Button 5)

STATION GROUPS

- ACD Groups (PGM 191)
- ACD Group Supervisor (PGM 191-Button 18)
- ACD Warning Tone (PGM 191-Button 12)

Web Admin.

SYSTEM ID &NUMBERING PLAN

- Flexible Numbering Plan➤ACD Help Code

STATION GROUP DATA

- Station Group Assignment ➤ ACD Group Type
- Station Group Attributes ➤Supervisor1 ~ 5,ACD Warning

Related Features

- Automatic Call Distribution
- Agent Help Request

Hardware

- iPECS IP or LDP Phone

2.7.12.6 Supervisor Login/Logout

Description

Each ACD group is assigned up to five (5) Supervisors to monitor and control real-time status of the ACD group and Agents assigned to the group. A Supervisor can be assigned to multiple groups. The Supervisor login/logout feature provides a means for a supervisor to log into one or more ACD groups and monitor calls.

Each Supervisor has a unique 4-digit Agent ID used to login and logout of active ACD group participation. Supervisors can login from any station in the system but only at one station at a time. If a Supervisor logs into a group from a station while logged in at another station, the active login is terminated and the new login is honored.

Operation

Supervisor iPECS IP & LDP Phone

To assign {ACD Supervisor Login} or {ACD Supervisor Logout} buttons

- **[TRANS/PGM]** + **{FLEX}** + Login (“*572”) or Logout (“*573”) code + ACD Group Number + **[HOLD/SAVE]**

To login

1) Dial “*572”, the ACD Supervisor Login code or press the Flex button.
2) Dial the ACD group number.
3) Dial supervisor Id code (“0000”-“9999”).

Or,

1) Press the Supervisor **{ACD Supervisor Login}**button.
2) Dial supervisor Id code (“0000”-“9999”)

To Logout

1) Dial “*573”, the ACD Supervisor Logout code or press the Logout Flex button.
2) Dial the ACD group number.
3) Dial supervisor Id code (“0000”-“9999”).

Or,

1) Press the Supervisor **{ACD Supervisor Logout}** button.
2) Dial supervisor Id code (“0000”-“9999”).

Conditions

- ✓ The system will output a Login/Logout message when a Supervisor logs-in or out.
- ✓ The Supervisor ID can be any 4-digit number (“0000”-“9999”).
- ✓ The system does not verify the Supervisor ID codes, other than requiring four digits are entered.
- ✓ Any Supervisor ID may be used to login through stations not assigned as a Hot Desk.
- ✓ The ACD Supervisor phone can be assigned an ACD Group Status Flex button used to view the activity for the group when the Supervisor logs in to multiple groups.

Programming

Keyset Admin.

NUMBERING PLAN

- ACD Supervisor Login (PGM 107-Button 3)
- ACD Supervisor Logout (PGM 107-Button 4)

STATION GROUPS

- ACD Groups (PGM191)
- ACD Group Supervisor (PGM 191-Button 18

Web Admin.

SYSTEM ID &NUMBERING PLAN

- Flexible Numbering Plan➤ACD Supervisor Login, ACD Supervisor Logout

STATION GROUP DATA

- Station Group Assignment ➤ ACD Group Type
- Station Group Attributes ➤Supervisor1 ~5

Related Features

- Automatic Call Distribution
- Hot Desk

Hardware

- iPECS IP or LDP Phone

2.8 Auto Called Number Redial (ACNR)

Description

This feature allows you to request and have the system retry a busy or no answer external call until the call is connected or the feature is cancelled.

Operation

iPECS IP & LDP Phones

To activate ACNR while receiving busy, no answer

1) Press the **[REDIAL]** button.
2) Hang-up handset, or press **[SPEAKER]**.

To cancel ACNR

1) Press flashing **[REDIAL]** button.

SLT(Single Line Telephone)

To activate ACNR while receiving busy, no answer

1) Place the outside call as normal.
2) If the called party is busy or does not answer, press and release the hook-switch, intercom dial tone should be heard.
3) Dial "*58#", the ACNR code.
4) Wait off-hook until the call is answered.

System

1) The system initiates the ACNR process, starting the ACNR Pause Timer.
2) At expiration of the timer, the system activates the station's speakerphone with the microphone in the mute mode.
3) The system attempts the previous call.
4) When the called party answers, the user may answer by lifting the handset or pressing the **[MUTE]** button to communicate with called party.

Conditions

ACNR Pause Time	Time allowed between ACNR attempts.
ACNR Delay Timer	At expiration of Pause Timer, if no Line is available, the system will wait for delay timer before attempting to access a Line.
ACNR Tone Detect	After dialing, the system will abandon the attempt if no tone or answer is detected within the Tone Detect time.
ACNR Retry Count	Count determines the number of times system will attempt the call before ACNR is automatically cancelled.

✓ Four timers and a retry counter can be programmed.

- ✓ The call will be placed on the same outside Line as originally used. If the Line is busy, an available Line in the same Line group will be seized.
- ✓ The ACNR Retry Counter decrements by one when the system completes the dialed number.
- ✓ When the ACNR Pause Timer expires, if the station is in a busy state, the ACNR Delay Timer is invoked.
- ✓ Upon completion of dialing, the system will monitor the call for progress signals.
- ✓ To preserve ACNR feature in activated state, SLT(Single Line Telephone) user does not go On-Hook.
- ✓ While ACNR dialing or waiting state, if SLT(Single Line Telephone) user goes On-Hook, then SLT(Single Line Telephone) ACNR is deactivated, automatically.
- ✓ While ACNR is active for a SLT(Single Line Telephone), the SLT(Single Line Telephone) user cannot receive or place a call.

Programming

Keyset Admin.

SYSTEM

- ACNR Delay Timer (PGM 180-Button 8)
- ACNR Pause Timer (PGM 180-Button 9)
- ACNR Retry Counter (PGM 180-Button 10)
- ACNR Tone Detect Timer (PGM 180-Button 11)

NUMBERING

- SLT(Single Line Telephone) ACNR Code (PGM 109-Button 18)

Web Admin.

SYSTEM ID & NUMBERING PLANS

- Flexible Numbering Plan ➤SLT(Single Line Telephone) ACNR

SYSTEM DATA

- System Timers ➤ACNR Delay Timer, ACNR Pause Timer, ACNR Retry Counter, ACNR Tone Detect Timer

Related Features

- LNR (Last Number Redial)
- Speakerphone
- Mute

Hardware

- iPECS IP or LDP Phone

2.9 Automatic Pause Insertion

Description

The system will automatically pause when dialing a Speed Dial number to allowfor potential connection delays. The pause will be inserted when any of the following occurs:

- After a Flash is encountered in a Speed Dial number,
- After a PABX access code is encountered in a Speed Dial or redial number,
- After a Pulse to Tone Switchover is encountered in a Speed Dial or Redial number.
- When a connect message is received on an ISDN Line.

Operation

System

The system automatically pauses dialing after an appropriate event.

Conditions

- ✓ An automatically inserted pause is not counted as a digit in a Speed Dial number.
- ✓ The LCD of an iPECS IP or LDP Phone will show a "P" when a pause is encountered. This indication is not shown if the system inserts the Pause automatically.

Programming

Keyset Admin.

SYSTEM

- Pause Timer (PGM 181-Button 10)

Web Admin.

SYSTEM DATA

- System Timers ➤Pause Timer

Related Features

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

Hardware

2.10 Automatic Privacy

Description

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

Operation

iPECS IP & LDP Phones

To intrude into a call when Privacy is disabled

✓ Press a busy (lit steady) individual **{LINE}** access button, user connected to the call with existing internal station user.

Conditions

✓ With Automatic Privacy disabled, privacy is still assured on all intercom and conference calls.

✓ To override privacy, Privacy must be disabled and the intruding Station must have Override enabled as well as a direct appearance for the desired {LINE}button.

✓ Only one station can intrude on an active call.

✓ An intrusion tone can be provided to the call indicating another station has accessed the Line.

✓ If either internal party presses another {LINE}, {DSS}, {PAGE}, [CONF] or other conflicting button, the party is removed from the "Conference" and must press the{LINE} button again to reenter the conversation.

Programming

Keyset Admin.

STATION

- Override Privilege (PGM 113-Button 4)

SYSTEM

- Automatic Privacy (PGM 161-Button 3)
- Privacy Warning Tone (PGM 161-Button 4)

Web Admin.

STATION DATA

- Common Attributes ➤Override Privilege

SYSTEM DATA

- System Attributes ➤Automatic Privacy, Privacy Warning Tone

Related Features

- Multi-Party Voice Conference
- Station Flexible Buttons

Hardware

- iPECS IP or LDP Phone

2.11 Auto Service Mode Control

Description

The service mode defines different ring assignments, COS (Class of Service) and answering privileges for theexternal network connections and Stations in the system based on the timeof day. The service mode can be controlled automatically through definitions in the Auto RingMode Selection Table(Weekly Time table), which defines the time of day for the Day, Night and Timed shift modes for each day of the week.

The Attendant may change the service mode selection from automatic to manual selecting the Day, Night, timed or Scenario mode.

Operation

System

Operation of this feature is automatic.

Programming

Keyset Admin.

STATION

- Station COS (Class of Service) (PGM 116)

CO/IP

- CO/IP Ring Assignment (PGM 144)
- CO Line COS (Class of Service) (PGM 141-Button 2)

SYSTEM

- DISA COS (Class of Service) (PGM 166)
- External Control Contact (PGM 168)
- PABX Access Codes (PGM 172)
- LD Call Digit Count (PGM 177-Button 4)
- Long Distance Code (PGM 177-Button 19)

TABLES

- Toll Exception Tables (PGM 224)
- Authorization Codes Table (PGM 227)
- Weekly Time Table (PGM 233)

Web Admin.

STATION DATA

- Station COS

CO LINE DATA

- CO/IP Ring Assignment
- Common Attributes ➤CO Line COS (Class of Service)

SYSTEM DATA

- DISA COS (Class of Service)
- External Control Contact
- PBX Access Codes
- Attendant Assignment
- SMDR Attributes ➤ Long Distance Call Digit Counter, Long Distance Code

TABLES DATA

- Toll Exception Table
- Authorization Codes Table
- Auto Ring Mode Table

Related Features

- Off-Hook Signaling
- Authorization Codes (Password)
- Direct Inward System Access (DISA)
- Day/Night/Timed/Scenario Ring Mode
- System Clock Set
- Outside Line Ring Assignment
- LBC (Loud Bell Control)
- Dialing Restrictions

2.12 Automatic System Daylight Saving 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 (1) hour at the DST Start time and back one(1) hour at the DST End time. The system time is sent for display to all devices and terminalsand is the basis of the system various time-based features (Auto Service Mode, Wake-up Alarm,etc.).

Operation

System

Operation of this feature is automatic.

Conditions

- ✓ The DST Start and End times are set through the Web Admin interface only.
- ✓ The interval between the DST Start and End times must be at least 7 days.

Programming

Keyset Admin.

SYSTEM

- System Set Time (PGM 178-Button 1)
- System Set Date (PGM 178-Button 2)
- DST Enable Mode (PGM 178-Button 3)
- DST Start & End Time (Web admin only)

Web Admin.

SYSTEM DATA

- System Date & Time ➤ Time, Date, DST (Daylight Saving Time) Mode, DST Start Time, DST End Time

Related Features

- Auto Service Mode Control
- Automatic Time Synchronization
- System Clock Set

2.13 Automatic Time Synchronization

Description

When enabled, the system automatically determines and sets the time of day employingNetwork Time Protocol (NTP) or ISDN time messages. When using NTP, the system, at ten(10) minute intervals, requests the time from the specified NTP time server and receives GMT(Greenwich Mean Time) time. This feature allows the System Time to synchronize with the NTP time serverautomatically. If the time deviates more than two (2) seconds, the system clock is adjusted tomatch the NTP server.

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

Operation

System

Operation of this feature is automatic.

Conditions

- ✓ NTP packets are expected over UDP port 123. Assure this port is open and available.
- ✓ A secondary NTP server address can be defined, should the first server not respond.
- ✓ The system adjusts for the local time zone assigned in the system as the Standard System Time as well as Daylight Savings Time (DST), if set.

Programming

Keyset Admin.

SYSTEM

- Network Time & Date (PGM 161-Button 12)
- Network Time & Date (PGM 195-Button 1)
- NTP Sever address (Web only)
- Standard System Time, Local Time Zone (Web only)

Web Admin.

SYSTEM DATA

- System Attributes ➤Network Time & Date
- NTP Attributes ➤Network Time & Date, NTP Primary, Server Address, NTP Secondary Sever Address, Standard Time Zone, NTP Server Service, DDoS Protector

Related Features

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

Hardware

2.14 Call Duration Warning Tone

Description

While on an outside call, you may receive a tone indicating the elapsed time of a call has reached the Warning Tone time. The station hears the tone after the CO Warning Tone Timer has elapsed.

Operation

System

Operation of this feature is automatic when assigned.

Conditions

✓ Warning tone is received 15 seconds prior to expiration of the timer and is repeated at intervals equal to the Warning Tone Timer.

Programming

Keyset Admin.

STATION

- CO Call Time Tone (PGM 112-Button 1)

SYSTEM

- Elapsed Call Tone Timer (PGM 180-Button 19)

Web Admin.

STATION DATA

- Common Attributes ➤Call Time Tone

SYSTEM DATA

- System Timers ➤Elapsed Call Timer

Related Features

- Outside Call Time Restriction

2.15 Call Forward

Description

You can have selected incoming calls re-routed to other stations (local or networked), station groups, the integrated Auto Attendant/Voice Mail, or over a system outside Line (Off Net).

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

Code 0	Remote Call Forward, forwards all calls to the station, except recalls, activated from a remote station, Call Forward, Follow-me.
Code 1	Unconditional, all calls to the station, except recalls, are forwarded internally or externally immediately upon receipt.
Code 2	Busy, if the station is busy, forwards all calls, except recalls, to the selected station.
Code 3	No Answer, forwards all calls, except recalls, to the selected station when the station does not answer within the No Answer timer.
Code 4	Busy/No Answer, forwards calls if the selected station is busy or does not answer within the No Answer timer.
Code 5	Attendant Off-Premise, forwards incoming outside calls to an outside number.
Code 6	Off-Net Unconditional, all calls to the station, except recalls, forward internally or externally, only SLT(Single Line Telephone).
Code 7	Off Net Busy, forwards all calls, except recalls, to the selected station when station is busy, only SLT(Single Line Telephone).
Code 8	Off Net No Answer, forwards all calls, except recalls, to the selected station when the station does not answer within the No Answer timer, only SLT(Single Line Telephone).
Code 9	Off Net Busy/No Answer, forwards calls if the selected station is busy or does not answer within the No Answer timer, only SLT(Single Line Telephone).

Operation

iPECS IP & LDP Phones

To activate Call Forward, (Unconditional, Busy, No Answer, or Busy/No Answer)

1) Lift the handset or press the **[SPEAKER]** button to receive dial tone.
2) Press the **[FWD]** button or Soft key. For the LIP8002, the **[DND]** button is used to activate Forward.
3) Dial desired Call Forward code (“1”~“4”).
4) Dial the station or station group to receive calls.
   Or, Dial an outside Line access code (9, 8xx, 88xx) and the desired external phone number.
   Or, Dial *587(for Station ICR).
   Or, Press the**[SPEED]** button and dial the desired bin number.
5) 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 or Soft key.

3) Dial Call Forward code “0”.
4) Dial the station’s Authorization Code (Station number and password).
5) Dial the Forward condition (“1”~“4”).
6) Dial the destination station or station group.
   Or, Dial an outside Line access code (“9”, “8xx”, “88xx”) and the desired external phone number.
   Or, Press the**[SPEED]** button and dial the desired bin number.
7) Replace the handset, return to idle.

To deactivate Call forward

1) Press flashing **[FWD]** button, Call Forward will deactivate and the **[FWD]** button LED is off.

Or, for an iPECS IP Phone.

1) Lift the handset or press the **[SPEAKER]** button to receive dial-tone.
2) Press the **[FWD]** Soft button, For the LIP-8002, the **[DND]** button is used to activate Forward.
3) Press the # dial pad button.

To activate/deactivate Call forward on LIP-9071

1) Tag a NFC card to 9071 NFC module.
2) NFC menu is displayed in 9071 display with NFC information.
3) NFC menu is chosen, NFC information is sent to system.
4) Call forward feature is activated or deactivated.

SLT(Single Line Telephone)

To activate Call Forward,(Unconditional, Busy, No Answer, or Busy/No Answer)

1) Lift the handset to receive dial tone.
2) Dial the Call Forward feature dial code “*554”.
3) Dial desired Call Forward code (“1”~“4”).
4) Dial the destination station, station group, or dial CO access code (9, 8xx, 88xx) and desired external phone number or, dial 587(for Station ICR).
5) Momentarily press the hook-switch and receive confirmation tone.
6) Replace the handset to return to idle.

To activate Call Forward to a Speed number

1) Lift the handset to receive dial tone.
2) Dial the Call Forward feature dial code “*554”.
3) Dial the Call Forward code (“6”~“9”).
4) Dial the Speed Dial bin number.
5) Replace handset to return to idle.

To activate Call Forward, Remote (Follow-me)

1) Lift the handset to receive dial tone.
2) Dial the Call Forward feature code “*554”.
3) Dial the Remote Forward code “0”.
4) Dial the station’s Authorization Code (Station and Password).
5) Dial the Forward condition (“1”~“4”).

6) Dial the destination station, station group, or dial CO access code (9, 8xx, 88xx) and desired external phone number.
7) Momentarily press the hook-switch and receive confirmation tone.
8) Replace handset return to idle.

To deactivate the Call forward

1) Lift the handset and receive stutter dial-tone.
2) Dial the Call Forward feature dial code “*554”.
3) Dial “#” to cancel Call Forward.

Conditions

- ✓ A station receiving a forwarded call can transfer the call to the forwarding station.
- ✓ A station, denied the use of Call Forward, will receive error tone in response to attempts to activate Call Forward.
- ✓ A forwarded intercom call will signal the receiving station in the Tone Signaling mode, regardless of the Intercom Signaling Mode at the station.
- ✓ You cannot forward calls to a station in DND and, if attempted, error tone is returned.
- ✓ Attempting activation of Call Forward will automatically deactivate any activate Display Text Message or DND status at your phone. Active Call Back or Queue requests are not canceled.
- ✓ When Call Forward is active, you can make outgoing calls (internal or external)but cannot activate a Call back or Queue request.
- ✓ Your private lines calls will forward if the receiving station has a direct appearance button for the Private line or if you forward to a Voice Mail group.
- ✓ For outside calls, manually activated Call Forward will override any Preset Call Forward assignment.
- ✓ You may forward calls to the integrated Voice Mail without recording a greeting, in which case, calls still forward to your Mailbox but callers receive the "No announcement recorded" message before they can leave a voice mail.
- ✓ Call Forward status is maintained in the system’s non-volatile memory for protection from power outage.
- ✓ A station in a Station Group (ACD, Circular or Terminal) can be assigned to receive incoming hunt calls, overriding any Call Forward. That is, the system recognizes the Forward condition and route shunt calls to the station based.
- ✓ Off-Net Call Forward of incoming outside calls is essentially an automated DISA call that establishes an Unsupervised Conference. Such calls are subject to the conditions of a DISA call and Unsupervised Conference, and may require entry of an Authorization Code.
- ✓ Off-Net forward calls are not answered by the system until the system completes dialing of the external call. The caller is then connected to the outgoing call.
- ✓ If the Speed Dial number used in Off-Net Call Forward contains a Flash, only digits prior to the Flash are dialed.
- ✓ An unlimited number of stations may be set-up in a Call Forward chain, forwarding calls from one station to the next. However, a station cannot forward calls to a station already a part of the chain.

- ✓ Calls to a Call Forward chain will progress as appropriate through the chain to the last station. If the last station enters DND, outside calls revert to the previous station while intercom calls receive DND tone.
- ✓ No Answer forward employs the Station No Answer Forward Timer unless it is set to zero in which case the System No Answer Timer is used.

Programming

Keyset Admin.

STATION

- Call Forward (PGM 111-Button 2)
- Station Call Forward No Answer Timer (PGM 123-Button 1)

SYSTEM

- System Call Forward No Answer Timer (PGM 181-Button 1)

STATION GROUPS

- Member Forward (PGM 191-Button 14-Circ/Term Group, 24-23-UCD/ACD Group, or 12-RingGroup)

Web Admin.

STATION DATA

- Common Attributes➤Call Forward
- Station Timer ➤Station Call Forward No Answer Timer

SYSTEM DATA

- System Timers ➤Call Forward No Answer Timer

STATION GROUP DATA

- Station Group Attributes ➤Allow Forward Member

Related Features

- Call Forward, Attendant
- Authorization Codes (Password)
- DND (Do Not Disturb)
- DND - One Time DND
- Direct Inward System Access (DISA)
- Unsupervised Conference
- Dialing Restrictions
- Station Groups
- Station Speed Dial
- System Speed Dial
- Intercom Signaling Mode
- Call Forward, Preset
- Station ICR(Individual Call Routing)

Hardware

2.16 Preset Call Forward

Description

With Preset Call Forward, your calls forward to a pre-determined destination assignedin the system database. Preset Station Call Forward can define separate treatment of outside and intercom calls. In addition, separate busy and no-answer treatments are defined.

Treatments available are:

- Internal Unconditional: Internal calls immediately forward.
- Internal Busy: Intercom calls that encounter busy, forward immediately.
- Internal No-Answer: Intercom calls that are not answered in the No-Answer time, forward. Note calls to a busy station also forward after the No-Answer time.
- Internal DND: if your phone is in the DND state, Intercom calls route to preset forward destination.
- External Unconditional: External calls immediately forward
- External Busy: external calls that encounter busy forward immediately.
- External No-Answer: external calls, not answered in the No-Answer time, forward. Note calls to a busy station also forward after the No-Answer time.
- External DND: if your phone is in the DND state, Incoming calls route to preset forward destination.

In addition, calls can directly forward to the users Voice Mail box using Preset Call Forward.

Operation

System

Operation of Preset Call Forward is automatic.

Conditions

- ✓ A station receiving a forwarded call can transfer the call to the forwarding station.
- ✓ Manual Forward has higher priority than Preset Forward and overrides any Preset Forward setting.
- ✓ Calls to a Preset Call Forward chain will progress as appropriate through the chain to the last station. If a station in manual Call Forward or DND is encountered, it is bypassed and the next station in the chain is signaled. If the last station has entered DND, outside calls revert to the previous station, signaling until answered or abandoned.
- ✓ Internal Busy or No-Answer will only operate when the internal call encounters a busy state or no answer, respectively. External Busy or External No-Answer will only operate when the external call encounters a busy state or no answer, respectively.
- ✓ Preset call forward status is not shown in the LCD of the Station.
- ✓ A station in a Station Group (ACD, Circular or Terminal) can be assigned to receive incoming hunt calls, overriding any Call Forward. That is, either the system recognizes the Forward condition and bypasses hunt calls around the station or route shunt calls to the station based on the system database.
- ✓ No-Answer forward employs the Station No-Answer Forward Timer unless it is set to zero in which case the System No-Answer Timer is used.

Programming

Keyset Admin.

STATION

- Preset Call Forward (PGM 120)
- Station Call Forward No Answer Timer (PGM 123-Button 1)
- Direct Transfer Mail Box Destination (PGM 120-Button 6)

STATION GROUPS

- Member Forward (PGM 191 Button-14-Circ/Term Group, 24-23-UCD/ACD Group, or 12-Ring Group)

SYSTEM

- System Call Forward No Answer Timer (PGM 181-Button 1)

Web Admin.

STATION DATA

- Preset Call Forward
- Station Timer ➤Station Call Forward No Answer Timer

SYSTEM DATA

- System Timers ➤Call Forward No Answer Timer

STATION GROUP DATA

- Station Group Attributes ➤Allow Forward Member

Related Features

- Call Forward
- Off-Hook Signaling
- External Auto Attendant/Voice Mail
- Integrated Auto Attendant/Voice Mail
- DND (Do Not Disturb)
- DND - One Time DND
- Private Line

2.17 Call Park

Description

You may place an active Internal & External call in a special holding location called a Park Orbit for easy accessfrom any station in the system.

There are 19 (eMG80/100) & 200 (eMG800/UCP) Park Orbits allowing multiple calls to be parked simultaneously.

Operation

iPECS IP & LDP Phones

To park an active Internal & External call

1) Press the **[TRANS/PGM]** button.
2) Dial the Park Orbit number (eMG80/100: #601~#619 / eMG800 (UCP) #601~#800).
3) Return to idle.

To retrieve a parked call

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial the Park Orbit number (eMG80/100: #601~#619 / eMG800 (UCP) #601~#800).

SLT(Single Line Telephone)

To park an active Internal & External call

1) Momentarily press the hook-switch.
2) Dial the Park Orbit number (eMG80/100: #601~#619 / eMG800 (UCP)#601~#800).
3) Return to idle.

To retrieve a parked call

1) Lift the handset.
2) Dial the Park Orbit number (eMG80/100: #601~#619 / eMG800 (UCP) #601~#800).

Conditions

- ✓ If the selected Park Orbit returns a busy signal, the user may simply dial another Park Orbit without the need to disconnect.
- ✓ 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.
- ✓ A Parked call will indicate busy at all appearances.

Programming

Keyset Admin.

SYSTEM

- Call Park Recall Timer (PGM 180-Button 2)

Web Admin.

STATION DATA

- Flexible buttons ➤ Programming (Numbering plan) ➤ Park Orbit number

SYSTEM DATA

- System Timers ➤Call Park Recall Timer

Related Features

- Hold Recall
- Attendant Recall

Hardware

2.18 Call Pick-Up

2.18.1 Directed Call Pick-Up

Description

You may answer (“Pick-Up”) incoming and transferred intercom and outside calls ringingat another station.

All ringing calls are subject to Directed Call Pick-up except Private Line calls,queue Callbacks and recalls from hold.iPECS IP and LDP Phone users may assign a Flex button as a **{Direct Call Pick-Up}** button.

Operation

iPECS IP & LDP Phones

To assign a {Direct Call Pick-Up} button

- **[TRANS/PGM]** + {FLEX} + “*77” + **[HOLD/SAVE]**

To Pick-up a call ringing at another station

1) Lift the handset or press the **[SPEAKER]**button.
2) Dial “*77”, the Directed Call Pick-up feature code.
3) Dial the ringing station’s number.

Or,

1) Lift the handset or press the**[SPEAKER]**button.
2) Press the {Direct Call Pick-Up} button.
3) Dial the ringing station’s intercom number.

SLT(Single Line Telephone)

To Pick-up a call ringing at another station

1) Lift the handset.
2) Dial “*77”, the Directed Call Pick-up code.
3) Dial the ringing station’s number.

Conditions

- ✓ To pick-up an outside call, you must have an idle appearance button available.
- ✓ When several calls are ringing at a station simultaneously, Call Pick-up will connect the oldest highest priority call (Ringing Line Preference, PGM 173).
- ✓ Queue callback and Private Line calls are not subject to Call Pick-up and such attempts receive error tone.
- ✓ Hands free announced intercom calls cannot be picked up by another station. Only ringing intercom calls are subject to Call Pick-up.

Programming

Keyset Admin.

SYSTEM

- Ringing Line Preference Priority (PGM 173)

Web Admin.

SYSTEM DATA

- RLP (Ringing Line Preference) Priority

Related Features

- Intercom Signaling Mode
- Ringing Line Preference
- Group Call Pick-Up
- Private Line

2.18.2 Group Call Pick-Up

Description

You can answer (“Pick-Up”) incoming and transferred intercom and outside calls ringing atanother station in the same Station group.

All ringing calls, except Private Line calls,queue Callbacks and recalls from hold, are subject to Pick-up by other stations in the same group.

iPECS IP and LDP Phone users may assign a Flex button as a **{Group Call Pick-Up}** button.

Operation

iPECS IP & LDP Phones

To assign a {Group Call Pick-Up} button

- **[TRANS/PGM]** + {FLEX} + “**” + **[HOLD/SAVE]**

To Pick-up a call ringing at another station

1) Lift the handset or press **[SPEAKER]**button.
2) Dial “**”, the Group Call Pick-up feature code.
3) Or, Press the**{Group Call Pick-Up}** button.

SLT(Single Line Telephone)

To Pick-up a call ringing at another station

1) Lift the handset.
2) Dial “**”, the Group Call Pick-up code.

Conditions

- ✓ To pick-up an outside call, you must have an idle appearance button available.
- ✓ When several calls are ringing simultaneously, Call Pick-up will connect the oldest highest priority call (Ringing Line Preference, PGM 173 or System Data ➤Ringing Line Preference Priority).
- ✓ Queue callback and Private Line calls are not subject to Call Pick-up and such attempts receive error tone.
- ✓ Hands free announced intercom calls cannot be picked up by another station. Only ringing intercom calls are subject to Call Pick-up.
- ✓ When a station belongs to multiple groups, calls to the group with the lowest number are answered first.

Programming

Keyset Admin.

STATION GROUPS

- Station Group(PGM 190-Button 1& 2)

SYSTEM

- Ringing Line Preference Priority (PGM 173)

Web Admin.

STATION GROUP DATA

- Station Group Assignment➤ Group Type, Pick-up Attribute

SYSTEM DATA

- RLP (Ringing Line Preference) Priority

Related Features

- Intercom Signaling Mode
- Ringing Line Preference
- Directed Call Pick-Up
- Station Groups
- Private Line

2.18.3 Pickup Group Pick-Up

Description

You can answer (“Pick-Up”) incoming and transferred intercom and outside calls ringing atanother station in the same Pick-up Group.

All ringing calls, except Private Line calls,queueCallbacks and recalls from hold are subject to Pick-up by other stations in the same Pick-up group.

iPECS IP and LDP Phone users may assign a Flex button as a **{Pickup Group Pick-Up}** button.

Operation

iPECS IP & LDP Phones

To assign a {Pickup Group Pick-Up} button

- **[TRANS/PGM]** + {FLEX} + “*588” + **[HOLD/SAVE]**

To Pick-up a call ringing at another station in the same Call Pick-up group

1) Lift the handset or press **[SPEAKER]**button.
2) Dial “*588”, the Pickup Group Pick-up feature code.
3) Or, Press the **{Pickup Group Pick-Up}** button.

SLT(Single Line Telephone)

To Pick-up a call ringing at another station in the same Call Pick-up group

1) Lift the handset.
2) Dial “*588”, the Pickup Group Pick-up code.

Conditions

- ✓ Pickup Group Pick-up applies to stations within the same Pick-up Group and does not apply to other types of Station Groups where Group Call Pick-up may be used.
- ✓ To pick-up an outside call, you must have an idle appearance button available.
- ✓ When several calls are ringing simultaneously, Call Pick-up will connect the oldest highest priority call (Ringing Line Preference, PGM 173 or System Data ➤Ringing Line Preference Priority).
- ✓ Queue callback and Private Line calls are not subject to Call Pick-up and such attempts receive error tone.
- ✓ Hands free announced intercom calls cannot be picked up by another station. Only ringing intercom calls are subject to Call Pick-up.
- ✓ When a station belongs to multiple Pick-up Groups, calls to the group with the lowest number are answered first.
- ✓ A Station must have “Pick up by Button” enabled to use a Flex button for Pickup Group Pick-up.

Programming

Keyset Admin.

STATION GROUPS

- Pick Up Group (PGM 192-Button 1)

STATION

- Pick up by Button(PGM 114-Button 20)

SYSTEM

- Ringing Line Preference Priority (PGM 173)

Web Admin.

STATION GROUP DATA

- Station Group Assignment➤ Group Type, Pick-Up Attribute

STATION DATA

- Common Attributes ➤Pick-Up by Flex Button

SYSTEM DATA

- RLP (Ringing Line Preference) Priority

Related Features

- Intercom Signaling Mode
- Ringing Line Preference
- Directed Call Pick-Up
- Pick Up Group, Private Line

Hardware

2.19 Call Re-Routing (CRR)

Description

The system can be assigned to route outside incoming calls based on the called number and the Line group receiving the call. Calls can be rerouted to an outside telephone number, a networked system or, using DISA to a Station. The reroute can be assigned for calls on a Line group or on all Line groups. The received called number is compared to entries in the database and, if a match occurs, the call is rerouted. The database called number entries may include an "*" as a wild card digit, any digit will match the wild card entry.

Operation

System

Operation of Call Re-Route is automatic once configured.

Conditions

✓ Call Re-Routing employs a Table with up to 170 entries.

Programming

SYSTEM

- CO Call Rerouting (PGM252-Web only)

2.20 Call Transfer

2.20.1 Call Transfer, Outside Line

Description

If allowed in the system database, you may transfer an outside call over an external line to another destination, establishing an Unsupervised Conference between the two external parties.

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

Operation

iPECS IP & LDP Phones

To transfer an active call with Screened Call Transfer

1) Press the **[TRANS/PGM]**button.
2) Place the outside call to the destination in normal manner.
3) At answer, announce the call.
4) Hang-up to complete the transfer.

To transfer an active call with Unscreened Call Transfer

1) Press the **[TRANS/PGM]**button.
2) Place the outside call to the destination in normal manner.
3) Hang-up to complete the transfer.

SLT(Single Line Telephone)

To transfer an active call with Screened Call Transfer

1) Momentarily depress the hook-switch.
2) Place the outside call to the destination in normal manner.
3) At answer, announce the call.
4) Hang-up to complete the transfer.

To transfer an active call with, Unscreened Call Transfer

1) Momentarily depress the hook-switch.
2) Place the outside call to the destination in normal manner.
3) Hang-up to complete the transfer.

Conditions

- ✓ For this feature, at least one of the two outside Lines (transferred or receiving) must provide detection of disconnect supervision and lost loop condition.
- ✓ Calls on Lines using digital or VoIP provide “Answer Supervision”, thus the iPECS system can provide Transfer Recall.
- ✓ If, during the transfer to an external party you press the outside Line of the original call, the outgoing call is disconnected and the original call is connected to your phone.

Programming

Keyset Admin.

STATION

- Off Net Forward (PGM 111-Button 14)

SYSTEM

- Transfer Recall Timer (PGM 180-Button 7)
- Off Net Prompt Usage (PGM 160-Button 11)
- CO to CO UC Timer Extension (PGM 160-Button 12)
- Unsupervised Conference Timer (PGM 182-Button 5)

Web Admin.

STATION DATA

- Common Attributes ➤Off-Net Forward

SYSTEM DATA

- System Attributes ➤Off-Net Prompt Usage, CO to CO Unsupervised Conference Time Extension
- System Timers ➤Unsupervised Conference Timer, Transfer Recall Timer

Related Features

- Hold Recall
- Call Transfer, Station, Unsupervised Conference

2.20.2 Call Transfer, Station

Description

After answering an outside call, you can send the call to another station in the iPECS system. Calls can be transferredannouncing the call (screened) or without an announcement (unscreened).

When a call is transferred, the Transfer Recall Timer is initiated. If the timer expires before thecall is answered, the Hold Recall process is initiated.A Flex button on iPECS IP and LDP Phones can be assigned for the DSS/BLF (Direct station Selection/Busy Lamp Field) function, which allows One-button Transfer.

Operation

iPECS IP & LDP Phones

*To assign a **{DSS/BLF}** button*

- **[TRANS/PGM]** + {FLEX} + Station Number + **[HOLD/SAVE]**

To transfer an active call with Screened Call Transfer

1) Press the**[TRANS/PGM]**button.
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,

1) Press the **{DSS/BLF}** button for the desired station.
2) At answer or splash tone, announce the call.
3) Hang-up to complete the transfer.

To transfer an active call with Unscreened Call Transfer

1) Press the**[TRANS/PGM]**button.
2) Dial the station to receive the transfer.
3) Hang-up to complete the transfer.

Or,

1) Press the **{DSS/BLF}** button for the desired station.
2) Hang-up to complete the transfer.

SLT(Single Line Telephone)

To transfer an active call with, Screened Call Transfer

1) Momentarily depress the hook-switch.
2) Dial the Station number to receive the transfer.
3) At answer or splash tone, announce the call.
4) Hang-up to complete the transfer.

To transfer an active call with Unscreened Call Transfer

1) Momentarily depress the hook-switch.
2) Dial the Station number to receive the transfer.
3) Hang-up to complete the transfer.

Conditions

- ✓ The transferring station may camp a call on to a busy station.
- ✓ The LED of a **{LOOP}** or **{POOL}** button will display the status of an outside call until the station no longer has call supervision i.e.; the call is successfully transferred.
- ✓ To prevent Toll abuse, outside Lines without an active call (either incoming or dialed digits on outgoing) cannot be transferred.
- ✓ For outgoing calls on analog Lines, the system will monitor the Line for dial tone to prevent Toll abuse.

Programming

Keyset Admin.

STATION

- No Touch Answer (PGM 111-Button 6)

SYSTEM

- Transfer Recall Timer (PGM 180-Button 7)

Web Admin.

STATION DATA

- Terminal Attributes ➤No Touch Answer

SYSTEM DATA

- System Timers ➤Transfer Recall Timer

Related Features

- Hold Recall
- Call Transfer, CO/IP
- Call Waiting/Camp-On
- Station Flexible Buttons

2.20.3 Call Transfer, System Announcement

Description

You can transfer outside calls to a System announcement allowing the system to play the announcement and, if configured, the party can then dial digits to route the call further.

After transferring to a System announcement the system establishes DISA (Direct Inward System Access) service for the outside party.

Operation

iPECS IP & LDP Phones

To transfer an active call to a System announcement

1) Press the **[TRANS/PGM]**button.
2) Dial "*55*" and the System announcement (001-200).
3) Hang-up to complete the transfer.

SLT(Single Line Telephone)

To transfer an active call to a System announcement

1) Momentarily depress the hook-switch.
2) Dial "*55*" and the System announcement number.
3) Hang-up to complete the transfer.

Conditions

✓ Only an outside call can be transferred to a System announcement.

Related Features

- DISA
- CCR
- External Auto Attendant/Voice Mail
- Integrated Voice Mail

Hardware

- iPECS IP or LDP Phone

2.20.4 Call Transfer, Voice Mail

Description

If permitted, you may transfer outside calls directly to another station's Voice Mailbox, either the integrated or anexternal Voice Mail system.

Operation

iPECS IP & LDP Phones

To transfer an active call to a user's Voice Mailbox

1) Press the**[TRANS/PGM]**button.
2) Press the**[MESSAGE/CALLBACK]** button.
3) Dial the Station number or press the **{DSS/BLF}** button for the desired station.
4) Hang-up to complete the transfer.

Conditions

- ✓ The LED of a {LOOP} or {POOL} button will display the status of a call until the station no longer has call supervision i.e.; the call is successfully transferred.

Programming

Keyset Admin.

STATION

- Direct Transfer to Mail Box Destination (PGM 120-Button 6)

SYSTEM

- Transfer Recall Timer (PGM 180-Button 7)

Web Admin.

STATION DATA

- Preset Call Forward ➤Transfer to Mail Box

SYSTEM DATA

- System Timers ➤Transfer Recall Timer

Related Features

- Hold Recall
- Call Waiting/Camp-On
- External Auto Attendant/Voice Mail
- Integrated Voice Mail

Hardware

- iPECS IP or LDP Phone

2.21 Call Waiting/Camp-On

Description

Call Waiting is used to notify a busy station that a call is waiting to be answered. The busystation is notified of the waiting call by a “Camp-On” tone. For users of an iPECS IP or LDP Phone, theLED of the **[HOLD/SAVE]** button will flash.

After receiving a busy signal, the calling station camps on to the called station.

The calledstation can respond by:

- answering the waiting call, which places the active call on hold first,
- sending a silent text or voice message,
- activating One-Time DND,
- ignoring the Camp-On tone.

Operation

iPECS IP & LDP Phones

To activate a Camp-On while receiving Intercom busy tone

1) Press the “*” button, the called and calling stations receive Camp-On tone.

SLT(Single Line Telephone)

To activate a Camp-On while receiving Intercom busy tone

1) Press the “*” button, the called and calling stations receive Camp-On tone.

Conditions

✓ The user may only Camp-On to a station in the busy mode. A user may not Camp-Onto a station in DND, in a conference, receiving a Page, etc.

✓ The Camp-On procedure is also employed by an Attendant or Secretary of an Executive/Secretary pair to activate DND Override.

✓ A Camp-On tone is sent each time the calling user presses the “*” button.

✓ You can respond to a Camp-on by activating a Pre-defined or Custom may be used to respond to a Camp-On, or a System prompt can be configured to play to the calling Station.

Related Features

- Pre-defined & Custom Text Display Messages
- DND (Do Not Disturb)
- Intercom Call (ICM Call)
- Silent Text Message
- Voice Over

2.22 Company Name Service

Description

When a user makes the outgoing call, system provides company name according to the option based on each station.

- ISDN – Facility or Display Information Element.
- SIP – From information.

Operation

Operation of the Company Name Service is automatic when configured.

Conditions

- ✓ In case of ISDN Line, The Company Name can be provided with Facility or Display IE element.
- ✓ In case of SIP line, The Company Name can be provided as From id.
- ✓ The company Name index 1~5 of station is defined by station CLI type of each CO line.
- ✓ If the company name index is existed in MSN table, system provides the company name of MSN table for MSN call.
- ✓ The company Name index 1~5 is 0, station name is provided.

Programming

Web Admin.

STATION DATA

- CLI Attributes ➤ Company Name Index 1~5

BOARD DATA

- T1/E1/PRI Attributes ➤ Caller Name Type

ISDN LINE DATA

- MSN Table ➤ Company Name Table index

TABLE DATA

- Company Name Table

Related Features

- Pre-defined & Custom Text Display Messages
- DND (Do Not Disturb)
- Intercom Call (ICM Call)
- Silent Text Message
- Voice Over

Hardware

2.23 Conference

2.23.1 Conference Group

Description

You can arrange stations and external contacts in groups so that you may create a conference with all members of the group through a single call. Each conference group can have up to 13 members(eMG80/100/eMG800) and 32 members(UCP) that can be a station or an external telephone number. Conference groups are assigned in the iPECS User Portal (Station Web Admin) by assigning members to a conference group.

Once the group is established, placing a call to the Conference Group initiates the Conference Group call. The system calls each member and a System announcement, which you pre-record, is played to members receiving the group call. Initiation of a Group call can be password protected with a 5-digit password that you assign when the group is created.

There are two (2) types of Conference Groups, Station and System. The Station Conference group is created by the station user, assigned as the group supervisor, and is the only member that can initiate a Conference group call. The System Conference Group is created by an Attendant or system administrator and any member can initiate the Conference Group call. The member initiating the call becomes the supervisor for that group call.

The supervisor can manage membership in the conference and can monitor the status of each member, in/out of conference. The Supervisor can remove members from the conference or add a non-member to the conference. An Absent Supervisor timer terminates the conference if the supervisor is not in the conference for the timer interval. Setting the interval to “0” disables this feature.

Once a group is created, the supervisor imitates the conference by calling the conference group. The system then attempts to call each member of the group either simultaneously or in turn using the Interval timer assigned for the group.

Members that answer the call receive the recorded voice announcement, if any, and can enter the conference, dial “1”, or reject the conference, hang-up. The system will attempt to contact a busy or no-answer member based on the assigned Retry Count. A call is consider no-answer using the No-answer timer set when the group was created.

Operation I

To create a conference group from the iPECS User Portal

1) Access **Station Program** in the Web Admin by entering Station number and Password and access “Station Web (old)” menu in User portal.
2) Select Conference Group button.
3) Create the Conference group entering
   - The name of group: up to 12 character
   - The password: Passwords must be 5-digits
   - Announcement number for the group (pre-recording the announcement is required)
   - The Absent Supervisor timer
   - The No answer timer

- The busy/no-answer Retry count
- The Interval time: Each member is called in turn at expiration of the interval timer

4) Set the Member Attributes by entering:
   - Index number
   - Type: Station, Individual CO, CO group, loop or Transit-out code (access loop for networking call)
   - The CO value: the supervisor can select a specific CO line or CO group for used to call external members
   - Dial number: The station or external party number
5) Press the Save button to store the Conference Group configuration.

User can join conference group according to below procedure if the group is made by PC UCS client (it is called as UCS conference room). The UCS conference room information is cleared when system is restarted.

System conference group 100~119 could be used for UCS conference room.

To activate conference group

1) Press **[PGM]** + “552”.
2) Dial system conference group number.
3) Dial password of the system conference group.

To deactivate conference group

1) Press **[PGM]** + “553”.
2) Dial system conference group number.
3) Dial password of the system conference group.

To join system conference group

1) Dial “Room Type Conf Group Join (“5*0”)” code.
2) Dial system conference group number.
3) Dial password of the system conference group.

Note) Conference group number range

- ✓ System conference group number: eMG80/100: 100~139, eMG800, UCP: 100~259
- ✓ Station conference group number: eMG80/100: 00~19, eMG800, UCP: 00~99

Operation II

IPECS phone & LDP Phones

To assign a {Conference Group Monitor} button

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “551” + **[HOLD/SAVE]**

To initiate a Conference Group call

1) Lift the handset.
2) Dial “68”xxx, the Conference Group code and group number (xxx).
3) Enter the Conference group password, if assigned, members receive the Conference Group call.

To enter a Conference Group call

1) Answer the ringing call and receive recorded announcement, if assigned and recorded.

2) Dial “1” to enter the conference or hang-up to reject the conference. For iPECS UCS clients, use Accept or Deny when notified of the conference.

To monitor conference member status or delete members while in the conference

1) Press the Monitor flex button, the state of the first member is displayed.
2) Press **[VOL UP/VOL DOWN]** button to display the state of other members.
3) Press the **[DND]** button to delete a participant from the conference.

To monitor the Group member conference status while Not in the conference

1) Press the **[TRANS/PGM]** button.
2) Dial “551”, the Monitor Conference Group code.
3) Press **[VOL UP/VOL DOWN]** button to display the state of other members.

To add a non-member to the Group

1) Press the **[HOLD/SAVE]** button to place the conference on hold.
2) Call the desired party.
3) Press the **[CONF]** button.

Conditions

- ✓ The conditions associated with Multi-Party Voice Conference apply.
- ✓ Members using the iPECS UCS client Soft phone may view the state of other members in the conference.
- ✓ A maximum of 20 (eMG80/100) and 100 (eMG800/UCP) Station Conference Groups can be created by any system user and 40 (eMG80/100) and 160 (eMG800/UCP) System Conference Groups can be created in iPECS system.
- ✓ The user must have access to the Station Web Admin with the proper password (Authentication code).
- ✓ If there are insufficient Conference channels available when initiating the conference group call, the initiator receives error tone.
- ✓ Two-way record can be used to record the Conference Group call.
- ✓ eMG80/100/800 can’t support Conference recording.
- ✓ For iPECS UCP, MCIM is required to support 32 party conferences. With the MCIU, the number of parties in the conference cannot exceed the number of MCIU channels . 2 channels of each MCIU/MCIM is reserved for link with other MCIU/MCIM.

Related Features

- Multi-Party Voice Conference
- Conference Room
- System Admin Programming

Hardware

2.23.2 Conference Member View & Delete

Description

When you establish a Multi-Party conference, you are the Conference supervisor. As the Supervisor, you can view, add, and remove parties from the conference. An iPECS IP or LDP Phone with 3-Soft buttons is required.

Operation

IPECS phone & LDP Phones with 3-Soft buttons

To view and modify the Conference members, while in the conference

1) Select the **{VIEW}** Soft button, the LCD displays the first Conference Member.

MEMBER 01		
STA 1001		
BACK	ADD	DELETE

2) Use the **[VOL UP]**/**[VOL DOWN]** buttons to scroll the list of members, use the **[ADD]** button to add a member to the Conference, **[DELETE]** removes the member from the Conference. The **[BACK]** button returns the display to the normal Conference display.

Conditions

- ✓ Only supervisor of Multi-Party conference can use this feature.
- ✓ Only the iPECS IP and LDP phones with 3 soft buttons can use this feature.

Related Features

- ■ Multi-Party Voice Conference

Hardware

- ■ IPECS IP or LDP Phone models w/3-Soft buttons

2.23.3 Conference Room

Description

In addition to ad-hoc Multi-party conferencing, you may open a Conference Room. Other internaland external parties are invited to the conference and can join the conference without furtheraction by you, the Conference Room Supervisor. A Conference Room can be password protected so that only parties thatenter the password are allowed to join the Room.As parties enter the Conference Room a tone is provided notifying members that a party has entered the conference.Conference room can be activated up to 200 (100 for eMG80/100) when conference room expansion feature is enabled.

In case of eMG, a maximum of 13 parties can join a Conference Room and three such conferences can be active simultaneously.

In case of UCP, up to 32 parties with the g.711 or g.729 codec, or 24 parties with the g.722 or g.723 codec can be join a Conference Room. Conference Rooms employchannels from an MCIM (Multi-party Conference Interface Module) or the MCIU built-in the iPECS UCP100 and UCP600.

A system Attendant can view the list of participants in an active Conference Room and close any Conference Room.

Operation

Attendant Phone

To view Room participant list

1) Press the **[TRANS/PGM]** button.
2) Dial “054”.
3) Dial Room number.

To delete a Conference Room

1) Press the **[TRANS/PGM]** button.
2) Dial “055”.
3) Dial Room number.

iPECS IP & LDP Phones

To open a Conference Room

1) Press the **[TRANS/PGM]** button.
2) Dial “53” to create a Conference Room.
3) Dial the desired Conference Room number.
4) If desired, enter a password (exactly 5 digits) for the Conference Room.
5) Press **[HOLD/SAVE]**button to open the Room.

To join a Conference Room

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial “59”, the Conference Room entry code.
3) Dial the Conference Room Number.
4) Dial the Conference Room password, if required.

To close a Conference Room

1) Press the **[TRANS/PGM]** button.
2) Dial “54”, the close Conference Room code.
3) Dial the Conference Room number.
4) Dial the Conference Room password.
5) Press **[HOLD/SAVE]** to delete the Conference Room.

To transfer an active call to a Conference Room

1) Press the **[TRANS/PGM]** button.
2) Dial “59”, the Conference Room entry code.
3) Dial the Conference Room Number.
4) Dial the Conference Room password.
5) Hang-up to complete the transfer.

SLT(Single Line Telephone)

To open a Conference Room

1) Lift the handset.
2) Dial “*561”, SLT(Single Line Telephone) Programming code.
3) Dial “53”, the Create Conference Room code.
4) Dial the desired Conference Room number.
5) Dial the Conference Room password.
6) Momentarily press the hook-switch.

To join a Conference Room

1) Lift the handset.
2) Dial “59”, the Conference Room entry code.
3) Dial the Conference Room Number.
4) Dial the Conference Room password.

To close a Conference Room

1) Lift the handset.
2) Dial “*561”, SLT(Single Line Telephone) Programming code.
3) Dial “54”, the close Conference Room code.
4) Dial the Conference Room number.
5) Dial the Conference Room password.
6) Momentarily press the hook-switch, receive confirmation tone.

Conditions

- ✓ Once established, a Conference Room will remain opened until the Room is closed.
- ✓ iPECS UCS Client Soft phones may also create, delete and join a Conference Room. For operation, refer to the iPECS UCS Client User Guide.
- ✓ In case of UCP, an MCIM is required to support 32 party conferences. With the MCIU, the number of parties in the conference cannot exceed the number of MCIU channels. Refer to the iPECS UCP

Hardware Description and Installation Manual for MCIU capacities. Note the capacity of the MCIU is reduced by approximately 1/3 when transcoding of complex Codes is needed.

- ✓ You must set the Open Loop Detect Timer (PGM 141-7th) when using 'transferring the conference room' through Analog CO Line.

Related Features

- Multi-Party Conference
- Automatic Speaker Select
- Hold Recall

2.23.4 Multi-Party Conference

Description

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

In case of eMG, you can join a maximum of thirteen (13) parties can be in a conference with a maximum of two such conferences simultaneously. There is no limit on the number of 3-party conferences.

In case of UCP, an unlimited number of 3-party conferences can be established by iPECS Phones. Inaddition, up to 32 parties with the g.711or g.729 codec and 24 parties with the g.722 or g.723 codec can be connected to a single voice conference using MCIM (Multi-party Conference Interface Module).

The MCIM of UCP will support any combination of parties and conferences to the maximum totalnumber of parties in conference. Multiple MCIMs are installed to supportmultiple multi-party conferences with a maximum of 32 parties in any single conference.

You can place a Conference on hold, allowing you to accomplish other tasks while permitting the other parties to continue to converse.

Operation

iPECS IP & LDP Phones

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 you receive 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.
6) Press the**[CONF]** button to establish conference.

To place a conference on hold

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

To retrieve held conference

1) Lift handset and Press the flashing**[CONF]** button, all parties reconnected.

To drop held conference

1) Press the flashing **[CONF]** button on idle, all parties disconnected.

SLT(Single Line Telephone)

To establish an ad-hoc conference

1) Establish the first call.
2) Momentarily press the hook-switch, the connected party is placed on exclusive hold and you receive dial tone.
3) Place the second call.

4) When completed adding parties, momentarily press hook-switch twice in 2 seconds to connect all parties.

Conditions

- ✓ The **[CONF]** button remains illuminated at the initiators phone for the duration of the conference.
- ✓ If the system receives a disconnect signal and no internal parties remain in the conference, the conference is terminated and all parties are disconnected. If an internal party is still connected when a disconnect signal is received, the connection to remaining parties is maintained.
- ✓ The normal Hold Recall process is applied to a conference on hold using the Unsupervised Conference recall Timer for recall timing.
- ✓ If while setting up a conference, system error tone is received, the initiator must press the **[CONF]** button to regain Intercom dial tone; a SLT(Single Line Telephone) user must hook-flash.
- ✓ A station that is busy, in DND or other non-idle state cannot be added to a conference.
- ✓ ASLT(Single Line Telephone) user can establish a 3-party conference only, but can be a member of other multi-party conferences.
- ✓ In case of UCP, an MCIM is required to support 32-party conferences. With the MCIU, the number of parties in the conference cannot exceed the number of MCIU channels. Refer to the iPECS UCP Hardware Description and Installation Manual for MCIU capacities. Note the capacity of the MCIU is reduced by approximately 1/3 when transcoding of complex Codecs is needed.

Related Features

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

Hardware

- ■ MCIU or MCIM for more than 3-party conference for UCP

2.23.5 Multi-Barge In Room

Description

Multiple Barge in Room permits authorized extensions to intrude into other existing outside or intercom calls. When you intrude on an existing call, a Multiple conference is established with you (the supervisor) and the parties on the existing call.

There are two modes for multiple barge in and it can be selected by intercom busy service

- (0) – Voice over
- (1) – Intrusion
- (2) – Barge in Monitor, the intruding station can listen only.
- (3) – Barge in Conference, the intruding station hears the conversation and may join and may join the conversation to speak to the other parties.

Multi Barge In room can be activated up to 50 (20 for eMG80/100) at the same time. In case of eMG, a maximum of 13 parties can join a Multi Barge In Room.

In case of UCP, up to 32 parties with the g.711 or g.729 codec, or 24 parties with the g.722 or g.723 codec can be join a Barge In Room. Multi Barge In Rooms employ channels from an MCIM (Multi-party Conference Interface Module) or the MCIU built-in the iPECS UCP100 and UCP600.

Operation

iPECS IP & LDP Phones

Barge in Monitor

1) Call the busy station and receive busy tone or queuing tone
2) Dial digit #.
3) You are connected to the call in a “listen-only”, **[MUTE]** button is activated.

Barge In Conference

1) Call the busy station and receive busy tone or queuing tone
2) Dial digit #.
3) You are connected to the call.

To exit Barge In

1) Hang-up the phone.

SLT(Single Line Telephone) & SIP Phones

Barge In Conference

1) Call the busy station and receive busy tone or queuing tone
2) Dial digit #.
3) You are connected to the call.

To exit Barge In

1) Hang-up the phone.

Conditions

- ✓ Multiple Barge In Monitor is only supported for iPECS IP and LDP Phone models
- ✓ This feature is only available for active calls; you cannot barge in on a ringing or held call.
- ✓ If 'Privacy' option is ON, then any person could not barge-in to the station.
- ✓ The override option should be enabled for barge in supervisor.
- ✓ In case of UCP, an MCIM is required to support 32-party conferences. With the MCIU, the number of parties in the conference cannot exceed the number of MCIU channels. Refer to the iPECS UCP Hardware Description and Installation Manual for MCIU capacities. Note the capacity of the MCIU is reduced by approximately 1/3 when transcoding of complex Codecs is needed.

Related Features

- ■ Conference room
- ■ Intercom busy service

Hardware

- ■ MCIU or MCIM for more than 3-party conference for UCP

2.23.6 Unsupervised Conference

Description

Using your iPECS IP or LDP Phone, you may establish a conference with external parties and exit the conferencewhile allowing the external parties to converse privately without the system supervising the connection through your phone.In this case, the system monitors the outside Lines to determine if either party disconnects. Also, the Unsupervised Conference timer is initiated when an Unsupervised Conference is established. Fifteen seconds priorto expiration of the timer, a Warning Tone is provided and at expiration, the call disconnects.

If enabled, either party in an Unsupervised Conference can extendthe UnsupervisedConference beyond the timer expiration. The party enters the Timer Extension feature code and a digit"1" to "9" indicating the Timer extension multiplier.

The system extends the timer based onthe dialed digit multiple of the Timer. For example, if the Unsupervised Conference timer is 5minutes and the user dials the digit 4, the timer will extend to 20 minutes (4 multiplied by 5minutes).

Operation

iPECS IP & LDP Phones

To set up an Unsupervised conference

1) Establish a conference with external parties as normal.
2) Press the **[CONF]** button. The **[CONF]** button LED flashes to indicate the Unsupervised Conference state.

To reenter the Unsupervised conference

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

Conference Party

To extend the Unsupervised Conference Time from a connected party

1) Dial “# #”.
2) Dial the Timer extension multiplier (1~9).

Conditions

- ✓ The Unsupervised Conference Timer also applies to an external call placed by to a DISA line in the system.
- ✓ An Unsupervised Conference will be terminated if the system receives a disconnect signal or the Unsupervised Conference Timer expires.
- ✓ An Unsupervised Conference will not recall the user.

Programming

Keyset Admin.

NUMBERING

- Unsupervised Conference Timer Extension (PGM 109-Button3)

STATION

- Off Net Forward (PGM 111-Button 14)

SYSTEM

- CO to CO UC Time Extension (PGM 160-Button 12)
- Unsupervised Conference Timer (PGM 182-Button 5)

Web Admin.

SYSTEM ID &NUMBERING PLANS

- Flexible Numbering Plan ➤Unsupervised Conference Timer Extension Code

STATION DATA

- Common Attributes ➤Off-Net Forward

SYSTEM DATA

- System Attributes ➤ CO to CO Unsupervised Conference Time Extension
- System Timers ➤Unsupervised Conference Timer

Related Features

- Direct Inward System Access (DISA)
- Multi-Party Voice Conference

Hardware

- iPECS IP or LDP Phone

2.24 Customer Site Name

Description

A Name, up to 24 characters, may be entered into the system database as the Site name. The Site Name, which is displayed on the SMDR and database outputs as well as during an Admin session, is commonly used as a site identifier.

Operation

System

Operation of this feature is automatic when a name is assigned.

Programming

Keyset Admin.

SYSTEM ID

- Site Name (PGM 100-Button 2)

Web Admin.

SYSTEM ID & NUMBERING PLANS

- System ID ➤Site Name

2.25 Data Line Security

Description

Data from ananalog Modem or FAX transmitted over analog Lines is subject to distortion and errors if system tones suchas Camp-On and Override are applied during transmission. To eliminate such errors, stationsthat use analog data (modems or Fax) can be assigned to block incoming system tones.

Operation

System

System tones are automatically blocked when Data Line Security is assigned.

Conditions

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

Programming

Keyset Admin.

STATION

- Data Line Security (PGM 111-Button 4)

Web Admin.

STATION DATA

- Terminal Attributes ➤Data Security

Related Features

- ■ Call Waiting/Camp-On
- ■ DND Override
- ■ Intrusion

Hardware

2.26 Delayed Line Ring

Description

Ring signals for an incoming call on a “Normal” type outside Linecan be sent to Stations immediately upon detection orafter an assigned ring cycle delay. The delay can be up to 9 system ring cycles, thus allowingother Stations to answer the call.

Note a “NormalLine is a standard non-DID/DISA line which use other routing techniques to determine the ring assignment.

Operation

System

Delay Ring operation is automatic when assigned.

Conditions

- ✓ Outside Lines must be assigned an immediate ring destination. As a default all incoming outside calls ring to the Attendant Station 100 and, if no immediate destination is assigned, the Attendant will receive ring for the call.
- ✓ Delay Ring can be assigned for a Station or Station Group.
- ✓ If no (Zero) delay is entered when programming Ring assignments, the Station will receive immediate ring.
- ✓ Private Lines may be assigned with delayed ring.
- ✓ If no Station or Station Group is assigned for immediate ring, the call will ring immediately at the first available Attendant.

Programming

Keyset Admin.

CO/IP

- CO Ring Assignment (PGM 144)

Web Admin.

CO LINE DATA

- CO/IP Ring Assignment

Related Features

- Private Line
- Station Groups
- Line Ring Assignment

Hardware

2.27 Delayed Auto Attendant

2.27.1 DISA Incoming Line-Russia-

Description

If the Country Code for the system is assigned as Russia, an incoming DISA call will be answered after the DISA Answer Timer expires. After the DISA Delay Timer expires, the caller is connected to the System Auto Attendant announcement number '001'.

Operation

System

Operation of this feature is automatic when assigned.

Conditions

✓ The DISA Answer and Delay Timers are only applicable when the Country Code is set to Russia.

Programming

Keyset Admin.

CO/IP

- DISA Answer Timer (PGM 142-Button 24)
- DISA Delay Timer (PGM 142-Button 22)

Web Admin.

CO LINE DATA

- Common Attributes ➤DISA Answer Timer, DISA Delay Timer

Related Features

- DISA Service Attribute
- DID Service Attribute

Hardware

2.27.2 Normal Line Ring Assignment

Description

An incoming outside call on a “Normal” outside Line can ring to the integrated Auto Attendant either immediatelyupon detection or after a delay of up to 30 seconds.

This allows other stations assignedimmediate ring the opportunity to answer before the call is routed to the Auto Attendant.

Once routed to the Auto Attendant the pre-recorded system announcement plays and the routing associated with the announcement will apply.

Note a “Normal” outsideLine is a standard non-DID/DISA line that employs “Ring Assignment” to route the call.

Operation

System

Operation of this feature is automatic when assigned.

Conditions

- ✓ When Delayed Auto Attendant Ring is assigned, after the delay, the call will no longer ring the assigned stations and will only ring to the Auto Attendant.
- ✓ If no delay is entered, the call will ring to the Auto Attendant immediately.
- ✓ To assign Delayed Auto Attendant ring, at least one Station or Station Group must be assigned immediate ring.
- ✓ Ring is assigned to station and AA ring time is assigned. The call is delivered to the station first after the AA ring time. The station ring is stopped and call is delivered to VSF announcement.

Programming

Keyset Admin.

CO/IP

- CO Ring Assignment (PGM 144)

Web Admin.

CO LINE DATA

- CO/IP Ring Assignment

Related Features

- Line Ring Assignment

Hardware

2.28 Diagnostics & Maintenance

Description

The system software incorporates various diagnostic and maintenance routines that may be“called” remotely or locally through the systems RS-232 serial ports, a TCP/IP connection usinga Web browser established over IP networks or a PPP connection over ISDN. Routines thatcan be accessed include trace functions at the device level, commands for diagnostics, database and software upload and download, andmaintenance, and tools for manipulation at the OS level.

2.29 Dial-by-Name (Directory Search & Dial)

Description

A name, up to 16 characters, may be assigned to each Station and System Speed dial. Inaddition, each station may be assigned a 20-character name. When Names are assigned, if using an iPECS IP or LDP phone,you mayplace an intercom call to another station or select a Station or System Speed dial using theName.

You select from one of three Dial-by-Name directories and enter characters employing2 dial pad buttons for each character. The system finds and displays the nearest match to your entries. You may continue entering characters or scroll the directory at any pointusing the **[VOL UP]/[VOL DOWN]** button and selecting a name to call.

Operation

iPECS IP & LDP Phones

*To use Dial by Name using a **[SPEED]** button*

1) Press the **[SPEED]** button twice or press the **[LIST]** soft button.
2) Choose the desired directory.
3) Search the directory using the **[VOL UP]/[VOL DOWN]** button or **[NAVI UP]/[NAVI DOWN]**button, or by entering characters.
4) Press the **[HOLD/SAVE]** button or press **[SEND]** soft button to place the call.

To toggle between the name and number display

1) Press the **[TRANS/PGM]** button or press **[NAME/TEL]** soft button.

To program your Station User Name

1) Press the **[TRANS/PGM]** button.
2) Dial “071” at Attendant station and enter desired station number, or Dial “74” at each station or the User name can be programmed via Web Admin – Station Data – Station Name Display.
3) Enter the name, up to 20 characters; refer to Station Speed Dial for The Character Entry Chart.
4) Press **[HOLD/SAVE]**.

*To search name through **[Directory]** button*

1) Press the **[Directory]** button or **[DIR]** soft button.

2) Enter the desired name for searching.
3) Press **[Search]** soft button or **[OK]** button.
4) Matched name is displayed on LCD.
5) The matched names can be scrolled by using the navigation key or the volume button.
6) To display the number of the searched name, press **[DETAIL]** soft button.
7) Press the **[HOLD/SAVE]** button or press **[SEND]** soft button to place the call.

Conditions

- ✓ Available characters are A to Z, space and period.
- ✓ The LCD will display multiple names, one per LCD line up to 20 characters.
- ✓ If you select a directory with no entries or there is no match to the entry, the "No Entries" message is displayed and error tone is provided.
- ✓ Dial-by-Name is only available to iPECS IP and LDP Phones with a display. Other users will receive error tone if an attempt is made to access Dial-by-Name.
- ✓ You may both scroll and enter characters to search a directory.
- ✓ To search name with [DIR] button, minimum 3 characters should be entered.
- ✓ When a name 'ABC' is searched, maximum 40 names that is matched can be searched & displayed.
- ✓ Maximum 50 keysets in eMG800/UCP (20 keysets in eMG80/100) can be the searched at the same time.
- ✓ The search is possible only from the first character of a name.
  Ex) if the user searches "bcd", the "bcdmfg" is searched. But the "kmbcda" is not searched.
- ✓ In Directory search, if LDAP is not set in system, minimum search character is 1 and if LDAP is set in system, minimum search characters are 2.

Programming

Keyset Admin.

STATION

- System Speed Dial Access (PGM 106-Button 16)

Web Admin.

STATION DATA

- Common Attributes➤Speed Dial Access

SYSTEM DATA

- System Attributes ➤Dial Back to Caller from Remote VM Access
- System Timers ➤Call log/Directory Auto Idle Timer

Related Features

- Station Speed Dial
- System Speed Dial

Hardware

- iPECS IP or LDP Phone models with Display

2.30 Dial Pulse to Tone Switchover

Description

On a pulse dial outside Line, you can request the system to change the signaling mode frompulse to DTMF. This allows you to access outside facilities that require DTMF signals suchas banking services, Voice mail, etc.

Operation

To switch from pulse to DTMF while on a pulse Line

- ✓ Dial "*", the signaling changes to DTMF.

Conditions

- ✓ This command is only recognized for analog pulse dial CO lines.
- ✓ Dial pulse to tone switchover is not available in the Redial features.
- ✓ When the system encounters an "*" in a Speed Dial digit string, dialing will pause then the remaining digit are sent using DTMF signaling.
- ✓ For VoIP calls, pulse dialing is not available thus; switchover is not required or supported.

Programming

Keyset Admin.

CO/IP

- CO Line Type (PGM 141-Button 4)
- CO Line Signal Type (PGM 141-Button 5)

Web Admin.

CO LINE DATA

- Common Attributes ➤CO Line Type
- Analog Attributes ➤CO Line Signal

SYSTEM DATA

- System Attributes ➤First digit * in SPD

Related Features

- Speed Dial
- Display Security

2.31 Dialing Restrictions

2.31.1 Class of Service

Description

Dialing privileges can be assigned for each station, DISA line and Authorization Code. Dialing privileges are assigned in “Allow” and “Deny” Exception tables that are associated with a COS.

Thedialing privileges are the result of the interaction of the Station and CO Class of Service (COS)assignments and Exception Tables as shown in the charts below.

Users placing an outgoing call or dialing afteranswering a call will be allowed the dialing privileges assigned.

Station/DISA/Code Class of Service – There are eleven (11) Station Class of Serviceassignments, which also apply to DISA and Authorization Code users based on the assignedCOS (Class of Service).

Station COS	Dialing Restriction
1	No restrictions are placed on dialing.
2	Assignments in Exception Table A are monitored for allow and deny numbers.
3	Assignments in Exception Table B are monitored for allow and deny numbers.
4	Assignments in Exception Tables A & B are monitored for allow and deny numbers.
5	The leading digit cannot be a Long Distance code and assignments in Exception Table C apply.
6	Number of digits cannot exceed LD digit count and assignments in Exception Table C apply.
7	Intercom and Emergency number calls are allowed. Incoming and transferred calls are allowed.
8	Assignments in Exception Table D are monitored for allow and deny numbers.
9	Assignments in Exception Table E are monitored for allow and deny numbers.
10	The assignments in the Exception Table D & E are monitored for allow and deny numbers.
11	The assignments in the Exception Table A & B and D & E are monitored for allow and deny numbers.

CO Class of Service – There are five (5) CO line COS (Class of Service) assignments.

Station COS	Dialing Restriction
1	Dialing privileges are governed by the Station COS only.
2	Assignments in Exception Table A govern dialing privileges for Station COS 2 & 4.
3	Assignments in Exception Table B govern dialing privileges for Station COS 3 & 5.
4	No LD calls (LD code & digit count) and assignments of Exception C apply to Station COS 2 & 6 only.
5	No dialing restrictions applied, overrides Station COS 2 through 6.

Station and CO COS (Class of Service) Interaction – The following chart displays the dialing capabilities based on theinteraction between the Station COS (Class of Service) and CO line COS (Class of Service) assignments.

Station COS	Restriction				
	CO COS 1	CO COS 2	CO COS 3	CO COS 4	CO COS 5
1	No restriction	No restriction	No restriction	No LD calls & Table C	No restriction
2	Exception Table A	Exception Table A	No restriction	No LD calls & Table C	No restriction
3	Exception Table B	No restriction	Exception Table B	No LD calls & Table C	No restriction
4	Exception Table A & B	Exception Table A	Exception Table B	No LD calls & Table C	No restriction
5	Local call & Table C	Local call & Table C	Local call & Table C	No LD calls & Table C	No restriction
6	Local call & Table C	Local call & Table C	Local call & Table C	No LD calls & Table C	No restriction
7	Intercom only	Intercom only	Intercom only	Intercom only	Intercom only
8	Exception Table D	Exception Table D	No restriction	No LD calls & Table C	No restriction
9	Exception Table E	Exception Table E	No restriction	No LD calls & Table C	No restriction
10	Exception Table D & E	Exception Table D & E	No restriction	No LD calls & Table C	No restriction
11	Exception Table A, B, D & E	Exception Table A, B, D & E	No restriction	No LD calls & Table C	No restriction

PBX Dialing Codes – Four (4) one or two-digit PBX Trunk Access Codes can be defined in thesystem database. When dialed as the first digit(s), these codes signal the system to apply theappropriate COS (Class of Service). If not dialed, the call is treated as an internal PBX call and dialing is notrestricted.

Exception Tables – Each Exception Table permits entry of 50 Allow codes and 50 Deny codes.Each code can contain up to 20 digits including digits 0-9, “#” as a wild card (any digit) and “*”as the end of entry mark. Reference the previous charts for application of the Exception Tables.

Exception Table process – As digits are dialed, they are compared to entries in the appropriateException Table. Based on the Allow and Deny entries, the system applies the following rulesto allow or deny the call.

Rule 1	If a table has no entries, no restrictions are applied.
Rule 2	If there are only Deny entries, restrictions are provided as Deny only.
Rule 3	If there are only Allow entries, restrictions are provided as Allow only.
Rule 4	If there are both Allow and Deny entries, the Deny entries are searched. If the dialed number matches a Deny entry, and is not part of an allow entry the call is restricted; if no match is found the call is allowed. For example, if 2223 is assigned in the Allow Table and 222 is assigned in the Deny Table numbers starting with 222 are denied except for 2223.

Operation

System

The system automatically applies the assigned COS (Class of Service)

Conditions

- ✓ The dialing privileges are based on the CO and Station COS (Class of Service).
- ✓ When you enter an Authorization code the COS associated with the code may be selected for the call.

Programming

Keyset Admin.

STATION

- Station COS (Class of Service) (PGM 116)

CO/IP

- CO Line COS (Class of Service) (PGM 141-Button 2)

SYSTEM

- DISA COS (Class of Service) (PGM 166)
- PABX Access Codes (PGM 172)
- LD Call Digit Count (PGM 177-Button 4)
- LD Digit Code (PGM 177-Button 19)

TABLES

- Toll Restriction Tables (PGM 224)

Web Admin.

STATION DATA

- Station COS (Class of Service)

CO LINE DATA

- Common Attributes ➤CO Line COS

SYSTEM DATA

- DISA COS (Class of Service)
- PBX Access Codes
- SMDR Attributes ➤Long Distance Call Digit Counter, Long Distance Call Code

TABLES DATA

- Toll Exception Tables

Related Features

- ■ Day/Timed & Night Station COS (Class of Service)
- ■ Direct Inward System Access (DISA)
- ■ Temporary Station COS (Class of Service)/Lock
- ■ Walking COS (Class of Service)

Hardware

2.31.2 Day/Timed & Night Station COS (Class of Service)

Description

Each station, DISA line and Authorization code is assigned a COS (Class of Service) for two modes: Day, whichincludes Timed, and Night service modes. The service mode is generally controlled by theSystem Attendant or automatically and, based on the mode, appropriate dialing privileges are established.

Operation

System

Dialing restrictions are automatically applied based on COS (Class of Service) assignments.

Programming

Keyset Admin.

STATION

- Station COS (Class of Service) (PGM 116)

CO/IP

- CO Line COS (Class of Service) (PGM 141-Button 2)

SYSTEM

- DISA COS (Class of Service) (PGM 166)
- PABX Access Codes (PGM 172)
- LD Call Digit Count (PGM 177-Button 4)
- LD Digit Code (PGM 177-Button 19)

TABLES

- Toll Exception Tables (PGM 224)
- Authorization Codes Table (PGM 227)

Web Admin.

STATION DATA

- Station COS (Class of Service)

CO LINE DATA

- Common Attributes ➤ CO Line COS (Class of Service)

SYSTEM DATA

- DISA COS (Class of Service)
- PBX Access Codes
- SMDR Attributes ➤ Long Distance Call Digit Counter, Long Distance Call Code

TABLES DATA

- Toll Exception Tables

Related Features

- Authorization Codes (Password)
- Class of Service
- Direct Inward System Access (DISA)

- Temporary Station COS (Class of Service)/Lock
- Walking COS (Class of Service)
- Auto Service Mode Control
- Day/Night/Timed/Scenario Ring Mode

Hardware

2.31.3 Temporary Station COS (Class of Service)/Lock

Description

You or an Attendant can change the Station's Class of Service to COS (Class of Service) 7, temporarilypreventing unauthorized outside calls from the station, i.e. "lock the station".

The Station can stillplace internal calls and Emergency number calls.

Operation

iPECS IP & LDP Phones

To activate Temporary COS (Class of Service)

1) Press the **[TRANS/PGM]** button.
2) Dial "21", the Temp COS (Class of Service) code.
3) Press the **[HOLD/SAVE]** button.

To restore the assigned COS (Class of Service)

1) Press the **[TRANS/PGM]** button.
2) Dial "22", the restore COS (Class of Service) code.
3) Dial Authorization Code (station number and code).
4) Press the **[HOLD/SAVE]** button.

SLT(Single Line Telephone)

To activate Temporary COS (Class of Service)

1) Momentarily press the hook-switch.
2) Dial "*561", the SLT(Single Line Telephone) Programming code.
3) Dial "21", the Temp COS (Class of Service) code.
4) Momentarily press the hook-switch.

To restore the assigned COS (Class of Service)

1) Momentarily press the hook-switch.
2) Dial "*561", the SLT(Single Line Telephone) Programming code.
3) Dial "22", the restore COS (Class of Service) code.
4) Dial Authorization Code (station number and code).
5) Momentarily press the hook-switch.

System Attendant

To activate Temporary COS (Class of Service)

1) Press the **[TRANS/PGM]** button.
2) Dial "021", the Temp COS (Class of Service) code.
3) Dial the Station range. If only one Station is to be assigned Temp COS, dial the Station number twice.
4) Press the **[HOLD/SAVE]** button.

To restore the assigned COS (Class of Service)

1) Press the **[TRANS/PGM]** button.
2) Dial "022", the restore COS (Class of Service) code.
3) Dial the Station range. If only one Station is to be assigned Temp COS, dial that Station number twice.
4) Press the **[HOLD/SAVE]** button.

Conditions

✓ When the COS of a Station is restored the Station COS (Class of Service)is changed to the COS assigned for the active service mode, Day, Night, or Timed.

Programming

Keyset Admin.

STATION

- Station COS (Class of Service) (PGM 116)

CO/IP

- CO Line COS (Class of Service) (PGM 141-Button 2)

SYSTEM

- DISA COS (Class of Service) (PGM 166)
- PABX Access Codes (PGM 172)
- LD Call Digit Count (PGM 177-Button 4)
- LD Digit Code (PGM 177-Button 19)

TABLES

- Toll Restriction Tables (PGM 224)
- Authorization Codes Table (PGM 227)

Web Admin.

STATION DATA

- Station COS (Class of Service)

CO LINE DATA

- Common Attributes ➤CO Line COS (Class of Service)

SYSTEM DATA

- DISA COS (Class of Service)
- PBX Access Codes
- SMDR Attributes ➤Long Distance Call Digit Counter, Long Distance Call Code

TABLES DATA

- Toll Exception Tables
- Station/System Authorization Codes Table

Related Features

- Class of Service
- Walking COS (Class of Service)

- Auto Service Mode Control
- Day/Night/Timed/Scenario Ring Mode
- Authorization Codes (Password)

2.31.4 Walking COS (Class of Service)

Description

You may temporarily override the toll restriction of a Station to make outside toll calls from a normallytoll restricted Station.

You must input an Authorization Code in order to activate WalkingCOS (Class of Service) and are subject to the assigned COS (Class of Service) of the Authorization Code.

The COS (Class of Service) associated withthe Authorization Code is applied for the next call.

Operation

iPECS IP & LDP Phones

To activate Walking COS (Class of Service)

1) Press the **[TRANS/PGM]** button.
2) Dial “23”, the Walking COS (Class of Service) code.
3) Dial the Station number and Authorization code, “*” and System Authorization code or“*”, the code index and System Authorization code.
4) Place your call as normal.

SLT(Single Line Telephone)

To activate Walking COS (Class of Service)

1) Momentarily press the hook-switch.
2) Dial “*561”, the SLT(Single Line Telephone) Programming code.
3) Dial “23”, the Walking COS (Class of Service) code.
4) Dial the Station number and Authorization code, “*” and System Authorization code or“*”, the code index and System Authorization code.
5) Place your call as normal.

Conditions

- ✓ The Station COS (Class of Service) applied for Walking COS (Class of Service) is the COS (Class of Service) of the Station associated with the Authorization code or, for a System Authorization code, COS1.
- ✓ Walking COS (Class of Service) applies the COS (Class of Service) for only one call. Terminating the call returns the station to the assigned Station COS (Class of Service). The user may reactivate Walking COS (Class of Service) to place another call or use Flash to maintain the Walking COS (Class of Service).
- ✓ Use of Authorization codes varies based on the Country Code assigned in the system database. In some regions, particularly the US and UK, a System Authorization code may be required for DISA access. Entering a Station code on a DISA line will fail in these areas. In other regions, a System Authorization code can be used with Walking COS (Class of Service).

Programming

Keyset Admin.

STATION

- Station COS (Class of Service) (PGM 116)

CO/IP

- CO Line COS (Class of Service) (PGM 141-Button 2)

SYSTEM

- DISA COS (Class of Service) (PGM 166)
- PABX Access Codes (PGM 172)
- LD Call Digit Count (PGM 177-Button 4)
- LD Digit Code (PGM 177-Button 19)

TABLES

- Toll Restriction Tables (PGM 224)
- Authorization Codes Table (PGM 227)

Web Admin.

STATION DATA

- Station COS (Class of Service)

CO LINE DATA

- Common Attributes ➤ CO Line COS (Class of Service)

SYSTEM DATA

- DISA COS (Class of Service)
- PBX Access Codes
- SMDR Attributes ➤ Long Distance Call Digit Counter, Long Distance Call Code

TABLES DATA

- Toll Exception Tables
- Station/System Authorization Codes Table

Related Features

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

2.32 Digit Conversion

Description

When a system receives digits from station, CO line, the dialed digits are converted according to the Digit ConversionTable before the Numbering Plan is checked.

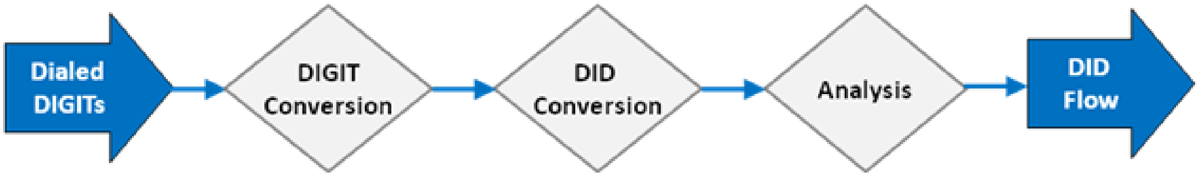
When you place an outside call the digits you dial are compared with the Digit Conversion Table. If a match is found, the digits are converted as defined in the table. Within the table are 15 sub-tables with up to 200 entries of up to sixteen digits for each sub-table for eMG80/100.With eMG800 and UCP, within the table are 32 sub-tables with up to 200 entries of up to sixteen digits for each sub-table

Different conversion can be assigned for the same dialed number based on the system service mode and LCR configuration. Conversion can also be limited to outside calls from a station or an outside Line. The latter may be used in networked environments where a system in the network employs Lines from another system to place a call.

Operation

CO.

Called party(DID) number is changed with digit conversion table.Then, system analyze converted number with DID process.



Station

Dialed digit from user is changed with digit conversion table.Then, system analyze converted number with outgoing process.

Conditions

- ✓ Each Station, CO line can have a Digit Conversion Tables.
- ✓ Emergency code takes priority over digit conversion.
- ✓ Dummy dial tone can be provided during digit conversion.
- ✓ There are two (2) special entries for configuring digit conversion:
  - X: Mask Digit, any digit is accepted
  - F: Ignore Digit
- ✓ In case Apply Time selects Follow Day/Night/Timed or Follow LCR Time, Auto Ring Mode Table should be assigned.

Programming

Keyset Admin.

SYSTEM

- Digit conversion Table (PGM 270)

STATION

- Digit conversion table index(114-Button 24-10)

CO

- Digit conversion table index (141-Button20)

Web Admin.

TABLE DATA

- Digit conversion Table (PGM 270)

STATION DATA

- Common Attribute ➤Digit conversion table

CO LINE DATA

- Common Attribute ➤Digit conversion table index

2.33 Alternative Route Selection for Digit conversion table

Description

This feature is enabled if there are several paths in order to connect toward a destinationSystem when a user use digit conversion feature.

If a selected path is not available for some reason (All Busy, Line Fault, etc) after digit conversion, Alternative RouteSelection (ARS) will connect calls using predefined designated path in digit conversion table.

Operation

If set, Alternative Route Selection operation is automatic.

Conditions

- ✓ ARS is optional and ARS digit in digit conversion table must be programmed to be operational.
- ✓ ARS digits should be contained in the CO Group Access Code
- ✓ ARS can be used in coordination with Last Number Redial, Station Speed Dial, and System Speed Dial.

Programming

Keyset Admin.

SYSTEM

- Digit conversion Table (PGM 270)

STATION

- Digit conversion table index(114-Button 24-Button10)

CO

- Digit conversion table index (141-Button 20)

Web Admin.

TABLE DATA

- Digit conversion Table (PGM 270)

STATION DATA

- Common Attribute ➤Digit conversion table

CO LINE DATA

- Common Attribute ➤Digit conversion table index

Hardware

2.34 Differential Ring

Description

Differential Ring allows any one of 1x (x means the number may be different) different audible Ring signals to be assigned to iPECS IP and1x (x means the number may be different) Ring signals for the LDP Phones, allowing you to determine which phone is ringing and the type of call (Intercom or external).

When the phone receives an incoming call, the selected ring signal is provided over thespeaker. Different selections are assigned for Intercom and outside calls.

Some different tones are stored in the iPECS IP Phones. These tones are permanent whilethe others are downloaded from iPECS system.In addition, each outside Line can be configured to provide one of different ring tones so that you can determine which outsideLine is ringing.

Operation

iPECS IP Phones

To select the desired ring tone

1) Press the **[TRANS/PGM]** button and Dial “1” for Ring Selection.
2) Dial “1” for Station Ring or “2” for CO Ring.
3) Dial Ring Tone selection, the ring tone is received.

To download a Ring tone from the system to an iPECS IP Phone

1) Press the **[TRANS/PGM]** button and Dial 1 for Ring selection.
2) Dial 5 for Ring tone download.
3) Dial the Ring tone storage bin.
4) Dial the Ring tone selection, the tone is received.
5) Press the **[HOLD/SAVE]** button.

LDP Phones

To select the desired ring tone

1) Press the **[TRANS/PGM]** button and Dial “1” for Ring Selection.
2) Dial “1” for Station Ring or “2” for CO Ring.
3) Dial Ring Tone selection, the ring tone is received.

Conditions

- ✓ To employ one of the system Ring tones, it must first be downloaded to a Ring tone storage bin in the iPECS IP.
- ✓ The iPECS UCS Client Soft phones do not have access to the system Ring tones. In the iPECS UCS Client phones multiple tones are available and you may load an *.wav file for use as a Ring tone.
- ✓ In case of LDP phone, 1xRing tones can be served from the memory in the phone.

Hardware

- ■ iPECS IP or LDP Phone

2.35 DND (Do Not Disturb)

Description

You can activate DND for a phone to prevent incoming calls from ringing at the station. This can be useful, for example, when you are in a meeting or busy. DND will also block page announcements.

Operation

iPECS IP & LDP Phones

To activate DND

1) Press the **[DND]** button, the **[DND]** button LED illuminates.

To remove DND

1) Press the **[DND]** button, the **[DND]** button LED extinguishes.

SLT(Single Line Telephone)

To activate DND

1) Dial “*553”, the DND feature code; stutter dial tone is received.

To remove DND

1) Dial “*553”, the DND feature code, dial tone is received.

Conditions

- ✓ A station will receive error tone if not allowed access to DND.
- ✓ If DND is allowed, pressing the **[DND]** button while ringing will activate One-Time DND.
- ✓ Only the Secretary of and Executive/Secretary Pair or an Attendant may override DND at the Executive’s station.
- ✓ An Attendant may cancel DND for other stations.
- ✓ DND service is not available to Attendants.
- ✓ Recalls for outside calls will override the DND feature.
- ✓ A station in DND is out-of-service for all incoming calls including Station Group calls.
- ✓ A station in DND is bypassed by calls forwarded to the Station. If the last Station in a Call Forward chain is in DND, the call will ring to the previous Station in the chain.
- ✓ When calling a Station in DND, an iPECS IP or LDP Phone display will indicate the DND status.

Programming

Keyset Admin.

STATION

- DND (PGM 111-Button 3)

Web Admin.

STATION DATA

- Common Attributes ➤DND

Related Features

- Feature Cancel
- Intrusion
- Call Forward
- DND Override
- Executive/Secretary Forward
- Station Groups
- DND - One Time DND

Hardware

2.36 Door Open

Description

The iPECS hardware is equipped with relays that activate External Control Contacts. Thecontacts can be assigned to one of several functions including a Door Open Contact. Whenused as a Door Open Contact, the contact is connected to a door-lock release mechanism.

When assigned stations receive the Door Bell signal, the user may dial the Door Open code toactivate the contact.

iPECS IP and LDP Phones may be assigned a Flex button as a **{DOOR OPEN}** button.

Operation

iPECS IP & LDP Phones

*To assign a **{DOOR OPEN}** button*

- **[TRANS/PGM]** + **{FLEX}** + Door Open code (“#*1”~“#*4”) + **[HOLD/SAVE]**

To activate the relay contact

1) Lift the handset or press the**[SPEAKER]** button.
2) Dial the Door Open code, “#*1”and “#*2” (eMG) or “#*1” in UCP100 or “#*1”to “#*4” in UCP600 and UCP2400.
3) Hang-up to return to idle.

Or,

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

Conditions

- ✓ The eMG has two External Control Contacts, one each located on the Miscellaneous connector of the KSU and EKSU.
- ✓ The UCP100 has a single External Control Contact while the UCP600 and UCP2400 each have four (4) External Control Contacts.
- ✓ The Door Open feature dial code is based on the assigned External Contact as below:
  - External Contact 1 = “#*1”
  - External Contact 2 = “#*2”
  - External Contact 3 = “#*3”
  - External Contact 4 = “#*4”
- ✓ The contacts are rated at 1 amp, 24 VDC.
- ✓ A station will receive error tone if not allowed access to the Door Open feature (PGM 113-Button 24-2) or in Station Data ➤Common Attributes.
- ✓ vUCP is not available for this feature.

Programming

Keyset Admin.

SYSTEM

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

STATION

- Door Open (PGM 113-Button 24-2)

Web Admin.

STATION

- Common Attributes ➤Door Open

SYSTEM DATA

- External Control Contacts
- System Timers ➤Door Open Timer

Related Features

- LBC (Loud Bell Control)
- Alarm Signal/Door Bell

Hardware

- External Control Contact connected to a door-lock release mechanism.

2.37 Emergency Call Service

2.37.1 Emergency Call

Description

Regardless of a Station's dialing restrictions (Class of Service), the user may dial any assigned Emergency numbers.

Operation

System

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

Conditions

- ✓ If there is no idle outside Line for the emergency call, a busy Line will be forced to release the current call and used for emergency call.
- ✓ An emergency call can be placed without a Line access code, except Australia. If an emergency number conflicts with the Station Numbering Plan, emergency calls must use a Line access code. For example, if the emergency number is "112" and the Station Numbering Plan is 100~, you must dial a Line access code before dialing the emergency number "112".
- ✓ If dialed number is matched with Emergency Code Table(PGM226) and Emergency CLI is programmed , Emergency CLI will be used as CLI (USA only)

Programming

Keyset Admin.

STATION

- Emergency CO Group (PGM 112-Button 18)

TABLES

- Emergency Service Call Table (PGM 226)

Web Admin.

STATION DATA

- Common Attributes ➤Emergency CO or Group

STATION DATA

- CLI Attributes(PGM113) – Emergency CLI

TABLES DATA

- Emergency Code Table

Hardware

2.37.2 Emergency Call Caller-Location Support

Description

Emergency Call - Caller Location Support integrates the iPECS to a PBX ANILink unit, such as from Tone Commander, to provide Caller Id and location information to anemergency center. When an Emergency call is placed from a Station, an assignedemergency outside Line is used to connect the ANI Link unit.

The iPECS sends the Station number,four (4) digits, to the unit as DTMF digits. The ANI Link unit places a call to the emergencycenter over a connected CAMA trunk. Caller ID and location information from the ANI Link unitdatabase is sent to the emergency center in the proper format and the Station is connected tothe call.

This feature also provides a means to identify a Power Failure signal from the ANI Link unit.Typically, an ANI Link unit will incorporate a normally open contact that closes when power tothe unit fails.

If a Power Failure signal is detected, iPECS routes emergency calls via PSTN/ISDN circuits other than those marked for emergency call use until the Power Fail signal isremoved.

Operation

Operation of this feature is automatic when assigned.

Conditions

- ✓ The system analog CO Lines connected to the ANI Link unit must be assigned to a separate Line group and the individual Lines assigned for emergency service.
- ✓ The ANI Link unit must be properly connected to a SLT(Single Line Telephone) port, an analog CO Line port and on the same LAN as the system KSU LAN port.
- ✓ The ANI Link unit must be properly installed and connected to a CAMA trunk from the carrier service provider to send location information to the local emergency center.
- ✓ It is the Customer’s responsibility to maintain the ANI Link unit number and location database; there is no integration provided between the iPECS database and the ANI Link unit.
- ✓ The system must be programmed for a three or four digit Station Numbering Plan. The system must send four digits to the ANI unit. When a three digit plan is used, a leading “0” is automatically added to the Station number.
- ✓ During a power failure, if the ANI Link unit Power Fail contacts, normally open, are connected to a system SLT(Single Line Telephone) port, the iPECS will recognize the fault and emergency calls are not sent to the ANI Link unit. In this situation, emergency calls are sent over normal system PSTN/ISDN circuits and the station digits are not sent.
- ✓ SMDR output will identify any call made via the Emergency call feature, including the success or failure of such calls.

Programming

Keyset Admin.

STATION

- Emergency CO/Group (PGM 112-Button 18)
- Proctor Monitor (PGM 113-Button 22)

CO/IP

- Proctor On/Off (PGM 141-Button 12)

TABLES

- Emergency Service Call Table (PGM 226)

Web Admin.

STATION DATA

- Common Attributes ➤ Emergency CO or Group, Proctor Monitoring Power-Failure

CO LINE DATA

- Analog Attributes ➤ PROCTOR SERVICE ON/OFF

TABLES DATA

- Emergency Code Table

Related Features

- Emergency Call
- Emergency Call Attendant Alert

Hardware

- Tone Commander PABX ANI Link unit or equal
- CAMA Trunk
- SLT(Single Line Telephone) port connected to ANI Link unit
- Analog Line connected to ANI Link unit

2.37.3 Emergency Call –Bomb Threat History

Description

When a station user is talking with an outside party, the user can save the telephone number of outside party to the Threat History log and record the call in the Emergency Mailbox.

Operation

iPECS IP & LDP Phones

To assign a{Bomb Threat History} button

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** button + 8*3 + **[HOLD/SAVE]**

*To activate a**{Bomb Threat History}**while on an outside call*

1) Press the**{Bomb Threat History}** button, the telephone number of outside party is saved and the call is recorded to the built-in Voice Mailbox of the assigned Emergency Mailbox Station.

Attendant iPECS IP & LDP Phones

To view the history log

1) Press **{Bomb Threat History}** button, information on latest Bomb Threat log displays with the telephone number and the time when the button was pressed.

Conditions

✓ The system can save up to eight (8)Bomb Threat History records. When the log is full,, the oldest of the eight records will be deleted then the new record is saved to the log.

Programming

Web Admin.

SYSTEM DATA

- System Attributes ➤Emergency Mailbox Station

Hardware

2.37.4 Emergency Mailbox

Description

When a station makesanEmergency call or the Threat History button pressed, the call can be recorded.

If a station number is assigned as the Emergency Mailbox Station in System Attributes, the emergency call will be recorded to built-inMailbox of the station.

All stations with an **{Emergency Mailbox}** button are notified of the recording and can access the Mailbox using the button.

Operation

iPECS IP & LDP Phones

*To assign a**{Emergency Mailbox}** button*

- **[TRANS/PGM]** + **{FLEX}** + Built-in VM Group number + Station Number + **[HOLD/SAVE]**

To listen to the Emergency Mailbox recording

1) Press the flashing **{Emergency Mailbox}** button
2) Follow the normal Voice Mail procedures to listen to the Mailbox recordings.

Conditions

✓ VSF must be installed and the VSF VM Station Group assigned in the system for recording to the Emergency Mailbox Station.

Programming

Web Admin.

SYSTEM DATA

- System Attributes ➤Emergency Mailbox Station

2.37.5 Emergency Call Monitor

Description

When a Station places a call to an assigned Emergency number, the System Attendant receives an alerting tone and can monitor the Emergency call.

Operation

Attendant iPECS IP & LDP Phones

To Monitor an Emergency call after the Alert tone

1) Press the **[MONITOR]** soft button, then the Attendant is connected to the Emergency call.

Programming

Web Admin.

STATION DATA

- Common Attributes ➤Emergency CO or Group

TABLES DATA

- Emergency Code Table

Related Features

- Emergency Call

Hardware

2.38 Executive/Secretary Forward

Description

iPECS IP and LDP Phones can be assigned as Executive/Secretary pairs.

By activating {Executive/Secretary Forward} button, a call that is routed to Executive can be forwarded to Secretary. This button is assgined in Executive & Secretary also.The button is pressed, then forward is activated, one more time pressed, then forward is deactivated.

Inaddition, if the Secretary is in DND, Executive calls sent to the Secretary route back to theExecutive if the “Call Exec If Sec in DND” option is enabled.

Each Executive can be assigned a “Grade” (01, highest ~12, lowest). Executives with a highergrade can call lower grade Executives overriding the Executive/Secretary forward.

With the “ICM Call to Secretary” option(PGM229-Button5) enabled, all intercom calls to theExecutive(except for calls from higher or same grade executive) forward to the Secretaryregardless of the Executive’s station status.

Callers to an Executive in DND can leave a Message Wait indication. The message waitingindication is given to the Executive or Secretary station based on the system database andUser Programming options.

Operation

iPECS IP & LDP Phones

To assign a{Executive/Secretary Forward} button

- **[TRANS/PGM]** + **{FLEX}** + **[TRANS/PGM]** button + 59 + xxx (Executive/Secretary table Index) + **[HOLD/SAVE]**
- Web Admin – Station Data – Flexible Buttons – Programming(PGM Code) - 59 + xxx(Executive/Secretary table Index) - Save

The **[Executive/Secretary forward]** button works ON (LED is red) or OFF as toggle.

- ✓ If all calls to Executive station want to forward to Secretary station, Executive station press the button to ON.
- ✓ If all calls to Secretary station want to forward to Executive station, Executive station press the button to OFF.

To activate/deactivate Executive/Secretary Forward from the Executive

2) Press the {Executive/Secretary Forward} flexible button to toggle Executive/Secretary Forward.

Conditions

- ✓ Up to 100 Executive/Secretary pairs can be configured in the iPECS UCP/eMG800.Up to 36 Executive/Secretary pairs can be configured in the iPECS eMG.
- ✓ An Executive may have multiple Secretaries and a Secretary may have multiple Executives. Each forms a separate Executive/Secretary pair.
- ✓ If the Secretary is busy when a call is received for the Executive, the caller will receive busy tone.
- ✓ The Secretary may override the DND status of the Executive to Camp-On and transfer calls to the Executive.

- ✓ A chain can be constructed by assigning the Secretary of one pair as an Executive of another. Although a chain may be constructed, a loop back is not allowed.
- ✓ If an Executive has multiple Secretaries, calls will automatically route to the Executive's first idle Secretary.
- ✓ The Executive may use Call Forward to send calls to stations other than the Secretary.
- ✓ Messages are left at the Executive or Secretary based on assignment of "Left Message Exec". When enabled (On), messages are left at the Executive's station.
- ✓ The Executive Grade can be assigned only for Country Code "82", Korea.
- ✓ The Executive can call to another executive who is same Group in ADM 229 – Button 7 even though they are different or lower grade.

Programming

Keyset Admin.

STATION

- DND (PGM 111-Button 3)
- Left Message Exec (PGM 113-Button 10)

TABLES

- Executive/Secretary Pair (PGM 229-Button 1)
- CO Call to Secretary (PGM 229-Button 2)
- Call Exec If Secretary in DND (PGM 229-Button 3)
- Exec Grade (PGM 229-Button 4)
- ICM Call to Secretary (PGM 229-Button 5)
- Secretary Auto Answer (PGM 229-Button 6)
- Executive Group (PGM 229-Button 7)

Web Admin.

STATION DATA

- Common Attributes ➤DND, Left Message to Executive

TABLES DATA

- Executive/Secretary ➤Executive Station, Secretary Station, CO Call to Secretary, Call Exec If Secretary in DND, Executive Grade, ICM Call to Secretary, Secretary Auto Answer, Executive Group

Related Features

- DND (Do Not Disturb)
- Call Forward
- Call Waiting/Camp-On
- CLI Message Wait
- Message Wait/Call Back
- Call Transfer

Hardware

- iPECS IP or LDP Phone

2.39 External Auto Attendant/Voice Mail

2.39.1 AA/VM Group

Description

The system provides support for an adjunct AA (Auto Attendant)/VM (Voice Mail) system via connectionto SLT(Single Line Telephone) ports. When a call arrives for the External AA/VM Group, the system will search thegroup for an idle port and deliver the call.

For AA/VM systems that employ SLT(Single Line Telephone) ports, signaling information between the iPECS and external AA/VM system may be assigned for in-bandDTMF signaling or the SMDI (Simplified Message Desk Interface) signaling protocol over theassigned iPECS system RS-232 port.

SIP based external AA/VM systems employ standard SIP signaling, and require a SIP server license and a license for each simultaneous call the AA/VM system will support.

Operation

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

Conditions

- ✓ Selection of SMDI or in-band signaling can be configured using Virtual Dip Switch settings (PGM 453-button 2 or the Maintenance admin).
- ✓ Only one AA/VM Group (Internal or External) can be defined in the system. Multiple definitions will cause erroneous system operation.

Programming

Keyset Admin.

STATION GROUPS

- Station Group Assignment (PGM 190)
- Station Group Attributes (PGM 191)

TABLES

- Voice Mail Dialing Table (PGM 234)

DIP SWITCH

- Virtual Dip Switch (PGM 453-Button 2)

Web Admin.

STATION GROUPDATA

- Station Group Assignment, VSF Group, Station Group Attributes

TABLES DATA

- Voice Mail Dialing Table

MAINTENANCE TRACE

- Trace ➤Virtual Dip Switch➤ SMDI Setting

Related Features

- In-band (DTMF) Signaling
- SMDI (Simplified Message Desk Interface)
- Auto Call Recording
- Two-Way Record
- Integrated Auto Attendant/Voice Mail
- SIP Phone Extension

Hardware

- External AA/VM system

2.39.2 In-band (DTMF) AA/VM 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(Single Line Telephone) port, the system will send DTMF signals informing theAA/VM of the characteristics of the call.

DTMF digit strings are assigned to various functionsallowing the AA/VM to respond appropriately to the call. These definitions are entered in theVoice Mail Dialing Table (PGM 234).

Operation

System

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

Conditions

- ✓ Selection of SMDI or in-band signaling can be configured using Virtual Dip Switch settings (PGM 453-button 2).
- ✓ Only one AA/VM Group (Internal or External) can be defined in the system. Multiple definitions will cause erroneous system operation.

Programming

Keyset Admin.

STATION GROUPS

- Station Group Assignment (PGM 190)
- Station Group Attributes (PGM 191)

TABLES

- Voice Mail Dialing Table (PGM 234)

DIP SWITCH

- Virtual Dip Switch (PGM 453-Button 2)

Web Admin.

STATION GROUP DATA

- Station Group Assignment, Station Group Attributes

TABLES DATA

- Voice Mail Dialing Table

MAINTENANCETRACE

- Virtual Dip Switch➤ SMDI Setting

Related Features

- AA/VM Group
- SMDI (Simplified Message Desk Interface)
- Auto Call Recording

- Two-Way Record
- Integrated Auto Attendant/Voice Mail

Hardware

- External AA/VM system

2.39.3 SMDI (Simplified Message Desk Interface) AA/VM Signaling

Description

The system may employ SMDI (Simplified Message Desk Interface) protocol to communicatewith an adjunct AA/VM system. When a call is routed to an AA/VM SLT(Single Line Telephone) port, the system willsend SMDI messages over the assigned SMDI RS-232 port, informing the AA/VM of thecharacteristics of the call.

SMDI Protocol

There are three types of SMDI messages. Within each message is an "Action Code", whichdefines the function or required action of the AA/VM system. Fields within the messages alsodefine the called/calling station and station status. The various message types and definitionof 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 SMDI Protocol

Field	Description
cr	carriage return
lf	line feed
MD	Message Desk
ggg	Message desk number, AA/VM system, default 001
mmmm	Message Desk terminal, 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 variousSMDI action codes.

Table SMDI Messages

Action Code	Reason	Purpose	Message Type	SMDI Message MD 001 0001-
A	Unconditional forward to VM	Put Mail	II	A xxxxx yyyyy
B	Called Station busy	Busy Mail	II	B xxxxx yyyyy
C	Disconnect, connected party	disconnect	II	C xxxxx yyyyy
D	Direct Fwd to VM group	Get Mail	II	D xxxxx yyyyy
E	Error, invalid number	Error	II	E xxxxx yyyyy
H	Two-way Record	record	II	H xxxxx yyyyy
I	DND	DND	II	I xxxxx yyyyy
N	No Answer	No Answer	II	N xxxxx yyyyy
R	Direct ring to VM group	AA	III	R xxxxx yyyyy

Operation

System

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

Conditions

- ✓ Selection of SMDI or in-band signaling can be configured using Virtual Dip Switch settings.
- ✓ Only one AA/VM Group (Internal or External) can be defined in the system. Multiple definitions will cause erroneous system operation.

Programming

Keyset Admin.

STATION GROUPS

- Station Group Assignment (PGM 190)
- Station Group Attributes (PGM 191)

TABLES

- Voice Mail Dialing Table (PGM 234)

DIP SWITCH

- Virtual Dip Switch (PGM 453-Button 2)

Web Admin.

STATION GROUP DATA

- Station Group Assignment, Station Group Attributes

TABLES DATA

- Voice Mail Dialing Table

MAINTENANCETRACE

- Trace ➤Virtual Dip Switch➤ SMDI Setting

Related Features

- AA/VM Group
- In-band (DTMF) Signaling
- Auto Call Recording
- Two-Way Record
- Integrated Auto Attendant/Voice Mail

Hardware

- External AA/VM system

2.39.4 Voice Mailbox Translation

Description

Your Station can be assigned to employ a Voice Mailbox number that is different from your Station number. When a call is routed to voice mail, your Mailbox number, which is assigned separately from your Station number, is sent to the Voice Mail system. The Voice Mailbox number can be any 8-digit number("00000001" to "99999999"). An entry of 00000000 or blank sends the Station number to the VM system.

Operation

System

Operation of this feature is automatic when programmed.

Conditions

✓ Voice Mailbox Translation applies to iPECS UMS and External In-band and SMDI Voicemail systems.

Programming

Keyset Admin.

STATION

- Voice Mail Id (PGM 113-Button 14)

STATION GROUPS

- Station Group Assignment (PGM 190)
- Station Group Attributes (PGM 191)

TABLES

- Voice Mail Dialing Table (PGM 234)

Web Admin.

STATION DATA

- Station VM Attributes ➤ VMID Number

STATION GROUP DATA

- Station Group Assignment, Station Group Attributes

TABLES DATA

- Voice Mail Dialing Table

Related Features

- AA/VM Group
- SMDI (Simplified Message Desk Interface)
- Two-Way Record
- Integrated Auto Attendant/Voice Mail

Hardware

2.40 Flexible Numbering Plan

Description

Your access to the iPECS system resources and features is accomplished through featurecodes or Flexible buttons on youriPECS IP or LDP Phone.

The Administrator selects from one of the nine (9) differentstandard numbering plans and, if desired, assigns codes for individual functions in the FlexibleNumbering Plan.

The feature codes are defined in the system's Flexible Numbering Plan; referto the iPECS Admin and Programming Manual.

Operation

System

System implements feature activation based on the Flexible Numbering Plan.

Conditions

✓ Feature codes can be one to four digits in length.

✓ During programming, conflicts in the Numbering Plan are not allowed. The existing non-conflicting Numbering Plan is used until correctly updated.

Programming

Keyset Admin.

SYSTEM ID

- System Id (PGM 100-Button 4)

NUMBERING

- Flexible Station Numbering Plan (PGM 105)
- Flexible Numbering Plan (PGM 106 – 109)

Web Admin.

SYSTEM ID & NUMBERING PLANS

- System ID
- Flexible Station Number
- Flexible Numbering Plan

Hardware

2.41 Forced Disconnect Intrusion

Description

You can force a busy outside Line or station to disconnect an active call, and connect to your phone. Prior to connecting, the outside Line is returned to idle.

When connecting to a station, the station will receive a short warning tone.

Operation

iPECS IP & LDP Phones

*To assign a**{Forced Release}** button*

- **[TRANS/PGM]** + [Flex button] + **[TRANS/PGM]** + [7#] + **[HOLD/SAVE]**.

To activate forced Line disconnect feature

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the busy Line button or dial a Line access code, you will receive busy tone.
3) Press the **{Forced Release}** button.
4) The call on the busy Line is terminated, the connected station receives error tone and you are connected to the Line.

Or,

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the busy Line button or dial a Line access code, you will receive busy tone.
3) Press the **[TRANS/PGM]** button and dial "*56*", Forced Channel Seize code.
4) The call on the busy Line is terminated, the connected station receives error tone and you are connected to the Line.

To activate forced intrusion to a busy station feature

1) Place an intercom call to the busy station and receive busy tone.
2) Press the **{Forced Release}** button.
3) The call on the busy Station is terminated, the station receives warning tone and you are connected to the station.

Or,

1) Place an intercom call to the busy station and receive busy tone.
2) Press the **[TRANS/PGM]** button and dial "*56*", Forced Channel Seize code.
3) The call on the busy Line is terminated, the connected station receives error tone and you are connected to the Line.

SLT(Single Line Telephone)

To activate forced Line disconnect feature

1) Lift the handset.
2) Dial a Line access code, you will receive busy tone.
3) Make a Hook Flash.
4) Dial "*56*".

5) The call on the busy Line is terminated, the connected station receives error tone and you are connected to the Line.

Conditions

- ✓ If the station was talking to other station or CO line, the call is disconnected when the station receives forced Intrusion call.
- ✓ A Forced Intrusion call is not possible to a station in DND state.
- ✓ A Forced Intrusion call is not possible to an Attendant.
- ✓ Any station with a Forced Intrusion button can activate this feature.
- ✓ If ‘Privacy’ option is ON, then any person can’t intrude into my conversation by barge-in and forcefully disconnect the feature.

Programming

STATION DATA

- Common Attributes ➤ Privacy

Hardware

2.42 Green Power Save for eMG

Description

The system can disable the power to LDP Phones and SLT(Single Line Telephone)s (Single Line Telephone) connected to the system at pre-determined times such as at night or during holidays when no one will be using the phones. In addition the automatic control, power to the phones can be manually controlled from the Web Admin.

Operation

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

Conditions

- ✓ When power is removed from an LDP Phone or SLT(Single Line Telephone), the phone cannot place or receive calls.
- ✓ In the event of a system reset while the Green Power mode is active, after the reset, the Green Power mode is disabled. If in the automatic mode, the Green Power mode will reactivate when the activation time is reached.
- ✓ Station ports 1 and 2 are not affected by the Green Power mode and these ports will continue to operate normally to receive and place calls.

Programming

Keyset Admin.

Green mode

- Green Mode Activation (PGM 500)
- Green Mode Time (PGM 501)

Web Admin.

Green Mode

- Green Mode Activation
- Green Mode Time Setting

Hardware

2.43 Headset Compatibility

Description

You can use an industry standard headset withiPECS IP and LDP Phones in place of or in additionto the handset. The station is programmed for Headset operation.

In the Headset mode, pressing the **[SPEAKER]** button will send audio to the Headset insteadof the speakerphone. In addition, when in the Headset mode, ring signals can be delivered tothe speaker or the headset as defined in the system database.

As an iPECS IP or LDP Phone users, you may assign a Flex button to select Headset or Speakerphone operation.The **{HEADSET SELECT}** button may be used to toggle the operation of the phone betweenHeadset and Speakerphone.

Operation

iPECS IP & LDP Phones

*To assign a Flex button for **{HEADSET SELECT}***

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “61” + **[HOLD/SAVE]**

To change operation from Speakerphone to Headset

1) Press the **[TRANS/PGM]** button.
2) Dial “61”, the Headset select code.
3) Dial “0” to select Headset, “1” to select Speakerphone.

Or,

1) Press the**{HEADSET SELECT}** button.

To change the device to receive ring signals

1) Press the **[TRANS/PGM]** button.
2) Dial “62”, the Ring select code.
3) Dial “1” for Speaker, “2” for Headset or “3” for both.

To place/answer calls using the headset

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

Conditions

- ✓ The Intercom Signaling Mode can be set in the Headset mode as with the Speakerphone mode.
- ✓ The station always receives Page announcements over the speaker of iPECS IP and LDP Phones.
- ✓ Although the phone is in the Headset mode, the system will monitor the hook-switch status. If the user lifts the handset to go off-hook, audio is delivered to the handset.
- ✓ If you are using a LIP-8012D, 8024D or 8040L phone in the Headset mode, the **{SPEAKER}**Soft button is available during an active call. You can switch to the speakerphone for the duration of the active conversation. If you take further action, such as pressing another feature or **{LINE}** button, the mode switches back to Headset.

Programming

Keyset Admin.

STATION

- Headset Ring Mode (PGM 111-Button 8)
- Speakerphone/Headset (PGM 111-Button 9)

Web Admin.

STATION DATA

- Terminal Attributes ➤Headset Ring, Speakerphone, Headset Usage

Related Features

- Speakerphone
- Paging

2.44 Hold

2.44.1 Automatic Hold

Description

While on an active outside call, the system will place the call on hold automatically if you press the **[FLASH]**, **[CONF]**, **{DSS/BLF}** or other feature buttons. In addition, your station can be programmed to support Line to Line Automatic Hold. In this case, pressing an outside Line button while on an outside call will place the active call on hold and access the selected Line.

Operation

iPECS IP & LDP Phones

To use Automatic Hold while on an active outside call

✓ Press the desired feature button or **{LINE}**; the active call is placed in the Preferred Hold state.

Conditions

✓ Outside Lines placed on hold with Automatic Hold are placed in the assigned Preferred Hold.

✓ There is no limit on the number of calls that can be placed on hold using Automatic Hold.

Programming

Keyset Admin.

STATION

- Automatic Hold (PGM 112-Button 2)

SYSTEM

- Hold Preference (PGM 160-Button 7)

Web Admin.

STATION DATA

- Common Attributes ➤Automatic Hold

SYSTEM DATA

- System Attributes ➤Hold Preference

Related Features

■ Hold Preference, Hold Recall

Hardware

■ iPECS IP or LDP Phone

2.44.2 Hold Preference

Description

Hold Preference defines either Exclusive Hold or System Hold as the preferred hold state,which is activated on the first depression of the **[HOLD/SAVE]** button.

Pressing the**[HOLD/SAVE]**button twice in succession, places the call in the non-preferred hold state. The selection of the preferred hold state is theassigned Hold Preference in the system database.

Operation

iPECS IP & LDP Phones

To activate the Preferred Hold while on a call

1) Press the **[HOLD/SAVE]** button, the connected party is placed in the preferred hold state.
2) Press the button a second time for non-preferred hold.

Conditions

- ✓ A transferred call is placed in the Exclusive hold state at the receiving station regardless of the assigned Hold Preference.
- ✓ Hold Preference is not available to a SLT(Single Line Telephone).

Programming

Keyset Admin.

SYSTEM

- Hold Preference (PGM 160-Button 7)

Web Admin.

SYSTEM DATA

- System Attributes ➤Hold Preference

Related Features

- Call Transfer, Outside Line
- Exclusive Hold
- System Hold

Hardware

- iPECS IP or LDP Phone

2.44.3 Hold Recall

Description

When you place anoutside call on hold, a hold timer is activated. If the timer expires, the heldcall will recall your station for the I-Hold Recall time. If the call remains unanswered, theAttendant also receives recall for the Attendant Recall time and, if the call is on Exclusive hold,the call is placed on System Hold.

If still unanswered after the Attendant Recall time, the call is disconnected and the appropriate circuits returned to idle.

Operation

Hold Recall operation is automatic.

Conditions

- ✓ Separate timers are assigned for the various types of hold: System, Exclusive, Transfer, Call Park, etc.
- ✓ If the I-Hold timer is set to zero, the station will not receive recall. If the Attendant Recall timer is set to zero, the Attendant will not receive recall.
- ✓ If the specific Hold timer is set to zero, recall is disabled.
- ✓ In case of Private Lines, recall ring is served to only the station that held the call and will recall until answered or abandoned.

Programming

Keyset Admin.

SYSTEM

- Hold Preference (PGM 160-Button 7)
- Attendant Recall Timer (PGM 180-Button 1)
- Exclusive Hold Recall Timer (PGM 180-Button 4)
- I-Hold Recall Timer (PGM 180-Button 5)
- System Hold Recall Timer (PGM 180-Button 6)
- Transfer Hold Recall Timer (PGM 180-Button 7)

Web Admin.

SYSTEM DATA

- System Attributes ➤Hold Preference
- System Timers ➤Attendant Recall Timer, Exclusive Hold Recall Timer, I-Hold Recall Timer, System Hold Recall Timer, Transfer Recall Timer

Related Features

- ■ Call Transfer, Line
- ■ Exclusive Hold
- ■ System Hold

Hardware

2.44.4 Exclusive Hold

Description

Outside lines may be placed in a waiting state such that other stations in the system are unableto access the line. Only the station placing the line on Exclusive hold can access the heldline.

If the call remains on hold at expiration of the Exclusive Hold Recall Timer, normal Hold Recallwill apply.

Operation

iPECS IP & LDP Phones

To place a call on Exclusive Hold

1) Press the **[HOLD/SAVE]** button twice.

To access a call on Exclusive Hold from the holding station

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the held line button.

Or,

1) Dial “8#”, Access Held Individual CO/IP code.
2) Dial the line number.

SLT(Single Line Telephone)

To place a call on Exclusive Hold

✓ Momentarily press the hook-switch.

To access a call on Exclusive Hold from the holding SLT(Single Line Telephone)

1) Lift the handset.
2) Dial “8#”, Access Held Individual CO/IP code.
3) Dial the line number.

Conditions

✓ When an outside line is placed on Exclusive Hold, the button LED will flash at 120 ipm at the holding station and will be On at all other stations.

✓ All Transferred outside calls are placed on Exclusive Hold for the receiving station.

✓ The LED of **{LOOP}** and **{POOL}** buttons will display the active outside line status.

Programming

Keyset Admin.

SYSTEM

- Hold Preference (PGM 160-Button 7)
- Attendant Recall Timer (PGM 180-Button 1)
- Exclusive Hold Recall Timer (PGM 180-Button 4)
- I-Hold Recall Timer (PGM 180-Button 5)
- Transfer Hold Recall Timer (PGM 180-Button 7)

Web Admin.

SYSTEM DATA

- System Attributes ➤Hold Preference
- System Attributes ➤Attendant Recall Timer, Exclusive Hold Recall Timer, I-Hold Recall Timer, System Hold Recall Timer, Transfer Recall Timer

Related Features

- Call Transfer, Line
- Hold Preference
- Hold Recall
- System Hold

2.44.5 System Hold

Description

OutsideLines may be placed in a waiting state such that other stations in the system are able toaccess the Line. Stations must have access to the Line in the system database toaccess the held call.

If the call remains on hold at expiration of the System Hold Recall Timer, normal Hold Recallwill apply.

Operation

iPECS IP & LDP Phones

To place a call on System Hold

1) Press the **[HOLD/SAVE]** button.

To access a call from System Hold

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the line button.

Or,

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial "8#", Access Held Individual CO/IP code.
3) Dial the line number.

SLT(Single Line Telephone)

To place a call on Hold

1) Momentarily press the hook-switch.
2) Dial "*560", SLT(Single Line Telephone) CO System Hold code.

To access a call from System Hold

1) Lift the handset.
2) Dial "8#", Access Held Individual CO/IP code.
3) Dial the line number.

Conditions

- ✓ When an outside Line is placed on System Hold, the button LED will flash at 30 ipm and wink at the holding station and will flash at all other stations.
- ✓ A call on System Hold can be retrieved from any station allowed access to the Line in the system database using the Line button or the Held Call access code.
- ✓ The LED of **{LOOP}** and **{POOL}** buttons will display the active outside Line status.

Programming

Keyset Admin.

SYSTEM

- Hold Preference (PGM 160-Button 7)
- Attendant Recall Timer (PGM 180-Button 1)

- I-Hold Recall Timer (PGM 180-Button 5)
- System Hold Recall Timer (PGM 180-Button 6)

Web Admin.

SYSTEM DATA

- System Attributes ➤Hold Preference
- System Attributes ➤Attendant Recall Timer, I-Hold Recall Timer, System Hold Recall Timer

Related Features

- Call Transfer, Outside Line
- Hold Preference
- Hold Recall
- Exclusive Hold

Hardware

2.45 Hot Desk

Description

The iPECS IP can be assigned as Hot Desk phones allowing users (Agents) to login with theiPECS. The Hot Desk will become active and will take on the attributes defined for the Agent'sStation number. When the Agent logs off, the Hot Desk phone becomes inactive and theAgent's calls are forward to the user-entered destination. A different Agent may then loginthrough the inactive Hot Desk phone.

Operation

iPECS IP Phones

To login through an inactive Hot Desk

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial "*0".
3) Dial the Agent's Station number and authorization code, Agent is logged in.

To logout through the active Hot Desk

1) Press the **[TRANS/PGM]** button.
2) Dial "* *" or press the **{AGENT LOGOUT}** Flex button.
3) Use the **[VOL UP]/[VOL DOWN]** buttons to select the call forward destination for Agent calls (speed dial, joined mobile phone, Internal or External VM group).
4) Press the **[HOLD/SAVE]** button.

To login / logout on LIP-9071

1) Tag a NFC card to 9071 NFC module.
2) NFC menu is displayed in 9071 display with NFC information.
3) NFC menu is chosen, NFC information is sent to system.
4) Hot desk feature is activated or deactivated.

Conditions

- ✓ Only iPECS LIP phones can be configured for Hot Desk.
- ✓ An active (logged in) Agent can login to another inactive Hot Desk. However, this will log the Agent off the previously active Hot Desk.
- ✓ A user may only logout from the Agent's active Hot Desk phone. Attempting to logout using a different Hot Desk phone will return error tone.
- ✓ The Flex button map of the Hot Desk station is fixed and does not take on the configuration associated with the Agent's station.
- ✓ When logged off, the Agent's active database, including the following items, are saved.
  - Station Number
  - Station Attributes, PGM 111~124
  - CO Routing, Ring assignments, DID routing, etc.
  - Voice Mail
  - Station Group assignment

- ✓ The number of Hot Desk phones and Hot Desk users is limited to the system capacity. Each Hot Desk phone and Hot Desk user (Agent) requires a separate station channel in the system.
- ✓ To identify the station number, only Station Authorization codes may be used for Hot Desk login.
- ✓ An ACD Agent must use the Agent Login procedure as defined under the ACD feature description.
- ✓ An agent that takes no action for the Hot Desk Auto Logout time is logged out of the Hot Desk and must login to use the Hot Desk.

Programming

Keyset Admin.

STATION

- Hot Desk Station (PGM 112-Button 13)

TABLES

- Hot Desk Attributes (PGM 250-Button 1~3)
- Auto Logout Timer (PGM 250-Button 3)

Web Admin.

STATION DATA

- Common Attributes ➤Hot Desk Station

SYSTEM DATA

- Hot Desk Attributes ➤ Auto Logout Timer

Related Features

- Station Groups
- Automatic Call Distribution
- Call Forward

Hardware

- iPECS IP Phone

2.46 ICLID Call Routing

Description

The system can employ ICLID (Incoming Calling Line ID) to determine the routing of incomingexternal calls. Each outside 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 and, if amatch is found, will route the call to the destination defined in the ICLID Ring Assignment Table.Destinations can be System announcement, integrated or external Voice Mail, a station or a station group.

An ACD group may be assigned to route calls employing the ICLID Tables. When configured,ACD calls re-route based on the “Caller Entered ICLID”.

Operation

System

System implements routing automatically based on database entries and the received ICLID.

Conditions

- ✓ If the received ICLID does not match an entry in the ICLID Route Table, the call is routed based on the type and other programming (Ring assignments, etc.) for Line.
- ✓ For analog CO Lines, the system will await receipt of 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.).
- ✓ The ICLID received from the Line must be a telephone number to match an ICLID Route Table entry.
- ✓ If ICLID routing is enabled for a DID Line, DID Call Wait is disabled.

Programming

Keyset Admin.

CO/IP

- ICLID Ring Timer (PGM 142-Button 14)

ISDN/ICLID

- ICLID Route Table Attributes (PGM 203)
- ICLID Ring Assignment Table (PGM 204)

STATION GROUPS

- ACD ICLID Use (PGM 191-Button 21)

TABLES

- Flexible DID conversion(PGM 231)

Web Admin.

CO LINE DATA

- Analog Attributes ➤ICLID Ring Timer

ISDN LINE & ICLID ROUTING DATA

- ICLID Route Table

- ICLID Ring Assignment

STATION GROUPDATA

- Station Group Attributes, ACD Group-Entered Caller ID(ICLID) Usage

TABLES DATA

- Flexible DID conversion

Related Features

- Outside Line Ring Assignment
- Automatic Call Distribution
- Direct Inward Dial (DID)

Hardware

2.47 In-Room Indication

Description

As an In-Room Supervisor, you can notify other station users that you are in the room and the phone is idle using a Flexible button on your phone. Activating the In-Room indicator illuminates the LED of the In-Room Indicator button assigned to the other stations. This feature is often used in small call centers or where the Supervisor's office is not visible by the supervised employees.

Up to ten (10) In-Room Supervisors can be assigned with up to twenty notified stations for each Supervisor.

You may assign a Flex button for In-Room indication however; the In-Room Supervisor can only be assigned in the system database.

Operation

iPECS IP & LDP Phones

To assign an{In-Room Indication No.} button"

- **[TRANS/PGM]** + **{FLEX}** + **[TRANS/PGM]** + "9*" + In-Rom Bin No (01~10) + **[HOLD/SAVE]**

To Active or Deactivate the In-Room Indication from the Supervisor phone while idle

1) Press the **[In-Room]** button.
2) Press the**[HOLD/SAVE]** button.

Conditions

✓ The In-Room Indication button operates only while the Supervisor's phone is idle.
✓ If the In-Room button on a station other than the Supervisor's is pressed, error tone is heard.
✓ The Supervisor must press the **[HOLD/SAVE]**button within 5 seconds of pressing the In-Room button to activate the indication otherwise; the station goes back to the idle state.

Programming

Keyset Admin.

SYSTEM

- In-Room Indication (PGM 183)

Web Admin.

SYSTEM DATA

- In-Room Indication

2.48 Integrated Auto Attendant/Voice Mail

2.48.1 Integrated AA/VM Overview

Description

iPECS eMG80/100

The integrated AA/VM application is provided through the Voice Store and Forward (VSF) Gateway incorporated in the KSU main board and includes an application processor, built-in default channels (refer to Specification table in Installation Manual) and 1 hour of storage. The storage can be increased to 16 hours with MEMU (Memory Expansion Unit) and 61 hours with MEMU2.

The memory is employed to store System 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.

eMG80: A second separate VVMU with eight (8) channels and 16 hours by license is available for the eMG.

iPECS eMG800

The integrated AA/VM application is provided through the VMIU in the MPB main board and includes an application processor, four (4) access channels and 1 hour of storage.

The memory is employed to store System 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.

A second separate VMIB with four (8) access channels and 100 hours of storage is available for the eMG800.

iPECS UCP

The integrated AA/VM application is provided through the Voice Store and Forward (VSF) Gateway incorporated in the iPECS UCP Module. The iPECS UCP100 provides eight (8) channelsand basic voice storage memory of four (4) hours expandable with license to 14 hours. The UCP600 provides eight (8) channels with six (6)hours of basic storage expandable to 16 hours. iPECS UVM modules may be included in the system and provide up to 8 channels expandable with license to 16 channels and 50 hours of storage expandable with license to 200 hours. The UVM is supported with all iPECS UCP systems.

The storage memory is employed to store user-recorded System 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.

Operation

System

Operation of the integrated AA/VM is automatic based on configuration.

Related Features

- Station Groups
- Automatic Call Distribution
- Delayed Auto Attendant

- Direct Inward System Access (DISA)
- Direct Inward Dial (DID)
- Remote Message Retrieval
- Multiple Language Support
- MOH (Music-On-Hold)
- Integrated Auto Attendant
- Integrated Voice Mail

Hardware

- eMG80/100 MEMU (15 hours) and MEMU2 (60 hours) Memory Expansion Unit
- eMG80 VVMU, 2nd VSF Gateway
- eMG800 VMIB
- UCP VSF, AA/VM 16 hour with VM Memory Expansion License
- UCP-UVM gateway

2.48.2 Integrated Auto Attendant

Description

When an outside call is received, on a DID or DISA line, the call may be routed directly to one of 200 user recorded System Announcements to function as an Auto Attendant announcement. Each System announcement has a corresponding CCR (Caller Controlled Routing) Table.

As an Auto Attendant, the announcement instructs the caller to dial a digit for a specific route ("for sales dial 2"). The CCR Table defines the routing for each digit the caller may dial. The routing may be a:

- Station
- Station group
- Speed Dial number
- Page Zone
- Voice Mail
- System Announcement

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 **[TRANS/PGM]** button and Dial "06", the System Announcement Record code.
2) Dial the sequence number of the voice storage unit.
3) Dial the Announcement number (001-200).
4) With Multi Language support, enter the Language number (1~6), the current announcement is played followed by the "Press # to record" prompt.
5) Dial "#".
6) After the beep-tone, record message.
7) Press the **[HOLD/SAVE]** button to stop recording and save the message.

To delete a recording

1) Press the **[TRANS/PGM]** button and Dial "06", the System Announcement Record code.
2) Dial the sequence number of the voice storage unit.
3) Dial the Announcement number (001-200).
4) With Multi Language support, enter the language number (1~6), the current announcement is played followed by the "Press # to record" prompt.
5) Dial "#".
6) Press the **[SPEED]** button during playback to erase message.

System

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

Conditions

- ✓ There are no individual time limits on an Auto Attendant announcement.
- ✓ The external caller may experience ring-back tone before the system answers and play the System announcement.
- ✓ A wave format audio file may be uploaded to the system for use as an Auto Attendant or other System announcement.
- ✓ The System Attendant must “Save” a recording before returning to the on-hook state, otherwise, the existing recording is used.
- ✓ To record or delete an Auto Attendant message, all of the AA/VM channels must be in the idle state.
- ✓ 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.
- ✓ If the external caller dials an invalid selection or station, the system will play the “Invalid Entry” prompt and allow re-entry. The DISA Retry Counter determines the number of times the caller may retry entries.
- ✓ If the external caller dials more than a single digit, the call is routed based on the Station Numbering Plan.
- ✓ Calls answered by an Auto Attendant (CCR) Announcement are interactive DISA calls and are subject to conditions of a DISA call.
- ✓ The “*” digit is reserved in the Audio/Text Tables to repeat the current or previous Auto Attendant announcement.
- ✓ The “#” digit is reserved for callers to access their Voice Mailbox remotely.
- ✓ A CCR Announcement may be programmed to disconnect after playing.
- ✓ The Auto Attendant Announcement feature is supported for DISA and DID calls as well as Normal Ring assigned outside Lines.
- ✓ With the optional second Voice Store Forward Gateway for eMG, System announcement should be recorded in both the main and second gateway. Also, each station and Line Voice Mailbox should be associated with one of the gateways.
- ✓ When an iPECS UVM gateway is employed, each System announcement should be recorded in both the built-in VSF gateway and the UVM gateway. In addition, each Station and outside Line Mailbox should be associated with one of the gateways.

Programming

Keyset Admin.

STATION

- VSF Access (PGM 113-Button 2)

CO/IP

- CO/IP Ring Assignment (PGM 144)
- DID Service Attributes (PGM 145)

STATION GROUPS

- Station Groups (PGM 190)
- Station Group Attributes (PGM 191)

SYSTEM

- DISA Retry Counter (PGM 160-Button 5)
- DID/DISA Destination (PGM 167)
- CCR Inter-digit Timer (PGM 180-Button 13)
- VSF User Record Timer (PGM 181-Button 3)
- VSF Valid User Message Timer (PGM 181-Button 4)

TABLES

- Customer Call Routing Tables (PGM 228)
- Flexible DID Conversion Table (PGM 231)

Web Admin.

STATION DATA

- VM Attributes ➤VSF Access

CO LINE DATA

- CO/IP Ring Assignment
- DID Service Attributes

STATION GROUP DATA

- Station Group Assignment
- Station Group Attributes

SYSTEM DATA

- Common Attributes ➤DISA Retry Counter, DID/DISA Destination
- System Timers ➤CCR Inter-Digit Timer, VSF User Maximum Record Timer, VSF Valid User Message Timer

TABLES DATA

- CCR (Customer Call Routing) Tables
- Flexible DID Conversion

Related Features

- Station Groups
- Automatic Call Distribution
- Delayed Auto Attendant
- Direct Inward System Access (DISA)
- Direct Inward Dial (DID)
- Remote Message Retrieval
- Multiple Language Support
- MOH (Music-On-Hold)

Hardware

2.48.2.1 Company Directory

Description

The integrated AA incorporates a Company Directory. A caller that reaches a configured CCR or Flexible DID Table can select to Dial-by-Name and will be prompted to spell the last three characters of the desired station user name using the dial pad.

If the entry matches a Name in the database, the call is transferred to the station. If multiple matches exist, the system plays the name greeting for the first nine (9) matches and the caller can select the appropriate station to receive the call.

If no matches exist the caller is prompted to reenter the name (three characters). The caller may retry the name entry based on the DISA Retry counter.

To be included in the directory, you must record a “Name Greeting” and your first and last name must be assigned in the system database.

Operation

iPECS IP & LDP Phones

To record a Company Directory Name

1) Dial the “*563”, the Company Directory Name record code.
2) Dial “1” to record your Name.
3) After recording prompt, say your name.

To delete Company Directory Name

1) Dial the [Record VM Subscriber Name] flexible number.
2) Dial 2 to delete the Name recording.

SLT(Single Line Telephone)

To record a Company Directory Name

1) Dial “*563” the Company Directory Name record code flexible number.
2) Dial 1, to record your Name.
3) After the recording prompt, say your Name.

To delete Company Directory Name

1) Dial “*563”, the Company Directory Name record code.
2) Dial 2 to delete the Name recording.

To use Company Directory Feature

1) In response to the system announcement, the caller dials the Company Directory index.
2) The system plays the First/Last name selection prompt.
3) The caller selects first or last name entry.
4) The system plays the enter 3-character prompt.
5) The caller dials the first three characters of the desired station user.
6) The System plays Name Greeting for each match (maximum 9).

7) If there is a single match, the called is transferred. If multiple matches exist, the system plays prompts for each with a selection digits below:
   - "for [subscriber name], press 1"
   - "for [subscriber name], press 2"
   - …
   - "for [subscriber name], press 9"

8) When the caller enters the desired digit, the system routes the caller to that station.

Conditions

✓ To be searched in company directory, the Station user name must be recorded and the first and last name also must be programmed properly.

Programming

Keyset Admin.

STATION

- VSF Access (PGM 113-Button 2)
- Company Directory First Name (PGM 127-Button 5)
- Company Directory Last Name (PGM 127-Button 6)

Web Admin.

STATION DATA

- VM Attributes ➤VSF Access, Company Directory - First Name, Company Directory - Last Name

Related Features

- CCR
- Integrated Auto Attendant

Hardware

2.48.2.2 Remote Announcement Recording

Description

An outside caller can be allowed to record a System announcement. This capability is often used in schools or other organizations to allow remote recording of school closing.

The administrator calls the Auto Attendant and dials the digit assigned for remote recording in the CCR Table.Note that this should not be included in the Auto Attendant announcement for security purposes.

The administrator enters the System Attendant password then can record any System announcement desired as shown in the Operation section.

Operation

To remote record a System Announcement

1) Call the number for the Auto Attendant.
2) During the announcement dial the digit assigned in the CCR Table for Remote Recording.
3) In response to the system prompt enter the Attendant's password.
4) In response to the system prompt, enter the System announcement number (001-200).
5) When prompted enter the Language type (1-6).
6) The system prompts "to record a new greeting, press pound" followed by any existing recording for the announcement and Language.
7) Press '#' and after the beep tone, start recording the new announcement.
8) Press '#' to complete and store the new recording, the system returns to step 5 so you can select another announcement to record. Conditions.

Programming

Keyset Admin.

TABLES

- Customer Call Routing Tables (PGM 228)
- Flexible DID Conversion Table (PGM 231)

Web Admin.

TABLES DATA

- CCR (Customer Call Routing) Tables
- Flexible DID Conversion

Related Features

- Integrated Auto Attendant
- Attendant

Hardware

2.48.3 Integrated Voice Mail

2.48.3.1 Administrator Mailbox

Description

A Mailbox may be assigned as an Administrator to allow access to other mailboxes and the administration menu to add, delete, or reset a mailbox password to default (station number). The administrator can record a Greeting and a Name for a mailbox and record and send a broadcast message to all mailboxes.

When you access an Administrative Mailbox the prompt "to access administrative options, press six" is added to the main VM prompt.

Operation

iPECS IP & LDP Phones

To access the Administrative Mailbox menu (From a Station with an Administrative Mailbox)

1) Call the integrated Voice Mail and enter the password.
2) Dial 6 to hear the Administrator menu.
3) Dial the desired option digit and follow the prompts to make the settings as below. (Administrator Mailbox Menus)
   - To add a mailbox, press 1
   - To delete a mailbox, press 2
   - To reset a mailbox password, press 3
   - To record a mailbox greeting, press 4
   - To record a broadcast message, press 5
   - To record a mailbox name, press 6

Conditions

- ✓ There are no limits on the number of mailboxes that can be marked as administrators.
- ✓ The broadcast message is sent to all mailboxes in the same tenancy group and is always retrieved first regardless message retrieve option.
- ✓ A station can delete broadcast message without confirming receipt.
- ✓ Only administrator can remove a recorded broadcast message.
- ✓ Administrator can record mail box greeting only for own system not TNET or networking.

Programming

Keyset Admin.

TABLES

- Customer Call Routing Tables (PGM 228)
- Flexible DID Conversion Table (PGM 231)

Web Admin.

STATION DATA

- VM Attributes ➤Administrator Mailbox

Related Features

- Integrated Auto Attendant/Voice Mail

Hardware

2.48.3.2 Announce Only Mailbox

Description

A mailbox can be configured to provide an announcement only and not provide incoming message storage. After the announcement is played, the call is disconnected or returned to the previous announcement.

Operation

Operation of an Announce Only Mailbox is automatic once configured.

Conditions

✓ There is no limit to the number of mailboxes that can be assigned as Announce only.

Programming

Keyset Admin.

STATION

- Announce Only Mailbox (PGM 127-button 3)
- Announce Only Option (PGM 127-button 4)

Web Admin.

STATION DATA

- VM Attributes➤Announce Only Mailbox, Announce Only Option

Related Features

- Integrated Auto Attendant/ Voice Mail

Hardware

2.48.3.3 Call Forward from VM

Description

If you are at a remote location, you can activate or deactivate Unconditional Call Forward from the integrated Voice Mail. Pressing “7” at the main VM menu will return the "Mailbox Set Forward" prompt.

Operation

To activate Call Forward while in the VM

1) Press “7”, for Mailbox set forward, the “Mailbox Set Forward” prompt is received.
2) Dial “1” and receive the “Forward” prompt (“Please enter the number to forward to ...").
3) Dial Station Number as follows:
   - To forward to another station, dial the station number.
   - To forward calls Off net, dial “*” and enter station speed number. If the station Speed bin is valid, confirmation announcement “forwarded to station (‘xxx’)” or “forwarded to speed bin number (yyyy)” is played.

To deactivate Call Forward

1) Press “7”, for Mailbox set forward, the “Mail Box Set Forward” prompt is received.
2) Dial “2” and receive the “Station forwarding is canceled” prompt.

To return to the Main menu

1) Dial “9” and receive the “Mailbox Settings” prompt.

Conditions

✓ Remote user dialing is subject to the inter-digit timer. If you do not input a response to the VM prompts in the inter-digit time, the call is released.

✓ The ability to activate Unconditional Call Forward from your Mailbox is only available when the Mailbox is accessed from a remote location.

Related Features

- Message Storage
- Message Retrieval
- Remote Message Retrieval
- Message Retrieval Options

2.48.3.4 Distribution Lists

Description

In your Voice Mailbox you can create a distribution list that will allow you to record and send or forward messages to all of the Mailboxes in the list.

You can create five (5) different Distribution Lists with up to 25 Mailboxes in each list.

Operation

To create a Distribution List

1) At the main Voice Mail prompt dial 5 for Personal Options and follow the voice instructions to create your list.

Conditions

✓ The maximum number of distribution lists you can create is five (5) and up to 25 mailboxes can be assigned in a distribution list.

Related Features

- Integrated Auto Attendant/Voice Mail
- Send a Message

Hardware

2.48.3.5 E-Mail Notification

Description

When the system stores a new voice message, it will check the notification settings for your mailbox. If E-mail notification is configured, the system will send you an E-mail notification. If configured, the voice message is attached as an *.wav file and the message may be erased from the mailbox since a copy has been delivered via e-mail.

Operation

System

System automatically sends e-mail to notify you of new voice message.

Conditions

- ✓ The E-mail is sent to the address assigned for the station with the “sender” address defined for the System. Note the latter is required, as many E-mail servers will reject anonymous e-mails.
- ✓ The E-mail address for the System and the station are defined under the Web Admin.
- ✓ To attach the Voice message to the E-mail notification, the Attach Message option (PGM 111– Button 24) must enabled.
- ✓ The “Sender E-Mail information (system E-mail account) is assigned in both the Station and System Data areas. If both are assigned, the Station assignment is used.

Programming

Keyset Admin.

STATION

- SMTP Mail Server Address (PGM113 – Button 18)
- Receiver Mail Address (PGM113 – Button 19)
- SMTP Mail Server ID (PGM113 – Button 23)
- SMTP Mail Server P/W(PGM113 – Button24-1)
- Delete Message After Add (PGM113 – Button24-5)
- SMTP Security (PGM113 – Button24-12)
- SMTP Port (PGM113 – Button24-13)
- Sender Mail Address (PGM113 – Button24-14)

BOARD

- VSF Sender Mail Address (PGM 132-Button 6)

SYSTEM

- VSF SMTP Port (PGM113 – Button24-13)

Web Admin.

STATIONDATA

- VM Attributes ➤VSF MSG – SMTP Mail Server Address
- VM Attributes ➤VSF MSG –Receiver Mail Address
- VM Attributes ➤VSF MSG –SMTP Mail Server ID
- VM Attributes ➤VSF MSG –SMTP Mail Server Password

- VM Attributes ➤VSF MSG – Attach Message
- VM Attributes ➤VSF MSG –Delete Message
- VM Attributes ➤VSF MSG –SMTP Security
- VM Attributes ➤VSF MSG –SMTP Port
- VM Attributes ➤VSF MSG –Sender Mail Address

Related Features

- Integrated Auto Attendant/Voice Mail

Hardware

2.48.3.6 Mark a Message as Private

Description

When you send or reply a message, you can mark the message as Private so that the receiver cannot forward the message.

Operation

To mark a message as private

1) After identifying the receiver, you will receive the send message options prompt.
“For regular delivery, press 1. To mark urgent, press 2. To mark private, press 3. To mark urgent and private, press 4. To request delivery receipt of the message for future, press 5.”
2) Dial “3” or “4” for Urgent and Private, the message is sent as a private message and you receive the “Your message has been sent”.

Conditions

✓ You must be permitted to mark a message as Private as part of the VM COS

Programming

Keyset Admin.

SYSTEM

- VM COS Attributes (PGM 253)

Web Admin.

SYSTEM DATA

- VM COS Attributes

Related Features

- Send a Message
- Distribution Lists

2.48.3.7 Mark a Message for Delivery Confirmation

Description

You can mark a message so that when the receiver listens to the message you are notified with a confirmation Voice message.

Operation

To mark a message for Delivery Confirmation

1) After identifying the receiver, you will receive the send message options prompt
“For regular delivery, press 1 to mark urgent, press 2 to mark private, press 3 to mark urgent and private, press 4 to request delivery receipt of the message for future, press 5.”
2) Dial “5”, the message is sent.
3) After the receiver listens to the message you receive a confirming Voice message with the Mailbox number that received the message and the time the message was received(listened to).

Conditions

✓ You must be permitted to mark a message for Delivery Confirmation as part of the VM COS.

Programming

Keyset Admin.

SYSTEM

- VM COS Attributes (PGM 253)

Web Admin.

SYSTEM DATA

- VM COS Attributes

Related Features

- Send a Message

Hardware

2.48.3.8 Message Cascade

Description

Message cascading copies or moves messages left for the originating mailbox to another mailbox. Once the message is copied or moved into the each mailbox, notification events can take place based on the mailbox settings. Multiple Mailboxes (up to 5 mailbox stations per a station) can be assigned as Cascade Mailbox destination.

1) Cascade Method: Decide Cascade Method
   A. Disable Cascade
   B. Copy Cascade
   C. Move Cascade
2) Cascade Message Type: Choose Message Type for Cascading
   A. All (Normal Message + Urgent Message)
   B. Normal Message Only
   C. Urgent Message Only
3) Cascade Timer: After Cascade Timer, Cascade is executed by sequentially form Cascade Mailbox 1 to Cascade Mailbox 5.
4) Cascade For Read Message
   Cascading stop or continue by following the message is checked by a user.
   A. Enable - regardless the message is read, cascading the message is continued.
   B. Disable - if the message is read, cascading the message is stopped.

Operation

Operation of a Cascade Mailbox is automatic once configured, with 2 options combination, Cascade way is decided.

Conditions

- ✓ Message Cascading is only allowed within programmed destination mailboxes in PGM 127, to protect unintentional cascade to other mailboxes.
- ✓ Same cascade mailbox number is not saved in PGM 127.
- ✓ Mailbox owner number is not saved in PGM 127.

Programming

Keyset Admin.

STATION

- Station VM Attribute (PGM 127)

Web Admin.

STATION DATA

- VM Attributes➤Cascade Mailbox 1~5
- VM Attributes➤Cascade Method
- VM Attributes➤ Cascade Message Type

- VM Attributes➤ Cascade Timer.
- VM Attributes➤ Cascade for Read Message.

Related Features

- Integrated Auto Attendant/Voice Mail

2.48.3.9 Message Retrieval

Description

You can access your Mail Box locally by placing a call to the integrated Voice Mail group or, from an iPECS IP or LDP Phone, by pressing the **[MESSAGE/CALLBACK]** button, or by pressing a **{VMAILBOX}** button, if assigned to the phone.

Prompts are then received to guide you in the Voice Mailbox operation. You must enter a Mailbox number, generally the 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.

If you enter valid and matching Mailbox and password numbers, the “Number of Messages" prompt (“You have xx new messages. You have yy saved messages.”) is received. At this point, you also receives the “VM long option prompt” (“To play new messages, press one, to play saved messages, press two, to set greeting or password, press eight, to disconnect, press pound, Press 0 for the operator, Press nine to hear this message again.”).

When you respond by dialing 1, the first new message is played. At the end of message playback, the “New Message option” prompt (“To replay message, press one, to listen to the next message, press two, to delete message, press three, to forward message, press four, to call the sender, press five, to skip message, press six, to return to main menu, press nine.”) is played. This process is repeated until the last new message is played and the "No Message" prompt (“No Messages”) is played.

When you dial 2 in response to the “Number of Messages” prompt, the oldest saved message is played. At the end of the message, the “Old Message option” prompt (“To replay message, press one, to listen to the next message, press two, to delete message, press three, to forward message, press four, to call the sender, press five, to return to main menu, press nine.”) is played. This process is repeated until the last saved message is played and the "No Message” prompt (“No Messages”) is played.

In addition to the options indicated in the prompt, you can record a memo, which is attached to the current voice mail by dialing the digit 7. The current voice mail and memo can then be forwarded to other iPECS users.

When you dial 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

iPECS IP & LDP Phones

*To assign a **{VMAILBOX}** Flex button*

- **[TRANS/PGM]** + **{FLEX}** + VM group + Mail-box (station) number + **[HOLD/SAVE]**

To retrieve Voice Mail locally

1) Lift the handset or press the **[SPEAKER]** button.
2) Press **[MESSAGE/CALLBACK]** button. The message contents summary is shown as below.

ST	CL	VS	VM	FS	MS
001	001	005	006	001	004

3) Dial digit “3” to select the integrated VM Messages to receive the “Mail Box & Password” prompts sequentially.
4) Dial the Mail Box and corresponding password to receive the “Number of Messages” prompt.
5) Dial desired option code.
6) At completion of session, hang-up to return to idle.

Or,

1) Lift the handset or press the **[SPEAKER]** button.
2) Press **{VMAILBOX}** button.
3) Dial the Mailbox password to receive the “Number of Messages” prompt.
4) Dial desired option code.
5) At completion of session, hang-up to return to idle.

To attach a memo to the current voice message

1) During or after the New or Old Message option prompt, dial “7”.
2) At the beep, record the 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.

To skip the date/time prompt while hearing

1) Dial ‘#’ while the date/time prompt is played.
2) System will stop playing the prompt and it will provide the message.

To hear the date/time prompt while hearing

1) Dial ‘0’ while the message is played.
2) System will provides data/time prompt of the message.

SLT(Single Line Telephone)

To retrieve Voice Mail locally

1) Lift the handset.
2) Dial the Voice Mail Group to receive the Mailbox number and Password prompts.
3) Dial the Mailbox and corresponding password to receive the “Number of Messages” prompt.
4) Dial desired option code.
5) At completion of session, hang-up to return to idle.

To attach a memo to the current voice message

1) During or after the New or Old Message option prompt, dial “7”.
2) At the beep, record the memo.
3) Dial * to stop and store the memo.
4) During or after the New Old option prompt, dial “4” to forward the message and memo.

To skip the date/time prompt while hearing

1) Dial ‘#’ while the date/time prompt is played.
2) System will stop playing the prompt and it will provide the message.

To hear the date/time prompt while hearing

1) Dial ‘0’ while the message is played.
2) System will provides data/time prompt of the message.

Conditions

- ✓ If no new/old messages are available, pressing “1” or “2”, is an invalid operation and you will receive the “Invalid Entry” prompt or “No Message” prompt.
- ✓ If the dialed number is not recognized, the "Invalid Entry" prompt is played. After the second invalid entry, you are disconnected.
- ✓ You may dial digits at any time during a voice mail playback, system prompt or silence. You must dial a digit in response to a system prompt within the CCR Analysis timer or the system will disconnect and return error tone.
- ✓ Messages can be retrieved in either a FIFO (First in First out) or LIFO (Last in First out) order to meet the desire of each user.
- ✓ If your Mailbox is configured with Auto Save New Message, after listening to the message, the message is stored and you must be expressly delete the message. To minimize the potential for large amounts of unwanted stored messages, Auto Save New message can be disabled then the message is deleted if not expressly saved.

Programming

Keyset Admin.

STATION

- Retrieve Message Order (PGM 113-Button 13)
- Auto save new message (PGM 161-Button 24-11)

Web Admin.

STATION DATA

- VM Attributes ➤Retrieve MSG Method

SYSTEM DATA

- System Attributes ➤Auto Save New Message

Related Features

- Message Retrieval Options
- Remote Message Retrieval
- Multiple Voice Mailbox Support

Hardware

2.48.3.10 Message Retrieval Options

Description

You 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 later may occur when you selects Message Retrieval Option 1/2 (Play New/Saved Message), 7 (Cancel or Forward message, for Remote Access Only) or 8 (Mail Box settings). The "VM long Options" prompt is:

"To play new messages, press one, to play saved messages, press two, to set station forwarding, press seven (This option is available only for remote access), to set greeting or password, press eight, to disconnect, press pound, Press 0 for the operator, Press nine to hear this message again."

The integrated Voice Mail will respond to incoming digits as shown in the following table.

Station VM Feature Usage: Off

Digit	Function
1	Play New message
2	Play Saved message
7	Set Cancel/ Fwd, available only for remote access
8	Mailbox Setting, "Mailbox Settings" prompt
9	VM Long options
#	Drop, "Goodbye"
0	Attendant Call, Call to System Attendant.

Station VM Feature Usage: On

Digit	Function
1	Play New message
2	Play Saved message
3	Play urgent message
4	Send a message
5	Personal options
6	Administrator options (administrator only)
7	Set Cancel/ Fwd, available only for remote access
8	Mailbox Setting, "Mailbox Settings" prompt
9	VM Long options
#	Drop, "Goodbye"
0	Attendant Call, Call to System Attendant.

Operation

To access a Message Retrieval option

✓ At any time after the "Number of Messages" prompt, dial the digit to play a message. The system initiates the selection providing any needed prompts.

Conditions

- ✓ You must begin dialing within the CCR Analysis timer in response to a system prompt. If the timer expires, the system will disconnect the call and you will receive error tone.
- ✓ When the “call sender” option results in an external call, dialing restrictions will be applied based on your Station COS (Class of Service).
- ✓ If you remain off-hook after a call placed through the Voice Mail is complete, you will be returned to the previous place in the Voice Mailbox. If you hangs up, the integrate Voice Mail will recall you and, upon answer will play “Request Mailbox Number” prompt.
- ✓ If your Mailbox is configured with Auto Save New Message, after listening to the message, the message is stored and you must be expressly delete the message. To minimize the potential for large amounts of unwanted stored messages, Auto Save New message can be disabled then the message is deleted if not expressly saved.

Programming

Keyset Admin.

SYSTEM

- Auto Save New Message (PGM 161-Button 24-11)

Web Admin.

SYSTEM DATA

- System Attributes ➤Auto Save New Message, Station VM Feature Usage

Related Features

- Message Retrieval
- Remote Message Retrieval
- Voice Mailbox Settings
- Class of Service

2.48.3.11 Message Storage

Description

When you activate Call Forward to the integrated Voice Mail, calls are transferred to your mailbox or recall the Auto Attendant, the call is handled by the iPECS Voice Mail application. The caller connects to your User Greeting followed by beep tone to notify the caller to leave a message.

The caller can record a message and hang-up or dial “*” for further options. At disconnect, the VM application stores the message in your Voice Mailbox and activates the Message Waiting indication at your station.

Operation

Remote Caller

To leave a voice message after Greeting and beep

1) State you message.
2) Hang up to quit recording or dial “*” for further options.

Conditions

- ✓ Two timers are provided to control Voice Message duration. The Valid User Message Timer establishes the minimum Voice Message duration. Voice Messages shorter than this timer are not stored. The User Record Timer establishes the maximum Voice Message duration. When the User Record Timer expires while recording a Voice Message, confirmation tone is heard and the message is saved for destination station.
- ✓ If all the AA/VM channels are in use, Ring Back tone is provided until a channel is available.
- ✓ All stations including a SLT(Single Line Telephone) can leave and receive voice messages.
- ✓ Individual User Greetings and Voice Mails are protected from loss of AC power.
- ✓ With the optional second Voice Store Forward Gateway, System announcement should be recorded in both the main and second gateway. Also, each station and Line Voice Mailbox should be associated with only one of the gateways.

Programming

Keyset Admin.

STATION

- VSF Access (PGM 113-Button 2)
- Direct Transfer to Mailbox (PGM 120-Button 6)
- Station Call Forward Timer (PGM 123)

SYSTEM

- Call Forward No Answer Timer (PGM 181-Button 1)
- VSF User Record Timer (PGM181-Button 3)
- VSF Valid User Message Timer (PGM 181-Button 4)

Web Admin.

STATION DATA

- VM Attributes ➤ VSF Access

- Preset Call Forward ➤Transfer to Mail Box
- Station Timer ➤Station Forward No Answer Timer

SYSTEM DATA

- System Timers ➤Call Forward No Answer Timer,
- VSF User Maximum Record Timer, VSF Valid User Message Timer

Related Features

- Call Forward
- Station Message Wait/Call Back
- Integrated AA/VM Overview
- Call Transfer, Voice Mail

Hardware

2.48.3.12 Outbound Message Notification

Description

When a new message is received, the system will check the new message notification settings.

If configured to notify your Mobile extension, the system will place a call to your mobile phone and, when answered, play the station number prompt followed by the new message prompt, "You have xx new messages." The new message prompt indicates the number of unheard messages.

After the new message prompt you will be prompted to enter your password, which will access your mailbox. You must listen to the new messages to confirm the notification.

If you take no action within the CCR Inter-digit timer, do not answer, the system receives busy or you hang-up, the call is disconnected and the system will retry the call after the retry timer expires.

The system will attempt to complete the notification until you listen to the message or the number of attempts reaches the DISA Retry counter.

Operation

To receive and confirm the notification

1) Answer Notification call and receive the New messages and password prompt.
2) Enter your Mailbox password.
3) Listen to the new messages.

Condition

- ✓ Caller Id will be the external caller who left the message or, for messages from another station, Caller Id will be the station receiving the message.
- ✓ If notification is disabled, any existing notification will be terminated after the initial notification call.
- ✓ For proper operation, the Station COS (Class of Service) and CO Group access for the station must be such as to allow the notification call.
- ✓ The Mobile telephone number to notify is assigned in PGM 236.
- ✓ The system will continuously attempt to seize an outside Line for the notification call until successful, if all lines in the assigned Line group are busy.
- ✓ The Retry counter is incremented after the system accesses the outside Line for notification.
- ✓ The Retry count is from 1 to 9; the retry interval is from 1 to 3 minutes.
- ✓ If a new message is logged before answer of the notification call, the message will be available to the user and a new notification is not invoked. If a new message is received after answering the notification call, the System will invoke another notification call. The user will receive the notification after returning to idle.
- ✓ An analog CO Line may be used for the notification however, notification over these Lines (LCO Lines) must be enabled and several timers must be set to compensate for the lack of answer and disconnect signals from an analog CO Line. The system will treat the call as answered after VM Notify Play Delay Over LCO Timer then play the new message prompt. The system will wait for you to enter a password for the VMIB Group Dial Time Out. You must enter the correct password or the call will be disconnected.

✓ If you enter an incorrect password, the system will allow you to reenter a correct password. You may retry entry for the DISA Retry Counter.

Programming

Keyset Admin.

TABLES

- Hunt Call Enable (PGM 236-Button 6)
- VSF Notify (PGM 236-Button 7)
- Notify Retry Count (PGM 236-Button 8)
- Notify Retry Interval (PGM 236-Button 9)

SYSTEM

- VM MEX Notify over LCO (PGM160-Button24-13)
- Notify Play Delay (PGM181-Btn18)

Web Admin.

TABLES DATA

- Mobile Extension Table ➤Hunt Call Enable, VSF/VMIM Notify, Notify Retry, Retry Interval

SYSTEM DATA

- System Attributes ➤VM Notify to Mobile Extension over CO
- System Timers ➤VM Notify Play Delay Over CO Timer

STATION GROUP DATA

- Station Group Attributes➤VSF Group Attributes ➤Dial Time Out

Related Features

- Mobile Extension
- Message Retrieval
- Remote Message Retrieval
- Message Retrieval Options

Hardware

2.48.3.13 Personal CCR/User Greeting DISA

Description

Your integrated Voice Mail User greeting can have an associated Personal CCR Table configured by your system Administrator. While listening to your greeting, internal or outside callers may dial a digit that that correlates to a destination in the Personal CCR Table.

When the digit is dialed, the caller is routed to the defined destination.

The available destinations, shown below, are the same as the system CCR tables employed for Auto Attendant.

- Station
- Station group
- Speed Dial number
- Conference Room
- Conference Group Join
- Page Zone
- Voice Mail
- System Announcement
- Company Directory
- Networking station number
- System VM greeting

In addition, the system will monitor next following digits for a station number.

If the user dials a station number, the system will transfer the caller to the station.

If [CCR Table Usage] option set to OFF, CCR table is not worked, and next following digits are available, and follow system DISA call flow.

Operation

Outside Caller or Internal Caller

To use Personal CCR

1) Call or be transferred to a Station with Personal CCR active.
2) During or within the CCR analysis timer, dial a digit from the User Greeting instructions.
3) The outside caller routes to the CCR destination or to the DID/DISA destination if the CCR destination is unavailable.
4) The internal caller routes to the CCR destination or error tone is served if the CCR destination is unavailable.

Conditions

- ✓ The caller can dial “*” or “#” to replay your greeting.
- ✓ If the caller dialed digit does not match an entry in the Personal CCR table the caller routes based on the numbering plan or DID/DISA Destination in System Data admin.

- ✓ If the destination defined in the CCR Table for the caller dialed digit is not available (Busy, No-answer or error) the call is sent to the appropriate destination defined by the DID/DISA Destination under System Data.
- ✓ Personal CCR use must be enabled, you must record your Voice Mail User Greeting and your calls must forward to your integrated Voice Mailbox for this feature to function properly.
- ✓ No retry count is applied to Personal CCR; the caller can attempt to enter a matching digit once. The caller routes to the DID/DISA Destination should the user dial an erroneous digit.
- ✓ The priority of Personal User Greeting is as followings.
  - Personal CCR
  - Outgoing Mailbox Destination
  - User Greeting DISA
- ✓ For internal call, the destination Company Directory / Record VM Greeting is not supported.
- ✓ For internal call, if the destination is VSF Announcement, the caller can dial by following system numbering plan, it does not follow System CCR(PGM 228).
- ✓ Although Announce Only mail box, Personal CCR is operated.

Programming

Keyset Admin.

STATION

- Station CCR Table (PGM 128)

SYSTEM

- CCR Interval Digit Timer (PGM 180-Button 13)

Web Admin.

STATION DATA

- Station Personal CCR

SYSTEM DATA

- DID/DISA Destination
- System Timers ➤CCR Inter Digit Timer

Related Features

- Integrated Auto Attendant
- Call Forward
- DISA

Hardware

2.48.3.14 Multiple Mailbox Greeting

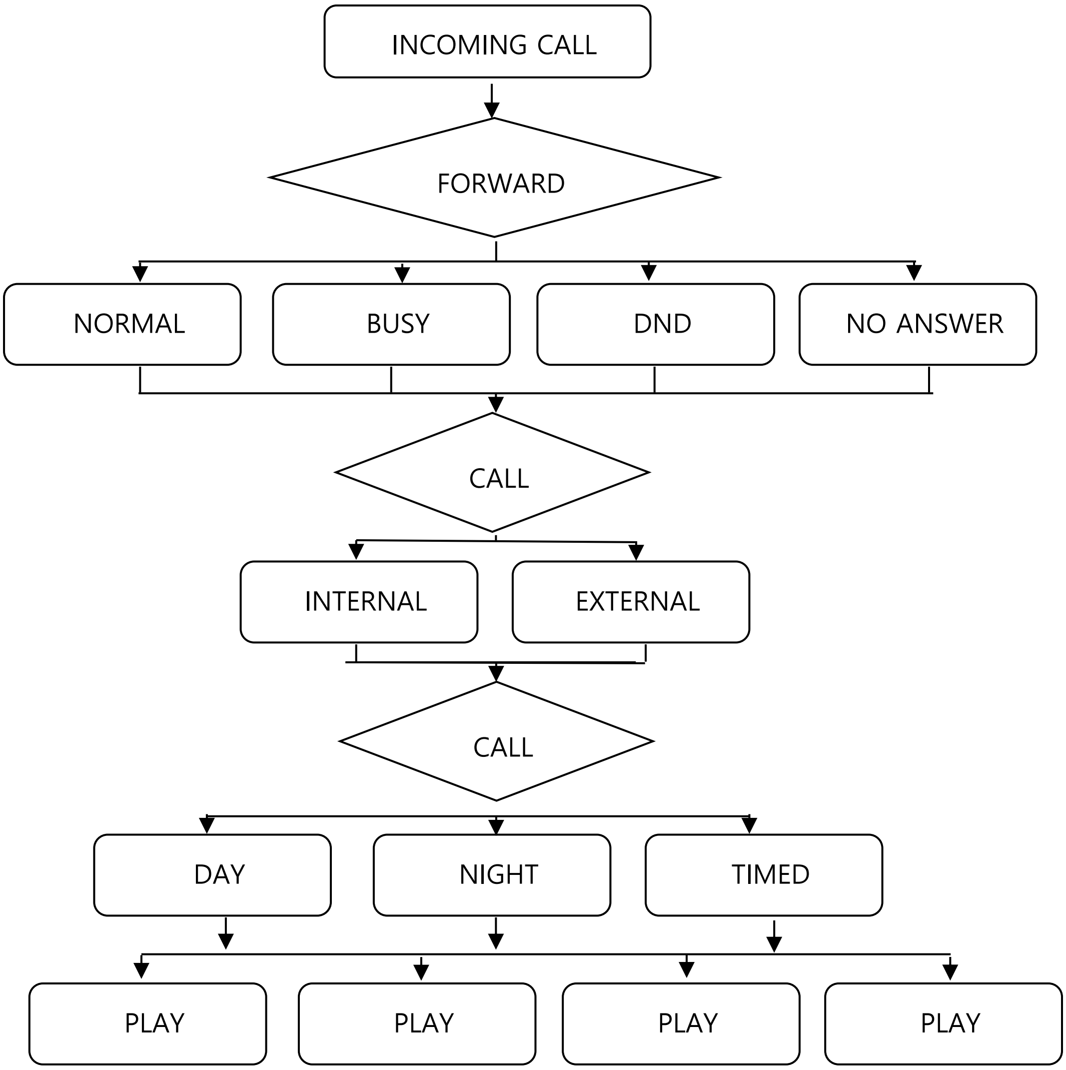
Description

System provides maximum 4 greeting for each mailbox user.

Each user can select greeting according to the following scenario in station program.

Operation

FLOW CHART



To change each mailbox greeting with station program

1) Press **[PGM]** button and dial 56.
2) Select case type 1-4, 0 (1: Unconditional, 2: Busy, 3 : No answer, 4: DND, 0:All).
3) Select Calling Party Type 1- 2, 0 (1: Internal, 2: External, 0: All).
4) Select D/N/T 1-3, 0 (1: Day, 2:Night, 3:Timed, 0:All).
5) Select greeting number 1-4.
6) Press **[SAVE]** button.

To change whole mailbox greeting with station program

1) Press **[PGM]** button and dial 56.
2) Select case type 0 (1: Unconditional, 2: Busy, 3 : No answer, 4: DND, 0:All).
3) Select greeting number 1-4.
4) Press **[SAVE]** button.

To change mailbox greeting type with station program

1) Press **[PGM]** button and dial 56.
2) Select case type 1-4, 0 (1: Unconditional, 2: Busy, 3 : No answer, 4: DND, 0:All).
3) Select 0 (1: Internal, 2: External, 0: All).
4) Select greeting number 1-4.
5) Press **[SAVE]** button.

To change mailbox greeting type, call type with station program

1) Press **[PGM]** button and dial 56.
2) Select case type 1-4, 0 (1: Unconditional, 2: Busy, 3 : No answer, 4: DND, 0:All).
3) Select 0 – 2 (1: Internal, 2: External, 0: All).
4) Select greeting number 1-4.
5) Press **[SAVE]** button.

To record user greeting

1) When a user enter the mailbox, the following menu can be provided.
   Main Menu
   - to play new messages, press one
   - to play saved messages, press two
   - to play urgent messages, press three
   - to send a message, press four
   - for personal options, press five
   - to access administrative options, press six
   - to set station forwarding, press seven
   - to set greeting or password, press eight
   - to disconnect, press pound
   - Press 0 for the operator
   - Press nine to hear this message again.
2) After pressing 8, the options that can be performed from the mailbox are:
   - to edit your greeting, press one
   - to edit you password, press two
   - to return to main menu, press nine
3) Press 1 to edit your greeting.
   - to listen to your current greeting, press five
   - to record a new greeting, press seven
   - to return to the main menu, press nine

4) Press 7 to edit your greeting.
   - Enter greeting

5) Select greeting number 1 to 4.
   - at the tone, record your greeting
   - press pound(#) when you have finished

Conditions

- ✓ First greeting can be provided as default.
- ✓ The following common prompt will be played if there are no user greeting recorded.
  - Prompt 601 is played if there is no User Greeting 1.
  - Prompt 602 is played if there is no User Greeting 2.
  - Prompt 603 is played if there is no User Greeting 3.
  - Prompt 604 is played if there is no User Greeting 4.
  - Prompt 80 is played if Prompt 601~604 are not loaded.

Programming

Web Admin.

SYSTEM DATA

- System Attributes (160~161) ➤ Station VM Feature Usage

Hardware

2.48.3.15 Remote Message Retrieval

Description

You can access your mailbox from a remote location. You will be required to enter your mailbox number then normal mailbox operation applies. The system will require configuration for a remote access telephone number.

Operation

Remote Caller

To access Voice Mailbox from a remote location

1) Call the telephone number for remote access to the integrated Voice Mail.
2) Upon answer, dial "#" to receive the "Request for Mail Box number" prompt.
3) Enter your Mailbox number, typically your station number.
4) Follow local access procedures.

Conditions

- ✓ The conditions associated with Message Retrieval and Message Retrieval Options apply.
- ✓ The conditions associated with DISA apply.

Related Features

- Message Retrieval Options
- Integrated Auto Attendant
- Message Retrieval

Hardware

2.48.3.16 Send a Message

Description

You can record and send messages to other Mailboxes or distribution list from your Voice Mailbox.

Operation

To send a message to another Mailbox

✓ At the main Voice Mail prompt, dial “4” and follow the voice instructions to record and send you message.

Related Features

- Integrated Auto Attendant/Voice Mail
- Distribution Lists
- Mark a Message as Private
- Mark a message for Delivery Confirmation

Hardware

2.48.3.17 Voice Mail Class of Service

Description

The system employs a Voice Mailbox Class of Service to provide control and management of Mailboxes. There are five VM Classes of service used to define the COS characteristics listed below.

- Greeting length: 00-99 seconds
- Message length: 000- 600 seconds
- Number of messages: 00-250
- Retention time: 00-99 days
- E-mail notification
- Future delivery messages
- Confirm message receipt
- Private message mark

Operation

Operation of a Voice Mailbox COS is automatic once configured.

Conditions

✓ The default Class of Service for all mailboxes is 1.

Programming

Keyset Admin.

SYSTEM

- VM COS Attributes (PGM 253)

Web Admin.

SYSTEM DATA

- VM COS Attributes

Related Features

- Integrated Auto Attendant/Voice Mail

Hardware

2.48.3.18 Voice Mailbox Settings

Description

You can program the Mailbox settings for your mailbox including a security password and record a greeting. When you press “8” while retrieving messages, the "Mailbox Setting" prompt, (“To edit your greeting, press one, to edit your password, press two. To return to main menu, press nine”) is played.

Operation

To program Mailbox settings while “in” the Voice Mailbox

1) Press “8”, for Mailbox settings, the “Mailbox Setting” prompt is received.

To edit your greeting

1) Dial “1” and receive the “Greeting Option” prompt (“To listen to your current greeting, press five, To record a new greeting, press seven, To return to the main menu, press nine.”).
2) Dial “5”, to hear your current greeting.
3) Dial “7” to record a new greeting (“At the tone, record your new greeting, press # when you have finished”.)

To edit your password

1) Dial “2” and receive the “Password Entry” prompt (“Please enter your new password and press pound when finished.”).
2) Dial new password.
3) Press “#” and receive the “Reenter Password” prompt (“Please re-enter your password to confirm and press pound when finished.”).
4) Dial new password again.
5) Press “#” and receive “Password Confirmation” prompt (“Your password is saved.”).

To return to main menu

1) Dial “9” to return to the main VM menu.

Conditions

- ✓ If you are accessing your Voice Mail remotely, you must begin dialing within the CCR Analysis time, if not the call is released.
- ✓ If the dialed number is not recognized, the “Invalid Entry” prompt is played.
- ✓ You must assign a password (Authentication code up to 12 digits) before access to the mailbox will be allowed. Note that a greeting need not be recorded.

Related Features

- Message Storage
- Message Retrieval
- Remote Message Retrieval
- Message Retrieval Options

2.48.4 System Voice Memo

Description

The integrated AA/VM memory incorporates Voice Memos for the system time and date as well as station number, and settings that can be played to your station upon request.

Operation

iPECS IP & LDP Phones

To hear the Date & Time Prompt

1) Dial “661”, the Play Time and Date code. The system plays the date and time Voice memo.

To hear the Station Number Prompt

1) Dial “662”, the Play Station number code. The system plays the Station Number voice memo.

To hear the Station Settings

1) Dial “663”, the Play Station Settings code. The system plays the Station setting, reporting the following items:
   - Station Number
   - Station IP Address
   - Station Mac Address
   - Station ICM Mode (Handsfree/Tone/Privacy)
   - Listed message x (x: number of all message waiting)
   - Wake-Up Time (hh:mm)
   - Do not disturb
   - Forwarded to station xxx
   - Forwarded to speed bin xxx
   - Queued Line xx
   - Locked (temporary COS (Class of Service) change)
   - COS (Class of Service) x

SLT(Single Line Telephone)

To hear the Date & Time Prompt

1) Lift the handset.
2) Dial “661”, the Play Time and Date code. The system plays the date and time Voice memo.

To hear the Station Number Prompt

1) Lift the handset.
2) Dial “662”, the Play Station number code. The system plays the Station Number voice memo.

To hear Station Settings

1) Lift the handset.
2) Dial “663”, the Play Station Settings code. The system plays the Station setting, reporting the following items:
   - Station number

- Station IP Address
- Station Mac Address
- Station ICM Mode Handsfree/Tone/Privacy)
- Listed message x (the number of all message waiting)
- Wake-Up Time (hh:mm)
- Do not disturb
- Forwarded to station xxx
- Forwarded to speed bin xxx
- Queued Line xx
- Locked (temporary COS (Class of Service) change)
- COS (Class of Service) x

Conditions

✓ For station status, items from “Listed message x” to “COS x” are not include unless active.

Programming

Keyset Admin.

STATION

- VSF Access (PGM 113-Button 2)

Web Admin.

STATION DATA

- VM Attributes ➤VSF Access

Related Features

- Integrated Auto Attendant/Voice Mail

Hardware

2.49 IP Bridge Mode

Description

UCS client Soft phones can automatically bridge the audio for a call to a “linked pair”Station.When the Soft phone places a call in the bridged mode, the audio for the call can delivered to theiPECS IP or LDP Phoneby simply lifting handset or pressing the **[SPEAKER]** button.

Operation

iPECS IP & LDP Phones

To use the IP Bridge mode

1) Place a call from Soft phone. The Soft phone must be inthe Bridged mode.
2) Lift the handset of the linked iPECS IP or LDP Phone to receive audio.

Conditions

✓ The Soft phone must be used local to the bridged iPECS IP or LDP Phone.
✓ Bridged operation must be selected at the Soft phone.
✓ The Soft phone must be linked to the associated iPECS IP or LDP Phone.

Programming

Web Admin.

STATION DATA

- Linked Station

Related Features

- Linked Station Pairs

Hardware

2.50 IP FAX Relay, T.38 Support

Description

Because of their nature, Fax tones do not transmit well through IP networks, particularly when compression is employed.

To address this, the external network interfaces of the system support T.38 protocol that defines the translation of analog fax tones to digital signals.

When Fax tone is detected on a port, the system will activate a T.38 Fax relay channel to the appropriate port.

Operation

Operation of this feature is automatic.

Programming

Web Admin.

Board Based Data

- Board Base Attributes ➤T.38 Enable, T.38 Port Usage

Hardware

- SLT(Single Line Telephone) port with FAX machine connected

2.51 IP System DECT

Description

The iPECS supports office building mobility employing Digital Enhanced Cordless Technology(DECT). Ericsson-LG Enterprise's Multi-channel DECT Base stations connect to the Wireless Interface board of the system to establish cells. With theDECT handsets, you can roam and maintain an uninterrupted communications link toiPECS features and resources.

For further information on installation and operation of the IP System DECT solution, refer tothe DECT Installation Manual.

iPECS eMG100 doesn't support this Terminals.

Operation

DECT operation is automatic when configured.

Conditions

✓ The maximum number of DECT phones (desktop or portable handsets) is 48 (eMG80) and 192 (eMG800/UCP).

Programming

Keyset Admin.

DECT

- DECT Registration (PGM 491-Web only)
- DECT Attributes (PGM 491)

Web Admin.

DECT DATA

- DECT Registration
- DECT Attributes

Hardware

- WTIB4/WTIB8 (eMG80/eMG800), WTIM4/8 (UCP)
- GDC-600BE Base stations
- GDC-450H/480H/500H handsets

2.52 IP TRANS-CODING

Description

The system employs the IEEE g.711, g.729, g.722 or g.723 codec to digitize and compress voicesignals for RTP packets between iPECS devices. iPECS terminals and gateway Modulesincorporate DSP functions to support codec conversion.

Available VoIP channelsinclude DSP circuitryused to support trans-coding (converting) codecs for incoming VoIP calls to devices such asthe VoIP channelsand the VSF channels, which have no built-incodec.

The VoIP DSPs will trans-code the incoming voice codec to the systemcodec and reverse the process for the outgoing packets.

When the DSP must implement a complextrans-coding operation (translation between g.729, g.723 or g.722), two (2) DSP channels are required. In all other cases, trans-coding onlyrequires a single channel per call.

Conditions

- ✓ The system codec toward the VoIP channel can be changed anytime during an IP call.
- ✓ The VoIP DSP can generate and detect in-band DTMF and Call Progress tones in support of DISA functionality.
- ✓ For complex trans-coding (g.723/g.729), two (2)VoIP DSP channels are required.
- ✓ If there are no available channels when trans-coding is required, the VoIP gateway will release call control.

Related Features

- System Networking
- Remote Devices

Hardware

- Built-in VoIP channels, VOIB (eMG)
- Built-in VoIP channels, VOIM8 or VOIM24 (UCP)

2.53 IP TRUNKING

2.53.1 H.323 v4 Service

Description

When assigned to support H.323 protocol, VoIP channels provide protocol conversion between H.323 v4 and the iPECS protocol. 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 are supported employing H.450.1 ~ H.450.12 standards, which define the following supplementary services:

- H.450.1: Supplementary Services Framework
- H.450.2: Call Transfer
- H.450.3: Call Diversion
- H.450.4: Call Hold
- H.450.5: Call Park & Pick-up
- H.450.6: Call Waiting
- H.450.7: Message Waiting
- H.450.8: Name Identification
- H.450.9:Call Back
- H.450.10: Camp-On
- H.450.11: Intrusion
- H.450.12: Additional Information

Operation

System

Operation of H.323 Service is automatic.

Programming

Keyset Admin.

STATION

- Station IP Attributes (PGM 122)

BOARD

- H.323 VoIP Attributes (PGM 130)

CO/IP

- CO/IP Line Group (PGM 141-Button 1)
- CO VoIP Mode (PGM 141-Button 11)

H.323

- IP Bind (PGM326-Button1)
- H.323 CO Group Attributes (PGM327)

- H.323 Routing Table (PGM328)

Web Admin.

STATION DATA

- Station IP Attributes

BOARD BASEDDATA

- H.323 VoIP Attributes

CO LINE DATA

- Common Attributes ➤CO/IP Group
- VoIP Attributes ➤CO VoIP Mode

H.323 DATA

- H.323 Basic Attributes
- H.323 CO Group Attributes
- H.323 Incoming route table

Related Features

- System Networking
- SIP Trunk Service

Hardware

- IP Line

2.54 LBC (Loud Bell Control)

Description

The iPECS hardware is equipped with relays that activate External Control Contacts. Each 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 ring in Day Ring mode (LBC)
- UNA (Universal Night Answer) outside line receives a call while in the Night or Timed Ring mode.

Operation

System

Operation of the relays is automatic when defined.

Conditions

- ✓ Two contacts are available, one each appearing in the Miscellaneous contacts of the KSU and the EKSU.
- ✓ When assigned to activate as a Loud Bell Control, outside Line ring, transfers, and tone ring Intercom calls to the assigned station will activate the contacts.
- ✓ The contacts are rated at 1 amp, 24 VDC.
- ✓ While in the Night or Timed Ring mode, the LBC function is not activated by ringing at the assigned station. If the system is in the Night or Timed ring mode with External Night Ring enabled, an incoming call on an outside Line assigned with UNA will activate the contact. This permits operation of an external night ring device.
- ✓ vUCP is not available for External Control Contacts.

Programming

Keyset Admin.

CO/IP

- Universal Night Answer (PGM 141-Button 7)

SYSTEM

- External Night Ring (PGM 160-Button 6)
- External Contact Control (PGM 168)

Web Admin.

CO LINE DATA

- Common Attributes ➤Universal Answer

SYSTEM DATA

- System Attributes ➤External Night Ring
- External Control Contacts

Related Features

- Universal Night Answer
- Door Open

Hardware

- External Control Contact connected to an external loud bell.

2.55 LCR (Least Cost Routing)

Description

The LCR Tables are employed to define appropriate routing for outgoing calls based on thedialed number. Generally, the LCR Tables are structured to define the Least Cost Route forLong Distance calls.

Your dialed digits are compared to table entries and modified based on time of day, day ofweek, and assigned routes. There are four LCR Tables: LCR Control Attributes, LCR LeadingDigit Tables, LCR Digit Modification and LCR Initialization Tables.

LCR Access modes are assigned in the LCR Control Attributes Table. These modes define themanner in which you access the LCR function. LCR may be disabled or one or more ofthe three access modes can be allowed to access LCR. The basic modes are:

Internal LCR

If you dial digits that match an Internal LCR code in the LDT (Leading Digit Table),an outside Line is selected and digits are modified using the DMT (Digit ModificationTable).

Loop LCR

If you dial 9 (the 1st available Line Group access code) or presses a **{POOL}** or**{LOOP}** button to place a call, the Loop LCR mode is accessed. Dialed digits arecompared to the Loop LCR codes in the LDT; the system will seize an outside Line fromthe assigned Line Group and sends digits from the DMT.

Direct CO LCR

If you select a **{LINE}** or **{LINE GROUP}** button, the Direct CO LCR mode isaccessed. If the dialed number matches a Direct LCR code in the LDT and theseized Line belongs to the assigned Direct CO LCR group, the system will senddigits modified based on the DMT.

Operation

System

Operation of LCR is automatic based on assignments in the system's LCR Tables.

Conditions

✓ There are a total of 6 LCR access mode combinations that can be defined as below:
- LCR Access Mode 00 (M00) – LCR is disabled.
- LCR Access Mode 01 (M01) – Loop LCR access is active.
- LCR Access Mode 02 (M02) – Loop and Internal LCR access are active.
- LCR Access Mode 11 (M11) – Direct CO and Loop access are active.
- LCR Access Mode 12 (M12) – Direct CO, Loop and Internal LCR access are active, seize Co line according to the LCR attributes.
- LCR Access Mode 13 (M13) – Direct CO, Loop and Internal LCR access are active, seize CO line according to the station attributes.

✓ Multiple Leading Digit Tables can be configured for each station and outside Line.

✓ Leading digit entries may be duplicated in the Leading Digit Table. Using a different "LDT Index" will make each entry unique. The system will use the lowest matching entry.

- ✓ When Direct CO LCR is used on an ISDN line, an ISDN information message with the Called party Info Element, which includes only the numbering plan and numbering type, is sent to the ISDN to maintain the ISDN connection while the user finishes dialing and the system modifies the digits.
- ✓ For Direct CO LCR, the number of digits for the LDT Table should be programmed considering the dial-tone time-out of the network.
- ✓ Since an outside Line is connected, Direct CO LCR does not support the Alternative DMT index, which allows the system to select a second or alternative outside Line to place the call.
- ✓ If the LCR CO group is not assigned, the system will not seize an outside Line and will make internal call.
- ✓ If the LCR CO group is assigned as the unused group, the system will seize an outside Line according to the station attributes.
- ✓ The system selects the LDT main index from {LDT Table Index} of Common Attributes(111) and the system compares the dialed data with {Compared Digits} of sub index tables which have same {LDT Zone number} with {LDT Zone number} of Common Attributes(111).

Programming

Keyset Admin.

TABLES

- LCR Control Assignment (PGM 220)
- LCR LDT Table (PGM 221)
- LCR DMT Table (PGM 222)
- LCR Table Initialization (PGM 223)

Web Admin.

STATION DATA

- Common Attributes ➤LDT Table Index
- Common Attributes ➤LDT Zone Number

TABLES DATA

- LCR Control Attributes
- LCR LDT
- LCR DMT
- LCR Table Initialization

Related Features

- Outside Line Access
- Outside Line Groups
- ISDN (Integrated Service Digital Network)
- Station Flexible Buttons

Hardware

2.56 Linked Station Pairs

Description

A station can be logically linked to a primary station so that the two stations function as a singlestation. When linked, the two stations effectively act as a single station with the stationattributes of the primary station.

The status of one station is reflected in the status of the otherand features activated at one are active at the other. All internal or external calls to a linked pairstation will ring both stations.

All features available to the primary station are available and controllable by the secondarystation, one station may activate Call Forward and the other may cancel the forward.

Thedisplays of the linked stations will display the status of the linked pair. When one is busy, thedisplay of the linked station will be as shown below.

IN USE AT LINK STA

Operation

System

Operation of Linked pairs is automatic when defined.

Conditions

- ✓ Support for Linked Pairs is available with the iPECS IP Phones and UCS client Soft phones and WIT 400HE WiFi phones only.
- ✓ Intercom calls to the Linked stations always signal in the Tone ring mode and cannot be changed using the Caller Controlled ICM Signaling feature.
- ✓ Linked pair stations are treated as having a single station number for all features including LCD displays, station programming, ADMIN access, ACD statistics, SMDR, etc.
- ✓ The station attributes of the secondary station will follow attributes of the primary station, i.e. Day/Night COS (Class of Service), CO Warning Tone, CO Auto Hold, CO Call Drop, DID Call Waiting, Speed Access, Alarm, Voice Mail Access, DND, FWD, Paging, outside Line Access, outside Line Ring Assign, etc.
- ✓ If one station of a Linked pair is busy, the other station of the linked pair is also considered as busy, thus use of the linked station to place a call is not supported.
- ✓ A station can be linked to another station without registration to the system. This allows a station to be linked without affecting the overall capacity of the system. In this case, only an iPECS IP or LDP Phone, or SLT(Single Line Telephone) can be used as the unregistered linked station. In other cases, the linked station must be registered with the system, reducing the system capacity by one.
- ✓ The IP Attendant application can be linked as the secondary phone only.
- ✓ A SIP Phone cannot be part of a linked pair.
- ✓ Linked pair station cannot call the other member of the pair. It is possible to place a call on hold at one and pick-up the call at the other member.

Programming

Keyset Admin.

SYSTEM

- Linked Station Pairs Table (PGM 124)

Web Admin.

STATION DATA

- Linked Station

Related Features

- Intercom Caller Controlled ICM Signaling

2.57 LNR (Last Number Redial)

Description

The last number dialed is stored in the Station's Last Number Redial buffer. You may request the system redial the last dialed number without the need to dial the number.

Operation

LDP/LIP Phone

To use Last Number Redial

1) Press the **[SPEED]** button.
2) Dial "*".

SLT(Single Line Telephone)

To use Last Number Redial

1) Lift the handset.
2) Dial "*552", the SLT(Single Line Telephone)Last Number Redial code.

To make redial using the {Redial} button

1) Press the **{REDAIL}**button.
2) Use the Navigation button or [VOL UP]/[VOL DOWN] to select the desired number.
3) Press the **[HOLD/SAVE]**or **{REDAIL}** button to place the call.

*To assign a **{Redial}** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + "97" + **[HOLD/SAVE]**

Conditions

- ✓ When the outside Line used for the original call is busy, the system will select an idle Line from the same Line group to place the call.
- ✓ Using Last Number Redial will cancel Automatic Called Number Redial if active.
- ✓ Manually dialing a Flash during an outgoing call will cause only those digits dialed after the Flash to be stored in the LNR buffer.

Related Features

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

Hardware

2.58 Mobile Extension

Description

Your mobile phone may be registered to a station allowing the mobile phone to place and receivecalls through the system.

DID calls are sent to youriPECS IP or LDP Phone and yourregistered mobile phone simultaneously. If the mobile phone is paired with a Station groupstation, Station group calls routed to the station can also ring to your mobile phone.

From your mobile phone you may access the facilities of the iPECS to place internal and externalcalls as well as activate and access features.

To access system facilities and resources, you call your DID number from your mobile. When the call is received, thesystem matches the Calling Line ID (CLI) to the mobile phone and provides you with system dialtone.

You may be permitted to register and activate a mobile phone for the Mobile Extension feature.

Operation

iPECS IP LDP Phones

To assign a Flex button to Register a Mobile

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]**+ “37” + **[HOLD/SAVE]**

To assign a Flex button to activate Mobile Extension

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “38” + **[HOLD/SAVE]**

To register the mobile phone number

1) Press the **[TRANS/PGM]** button.
2) Dial “37”.
3) Dial the mobile phone number.
4) Press the **[HOLD/SAVE]** button.

To activate a registered mobile phone from the user’s station

1) Press the **[TRANS/PGM]** button.
2) Dial “38”.
3) Dial digit “1” to activate, “0” to deactivate.
4) Press the **[HOLD/SAVE]** button.

To place a call from the mobile extension using the iPECS

1) Dial the DID number of your station, the system will check the CLI, answer the call and you receive internal dial tone.
2) Place an internal or external iPECS call as normal.

To Transfer a call your mobile received from iPECS

1) Dial “*” while on the iPECS call.
2) Dial the desired station, the call is transferred and the mobile phone returns to idle.

3) Or, Dial a Park code to Park the call.
4) Or, Dial a Page zone to place an Internal/External or All Call page within the iPECS system.
5) Or, Dial a Conference Room to transfer the external party to a Conference Room.

Note: The mobile may reconnect to the call by dialing "#".

To move from Mobile phone to Desktop phone

1) The mobile phone of MEX is on MEX call.
2) Press **{my station number}** flexible button in desktop phone.
3) The user can do the MEX call with the desktop phone.

To move from Desktop phone to Mobile phone

1) The desktop phone of MEX is on MEX call.
2) Press **{my station number}** flexible button in desktop phone.
3) Then, the mobile phone is ringing.
4) The desktop phone ends the call.
5) When the user answers with the mobile phone, the user can do the MEX call with the mobile phone.

System

Incoming DID calls are sent to active mobile phones automatically.

Conditions

- ✓ When the mobile phone places an external call through iPECS, the CLI of the corresponding station is used.
- ✓ The Mobile Extension features are not supported on analog CO Lines as answer and disconnect supervision is not reliable on these Line types.
- ✓ Message Wait and Callback cannot be activated to a mobile phone.
- ✓ The Mobile Extension feature is not supported over a distributed networked environment.
- ✓ When an incoming ISDN DID call is received, the system will access an ISDN line and place a call to the mobile phone. Thus, an ISDN line must be available for the system to notify the mobile user of the incoming call.
- ✓ Hold and Transfer Recalls to the mobile phone are sent to mobile phone and the associated station.
- ✓ ACD, Circular and Terminal Hunt Group calls can be routed to the active Mobile Extension.
- ✓ To move between Mobile and Desk phone is applied to only keysets with flexible buttons.

Programming

Keyset Admin.

SYSTEM

- Mobile Extension Table (PGM 236)

Web Admin.

TABLES DATA

- Mobile Extension Table

Related Features

- DND (Do Not Disturb)
- Station Message Wait/Call Back
- Attendant Recall
- Distributed Control Network
- Station User Programming & Codes

Hardware

- iPECS IP or LDP Phone

2.59 MOH (Music-On-Hold)

Description

When you place a call in a hold state, 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 delivers audio from one of several sources that are defined for the system and can be defined for each Line and DID number. Also it can be defined for each station user.

With multiple sources, different organizations can have different on-hold messages.Audio sources available include:

- An External audio connection through the Miscellaneous connector of the KSU,
- An Internal source with 13 canned melodies, one is selected as the internal source,
- Up to five (5) SLT(Single Line Telephone) ports can be assigned as SLT(Single Line Telephone) MOH ports and connected to an audio source and
- Up to thirty System announcements can be assigned, recorded and used for MOH, one announcement, number 201, is dedicated for MOH and twenty-nine using any desired System announcement number.

When using a System announcement, separate messages can be recorded for each of the languages, maximum six (6), supported by the system.

Operation

System

Operation of MOH is automatic

To record a System announcement for MOH

1) Press the **[TRANS/PGM]** button.
2) Dial “06”, the System Announcement Record code.
3) Dial the sequence number of the voice storage unit.
4) Dial the System Announcement number, if other than 201, assure the announcement is configured for MOH use.
5) Dial the Language number, only required with multi-language support. The current announcement is played followed by the “Press # to record” prompt.
6) Dial “#”.
7) After the beep-tone, record message.
8) Press the **[HOLD/SAVE]** button to stop recording and save the message.

Conditions

- ✓ The External music source should only be connected to one of the inputs.
- ✓ The System announcements can be externally recorded and uploaded to the system.
- ✓ Separate MOH announcements should be recorded for each Language when multiple languages are supported.

Programming

Keyset Admin.

CO LINE

- MOH (PGM 142-Button 6)

Flexible DID Table

- MOH (PGM 231-Button 8)

T-NET

- MOH (PGM 335-Button 2)

SYSTEM

- MOH Type (PGM 171-Button 2)

Web Admin.

CO LINE DATA

- Common Attributes ➤CO Line MOH

Flexible DID Table

- Table Data ➤Flexible DID Conversion(231) ➤MOH

T-NET

- T-NET Data➤T-NET Music/Alarm(335) ➤MOH Type

Zone

- Zone Data➤Zone Attributes(439) ➤Zone MOH Assign

SYSTEM DATA

- Music Sources

Related Features

- Hold
- Back Ground Music
- Multiple Language Support

Hardware

- External Music source connected to KSU music source input.
- External Music source connected to UCP music source input.

2.59.1 Programmable MOH Based On Station User & Call

Description

Normally, MOH(Music on Hold) is served by system overall or incoming DID destination number or CO channel based. Additionally, per a call, press of a flexible button that has a specific MOH with HOLD or TRANSFER function by a Station user can serves differential MOH that user needs for covering multiple department, business, marketing or advertising. Also, by setting of default MOH type for a Station, differential MOH is served when press fixed **[HOLD]** or **[TRANS]** button. The differential MOH is serviced for both of incoming and outgoing calls.

Operation

iPECS IP & LDP Phones

Overview of MOH attributes and services

1) Number of MOH types :
   2 of MISC(Internal/External Music1, External Music2) + 30 of VSF MOH + 5 of SLT MOH = 37
2) Case of MOH play
   - Hold or Transfer call by press of fixed HOLD and TRANS button
   - Hold or Transfer call by press of a flexible button that has HOLD or TRANS feature and a specific MOH type
   - Automatic HOLD by press of LOOP button.
   - Queuing to a Station group
3) Priority of MOH type related programming services
   - Station User(P111) **>** Original Called Number for Incoming Call (P231) **>** CO channel (P140) **>** Zone (P439) **>** System (P171)
   - Or, Original Called Number for Incoming Call (P231) **>** CO channel (P140) **>** Station User(P111) **>** Zone (P439) **>** System (P171)

Hold or Transfer using Flexible Buttons

1) Usage : Hold or Transfer an External or Internal Call using a Flexible button
2) Programming of Flexible buttons
   - Web : Station Data ➤ Flex Buttons (115/129)
     . Type : 6th - Programming (PGM Code)
     . Input : PGM Code (HOLD:1*, TRANS:1#) + MOH type (00-37)
   - Phone : by User
     . [PGM] + flex button + [PGM] + Code (HOLD:1*, TRANS:1#) + MOH type (00-37) + [Save]
   - Example
     . Hold a Call with MOH 03 : 1*03 (1*:PGM Code HOLD Key, 03:MOH Type)
     . Transfer a Call with MOH 03 : 1#03 (1#:PGM Code TRANS Key, 03:MOH Type)
3) Call Process (Same process as using of fixed [HOLD] and [TRANS] buttons)
   - For an incoming or outgoing call on talking
   - Press flexible {HOLD MOHxx}/{TRAN MOHxx} button
   - The call will be held hearing the specific MOH type ‘xx’ of the holding party Station

4) Type of MOH for Hold/Transfer Flex button

- **00** : Default MOH Type of a Station. Same MOH source as fixed [HOLD] and [TRANS] buttons
- Music Types

<Music Type#>	<Music Type Name>
01	*Internal / External Music1*
02	*External Music2*
03	*VSF MOH*
04	*SLT MOH1*
05	*SLT MOH2*
06	*SLT MOH3*
07	*SLT MOH4*
08	*SLT MOH5*
09	*VSF MOH2*
10	*VSF MOH3*
11	*VSF MOH4*
12	*VSF MOH5*
13	*VSF MOH6*
14	*VSF MOH7*
15	*VSF MOH8*
16	*VSF MOH9*
17	*VSF MOH10*
18	*VSF MOH11*
19	*VSF MOH12*
20	*VSF MOH13*
21	*VSF MOH14*
22	*VSF MOH15*
23	*VSF MOH16*
24	*VSF MOH17*
25	*VSF MOH18*
26	*VSF MOH19*
27	*VSF MOH20*
28	*VSF MOH21*
29	*VSF MOH22*
30	*VSF MOH23*
31	*VSF MOH24*
32	*VSF MOH25*
33	*VSF MOH26*
34	*VSF MOH27*
35	*VSF MOH28*
36	*VSF MOH29*
37	*VSF MOH30*

Default MOH Type of a Station

1) Usage : Hold or Transfer an External or Internal Call using Fixed [HOLD] or [TRANS] button, or using a Flexible button that has MOH Type 00(MOH00).
2) Programming of Default MOH Type (MOH**00**)
   - Default MOH Type for External Call : select one of following list
     . Follow Zone(P439) or System(P171)
     . Internal / External Music1
     . External Music2
     . VSF MOH
     . SLT MOH1~5
     . VSF MOH2~30
   - Default MOH Type for Internal Call : select one of following list
     . Follow Zone(P439) or System(P171)
     . Internal / External Music1
     . External Music2
     . VSF MOH
     . SLT MOH1~5
     . VSF MOH2~30
   - Override MOH of Original Called Number (P231)
     . OFF(default) : apply MOH type that was defined by MOH type of Flexible DID Routing Table.
     . ON : apply MOH type that was defined by Station User.
   - Override MOH of CO Channel (P140)
     . OFF(default) : apply MOH type that was defined by MOH type of CO channels.
     . ON : apply MOH type that was defined by Station User.
   - Zone MOH – Follow Holding Party Zone :
     If MOH of device specific (CO or Station) is set to “Refer To Zone or Sys MOH” and Zone MOH type is set to a specific MOH that is not following “Refer To Sys MOH”, MOH is type is decided by Zone MOH(P439).
     . OFF(default) : apply held party’ Zone MOH
     . ON : apply holding party’ Zone MOH
   - Sys MOH – Follow Holding Party Sys(TNET) :
     If MOH of device specific (CO or Station) is set to “Follow Zone or Sys MOH” and Zone MOH is set to “Refer To Sys MOH”, MOH type decided by System(P171) or TNET(P335) Music Source setting.
     . OFF(default) : apply held party’ System MOH
     . ON : apply holding party’ System MOH
3) Call Process
   - For an incoming or outgoing call on talking
   - Press fixed [HOLD] / [TRANS] button, or press flexible {HOLD MOH00}/{TRAN MOH00} button
   - The call will be held hearing the default MOH type of the holding party Station

Conditions

✓ MOH Type - "Follow Zone or System" means that the MOH type is not defined by a Station or CO Line or Zone, etc.

Programming

Keyset Admin.

SYSTEM

- MOH Type (PGM 171-Button 2)
- Internal MOH Type (PGM 171-Button 3)
- Assign SLTMOH Port (PGM 171-Button 4)
- VSF MOH2 (PGM 171-Button 5)
- VSF MOH3 (PGM 171-Button 6)
- VSF MOH4~30 (PGM 171-Button 7~33)

Web Admin.

SYSTEM DATA

- Music Sources ➤BGM Type, Internal Music Type
- Music Sources ➤SLTMOH 1-5
- Music Sources ➤VSF MOH2-30

Related Features

- Call Transfer, Outside Line
- Exclusive Hold
- System Hold

Hardware

- iPECS IP or LDP Phone

2.60 Multiple Language Support

Description

With the integrated Auto Attendant and Voice Mail, iPECS can support six (6) languages simultaneously.

Prompts in thedesired languages are loaded into the integratedAA/VM memory along with the Language Selectionprompts.

To assure the proper language is employed, the Language Selection prompts areplayed when an incoming call is assigned to be answered by a DID, DISA, System Announcement or Station Group announcement.

The Language Selection prompts are played in multiple phrases,one in each of the equipped languages, with a request for the caller to input a digit to select theappropriate language.

The system then employs the defined announcement (DID, DISA, etc.)recorded for the selected language.

Operation

System

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

To record a Multi-Language Selection prompt

1) Press the **[TRANS/PGM]** button.
2) Dial “06”, the System Announcement Record code.
3) Dial the slot number of the voice storage unit.
4) Dial the Multi Language selection Announcement number, 202.
5) Dial the Language number, only required with multi-language support. The current announcement is played followed by the “Press # to record” prompt.
6) Dial “#”.
7) After the beep-tone, record your message.
8) Press the **[HOLD/SAVE]** button to stop recording and save the message.

Conditions

- ✓ The Language Selection prompts must be recorded as announcement number 202.
- ✓ Multi-language support is only available using the integrated Auto Attendant.
- ✓ Separate announcements must be recorded by the Attendant for each language supported.
- ✓ The System announcements can be externally recorded and uploaded to the system in a telephony quality (64 Kbps) wav file format.

Programming

Keyset Admin.

SYSTEM

- Multi-language (PGM 179)

Web Admin.

SYSTEM DATA

- System Multi Language

Related Features

- Integrated Auto Attendant/Voice Mail
- Direct Inward Dial (DID)
- Station Groups
- Announcements
- Direct Inward System Access (DISA)

2.61 Multiple Voice Mailbox Support

Description

A station can access any Voice Mailbox by dialing the VM group, the mailbox number andpassword. If you use aniPECS IP and LDP Phone, one or more Flex buttons of your phone can be configured to access a specificmailbox and receive a visual indication (LED flashes) of message status in the mailbox. A Flex**{VMAILBOX}** button is assigned to allow access to each Voice Mailbox.

Operation

iPECS IP & LDP Phones

*To assign a **{VMAILBOX}** Flex button*

- **[TRANS/PGM]** + **{FLEX}** + VM group + Mail-box (station) number + **[HOLD/SAVE]**

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

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

Conditions

- ✓ A Flex button can be assigned to access the Mailbox for a Station Group or outside Line Voice Mailbox.
- ✓ The **{VMAILBOX}** button LED will flash indicating new messages have been received in the associated mailbox.
- ✓ A Flex button cannot be assigned to access the mailbox of a networked station.
- ✓ An external Voice Mail system may not provide proper notification to the system of message status and thus, the **{VMAILBOX}** button LED may not properly indicate new messages.

Related Features

- External Auto Attendant/Voice Mail
- Integrated Voice Mail

Hardware

- iPECS IP or LDP Phone
- External or Integrated Voice Mail

2.62 Network Security & Priority

Description

Each IP device that connects to the eMG supports several security and priorityprotocols. IP devices incorporate a Web server, which will deliver the Module Web Adminpages to a standard browser.

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

Operation of Security and priority is automatic once configured.

Conditions

- ✓ For Web Admin, the password is encrypted.
- ✓ Security and priority characteristics can be set for all devices, local or remote.
- ✓ The iPECS implementation of IPSec employs a proprietary Key exchange protocol from the MPB/UCP to the iPECS device.
- ✓ IEEE 802.1 VLAN priority and ID are set at each device interface. Diffserv, IPSEC and SRTP treatments may also be set in the device via the local Web interface. For the UCP, VLANs are configured via the maintenance interface of the RS-232 port.

Programming

Keyset Admin.

BOARD

- RTP Security (PGM 132-Button 4)

Web Admin.

BOARD BASEDDATA

- Board Based Attributes ➤RTP Security

2.63 Outside Call Redirect

Description

You can assign a Flex button of your iPECS IP or LDP Phone to send an incoming outside call to a pre-defined destination such as the Attendant, Voice Mail, another Station, or Hunt group without answering the call. Call Redirect acts as a one-time Call Forward.

Operation

iPECS IP & LDP Phones

*To assign a Flex button for **{Call Redirect}***

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “58” + destination + **[HOLD/SAVE]**

To activate Call Redirect while receiving an outside call

1) Press the **{Call Redirect}** button; the call is redirected to the defined destination.

Conditions

✓ Conditions associated with Call Forward apply.

✓ Only the call ringing at the time Call redirect is activated will be redirected. The next incoming outside call is routed as normal.

Hardware

- iPECS IP or LDP Phone

2.64 Outside Call Time Restriction

Description

The system can be programmed to limit the length of outgoing calls by specified stations. If Call Time Restriction is assigned to a station, when the station places an outgoing call, the system initiates the Cut Off timers for the Station and the outside Line.

Fifteen (15) seconds prior to expiration of the shortest timer, a warning tone is delivered to the station. At expiration, the system terminates the call returning the external Line to idle.

Operation

System

Operation of this feature is automatic when assigned.

Conditions

- ✓ Once activated, the Cut Off timer continues timing while the call is connected to the system even if the call is transferred or picked up at another station.
- ✓ Two Cut Off timers are provided, one assigned on a Station basis, the other assigned CO basis. The shortest of the two timers will control the Cut-Off function

Programming

Keyset Admin.

STATION

- CO/IP Call Time Restriction (PGM 112-Button 3)
- Cut Off Timer (PGM 123-Button 2)

CO/IP

- CO Cut Off Timer (PGM 142-Button 21)

Web Admin.

STATION DATA

- Common Attributes ➤Call Time Restriction
- Station Timer ➤Cut Off Timer

CO LINE DATA

- Common Attributes ➤CO Cut Off Timer

Related Features

- ■ Call Duration Warning Tone Timer

2.65 Outside Line Access

Description

Stations can access outgoing Lines if allowed Line allowed access in the System database. The iPECS IP and LDP Phones may use flexible buttons assigned to access a specific Line or Line group button for outgoing calls or a **{LOOP}** button.

Individual users may be allowed to assign Flexible buttons to access a Line or Line group.

Operation

iPECS IP & LDP Phones

*To assign a **{LOOP}** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + "85" + **[HOLD/SAVE]**

*To assign a **{POOL}** button*

- **[TRANS/PGM]** + **{FLEX}** + CO Group Number + **[HOLD/SAVE]**

To place an outgoing outside call

1) Lift the handset or press the **[SPEAKER]** button.
2) Press desired **{LINE}**, **{POOL} or{LOOP}** button, or dial the Line or Line group access code.
3) Dial the desired outside number.

To answer an incoming outside call

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

Or,

1) Press flashing **{LINE}**, **{POOL}** or **{LOOP}** button; which will answer the call over the speakerphone. You may lift the handset to speak privately.

SLT(Single Line Telephone)

To place an outgoing outside call

1) Lift handset.
2) Dial the CO line or Group access code.
3) Dial the desired number.

To answer an incoming outside call

1) Lift handset.

Conditions

- ✓ When a user dials "9", the system will search the preferred Line Group for an idle Line, if there is no idle line then the system will search the 1st Line Group for an idle Line. The system may continue the search through all Line Groups for an available line if "1st CO/IP Group Override" is enabled.
- ✓ If not allowed access to a specific outside Line, you receive error tone when access is attempted. The station may receive transferred calls on such denied access Lines but will not be able to flash or use the Line for an outgoing call.

- ✓ A station denied access to an outside Line but assigned to have ring for the Line will receive ring, a flashing **{LINE}** button LED and may answer the call. The user may transfer the call but cannot make an outgoing call on the Line.
- ✓ When you place an outside call, the audio transmit path to your station is muted until the system has verified Toll Restriction for the Line.
- ✓ When an analog Line is seized, the system will monitor the Line for dial tone.
- ✓ The system selects Lines from a group using the Round-robin or Last-choice method as assigned in the system database.
- ✓ When an iPECS IP or LDP Phone is not assigned Ringing Line Preference, the user must press the ringing **{LINE}** button to answer the call.
- ✓ iPECS IP and LDP Phone users may be assigned exclusive use of a Line as a Private Line.

Programming

Keyset Admin.

STATION

- Individual CO Access (PGM 112-Button 4)
- CO Line Programming (PGM112-Button 6)
- Ringing Line Preference (PGM112-Button7)
- Prefer CO/Group (PGM112-Button 14)
- CO/IP Group Access (PGM 117)

CO/IP

- CO/IP Group (PGM 141-Button 1)
- CO Ring Assignment (PGM 144)

Web Admin.

STATION DATA

- Common Attributes ➤Individual CO Access
- Common Attributes ➤CO PGM Ringing Line Preference
- Common Attributes ➤Prefer CO or Group
- Common Attributes ➤CO/IP Group Access

CO LINE DATA

- Common Attributes ➤CO/IP Group
- CO/IP Ring Assignment

Related Features

- Outside Line Groups
- Line Ring Assignment
- Ringing Line Preference
- Private Line

2.66 Outside Line Queuing

Description

When outside Lines are busy, if permitted, you can request to be placed in a queue awaiting the Line or a Line in the same group to become available.

When an appropriate Line becomes available, the system calls the waiting station on a first in first out basis.

Operation

iPECS IP & LDP Phones

To request to be placed in queue for a busy Line

1) Press the busy **{LINE}** or **{POOL}** button.
2) Press the **[MESSAGE/CALLBACK]** button, confirmation tone is received.
3) Hang-up, the **[MESSAGE/CALLBACK]** LED flashes.

To cancel the queue from the queued station

1) Press the **[MESSAGE/CALLBACK]** button, the **[MESSAGE/CALLBACK]** LED extinguishes.

SLT(Single Line Telephone)

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

1) Momentarily press the hook-switch.
2) Dial "*556", Activate Message Wait/Callback code.

To cancel the queue from the queued station

1) Lift the handset.
2) Dial "*556", Activate Message Wait/Callback code.

System

When a Line becomes available

1) Send distinctive Queue recall to the station with oldest queue, flash appropriate **{LINE}** button LED. Make the Line and station busy to all other users.

Conditions

- ✓ An outside Line can have any number of simultaneous queue requests.
- ✓ A station may only have a single active outside Line queue request. Activating a new queue request will replace, and thus cancel, an existing queue.
- ✓ A Queue recall will always signal the station with tone ring, ignoring the station's assigned Intercom Signaling mode.
- ✓ Queue recall will bypass a busy station, and place the station at the bottom of the queue list.
- ✓ Queue recall will signal a station for 15 seconds, after which, the station is removed from the queue; the queue is cancelled.

Programming

Keyset Admin.

STATION

- CO/IP Queuing (PGM 112-Button 5)

Web Admin.

STATION DATA

- Common Attributes ➤CO/IP Line Queuing

Related Features

- Outside Line Access

Hardware

2.67 Personal Groups

Description

A Personal Group consisting of your master station and group member stations (maximum 32 including your master station) can be configured by the system Administrator.

When your master station receives a call all members also receive the call and when placing a call the master station number, access and dialing restrictions are used. Each member can still receive call to the member station number.

Certain attributes of the master station can also be delivered to member stations, if configured, including:

- Master Wake-Up will send the Wake-Up alarm set at the master station to all station members.
- Master Call Forward will forward member calls as well.
- Master DND activates DND at the master and all member stations.
- Linked Option.

When 'On', the members of the Personal Group act as a Linked pair station of the master station number. When one station of the group is busy all are busy; Linked stations will receive the 'linked pair in-use' display.

When 'Off', The members receive calls for the master employing Call Coverage, attributes of the master (Wake-Upand DND) can be enabled for the members, the LCD of member stations display outgoing call information for the master, each member maintains separate calling capabilities and the master will be busy only when all members are busy.

Operation

Operation of this feature is automatic when configured.

Conditions

- ✓ Unlike the 'Linked Pair stations, the stations in Personal Group with the Linked option enabled maintain a separate station database and station number.
- ✓ DECT phone can't be the master station.
- ✓ When DECT phone press **[Talk]** button, they can't pick up the call of the talking personal group member.

Programming

Keyset Admin.

SYSTEM

- Personal Group Assignment (PGM 260)
- Personal Group Attributes (PGM 261)

Web Admin.

STATION GROUP DATA

- Personal Group
- Personal Group Attributes

Related Features

- SMDR
- Call Coverage
- Wake-Up
- Message Wait/Call Back
- Linked Station Pairs

Hardware

2.68 PPP over MODU for eMG

Description

PPP (Point-to-Point Protocol) is a protocol for communication between two computers using a serial interface, typically a personal computer connected by phone line to a server.

MODU stands for MODEM Unit which is installed in eMG system for data communication using CO line which can be analogue or digital(ISDN or SIP trunk etc.).

eMG system acts as a PPP server and PPP client can connect to the system using dial-up modem.After PPP connection established, users can access the system through Web admin or Telnet service.

There is MODEM associated(ASC) device which maybe a CO line or a station.MODEM ASC should be configured for PPP over MODU.

Operation

When MODEM ASC is a CO line

1) Incoming call to the MODEM ASC CO line is answered by MODU automatically and PPP connection is established.

When MODEM ASC is a station

1) If an incoming call is transferred to the MODEM ASC station, MODU will answer to the station and then PPP connection is established.
2) If an incoming call is routed to the MODEM ASC station through DISA or DDD, MODU will answer to the station and then PPP connection is established.

Programming

Web Admin.

SYSTEM DATA

- PPP Attributes
- System Attributes ➤ MODEM Associated Station / CO line

Hardware

- MODU(MODEM Unit)

2.69 Pre-Defined & Custom Text Display Messages

Description

You can select a text message that will display in the LCD of iPECS IP and LDP Phones when they call you. When you activate Text Display Messages, incoming intercom calls will signal with muted Ring in-place of normal ring and the LCD of the calling user displays themessage you selected.

There are ten Pre-defined messages (01-10), ten system-wide Custommessages and one user defined Custom message that you can enter. Your Custom message, the user defined message, is message number 00, messages 01 to 10 are the Pre-defined messages and messages 11 to 20 are the system-wide messages. Several of the ten Pre-defined messagesallow for auxiliary information such as a time, date or number.

System level Custom Messages are entered from the Attendant or Administrator's phoneor via the Web Admin. Your Custom Message is entered from your phone, the Attendant or the Administrator.As a default, when Text Display Message is active, your phone is effectively in DND to internal callers. The system can be configured to deliver ring to your phone by disabling the Message DND attribute for your phone. The iPECS IP and LDP Phone may have a Flex button defined for**{DISPLAY MESSAGE}** button and the button mayinclude the message code to act as a **{ONE-TOUCH DISPLAY MESSAGE}** button.The Pre-defined messages are:

Message 01

LUNCH RETURN AT HH:MM

Message 02

ON VACATION
RETURN AT DATE DD:MM

Message 03

OUT OF OFFICE
RETURN AT TIME HH:MM

Message 04

OUT OF OFFICE
RETURN AT DATEDD:MM

Message 05

OUT OF OFFICE
RETURN UNKNOWN

Message 06

CALL (enter up to 17 digits)

Message 07

IN OFFICE STA xxxx

Message 08

IN MEETING RETURN AT TIME hh:mm

Message 09

AT HOME

Message 10

AT BRANCH OFFICE

Operation

iPECS IP & LDP Phones

To assign a Flex button for Display Messages

{DISPLAY MESSAGE} button

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “51” + **[HOLD/SAVE]**

{ONE-TOUCH DISPLAY MESSAGE} button:

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “51” +message number + **[HOLD/SAVE]**

To activate Display Message

1) Press the **[TRANS/PGM]** button.
2) Dial “51”, the Display Message code.
3) To view available messages, press **[VOL UP]**/**[VOL DOWN]** button.
4) Dial the Message number (“00”-“20”).
5) Dial auxiliary input, as desired.
6) Press the **[HOLD/SAVE]** button.

Or,

1) Press the **{DISPLAY MESSAGE}** button.
2) To view available messages, press **[VOL UP]**/**[VOL DOWN]** button.
3) Dial the Message number (“00”-“20”).
4) Dial auxiliary input, as desired.
5) Press the **[HOLD/SAVE]** button.

Or,

1) Press the {ONE-TOUCH DISPLAY MESSAGE} button.
2) Dial auxiliary input, as desired.
3) Press the **[HOLD/SAVE]** button.

To cancel an active Display Message

1) Press the flashing **[FWD]** button.

To define the your Custom Text Message (00)

1) Press the **[TRANS/PGM]** button.
2) Dial “52”, the Custom Message program code.

3) Enter the Message contents, up to 24 characters using two (2) dial pad digits for each character.
4) Press the **[HOLD/SAVE]** button, confirmation tone is received, your new Custom Text Display Message is stored and the station returns to idle.

SLT(Single Line Telephone)

To activate a Display Message

1) Lift the handset.
2) Dial "*561", the SLT(Single Line Telephone) Programming code.
3) Dial "51", the Display Message code.
4) Dial Message number ("00"-"20").
5) Dial auxiliary data as desired.
6) Momentarily press the hook-switch, confirmation tone is received.

To cancel an active Display Message

1) Lift the handset.
2) Dial "559", the SLT(Single Line Telephone) Feature Cancel code.

To enter the User Custom Message (00)

1) Lift the handset.
2) Dial "*561", the SLT(Single Line Telephone) Programming code.
3) Dial "52", the Custom Message program code, confirmation is received.
4) Enter the Message contents, up to 24 characters.
5) Momentarily press the hook-switch, confirmation tone is received.

System Attendant

To activate Display Messages for other stations

1) Press the **[TRANS/PGM]** button.
2) Dial "051", Attendant Display Message code.
3) Dial the station range, for a single station dial the same station number twice.
4) To view available messages, press **[VOL UP]**/**[VOL DOWN]** button.
5) Dial Message number ("00"-"20").
6) Dial auxiliary data as desired.
7) Press the **[HOLD/SAVE]** button.

To cancel active Display Messages for other stations

1) Press the **[TRANS/PGM]** button.
2) Dial "052", Attendant Display Message Cancel code.
3) Dial the station range, for a single station dial the same station number twice.
4) Press the **[HOLD/SAVE]** button.

To enter a System-wide Custom Message

1) Press the **[TRANS/PGM]** button.
2) Dial "053", the System-wide Custom Message program code.
3) Dial desired Message code ("11"-"20").

4) Enter the Message contents, up to 24 characters.
5) Press the **[HOLD/SAVE]** button, confirmation tone is received, the new Custom Message is stored and the station returns to idle.

Conditions

- ✓ The iPECS system can automatically activate DND to internal callers when you activate Pre-selected or Custom Text Message Display. Enabling P-Msg DND activates DND when a Text Display Message is activated.
- ✓ Only the System Attendant or Administrator can assign the contents of Custom Messages 11-20.
- ✓ The alphanumeric characters are displayed as they are entered; refer to Station Speed Dial for entry procedures.
- ✓ Display Message is cancelled if you activate DND or Call Forward.
- ✓ Custom Text Display Messages and Display Message status are stored in non-volatile memory to protect against loss during power failure.
- ✓ The calling station must be an iPECS IP or LDP Phone with display to receive the message.
- ✓ SLT(Single Line Telephone)s are notified of an active Display Message with stutter dial tone. The iPECS IP and LDP Phones will have a flashing **[FWD]** button when Text Display Message is active.
- ✓ Activating Text Display Message does not affect normal operation of the station.
- ✓ Pre-defined Messages 01 to 04 and 06 to 08 permit you to input auxiliary information such as time, date or number.
- ✓ The Attendant station can activate Text Display Messages for other stations. However, the Text Display Message feature is not available to an Attendant.

Related Features

- Silent Text Message
- DND (Do Not Disturb)
- Call Forward
- Speed Dial

Hardware

- iPECS IP or LDP Phone required to receive Display Messages

2.70 Prefix Dialing Table

Description

The Prefix Dialing Table supports several features. On analog CO Lines, the table is used to determine the cost of the call. For ISDN Lines assigned Enblock dialing and SIP Trunks, Enblock dialing can be supported so that you don't need to wait for the inter-digit time-out to send digits as a block to the ISDN or SIP service provider. Up to 500 Prefix numbers and associated entries are available in the table.

When employed for calculating the cost of a call, the system will compare the outgoing digits dialed with the Prefix Table entries (maximum 8 digits). If the first dialed digits match the entry, the cost of the call is determined using the Prefix Dialing Table. The Call Charge timer assigned in the table for the Line is used in-place of meter pulses along with the Cost per Meter Pulse assigned in the system database. The cost is then calculated as: (Call Duration/Call Charge Timer) * Cost per Meter Pulse.

When used for Enblock dialing, the first digits you dial are compared to the Prefix digits. If the digits match, when you dial the maximum number of digits assigned, the number is immediately sent to the service provider as a block.

If there is no match or the maximum digits count is not reached, the system will wait for the inter-digit time before sending the digits. For an ISDN Line, a minimum number of digits can be defined. Even if the Prefix digits match, the call will not be processed until the minimum number of digits is dialed.

Each Line is assigned a Table ID which must match the Prefix dialing Table entry to be considered for the specific Line.

Operation

Operation of this feature is automatic when configured.

Conditions

- ✓ For ISDN Enblock dialing, ISDN messages will include the Type of Number, Numbering plan type and sending complete messages, if assigned in the matching Prefix Dialing Table entry.

Programming

Keyset Admin.

ISDN LINE DATA

- Prefix Dialing Table (PGM 206)

CO LINE DATA

- Prefix Table ID (PGM 142-Button 20)

Web Admin.

ISDN LINE DATA

- Prefix Dialing Table

CO LINE DATA

- Common Attributes ➤ Prefix Table ID

2.71 Prepaid Call

Description

Prepaid refers to services paid for in advance.There are two kinds of prepaid call. One is CO prepaid call and the other one is Station prepaid call. If a CO line is configured to use prepaid call, the CO line is available for outgoing call until prepaid money becomes empty.When you make prepaid calls, the prepaid money is deducted based on call metering pulse.

Eventually, if prepaid money of a CO line is consumed completely, users can not make an outgoing call with that CO line.If a station is configured to use prepaid call, it can use outgoing call until its prepaid money is used up.

Operation

Operation of this feature is automatic when configured.

Conditions

- ✓ CO prepaid call has a higher priority than station prepaid call.
- ✓ If a CO is allowed to use prepaid call, a station of which station prepaid call is not possible can make an outgoing call with that CO.
- ✓ On the contrary, if a CO is not allowed to use prepaid call, a station of which station prepaid call is possible can't access to that CO for outgoing call.
- ✓ Emergency call will be possible even if Station or CO prepaid money is empty.

Programming

Keyset Admin.

CO/IP

- Prepaid Call (PGM 141-Button 21)
- Prepaid Money (PGM 141-Button 22)

Related Features

- Station prepaid call
- SMDR (Station Message Detail Recording)- Call Cost display
- Outside Call Warning Tone Timer

2.72 Redundant System Processor for UCP

Description

The iPECS UCP system supports redundant processor operation. The redundant system processoris a hot standby processor.

The LAN2 port of the master iPECS UCP moduleis connected directly to theLAN2 port of the standby UCP module for local redundancy, in which case the LAN2 ports may be connected over the LAN or WAN.

The standby iPECS UCP module monitors the active iPECS UCP module LAN1 and LAN2 port and takes oversystem control if the:

- active UCP module power fails.
- active UCP module is reset.
- LAN1 or 2connection to the active iPECS UCP module fails or is removed, or,
- standby UCP module does not receive a polling message from the active UCP module for 1 minute.

If failure occurs, the standby UCP module will take over and maintain control of the system. When theoriginal master recovers from failure, it functions in the standby mode.

The active UCP module will maintain the system database for both the active and standby processor.

Operation

Operation of redundant processors is automatic when installed.

Conditions

- ✓ The Master/Slave switch on the main iPECS UCP module must be set in the Master position and the stand-by UCP module must be set in the Slave position to match the system Admin settings.
- ✓ All eight wires of the Cat 5 cable must be terminated on the RJ45 connectors.
- ✓ It is not possible to use the built-in Voice Mail, a UVM or external VM should be installed when CPU redundancy is required. If you enable redundancy (Redundancy Attributes ➤CPU Redundancy Usage),the built-in Voice Mail is placed “OUT of Service”.
- ✓ With Geographic Redundancy active, both the Master and Slave can be active at the same time.
- ✓ With local redundancy both the Master and Slave can be active at the same time if theLAN2cable is out of order. These dual active processor conditions can be addressed using the IP Watch time. When the timer is set, the Master sends an ARP packet from the LAN1 port to the Slave at the timer interval allowing the Slave can to determine the state of the Master. If the Slave receives an ARP packet, even though the LAN2 port fails it will not become the active processor thus preventing the potential for a dual processor active condition.
- ✓ With vUCP, Redundant System Processor (Local Redundancy) is not available, but Geographical Redundancy is available.
- ✓ vUCP/UCP Hybrid redundancy is supported from unified 5.0 or later version.
- ✓ vUCP and UCP2400 Hybrid Geographic Redundancy is supported if built-in vVOIM/vUVM is disabled
- ✓ vUCP and UCP600 Hybrid Geographic Redundancy is supported if built-in vVOIM/vUVM is enabled

Programming

Web Admin.

REDUNDANCY DATA

- Redundancy Attributes ➤CPU Redundancy Usage, Change Active UCP by Power Fail, Geographic Redundancy

SYSTEM DATA

- System Timers ➤IP Watch timer

Hardware

- For Main and stand-by UCP installation, please refer to the iPECS UCP Hardware Description and Installation Manual.

2.73 Registration & Fractional Module Tables for UCP

Description

Normally, local iPECS devices can register with the system only when the “RegistrationSwitch”, 3rd Dip switch on the UCP module, is ON. In this state, the system will allow any local iPECSdevice to register, providing a convenient “plug & play” initial installation. After initial installation, the Registration Switch is placed in the OFFposition, disabling registration of additional localdevices.

To eliminate the potential for unintended device registration, particularly where multiplesystems exist on the same LAN, the system can be programmed to allow local deviceregistration employing MAC addresses. Using the defined MAC address registration, thesystem allows a device with a matching MAC address to register regardless of the RegistrationSwitch position.

Three tables are provided to enter MAC addresses. The System & Device IP Plan allows entry of two MAC address ranges. Any device with a MAC address in either of theranges will be permitted to register with the system.

The second is the Registration Table. This Table permits entry of five MACaddresses and the number of channels to be registered for each device. Entering the MACaddress permits the device to register with the system regardless of the Registration Switchposition. The number of channels available to the device is limited to the number of channelsentered in the Table. This function is commonly used to limit the number of channels availableover an E1/T1 or PRI ISDN circuit (Fractional T1 line).

The third is Device Login ➤ Remote Device Registration thatallows entry of the MAC address of a remote phone or gateway Module as wellas the Nation code and Zone. During registration, the system will compare the MAC addressreceived from the remote device and, if matched, will permit registration of the device. Oncethe device is registered, the data for the device is placed in the appropriate locations and thedata is removed from the Remote Device Registration Table.

Operation

Operation of registration is automatic based on the system database and Registration Switch position.

Programming

SYSTEM ID &NUMBERING PLANS

- System& Device IP Plan ➤First MAC Range, Second MAC Range

TABLES DATA

- Registration Table

DEVICE LOGIN

- Remote Device Registration

Hardware

2.74 Remote Control from Mobile Phone

Description

In addition to the call features available to your Mobile phone, you can control certain settings through your Mobile phone. The Mobile has access to the Remote Control menu shown below.

- 1: Activate Mobile extension
- 2: Deactivate Mobile extension
- 3: Unconditional Forward to VM group
- 4: Cancel Fwd to VM group
- 5: ACD Agent Off-Duty
- 6: ACD Agent On-Duty
- 7: Callback to Mobile extension
- 8: Turn On CCR Temporary Announcement – [8 + CCR table number]
- 9: Turn Off CCR Temporary Announcement – [9 + CCR table number]

A System announcement can be recorded and configured to play when you enter the Remote Mobile Extension Code. The announcement can be used to list the Remote Control Menu.

Operation

Mobile Phone

To use the Remote Control Menu

1) Call your system DID number.
2) Upon answer, enter "*580", the Remote Mobile Extension Control code.
3) Dial the desired menu, dial tone is returned as confirmation.

Conditions

- ✓ For proper operation, your phone must be allowed to Program the Mobile Extension parameters, permitted Call Forward and access to the integrated Voice Mail must be enabled.
- ✓ Selecting ACD Agent Off-Duty will place you Off-duty with Reason code 1.
- ✓ If you select Call Back from the Remote Menu, the call will be disconnected after the confirmation tone then the system calls the mobile extension.

Programming

Keyset Admin.

NUMBERING PLAN

- Remote MEX Control (PGM109-Button 16)

STATION

- Call Forward (PGM 111-Button 2)
- VSF Access (PGM 113 Button 2)

TABLES

- Mobile PGM Authority (PGM 236-Button 1)
- Announce (PGM 236-Button 13)

Web Admin.

SYSTEM ID & NUMBERING PLANS

- Flexible Numbering Plan ➤Remote MEX Control

STATION DATA

- Common Attributes ➤Call Forward
- VM Attributes ➤VSF Access

TABLES DATA

- Mobile Extension Table ➤PGM Authority, Announcement

Related Features

- Call Forward
- Mobile Extension

Hardware

2.75 Ringing Line Preference

Description

When Ringing Line Preference is enabled, you answer incoming callsby lifting the handset or pressing the**[SPEAKER]** button.

You may select a Line before lifting the handset or pressing the **[SPEAKER]** button, which will override Ringing Line Preference and connect you to the Line you selected.

Operation

iPECS IP & LDP Phones

To answer a call while the station is ringing

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

Conditions

- ✓ When multiple calls are ringing at the station, a priority defines the order in which calls are answered. The default priority is: Transfer > recalls > incoming calls > queued calls.
- ✓ Intercom calls are always given the lowest answering priority.
- ✓ Ringing Line Preference overrides Prime Line assignment.
- ✓ SLT(Single Line Telephone)s operate only in the RLP mode; when ringing, lifting the handset connects the SLT(Single Line Telephone) to the ringing call.

Programming

Keyset Admin.

STATION

- Auto Speaker Select (PGM111-Button 1)
- Ringing Line Preference (PGM 112-Button 7)

CO/IP

- CO Ring Assignments (PGM 144)

SYSTEM

- Ring Line Preference Priority (PGM 173)

Web Admin.

STATION DATA

- Terminal Attributes ➤Auto Speaker Selection
- Common Attributes ➤Ringing Line Preference

CO LINE DATA

- CO/IP Ring Assignment

SYSTEM DATA

- RLP (Ring Line Preference) Priority

Related Features

- Prime Line Immediately/Delayed
- Automatic Speaker Select

Hardware

2.76 Speed Dial

2.76.1 Display Security

Description

Station and System Speed Dial numbers may be programmed so that the telephone number digits are not displayed on the LCD of iPECS IP and LDP Phones when the Speed Dial is selected.

Operation

To assign Display Security to a Speed Dial number

1) Dial "*" as the first digit of the Speed Dial number.

Conditions

- ✓ The number is displayed when programming the Speed Dial number.
- ✓ Display Security does not affect the SMDR output.
- ✓ Display Security is provided on all outside calls including calls that are transferred or recall.
- ✓ An "*" in any digit position other than the first, will activate Pulse to Tone Switchover.

Related Features

- Dial Pulse to Tone Switchover
- Speed Dial

Hardware

2.76.2 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. You can insert multiple pauses into a Speed Dial number. The pause feature is often used to allow external network equipment time to connect and deliver dial-tone.

Operation

System

Pause operation is automatic when encountered; see Station or System Speed Dial for pause entry.

Conditions

- ✓ The timed pause is used only with analog CO Lines.

Programming

Keyset Admin.

SYSTEM

- Pause Timer (PGM 181-Button 10)

Web Admin.

SYSTEM DATA

- System Timers ➤Pause Timer

Related Features

- ✓ Station Speed Dial
- ✓ System Speed Dial

Hardware

2.76.3 Station Speed Dial

Description

You can store commonly dialed numbers for easy access using Station Speed Dial.

With the system, your station has access to 100 Speed Dial numbers each stored in a “bin”.

Each Speed Dial number can be up to 25 digits in length and may include special instruction codes.

Special instruction codes available are:

‘*’ as 1st digit	Activate Display Security, do not display number.
‘*’ not 1st digit	Switch from Pulse to DTMF dialing
[DND]	Insert a Flash command
Flash as 1st digit	Activate dial tone detect.
[MESSAGE/CALLBACK]as 1st digit	Activate ISDN Keypad Facility
[MESSAGE/CALLBACK]	Insert pause dialing command.

Your iPECS IP and LDP Phonemay have a Flex button assigned for One-Touch access to a specific Speed Dial number.

In addition, you may assign a telephone number directly to a Flex button on your iPECS IP and LDP Phone.

In this case, the telephone number is allocated to the highest numbered Station Speed Dial bin you have available.

Operation

iPECS IP & LDP Phones

To assign a Flex button as a ***{STATION SPEED DIAL}*** *button*

- **[TRANS/PGM]** + {FLEX} + [SPEED] + Station Speed Dial bin number + **[HOLD/SAVE]**

To dial using a Station Speed Dial with a ***[SPEED]*** *button*

1) Lift handset or press the **[SPEAKER]** button.
2) Press the **[SPEED]** button.
3) Dial the desired bin number (00-99).

To program a Station Speed Dial number using the ***[SPEED]*** *button*

1) Press the **[TRANS/PGM]**button.
2) Press the **[SPEED]** button.
3) Dial the Speed Dial bin number (00-99).
4) Press select a Line button or dial a Line access code.
5) Press a Line button or dial a Line access code and Dial the number to be stored.
   Or, Dial Net station number.
6) Dial the number to be stored.
7) Press the **[HOLD/SAVE]** button.
8) If desired, enter a name, see alphanumeric entry chart below.
9) Press the **[HOLD/SAVE]** button.

iPECS IP Phones with 3-Soft buttons

To dial using a Station Speed Dial using an LIP 8000/8000E/LIP-9000/1000i series phones with soft buttons

1) Press the **[SPEED]** soft button.
2) Dial the desired bin number (00-99).

To program a Station Speed Dial number using an LIP-8000/8000E/LIP-9000/1000i series phones with soft buttons

1) Press the **[SPEED]** soft button.
2) Press the **[ADD]** soft button.
3) Dial the Speed Dial bin number (00-99).
4) Press a Line button or dial a Line access code and Dial the number to be stored.
   Or, Dial Net station number.
5) Press the **[HOLD/SAVE]** button.
6) If desired, enter a name, see alphanumeric entry chart below.
7) Press the **[HOLD/SAVE]** button.

SLT(Single Line Telephone)

To dial using a Station Speed Dial

1) Lift handset.
2) Dial "*558", the SLT(Single Line Telephone) Speed Dial access code.
3) Dial the desired bin number (00-99).

To program a Station Speed Dial number

1) Dial "*561", the SLT(Single Line Telephone) Programming code.
2) Dial "*558", the SLT(Single Line Telephone) Speed Dial access code.
3) Dial the Speed Dial bin number (00-99).
4) Dial the Line access code and Dial the number to be stored.
   Or, Dial net station number.
5) Momentarily press the hook-switch.
6) If desired, enter a name, see alphanumeric entry chart below.
7) Momentarily press the hook-switch.
8) Alphanumeric characters may be entered to name the Speed Dial number using the chart below.

ALPHANUMERIC DIAL-PAD ENTRIES

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

Conditions

- ✓ Accessing an empty Speed Dial bin will return error tone.
- ✓ A Speed Dial number uses a specific Line/or Line from a Group assigned in the Speed Dial bin. 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.
- ✓ All Speed Dial numbers are stored in memory protected from power loss.
- ✓ For iPECS IP and LDP phones, a call placed with Speed Dial will appear under the Flex button assigned for the Line, Line Group, or the general Line appearance Loop button, as appropriate for the station Flex button configuration.
- ✓ You may pre-select a Line for a Speed Dial number, overriding the Line assignment in the Speed Dial bin.
- ✓ A name can be entered for a Speed Dial number to permit access from the Dial-by-Name directory.
- ✓ If a Speed Dial number contains a “Dial Tone Detect” command, Flash as the first entry, and dial tone detect is enabled for the Line, the system must detect dial tone before dialing the Speed Dial number.
- ✓ **[SPEED]**fixed button is only allowed in LIP 9000 series.

Programming

Keyset Admin.

SYSTEM

- CO Dial Tone Detect (PGM 160-Button 3)

Web Admin.

SYSTEM DATA

- System Attributes ➤CO Dial Tone Detect

Related Features

- CO Line Flash
- Dial-by-Name
- Display Security
- Keypad Facility
- LNR (Last Number Redial)
- Dial Pulse to Tone Switchover
- Save Number Redial (SNR)
- Speed Dial Pause Insertion
- System Speed Dial
- Flex Button Direct Speed Dial Assignment

2.76.4 System Speed Dial

Description

Commonly dialed numbers can be stored by the System Attendant or by the Administrator in Web Admin for easy access by stations allowed use of System Speed Dial bins.

With eMG80/100, up to 3000 System Speed Dial numbers are available and with eMG800, up to 8000 System speed dial numbers are available and with UCP, up to 1200 systems speed dial numbers are available.

Each Speed Dial number can be up to 25 characters in length and may include special instruction codes.Special instruction codes available are:

'*' as 1st digit	Activate Display Security.
'*' not 1st digit	Switch from Pulse to DTMF dialing.
[DND]	Insert a Flash command.
Flash as 1st digit	Activate dial tone detect.
[MESSAGE/CALLBACK] as 1st digit	Activate ISDN Keypad Facility
[MESSAGE/CALLBACK]	Insert a pause dialing command.

You may assign a Flex button of your iPECS IP or LDP Phone for One-Touch access to a specific System Speed Dial bin.

Operation

iPECS IP & LDP Phones

To assign a Flex button as a ***{SYSTEM SPEED DIAL}*** *button*

- **[TRANS/PGM]** + **{FLEX}** + **[SPEED]** + System Speed Dial bin number + **[HOLD/SAVE]**

To dial using a System Speed Dial using a ***[SPEED]*** *button*

1) Lift handset or press the **[SPEAKER]** button.
2) Press the **[SPEED]** button.
3) Dial the desired bin number(eMG80/100:2000-4999 / eMG800: 2000-9999 / UCP: 20000-31999).

To dial a System Speed Dial number using an LIP-8000/LIP-9000/1000i series phone with soft buttons

1) Press the **[SPEED]** soft button.
2) Dial the desired bin number (eMG80/100:2000-4999 / eMG800: 2000-9999 / UCP: 20000-31999).

SLT(Single Line Telephone)

To dial using a System Speed Dial

1) Lift handset.
2) Dial "*558", the SLT(Single Line Telephone) Speed Dial access code.
3) Dial the desired bin number(eMG80/100:2000-4999 / eMG800: 2000-9999 / UCP: 20000-31999).

Attendant

To program a System Speed Dial number

1) Press the **[TRANS/PGM]** button.
2) Press the **[SPEED]** button.

3) Dial the Speed Dial bin number(eMG80/100:2000-4999 / eMG800: 2000-9999 / UCP: 20000-31999).
4) Press the Line button or dial the Line access code and Dial the number to be stored.
Or, Dial Net station number.
5) Press the **[HOLD/SAVE]** button.
6) If desired, enter a name, see alphanumeric entry chart under Station Speed Dial.
7) Press the **[HOLD/SAVE]** button.

To enter a System Speed Dial number using an LIP-8000/LIP-9000/1000i series phone with soft buttons

1) Press the **[SPEED]** soft button.
2) Press the **[ADD]** soft button.
3) Dial the Speed Dial bin number(eMG80/100:2000-4999 / eMG800: 2000-9999 / UCP: 20000-31999).
4) Press the **{LINE}** button or dial the Line access code and Dial the number to be stored.
Or, Dial Net station number.
5) Press the **[HOLD/SAVE]** button.
6) If desired, enter a name, see alphanumeric entry chart under Station Speed Dial.
7) Press the **[HOLD/SAVE]** button.

Conditions

- ✓ Accessing an empty Speed Dial bin will return error tone.
- ✓ A Speed Dial number uses a specific Line/or Line from a group assigned in the Speed Dial bin. 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.
- ✓ All Speed Dial numbers are stored in memory protected from power loss.
- ✓ For iPECS IP and LDP phones, a call placed with Speed Dial will appear under the Flex button assigned for the Line, Line Group, or the general Line appearance **[LOOP]** button, as appropriate for the station Flex button configuration.
- ✓ The user may pre-select an outside Line for a Speed Dial number, overriding the assignment in the Speed Dial bin.
- ✓ A name can be entered for a Speed Dial number to permit access from the Dial-by-Name directory.
- ✓ If a Speed Dial number contains a “Dial Tone Detect” command (Flash as the first entry) and dial tone detect is enabled for the Line, the system must detect dial tone before dialing the Speed Dial number.

Programming

Keyset Admin.

STATION

- Speed Dial Access (PGM 112-Button 8)

SYSTEM

- CO Dial Tone Detect (PGM 160-Button 3)

TABLES

- System Speed Zone (PGM 232)

Web Admin.

STATION DATA

- Common Attributes ➤Speed Dial Access

SYSTEM DATA

- System Attributes ➤CO Dial Tone Detect

TABLES DATA

- System Speed Zone

Related Features

- CO Line Flash
- Dial-by-Name
- Display Security
- Keypad Facility
- LNR (Last Number Redial)
- Dial Pulse to Tone Switchover
- Save Number Redial (SNR)
- Speed Dial Pause Insertion
- System Speed Dial
- System Speed Zone (Groups)

2.76.5 System Speed Zone (Groups)

Description

System Speed Dial bins can be divided into zones or groups. iPECS systems have 10 zonesavailable. To access a SpeedDial bin assigned to a zone, the station must be allowed access to the Speed Dial Zone.

Each zone can be assigned to apply COS (Class of Service) dialing restrictions to the Speed Dialnumbers in the zone. When assigned, the system will apply the Station and LineCOS (Class of Service) to callsusing Speed Dial numbers in the zone.

Operation

Refer to System Speed Dial for operation.

Conditions

- ✓ The conditions of System Speed Dial apply to System Speed Dial numbers assigned to a zone.
- ✓ A station not assigned access to a zone will receive error tone when trying to access a System Speed Dial number in the zone.
- ✓ The COS (Class of Service) is applied to the Speed Dial number based on the active service mode, Day/Timed or Night.

Programming

Keyset Admin.

STATION

- Speed Dial Access (PGM 112-Button 8)

TABLES

- System Speed Zone (PGM 232)

Web Admin.

STATION DATA

- Common Attributes ➤Speed Dial Access

TABLES DATA

- System Speed Zone

Related Features

- System Speed Dial
- Class of Service

2.76.6 Flexible Speed

Description

System has the default range of speed code for individual Station speed and System speed codes.

To lessen length of speed code for user's convenience, system provides programmable options for selection of speed code range and length.To program/edit or dial speed code dial, the input of speed code should be in range of selected speed code type. Database compatibility is maintained because the internal data of speed code is not changed in system database. Only visual range of speed code is adjusted when selecting the speed code type.

Operation

Administrator can select one of speed code types

- Web Admin (System ID & Numbering Plans > System ID(100) > speed Numbering = select the desired type.

The default is Type 0 for each system as below:

System / Type		Station(individual) speed code			System(Shared) speed code		
		Code range	Digit length	How many	Code range	Digit length	How many
UCP	Type 0	000 - 099	3	100	20000 - 31999	5	12000
	Type 1	000 - 099	3	100	2000 - 9999	4	8000
	Type 2	000 - 099	3	100	200 - 999	3	800
	Type 3	00 - 19	2	20	200 - 999	3	800
	Type 4	00 - 19	2	20	20 - 99	2	80
	Type 5	000 - 099	3	100	1000 - 9999	4	9000
	Type 6	000 - 099	3	100	100 - 999	3	900
	Type 7	00 - 09	2	10	100 - 999	3	900
	Type 8	00 - 09	2	10	10 - 99	2	90
	Type 9	0 - 0	1		1 - 9	1	9
eMG800	Type 0	000 - 099	3	100	2000 - 9999	4	8000
	Type 1	000 - 099	3	100	200 - 999	3	800
	Type 2	00 - 19	2	20	200 - 999	3	800
	Type 3	00 - 19	2	20	20 - 99	2	80
	Type 4	000 - 099	3	100	1000 - 8999	4	8000
	Type 5	000 - 099	3	100	100 - 999	3	900
	Type 6	00 - 09	2	10	100 - 999	3	900
	Type 7	00 - 09	2	10	10 - 99	2	100
	Type 8	0 - 0	1		1 - 9	1	9
eMG80/100	Type 0	000 - 099	3	100	2000 - 4999	4	3000
	Type 1	000 - 099	3	100	200 - 999	3	800
	Type 2	00 - 19	2	20	200 - 999	3	800
	Type 3	00 - 19	2	20	20 - 99	2	80
	Type 4	000 - 099	3	100	1000 - 3999	4	3000

	Type 5	000 - 099	3	100	100 - 999	3	900
	Type 6	00 - 09	2	10	100 - 999	3	900
	Type 7	00 - 09	2	10	10 - 99	2	90
	Type 8	0 - 0	1		1 - 9	1	9

Speed Dial Phone Number Programming

User enter the speed code to program/edit the speed dial telephone number.
The speed code should be in range of selected the speed code range.

SPEED Dialing

User dials Speed Dial Code to make a phone call.
The speed code should be in range of selected the speed code range.

Conditions

- ✓ Suggest to change Type from default before program/use SPEED Dials. If change Type, during working with exist SPEED data, then range-out codes referred to by preset-forward etc. will be seen as an empty or range-out-assigned(65535) that need to be changed to a valid or need to be cleared by manual programming. If not, features or data referring the range-out speed code will go to failure.
- ✓ Be careful, if change speed type from 3 or more digit system-speed(xxx/xxxx/xxxxx) to 2 or less digit system-speed(x or xx) then the feature(off-net forward, etc.) data that have 2digit station-speed(xx) previously, may be wrong-identified as location of system speed xx or x. And vice versa in case of change from 2 digit or less system-speed(xx or x) to 3 or more digit speed code (xxx/xxxx/xxxxx).

Programming

Keyset Admin.

STATION

- Speed Dial Access (PGM 112-Button 8)

TABLES

- System Speed Zone (PGM 232)

Web Admin.

STATION DATA

- Common Attributes ➤Speed Dial Access

TABLES DATA

- System Speed Zone

Related Features

- System Speed Dial
- Class of Service

2.77 Station Call Coverage

Description

The Call Coverage feature permits your iPECS IP or LDP Phone to receive ring and answer callsdirected to a covered station. This feature is generally employed to allow a Secretarialanswering position to cover calls to other stations. When a covered station rings, the **{CALLCOVERAGE}** button LED will flash and the covering station may receive ring (immediate ordelayed) for the call. The covering station can answer the call using the **{CALL COVERAGE}**button, terminating ring at other stations. Once answered, the LED of **{CALL COVERAGE}**buttons for the station at other covering stations will extinguish.

Operation of this feature requires a **{CALL COVERAGE}** button at the covering iPECS IP or LDP Phoneand the covered station must activate call coverage. A station can have multiple Call Coveragebuttons each covering a different station and multiple stations can have a Call Coverage buttonfor a given station.

Call Coverage has several options, as below that are configured in the system database.

- Call Coverage On Busy
- Call Coverage Through Mobile
- Call Coverage Wake-Up call
- Call Coverage Ring and Delay Ring

Operation

iPECS IP & LDP Phones

*To assign a **{CALL COVERAGE}** button at the covering station*

- **[TRANS/PGM]** + **{FLEX}** + “*76” + covered Station number + **[HOLD/SAVE]**

To activate Call Coverage at the covered station

1) Press the **[TRANS/PGM]**button.
2) Dial “141”, the Call Coverage code.
3) Dial “1” to enable or “0” to disable Call Coverage.

*To assign ring for a **{CALL COVERAGE}** button*

1) Press the **[TRANS/PGM]** button.
2) Dial “142”, the Call Coverage Ring code.
3) Dial the delay in ring cycles (“00”-“15”).
4) When a covered station receives a call, the covering station will receive the following display:

CALL FOR STA xxxx FROM yyyy time

Conditions

✓ An iPECS IP or LDP Phone user may cover for a SLT(Single Line Telephone) or other stations. However, since a Flex button is required, a SLT(Single Line Telephone) cannot provide coverage for other stations.

- ✓ When off-hook or in DND, the covering station will only receive a visual indication of the call from the LED of the **{CALL COVERAGE}** button and display, no off-hook ring is provided.
- ✓ The **{CALL COVERAGE}** button will provide an appearance for Lines that do not appear on the covering station except for Private Lines. To cover for Private Lines, the covering station must have an appearance and be allowed access to the Private Line.
- ✓ Call Coverage attributes can be assigned either by the covered station user or in the system database.
- ✓ A station cannot provide call coverage for a station to which it is linked.

Programming

Keyset Admin.

STATION

- Call Coverage Enable (PGM 111-Button 12)
- Call Coverage Delay Ring (PGM 111-Button 13)
- Call Coverage On Busy (Web PGM 111, OFF/ON)
- Call Coverage Through Mobile Ext (Web 111-113, OFF/ON)
- Call Coverage On Busy Range (Web 111-113, External Call Only/ External and Internal Call)
- Call Coverage Delay Ring Method (Web 111-113, by Originator/by Member)
- Call Coverage For Wakeup Ring (Web 111-113, OFF/ON)
- Call Coverage Ring Type on Member (Web 111-113, Silence/Normal Ring)

Web Admin.

STATION DATA

- Common Attributes ➤Call Coverage Mode
- Common Attributes ➤Call Coverage Delay Ring
- Common Attributes ➤Call Coverage On Busy
- Common Attributes ➤Call Coverage Through Mobile Extension
- Common Attributes ➤Call Coverage On Busy Range
- Common Attributes ➤Call Coverage Delay Ring Method
- Common Attributes ➤Call Coverage For Wakeup Ring
- Common Attributes ➤Call Coverage Ring Type on Member

2.78 Station Groups

Description

Stations can be grouped for incoming call routing and Call Pick-up purposes. Ten types ofgroups can be defined:

- Circular
- Terminal
- ACD
- Ring
- Pick-Up
- External Voice Mail
- Integrated Voice Mail
- Feature Server UMS Group
- Net VM (Centralized External VM)
- Unified Communication Solution Server

Circular Station Group

In Circular group, calls to a station in the group will go to the station, if unavailable orunanswered in the hunt no answer time; the call will be directed to the next station defined inthe group.

The call will continue to hunt until each station in the group has been tried. The callremains at the last station or passes to a designated overflow station or group.

A Circular Station Group can be assigned with a pilot number (the Station Group Number) sothat calls to the pilot number will hunt. In this case, the call will be directed to the first station inthe group and, if needed, hunt through each station in the group until reaching the last station.The call may remain at the last station, passed to an overflow destination or sent to a voicemailbox.

Terminal Station Group

Calls to a station in a Terminal Station Group that encounter an unavailable or unansweredstatus will be routed through the hunt process.

The call will proceed to the next listed station inthe group until reaching the last listed station in the group. The call may remain at the laststation or be routed to an Overflow destination.

A Terminal Station Group can be assigned with a pilot number (the Station Group number) so thatcalls to the pilot number will hunt.

ACD Station Group

Calls can be sent to an ACD group by dialing the Station Group Number or assigning outside Lines to ring directly to the Station Group.

Calls are directed to the station in the group that hasbeen idle for the longest continuous time, Uniform Call Distribution. If all stations are busy orunavailable when the call is received, the call may be routed to an alternate location or maycontinue to wait (queue) for an available station in the group.

After queuing to the group, thecaller may be routed to an overflow destination, which can be a Station, Station Group or VoiceMailbox.

An ACD supervisor can be assigned to monitor the group and act to oversee operations of thegroup. The ACD Supervisor can print group statistics and activate alternate routing as well asassist agents.

Ring Station Group

A call to any station in the Group will cause all stations in the group to ring and any station inthe group may answer the call. If the call remains unanswered beyond the Overflow timer, thecall is sent to the Overflow destination, which can be a Station, Station Group or Voice Mailbox.

A Ring Group station can receive indication of Ring Group call while busy on another call in the form of Muted Ring. The call can be answered and Automatic Hold is supported.

Multiple calls can be received by a Station Ring Group and can be serviced in any order.

External AA/VM Station Group

This group is assigned to support an external Auto Attendant Voice Mail system that employsSLT(Single Line Telephone) ports to interface to the iPECS.

An External AA/VM group is assigned for either circular orterminal hunt. The External AA/VM may employ either in-band signaling over the audio channelor SMDI protocol with a signaling connection to the system RS-232 channel.

Pick-Up Station Group

A station can be assigned to a Call Pick-Up group and then may pick-up (answer) calls to otherstations in the group employing the system's Group Call Pick-Up feature.

Integrated AA/VM Group

Incomingcalls can be directed to one of 200 user-recorded System announcements, which may request furtherrouting instructions from the user in the form of caller dialed digits. These digits are employedto route the caller as defined in the system CCR (Caller Controlled Routing) Tables.

The integrated AA/VM Group Voice Mail application receives calls forwarded or recalling froma station. Such calls will receive the user's pre-recorded greeting and may leave voicemessages. The user may call the integrated AA/VM Group to review and manage theintegrated Voice Mail application.

Feature Server UMS Group

The Feature Server is a PC based TAPI application with high-end Auto Attendant, Voice Mailand Unified Messaging Service (Voice/Fax and e-mail). The iPECS Feature Server receivescalls, plays announcements, stores voice messages and forwards them as wave fileattachments to the user's e-mail. The application also receives Faxes and forwards them asattachments to e-mail. The Text-to-Speech option permits listening to e-mails as well as voicemails.

Net VM

This group is defined to support a Centralized Voice Mail system for a networked environment.At supported systems, the group is used to handle the AA/VM requirements from the centraliPECS. The Net VM group may be an external VM system or the iPECS Feature Server.

UCS Group

This group is defined to support the Unified Communication Solution available with the iPECSsystems.

Group Announcements

Station Group routing can be augmented with Systemannouncements recorded in the integrated AA/VM memory. Callers are routed to one of several user-recorded System announcements. The systemanswers the call and plays the defined 1st announcement to the caller. The announcement mayprovide the caller with routing options for Caller Controlled Routing. The 1st announcementmay be “Guaranteed” meaning it will play in full before routing the call. A 2nd announcementcan be provided to the caller should queue timers expire.

Conditions

- ✓ eMG80/100 supports 40 Station Groups with up to 70 members for each group.
- ✓ eMG800/UCP supports 100/200 Station Groups with up to 200 members for each group.
- ✓ Station Group calls are not routed to member stations that are in DND.
- ✓ When a member of a Circular, Terminal, ACD or Ring Group activates Call Forward, calls to the group may still route to the member based on the Member Forward option.
- ✓ A call transferred to a Station Group will follow the routing for the group and will not initiate the Transfer Recall process.
- ✓ Calls to a Station Group receive either ring-back tone or MOH while queued to the group.
- ✓ Calls, which are not answered in the Overflow time, are routed to the defined Overflow destination, station, group, etc. If no Overflow destination is defined, the call is dropped after expiration of the Overflow timer.
- ✓ One of the 200 System announcements may be assigned as the Overflow destination. These announcements can have a corresponding Caller Controlled Routing table allowing the caller to enter digits for further routing.

Programming

Keyset Admin.

STATION GROUPS

- Station Group Assignment (PGM 190)
- Station Group Attributes (PGM 191)
- Pickup Group Assignment (PGM 192)

Web Admin.

STATION GROUP DATA

- Station Group Assignment
- Station Group Attributes
- Pick-Up Group Assignment

Related Features

- Automatic Hold
- Automatic Call Distribution
- External Auto Attendant/Voice Mail
- Group Call Pick-Up
- MOH (Music-On-Hold)
- Integrated Auto Attendant/Voice Mail

2.79 Station Flexible Button Default Table

Description

When a station is registered, administrator can assign preset flexible button table to the station automatically by default.

Operation

1) Prepare default flexible button table for station in PGM "Flexible Button Defaults".
2) Configure conditions for each priority. You can choose 3 conditions. These conditions will be checked AND operation if the conditions are set.
3) Station model type
4) Number of flexible button(range)
5) Station number(range)
6) Register Stations.

Conditions

✓ If a station is not matched with conditions, the flexible buttons will be assigned by default according to number of buttons.

Programming

Web Admin.

TABLE DATA

- Flexible button Default Table (239)
- Preset Flexible Button Default (240)

STATION DATA

- Flexible Button (115/129)

2.80 SMDR (Station Message Detail Recording)

2.80.1 Call Cost Display

Description

Each SMDR call record includes a “Cost” field, which is a calculated estimate for the cost of thecall.

The call cost updates in real-time and displays in the LCD ofiPECS IP and LDP Phones in place of thecall duration.

The cost is determined by:

- Fixed charge per “Call Meter Pulse”,
- ISDN Advice of Charge or,
- Estimated cost updated based on Elapsed Call Timer and assigned costing.

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

Analog CO

Where “Call Metering Pulse” service is available from the carrier, the system will apply the“SMDR Cost per Unit Pulse” and the “SMDR Decimal” to the Call Metering received to estimatethe call cost.

When no “Metering Type” is selected, the system call duration is used with the cost/pulse anddecimal 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. Ifassigned, 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 establishthe call cost estimate. The cost is updated periodically at intervals of the “Elapsed Call Timer”.

Operation

System

Call cost is estimated automatically and output to iPECS IP and LDP Phone displays, and included in the SMDR output.

Conditions

✓ The call cost display begins after the “SMDR Start Timer” expires, if enabled, or at receipt of the first Call Meter Pulse.

✓ Once connected to the system, the call duration includes the total time the call is connected including periods when the call is on hold, in queue, etc.

✓ To enable Call Cost Display, the “SMDR Cost per Unit Pulse” and “SMDR Decimal” must be assigned. When not assigned, the call duration is provided by the system.

Programming

Keyset Admin.

CO/IP

- Metering Unit (PGM 142-Button 3)

SYSTEM

- SMDR Cost per Unit Pulse (PGM 177-Button 10)
- SMDR Decimal Location (PGM 177-Button 11)
- SMDR Start Timer (PGM 177-Button 12)
- Elapsed Call Timer (PGM 180-Button 19)

Web Admin.

CO LINE DATA

- Analog Attributes ➤SMDR Metering Unit

SYSTEM DATA

- SMDR Attributes ➤SMDR Cost Per Metering Pulse, SMDR Decimal Location, Start Timer
- System Timers ➤ Elapsed Call Timer

Related Features

- SMDR (Station Message Detail Recording)
- Lost Call Recording
- Traffic Analysis
- Outside Call Warning Tone Timer

Hardware

- RS-232 device to capture SMDR

2.80.2 Lost Call Recording

Description

Incoming calls where the caller hangs up before answer or while in a hold state are considered Abandoned or Lost calls. Special SMDR call records are provided for lost calls in real-time, as they occur, and a summary Lost Call count report is available on demand.

The real-time Lost Call records provide details on the called party, when and how long the call rang or was on hold before being abandoned, etc.

Description of the record details is provided in the following charts. As noted in the charts, the dialed number field indicates the type of call and the ring before the call was abandoned.

The first character in the NUM I field is the status of the call when abandoned:

R	normal ring to a station,
G	ring to a station group and
H	call placed in a hold state, including Transfer hold.

STA	CO	TIME	START	DIAL/CLI/CPN NUM-1	COST	ACCOUNT CODE	DIAL/CLI/CPN NUM II
EXT	31	00:00	14/05/02 15:45	R RING 01:35			

- Incoming call on CO Line 31 received on May 14, 2002 at 3:45 pm, rang the assigned stations for 1 minute and 35 seconds.

STA	CO	TIME	START	DIAL/CLI/CPN NUM-1	COST	ACCOUNT CODE	DIAL/CLI/CPN NUM II
101	31	00:00	14/05/02 16:45	R RING 02:03			

- Station 101 rang for an incoming call on CO Line 31 on May 14, 2002 at 4:45 pm, rang for 2 minutes and 3 seconds.

STA	CO	TIME	START	DIAL/CLI/CPN NUM-1	COST	ACCOUNT CODE	DIAL/CLI/CPN NUM II
101	02	00:00	15/05/02 09:35	R 100 RING 00:49			

- Incoming call on CO Line 02 on May 15, 2002 at 9:35 am forward from station 101 to station 100 and ring for 49 seconds.

STA	CO	TIME	START	DIAL/CLI/CPN NUM-1	COST	ACCOUNT CODE	DIAL/CLI/CPN NUM II
104	05	00:00	16/05/02 11:06	G401 RING 01:32			

- Incoming call on CO Line 05 on May 16, 2002 at 11:06 am routed to station 104 of Station Group 401 and ring for 1 minute and 49 seconds.

STA	CO	TIME	START	DIAL/CLI/CPN NUM-1	COST	ACCOUNT CODE	DIAL/CLI/CPN NUM II
401	17	00:00	16/05/02 14:03	G401 RING 00:39			

- Incoming call on CO Line 17 on May 16, 2002 at 2:03 pm routed to Station Group 401 and ring for 39 seconds.

STA	CO	TIME	START	DIAL/CLI/CPN NUM-1	COST	ACCOUNT CODE	DIAL/CLI/CPN NUM II
100	01	03:32	16/05/02 15:30	H100 03:02			

- Call on CO Line 1 on May 16, 2002 at 3:30 pm placed on hold by station 100 for 3 minutes and 2 seconds had total duration of 3 minutes and 32 seconds.

STA	CO	TIME	START	DIAL/CLI/CPN NUM-1	COST	ACCOUNT CODE	DIAL/CLI/CPN NUM II
129	23	00:45	18/05/02 08:40	H100 RING 00:33			

- Call on CO Line 23 on May, 18, 2002 at 8:40 am was transferred by station 100 to station 129 was on hold for 33 seconds.

The output for the Lost Call summary count report is shown in the figure below:

```
Lost call count start time: 05/01/02 09:31
Current time 26/04/02 16:32
Total Lost call count until now: 121
```

Operation

System Attendant

To print the summary Lost Call Count report

1) Press the **[TRANS/PGM]** button.
2) Dial “0115”, the Lost Call Count report code.
3) Press the **[HOLD/SAVE]** button.

To reset the Lost Call summary Count

1) Press the **[TRANS/PGM]** button.
2) Dial “0116”, the Lost Call Count Reset code.
3) Press the **[HOLD/SAVE]** button.

Conditions

- ✓ When the Lost Call Count is reset, the SMDR port will provide a “count reset” message.
- ✓ Individual Lost Call records are only available in real-time and not on-demand.
- ✓ “Print Incoming Calls” and “Print Lost Calls” must be enabled in the SMDR Attributes for the system to output real-time Lost Call records and for the Lost Call Count summary report.
- ✓ The fields of a Lost Call Record are the same as a normal SMDR Call Record.

Programming

Keyset Admin.

SYSTEM

- SMDR Attributes (PGM 177)

Web Admin.

SYSTEM DATA

- SMDR Attributes

Related Features

- Call Cost Display
- SMDR Call Records
- Traffic Analysis

Hardware

- RS-323 device to capture SMDR

2.80.3 SMDR Call Records

Description

SMDR (Station Message Detail Recording) provides detailed information on incoming andoutgoing calls.

Assignable options in the system database permit recording of all external calls,all external outgoing calls or toll calls, calls that exceed a fixed duration and intercom calls.

Callrecords are output either upon completion of the call (real-time) or in response to a request fromthe System Attendant. SMDR may be sent periodically via e-mail to a defined e-mail address.

The SMDR record output for external calls is as shown in the figure below. There are twoflexible fields, Field I and Field II. Each Field is defined to show Ring duration, CLI (Caller Id)or CPN (Called Party Number).

NO	STA	CO	TIME	START	DIAL/CLI/ CPNNUM1	COST	ACCOUNT CODE	DIAL/CLI/ CPN NUM II	DC
XXXX	SSSS	BBB	DD:DD	FF/FF/FF EE:EE	HCCCCCCCCCC CCCCCCCC	sssss	aaaaaaaa aaaa	hcccccccccc cccccccc	rr

The various fields or items for a Call Record are:

NO	SMDR record number, optional output, (PGM 177-Button 23)
STA	3 or 4 digit station number
CO	2 or 3 digit CO Line number
Time	Call duration in minutes and seconds
Start	Date and time call was placed/received
NUM I	Flex Field I, for outgoing call displays dialed number, for incoming call displays Ring duration, CLI or CPN (PGM 177-Button 20)
Cost	Cost of Call
Account Code	Account code entered for call
NUM II	Flex Field II for incoming call, displays Ring duration, CLI or CPN or blank, (PGM 177-Button 23)
DC	Disconnect Cause, optional output (PGM 177-Button24-5), with value: 01~7F: Follow cause value of Q.931 specification, 81: Disconnected, toll restricted 84/85: Disconnected, disconnect supervision 0: Other

The SMDR record output for internal calls is as shown in the figure and explained in the chart below.

```
---- Site Name :                    ------------------------------------------------------------------------
STA     CO      TIME      START               DIALEDNUMBER   COST           ACCOUNTCODE    CPNNUMBER
------- ------- -------   ------------------- -------------- -------------- -------------- --------------
100             00:00:01  06/11/08 10:53      E100
105             00:00:00  06/11/08 14:13      S103
110             00:00:10  06/11/08 11:05      E Conf Rm 1
120             00:00:03  06/11/08 11:11      P123
```

STA	DIALED	Meaning
100	E101	Station 100 called station 101 and station 101 answered. In a conference, station 100 was a conference master, station 101 was a conference member.
105	S103	Station 105 sent an internal SMS to station 103.
110	E Conf Rm 1	Station 110 was the member of conference room 1.
120	P123	Station 120 paged station 123.

Operation

System

For real-time SMDR, records are output after completion of the call as shown in the above

System Attendant

To print SMDR records

1) Press the **[TRANS/PGM]** button and Dial “0111”, the SMDR print code.
2) Enter the desired station range.
3) Press the **[HOLD/SAVE]** button.

To delete stored records

1) Press the **[TRANS/PGM]** button and Dial “0112”, the SMDR delete code.
2) Enter the desired station range.
3) Press the **[HOLD/SAVE]** button.

To abort SMDR printing

1) Press the **[TRANS/PGM]** button and Dial “0114”, the SMDR abort code.
2) Press the **[HOLD/SAVE]** button.

Conditions

- ✓ For SMDR, if the first dialed digit(s) match the programmed LD (Long Distance) code or the number of dialed digits exceeds the LD digit count, the call is considered an LD call. When behind a PBX, LD determination is made only if a PBX Trunk Access code is dialed as the first digit(s).
- ✓ Except for DISA calls, the duration of ring for an incoming call is provided in the Dialed number field.

- ✓ A header, including the assigned “Customer Site Id” is output after two blank lines and is repeated every 66th line.
- ✓ The SMDR output is a simple ASCII stream of up to 80 characters per line.
- ✓ When enabled, SMDR call record timing begins after the “SMDR Start Timer” expire sand ends at call completion.
- ✓ For security, if an Authorization code is entered as the Account code, the call record will show “STA-P” and the station number, or “SYS-P” and the Authorization code index number in place of the Account code.
- ✓ For incoming calls, the “NUM I” and “NUM II” fields will display the assigned data item– Ring Service time, CLI, or CPN. For outgoing calls, the NUM I field will always display the dialed number, user or system.
- ✓ In case of Networking call, Nxxxx(nxxxx) can be displayed by [Display Nxxxx for Net Number] option.

Programming

Keyset Admin.

SYSTEM

- SMDR Attributes (PGM 177)
- Field I (PGM 177-Button 20)
- Field II (PGM 177-Button 22)
- Print Serial No (PGM 177-Button 23)
- SMDR DISC Cause (PGM 177-Button 24-5)
- SMDR ICM SAVE (PGM177-Button 24-3)
- SMDR ICM PRINT (PGM177-Button 24-4)

Web Admin.

SYSTEM DATA

- SMDR Attributes ➤SMDR Ring/CLI/CPN Service-I
- SMDR Attributes ➤SMDR Ring/CLI/CPN Service-II
- SMDR Attributes ➤Print Serial No
- SMDR Disconnect Cause
- SMDR ICM Save
- SMDR ICM Print

Related Features

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

Hardware

- RS-323 device to capture SMDR

2.80.4 SMDR Enhanced Options for Internal Call

Description

Basically, for SMDR of internal call, unanswered call is printed and recorded without distinguishable information from an answered call.

And there is no way to show what the original dialed number is dialed in answered by another user scenario because the basic SMDR for internal call shows only the last connected station number.

Enhanced option of SMDR to show with distinguishable information for an unanswered call is available. Also an enhanced option to show the original dialed number is available.

NO	STA	CO	TIME	START	DIALED NUMBER	COST	ACCOUNT CODE	CPN NUMBER
XXXX	SSSS		hh:mm :mm	MM/DD/YY hh:mm	EXXXX RING mm:ss	sssss	aaaaaaaa aaaa	XXXX

The various fields or items for a Call Record are:

NO	SMDR record number, optional output, (PGM 177-Button 23)
STA	3 or 4 digit station number
CO	empty
TIME	Talk duration of answered call
Start	Date and time call was placed/received
DIALED NUMBER	ESSSS : SSSS is Ringing or Connected Station number **RING mm:ss for Ringing duration**
Cost	Cost of Call
Account Code	Account code entered for call
CPN NUMBER	**Original Dialed Digits**

Operation

Utilization of SMDR ICM Ring Information Option

Set the enhanced SMDR option for internal call to distinguish unanswered and answered.

Operation of SMDR ICM Ring Information option

P.177 - SMDR ICM Ring Info = OFF

1) Basic print and record of SMDR for internal call
2) TIME information is ringing duration for unanswered call, and it is talking duration for answered call.

Example

Unanswered call from 1013 to 1011 with ringing duration 00:00:08

STA	CO	TIME	START	DIALED
1013		00:00:08	02/28/20 20:38	E1011

Answered call from 1013 to 1011 with talking duration 00:00:25

STA	CO	TIME	START	DIALED
1013		00:00:25	02/28/20 20:38	E1011

P.177 - SMDR ICM Ring Info = ON

1) Show additional ringing time information.
2) Distinguishes unanswered and answered call by TIME information.
3) TIME information is talking duration, and it is 00:00:00 for unanswered call.

Example

Unanswered call from 1013 to 1011 with ringing duration 00:00:08

STA	CO	TIME	START	DIALED
1013		00:00:08	02/28/20 20:38	E1011

Answered call from 1013 to 1011 with talking duration 00:00:25

STA	CO	TIME	START	DIALED
1013		00:00:25	02/28/20 20:38	E1011

Utilization of SMDR ICM CPN Information Option

Set the enhanced SMDR option for internal call to show an original called party number.

Operation of SMDR ICM CPN Information option

P.177 - SMDR ICM CPN Info = OFF

1) Basic print and record of SMDR for internal call
2) Only the last connected user information is available.

Example

Call from 1013 to 1034, 1011 picked up the call

STA	CO	TIME	START	DIALED
1013		00:00:40	02/28/20 20:38	E1011

*Call from 1013 to group *403, member 1011 answered the group call*

STA	CO	TIME	START	DIALED
1013		00:00:40	02/28/20 20:38	E1011

P.177 - SMDR ICM CPN Info = ON

1) Show additional CPN (Called Party Number) information.
2) This is useful for answered by another user scenario because with this option calling party, called party, connected party is printed.
3) TIME information is talking duration, and it is 00:00:00 for unanswered call.

Example

Call from 1013 to 1034, 1011 picked up the call

STA	CO	TIME	START	DIALED	. . .	CPN
1013		00:00:40	02/28/20 20:38	E1011		1034

*Call from 1013 to group *403, member 1011 answered the group call*

STA	CO	TIME	START	DIALED	. . .	CPN
1013		00:00:40	02/28/20 20:38	E1011		*403

Programming

Keyset Admin.

SYSTEM

- SMDR Attributes (PGM 177)
- SMDR ICM Ring Info (PGM 177-Button 24-20)
- SMDR ICM CPN Info (PGM 177-Button 24-21)

Web Admin.

SYSTEM DATA

- SMDR Attributes ➤ SMDR ICM Ring Info
- SMDR Attributes ➤ SMDR ICM CPN Info

Related Features

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

Hardware

- RS-323 device to capture SMDR

2.81 System Administration

2.81.1 Keyset Administration

Description

The system database can be accessed and modified with the keypad and Flex buttons of aniPECS IP or LDP Phone. The display of the Phone is employed to view items in the iPECSdatabase.

You may be required to enter a password for access to Admin. Basedon the Multi-level password, you will have access to specified system database program codes in the Web Admin.

For detailed information on database administration and maintenance, refer to the iPECSAdministration&Programming Manual.

Operation

Operation is detailed in the iPECS eMG Administration and Programming Manual.

Conditions

- ✓ Only stations assigned with Admin access can enter and change the system database items. As a default, all iPECS IP and LDP Phones can access the database.
- ✓ The Keyset has access to most of the system database. However, certain characteristics can be modified through Web Admin.

Programming

Keyset Admin.

STATION

- Admin Access (PGM 113-Button 1)

SYSTEM

- Admin Access Authority (Web only)
- System Password (PGM 162)

Web Admin.

STATION DATA

- Common Attributes ➤Admin

SYSTEM DATA

- System Password

Related Features

- Web Administration
- Multi-Level Admin Access

Hardware

2.81.2 Multi-Level Admin Access

Description

Access to the Web Admin database is protected by password. Up to three (3) levels of access to thedatabase can be established by assigning a different password to each level.

The Maintenancelevel has access to the entire database, all maintenance routines and defines the AdminAccess Authority of the two remaining passwords. The User password can only access thedefined database items and cannot access the Station Program pages.

The Admin passwordhas access to the defined database items as well as Station Program pages. In Web Admin,a user of the Maintenance password establishes the Admin Program Codes and Web pages available to each password level.

Operation

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

Conditions

- ✓ Admin Access Authority is defined only in the Web Admin; it cannot be defined when using Keyset Admin.
- ✓ Admin Access Authority applies to all Admin access whether accessed via an iPECS IP or LDP Phone, ISDN, or IP channel.

Programming

Keyset Admin.

STATION

- Admin Access (PGM 113-Button 1)

SYSTEM

- Admin Access Authority (Web only)
- System Password (PGM 162)

Web Admin.

STATION DATA

- Common Attributes ➤Admin

SYSTEM DATA

- System Password
- Station Web Authorization
- Web Access Authorization

Related Features

- Keyset Administration
- Web Administration

2.81.3 Web Administration

Description

The system database is accessed and modified via an iPECS IP or LDP Phone, the LAN interface, MODU, or anISDN BRI/PRI line. The LAN, MODU, ISDN access the system’s Web server, which deliversthe database as a set of Web pages to the user’s Web browser. Under the proper conditions,both also allow for remote access to the system database.

For remote access with the LAN interface, the system must be assigned a remotely accessibleIP address. The IP address should be fixed either as a public IP address or through a NAPTserver with port forwarding. The browser should be pointed to the system's IP address and theTCP port assigned in the system database.

For the MODU, a PPP (Point-to-Point Protocol) connection can be established betweena standard modem and the MODUboard. The system negotiates access using 2 User Ids and matchingpasswords assigned in the database. In addition, for DID access the Called Party number mustmatch the assigned system PPP destination. Once a connection is established over the MODU, the user’s Web browser can be opened and pointed to the system’s IP address andassigned port for access to the database.

For the ISDN BRI/PRI, a PPP (Point-to-Point Protocol) connection can be established betweena standard ISDN modem and the iPECS BRI/PRI board. The connection can be set-up overa normal or DID BRI/PRI line. The system negotiates access using twoUser Ids and matchingpasswords assigned in the database. In addition, for DID access the Called Party number mustmatch the assigned system PPP destination. Once a connection is established over the BRI/PRI, the user’s Web browser can be opened and pointed to the system’s IP address andassigned port for access to the database.

When accessed, the system will deliver the iPECS Admin and Maintenance Web page. Fromthis page, selecting Admin & Maintenance will return the login page where the user must entera password that matches an assigned password. Based on the password entered, the user ispermitted access to specified system program codes.For detailed information on database administration and maintenance, refer to the iPECSAdministration& Programming Manual.

Operation

Operation is detailed in the iPECS Administration and Programming Manual.

Conditions

- ✓ For Web Admin, the password can be 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.
- ✓ If Web admin login fails five consecutive(5) times within 1-minute, then login is blocked for a ten (10) minute period and a “blocked access” log is stored in Http Admin Log.
- ✓ MODU is available only in eMG80/100.

Programming

Keyset Admin.

NUMBERING PLAN

- System IP Address Plan (PGM Code 102)

SYSTEM

- Web Server TCP Port (PGM 161-Button 14)
- WEB Password Encryption (PGM 161-Button 15)
- Admin Access Authority (Web only)
- Web Login ID Usage (PGM 160-Btn24-Btn12)
- System Password (PGM 162)

ISDN LINE

- PPP Attributes (PGM 205)

Web Admin.

SYSTEM ID &NUMBERINGPLANS

- System & Device IP

SYSTEM DATA

- System Attributes ➤Web Server Port, Web Admin Password Encryption
- System Password

ISDN LINE DATA

- PPP Attributes

Related Features

- Keyset Administration
- Multi-Level Admin Access

2.82 System Alarm

2.82.1 System Alarm Notification

Description

When an Alarm event is initiated, system serves several types of notification to:

- Ring to multiple internal station users
- Ring to multiple external parties
- Page to multiple groups
- Send email to multiple address
- Triggering of external control contact ports

Operation

Features

1) Alarm input to multiple internal ring
2) Alarm input to multiple external ring ("Alarm To External Call" table)
   - For simultaneous ring (with same “Scenario Priority” per each destination)
   - For failover scenario (with different “Scenario Priority” per each destination)
   - With day/time condition
   - With selection of alarm types to serve
3) Alarm input to multiple paging ("Alarm Paging" table)
   - For simultaneous paging (with same “Scenario Priority” per each destination)
   - For failover scenario (with different “Scenario Priority” per each destination)
   - With weekday and time condition
   - With selection of alarm types to serve
4) Alarm input to multiple email address per alarm types ("Alarm Email Sending" table)
   - With selection of alarm types to serve
   - With internal implementation of retrial & recovery of email sending
5) Play VSF recording
   - Play different message based on alarm type when internal and external party answers a ringing alarm
6) Alarm input to triggering of external control contact ports
   - Triggering of an external control port according to alarm input start/stop

2.82.1.1 Alarm System Notification

Internal Ring

Alarm Type	Internal Noti. Alarm Ring	Internal Noti. Annc. No(0 or 1 – 200)
Alarm1	OFF/ON	1
Alarm2	OFF/ON	2
Emergency	OFF/ON	3
(others …)	OFF/ON	. . .

Internal Noti. Alarm Ring (OFF/ON)

- Alarm ring to alarm ring enabled stations
- Alarm enabled stations : system-attendant, Stations who have Alarm Ring Coverage button(s)
  Alarm / Door Bell Enabled Stations for alarm port 1/2.
  Emergency Call Notification Stations for emergency call.
  DCOB Fault Notify Station for DCOB(E1/T1/PRI) fault notification
- {Alarm Coverage} flexible button assignment
  - Numbering Plan Code for Coverage Ring + “##” + Alarm Type Code
    (ex.: *76 + ## + 0) (ex.: *76 + ## + 15)
  - Alarm Type Code

Code No.	Alarm notification	Code No.	Alarm notification	Code No.	Alarm notification
0	All alarm	10	VM FULL	20	License Overflow
1	Emergency call	11	WTIM/WITB Base fail	21	TNET License Expire
2	DCOB Fault	12	WTIM/WTIB Chain fail	22	TAPI fail
3	BATH	13	ISMDR full	23	PMS fail
4	DOOR-Bell	14	CABINET alarm (for UCP)	24	UCS Server Link Down
5	Alarm-Bell	15	BAR full	25	PUSH Server Link Down
6	Station Overflow	16	IPCR fault	26	System Act on Slave
7	CO Overflow	17	IPWATCH fail	27	Web API SMDR Full
8	Alarm-Bell	18	Temp License Expire	28	Timeslot Full
9	Station Overflow	19	License Expire		

- To stop an alarm (alarm stop is supported only by alarm ringing station or external called party after answer. So, at least one of station for alarm internal ring needed to be assigned and "Internal Noti. Alarm Ring = ON" is strongly suggested currently)
  - By dial "alarm reset" code on a station who has the alarm ring
  - By press "stop" soft button on a station who has the alarm ring
  - By press "alarm coverage" button and select '1(me)/0(all)' on a station who has the alarm ring
  - By dial [alarm reset code] by external party after answer the alarm ring

Internal Noti. Annc. No (0 or 1 – 200)

- Play& Repeat VSF announcement when a station answers an alarm-to-internal ring

External Ring

Alarm Type	External Noti. Annc. No (0 or 1 – 200)	External Noti. Initial Delay= 0 or 1-3600
Alarm1	1	1
Alarm2	1	2
Emergency	1	3
(others …)	1	. . .

External Noti. Annc. No (0 or 1 – 200)

- Play& Repeat VSF announcement when an external party answers an alarm-to-external ring
- Even without a valid announcement, external ring is implemented
- '0' means not assign

External Noti. Initial Delay (0 or 1-3600)

- Duration of delay before external ring starts

External Noti. Cancel at Answer

- Option to cancel simultaneous ring or cancel on-going failover scenario when an external party answers the alarm ring
- Do not Cancel Other Trying Calls : leave all of on-going ring or retying calls
- Cancel All Other Trying Calls : cancel all of on-going ring or retying calls except answered calls
- Cancel All Except Prior Trying : cancel all except on-going ring or retying calls in prior Priority and answered calls

External Noti. Next Prio on Failover

- Option for condition to start of failover scenario to next priority (from 0 to 9)
- Immediately Go To Next : immediately implement next(less) priority at first busy/error/unanswered status of previous(prior) calls
- After Retry All Counts : implement next(less) priority after previous(prior) calls are all (re)tried and stopped by failure(busy/error/unanswered).

External Noti. Next Prio after Answer (0 or 1-3600)

- Normally failover scenario is not implemented anymore if anybody answers the alarm ring.
- But if this timer is set(1-3600 sec.) then failover scenario is started/resumed after this duration even after somebody answers the alarm ring

Paging

Alarm Type	Page Noti. Annc. No(0 or 1 – 200)	Page Noti. Initial Delay= 0 or 1-3600
Alarm1	1	1
Alarm2	1	2
Emergency	1	3
(others …)	1	. . .

Page Noti. Annc. No (0 or 1 – 200)

- Play& Repeat VSF announcement to page group(s)
- Without a valid announcement, paging is not implemented
- ‘0’ means not assign

Page Noti. Initial Delay (0 or 1-3600)

- Duration of delay before paging starts

Page Noti. Next Prio on Failover

- Option for condition to start of failover scenario to next priority (from 0 to 39)
- Immediately Go To Next : immediately implement next(less) priority at first busy status of previous(prior) paging
- After Retry All Counts : implement next(less) priority after previous(prior) paging are all (re)tried and stopped by failure(busy).

Page Noti. Next Prio after Answer (0 or 1-3600)

- Normally failover scenario is not implemented anymore if a page to a group is played to external page or listened by at least one station internally.
- But if this timer is set(1-3600 sec.) then failover scenario is started/resumed after this duration even after paging is done successfully.

Page Noti. Page Urgency Level (0-255)

- '0' is highest and '255' is lowest level
- Privilege to override other on-going page
  - Case 1) page group busy
    If page group is occupied by other paging, higher level page will override the lower page, so the previous on-going page will be stopped.
  - Case 2) page member busy
    If a station is receiving a lower paging then it is overrided by a higher level paging, so the station will hear new paging.
- Emergency, SoS pages are fixed to level '0'
- Normal page are fixed to level '255'
- Alarm page are programmable from level '0' to '255'

Page Noti. Override By Urgency

- Override paging call only or override all calls

Page Noti. Override DND

- Override DND/Pre-selected-msg. status of a station

Page Noti. Page Repeat Count

- Repeat count of paging
- Total Play count: default(1) + repeat count(0-99)

Page Noti. Page Repeat Interval

- Interval of paging repeat

Page Noti. Annc. Repeat Count

- Repeat count play announce per one repeat of paging
- Total Play count: default(1) + repeat count(0-99)

2.82.1.2 Alarm To External Call

SIP Station (Caller)

Registration of a Third Party SIP Extension

- License: “Third Party SIP Extension”
- It is a virtual Extension without a physical phone
- New Registration
  - Refer to “Flexible Station Number (P.105)” and get an unused Station Number
  - Input the Station Number to “SIP Station (Caller)” and press [Save]
  - A new Third Party SIP Extension will be registered automatically
- Replace of an exist Third Party SIP Extension
  - Refer to “Device IP Plan (P.103)” and get an used Third Party SIP Station Number
  - Input the Station Number to “SIP Station (Caller)” and press [Save]
  - The Third Party SIP Extension will be re-registered as a virtual Extension mode automatically
- Delete the virtual Third Party SIP Extension
  - Empty the “SIP Station (Caller)” and press [Save]
  - The Third Party SIP Extension will be automatically removed from system
- Replace of the previous virtual Third Party SIP Extension
  - Input different Station number in the “SIP Station (Caller)” and press [Save]
  - The previous Third Party SIP Extension will be removed from system automatically and the input is (re)registered as the virtual Third Party SIP Extension automatically.

Destinations (10 tables)

Index	Attributes	Value	Range
1	Scenario Priority		
	Telephone Number		
	Comment		
	No Answer Timer		
	Retry on Busy		
	Retry on No Answer		
	Retry Count		
	Retry Interval		
	Weekday		
	Service Time		
	Service for Alarms		
2			
3			
. . .			
10			

Scenario Priority (0-9)

- '0' is highest, '9' is lowest priority
- For single destination : anything of 0-9 (meaningless)
- For multiple destinations for simultaneous ring : same priority value
- For failover (busy/error/unanswered) scenario : different priority

Telephone Number

- CO access code + telephone number
- (example, 911112222, 9 is CO access code, 11112222 is telephone number)
- Mandatory input
- Without this, no service

Comment

- Just for memorization by naming

No Answer Timer (30-180 sec.)

- Unanswered duration of the external call, after this timer the external is disconnected if not answered

Retry on Busy (Yes/No)

- Retry to make the external call if calling is disconnected by busy cause
- If called party system serves 'busy prompt' after connect the call, it is not regarded as busy cause but is regarded as answered.

Retry on No Answer

- Retry to make the external call if call is disconnected after 'No Answer Timer'.

Retry Count (1-9)

- Additional to first trial, system will try to make call till this count on fail(busy/error/unanswered) condition.

Retry Interval (10-180 sec.)

- Pause duration per a retrial

Weekday

- Monday to Sunday

Service Time

- 'start time' to 'end time' (start time can be bigger than end time, etc. 2300 - 0559)

Service for Alarms

- Selection of alarm types to serve by this table
- Mandatory input
- Without selection of an alarm type, the alarm type is served

2.82.1.3 Alarm EmailSending

Destinations (10 tables)

Index	Attributes	Value	Range
1	E-mail Address		
	Comment		
	Service for Alarms		
2			
3			
. . .			
10			

E-Mail Address

- Mandatory input
- Without this, no service

Comment

- Just for memorization by naming

Service for Alarms

- Selection of alarm types to serve by this table
- Mandatory input
- Without selection of an alarm type, the alarm type is serve

2.82.1.4 Alarm Paging

Destinations (40 tables)

Index	Attributes	Value	Range
1	Scenario Priority		
	Page Zone		
	Comment		
	Retry on Busy		
	Retry Count		
	Retry Interval		
	Weekday		
	Service Time		
	Service for Alarms		
2			
. . .			
40			

Scenario Priority (0-39)

- '0' is highest, '39' is lowest priority
- For single destination : anything of 0-39 (meaningless)
- For multiple destinations for simultaneous paging : same priority value
- For failover (busy) scenario : different priority

Page Zone

- Mandatory input
- Without this, no service

Comment

- Just for memorization by naming

Retry on Busy (Yes/No)

- Retry to make paging

7) If page group is occupied by other paging
8) Or, all of page receiving member stations are not on valid state to receive page

Retry Count (1-9)

- Additional to first trial, system will try to make paging till this count on fail(busy) condition.

Retry Interval (10-180 sec.)

- Pause duration per a retrial

Weekday

- Monday to Sunday

Service Time

- 'start time' to 'end time' (start time can be bigger than end time, etc. 2300 - 0559)

Service for Alarms

- Selection of alarm types to serve by this table
- Mandatory input
- Without selection of an alarm type, the alarm type is served

2.82.1.5 Scenario & Programming

Internal Ring

Alarm to internal ring serve

Mandatory Programming

- For all of alarm types to serve
  - Alarm System Notification➤ Internal Noti. Alarm Ring = ON

Alarm to internal destinations

(Alarm contact 1/2)

- Alarm Attributes(P.163) ➤ Alarm Enable = ON
- Station Assignment
  - Station Data ➤ Common Attributes(P.111) ➤ Alarm / Door Bell Enabled
  - Or, Alarm Ring Coverage button assignment (including SLT(Single Line Telephone) by flexible button programming virtually in web admin)
  - Or, for only System Attendant station
    System Data ➤ System Attributes(P.160-161) ➤ Emergency Call ATD Notify = Use

Emergency Call

- Emergency Dialing Code Programming : Tables Data ➤ Emergency Code Table(P.226)
- Station Assignment
  - Zone Data ➤ Zone Attributes(P.439) ➤ Emergency Call Notify Station , Emergency Call Notify 2nd Station
  - Or, Alarm Ring Coverage button assignment (including SLT(Single Line Telephone) by flexible button programming virtually in web admin)
  - Or, for only System Attendant station
    System Data ➤ System Attributes(P.160-161) ➤ Emergency Call ATD Notify = Use

DCOB fault

- Station Assignment
  - Zone Data ➤ Zone Attributes(P.439) ➤ DCOB Fault Notify Station
  - Or, Alarm Ring Coverage button assignment (including SLT(Single Line Telephone) by flexible button programming virtually in web admin)
  - Or, for only System Attendant station
    System Data ➤ System Attributes(P.160-161) ➤ Emergency Call ATD Notify = Use

SIP Trunk Registration failure

- Station Assignment
  - Zone Data ➤ Zone Attributes(P.439) ➤ SIP Registration Fault Notify Station
  - Or, Alarm Ring Coverage button assignment (including SLT(Single Line Telephone) by flexible button programming virtually in web admin)
  - Or, for only System Attendant station
    System Data ➤ System Attributes(P.160-161) ➤ Emergency Call ATD Notify = Use

Redundancy Switching

- Redundancy Condition

- Redundancy Data ➤Redundancy Attributes(502)➤CPU Redundancy Usage = ON
- Change Active UCP By Power Fail = ON or Geographical Redundancy = ON

- Slave/Act – Slave UCP is Activated
  - Alarm(Internal Ring, Relay Contact Control, External Call, Paging) and Email
- Master/Act – Master UCP is Activated
  - Email

[For other kind of Alarms]

- Station Assignment
  - Alarm Ring Coverage button assignment (including SLT(Single Line Telephone) by flexible button programming virtually in web admin)
  - Or, for only System Attendant station
    System Data ➤ System Attributes(P.160-161) ➤ Emergency Call ATD Notify = Use

Alarm announcement

Alarm System Notification

- Internal Noti. Annc. No = 0 or a valid announcement#

External Ring

Simple scenario with single destination

Mandatory Programming

- Input one of "Alarm To External Call" table
  - Alarm To External Call➤ Scenario Priority : anything from 0 to 9 (meaningless)
  - Alarm To External Call➤ Telephone Number
  - Alarm To External Call➤ Service for Alarms

Optional Programming

- Basic options
  - Alarm System Notification➤ External Noti. Annc. No
  - Alarm System Notification➤ External Noti. Initial Delay
  - Alarm To External Call➤ No Answer Timer (30-180 sec.)
  - Alarm To External Call➤ Retry on Busy (Yes/No)
  - Alarm To External Call➤ Retry on No Answer
  - Alarm To External Call➤ Retry Count (1-9)
  - Alarm To External Call➤ Retry Interval (10-180 sec.)
  - Alarm To External Call➤ Weekday
  - Alarm To External Call➤ Service Time

Simple scenario with multiple destination for simultaneous ring

Mandatory Programming

- Input several "Alarm To External Call" tables
  - Alarm To External Call➤ Scenario Priority
    Each table should have same one of 0-9
  - Alarm To External Call➤ Telephone Number
  - Alarm To External Call➤ Service for Alarms

Optional Programming

- Basic options
  - Alarm System Notification➤ External Noti. Annc. No
  - Alarm System Notification➤ External Noti. Initial Delay
  - Alarm System Notification➤ External Noti. Cancel at Answer (cancel or maintain simultaneous calls)
  - Alarm To External Call➤ No Answer Timer (30-180 sec.)
  - Alarm To External Call➤ Retry on Busy (Yes/No)
  - Alarm To External Call➤ Retry on No Answer
  - Alarm To External Call➤ Retry Count (1-9)
  - Alarm To External Call➤ Retry Interval (10-180 sec.)
  - Alarm To External Call➤ Weekday
  - Alarm To External Call➤ Service Time

Complex scenario with multiple destination including failover(busy/error/unanswered)

Mandatory Programming

- Input several "Alarm To External Call" tables
  - Alarm To External Call➤ Scenario Priority
    Each table should have same one of 0-9 for simultaneous ring
    Each table should have less priority (bigger value) than original for failover ring
  - Alarm To External Call➤ Telephone Number
  - Alarm To External Call➤ Service for Alarms

Optional Programming

- Basic options
  - Alarm System Notification➤ External Noti. Annc. No
  - Alarm System Notification➤ External Noti. Initial Delay
  - Alarm System Notification➤ External Noti. Cancel at Answer (cancel or maintain simultaneous calls)
  - Alarm To External Call➤ No Answer Timer (30-180 sec.)
  - Alarm To External Call➤ Retry on Busy (Yes/No)
  - Alarm To External Call➤ Retry on No Answer
  - Alarm To External Call➤ Retry Count (1-9)
  - Alarm To External Call➤ Retry Interval (10-180 sec.)
  - Alarm To External Call➤ Weekday
  - Alarm To External Call➤ Service Time
- Failover options
  - Alarm System Notification➤ External Noti. Cancel at Answer (cancel or maintain failover calls)
  - Alarm System Notification➤ External Noti. Next Prio on Failover
  - Alarm System Notification➤ External Noti. Next Prio after Answer
  - Alarm To External Call➤ [Retiral conditions and count ...]

Paging

Simple scenario with single page group

Mandatory Programming

- Assign a valid(recorded) announcement
  - Alarm System Notification➤ Page Noti. Annc. No (1-200)
- Input one of "Alarm Paging" table
  - Alarm Paging➤ Scenario Priority : anything from 0 to 39 (meaningless)
  - Alarm Paging➤ Page Zone
  - Alarm Paging➤ Service for Alarms

Optional Programming

- Basic options
  - Alarm System Notification➤ Page Noti. Initial Delay
  - Alarm System Notification➤ Page Noti. Page Urgency Level (0-255)
  - Alarm System Notification➤ Page Noti. Override By Urgency
  - Alarm System Notification➤ Page Noti. Override DND
  - Alarm System Notification➤ Page Noti. Page Repeat Count
  - Alarm System Notification➤ Page Noti. Page Repeat Interval
  - Alarm System Notification➤ Page Noti. Annc. Repeat Count
  - Alarm Paging➤ Retry on Busy (Yes/No)
  - Alarm Paging➤ Retry Count (1-9)
  - Alarm Paging➤ Retry Interval (10-180 sec.)
  - Alarm Paging➤ Weekday
  - Alarm Paging➤ Service Time

Simple scenario with multiple page groups for simultaneous paging

Mandatory Programming

- Assign a valid(recorded) announcement
  - Alarm System Notification➤ Page Noti. Annc. No (1-200)
- Input several "Alarm Paging" tables
  - Alarm Paging➤ Scenario Priority
    Each table should have same one of 0-39
  - Alarm Paging➤ Page Zone
  - Alarm Paging➤ Service for Alarms

Optional Programming

- Basic options
  - Alarm System Notification➤ Page Noti. Initial Delay
  - Alarm System Notification➤ Page Noti. Page Urgency Level (0-255)
  - Alarm System Notification➤ Page Noti. Override By Urgency
  - Alarm System Notification➤ Page Noti. Override DND
  - Alarm System Notification➤ Page Noti. Page Repeat Count

- Alarm System Notification➤ Page Noti. Page Repeat Interval
- Alarm System Notification➤ Page Noti. Annc. Repeat Count
- Alarm Paging➤ Retry on Busy (Yes/No)
- Alarm Paging➤ Retry Count (1-9)
- Alarm Paging➤ Retry Interval (10-180 sec.)
- Alarm Paging➤ Weekday
- Alarm Paging➤ Service Time

Complex scenario with multiple page groups including failover(page group busy or none of idle station members)

Mandatory Programming

- Assign a valid(recorded) announcement
  - Alarm System Notification➤ Page Noti. Annc. No (1-200)
- Input several "Alarm Paging" tables
  - Alarm Paging➤ Scenario Priority
    Each table should have same one of 0-39 for simultaneous ring.
    Each table should have less priority (bigger value) than original for failover page
  - Alarm Paging➤ Page Zone
  - Alarm Paging➤ Service for Alarms

Optional Programming

- Basic options
  - Alarm System Notification➤ Page Noti. Initial Delay
  - Alarm System Notification➤ Page Noti. Page Urgency Level (0-255)
  - Alarm System Notification➤ Page Noti. Override By Urgency
  - Alarm System Notification➤ Page Noti. Override DND
  - Alarm System Notification➤ Page Noti. Page Repeat Count
  - Alarm System Notification➤ Page Noti. Page Repeat Interval
  - Alarm System Notification➤ Page Noti. Annc. Repeat Count
  - Alarm Paging➤ Retry on Busy (Yes/No)
  - Alarm Paging➤ Retry Count (1-9)
  - Alarm Paging➤ Retry Interval (10-180 sec.)
  - Alarm Paging➤ Weekday
  - Alarm Paging➤ Service Time
- Failover options
  - Alarm System Notification➤ Page Noti. Next Prio on Failover
  - Alarm System Notification➤ Page Noti. Next Prio after Answer
  - Alarm Paging➤ [Retiral conditions and count ...]

Email Sending

Simple scenario with single email address

Mandatory Programming

- Input of SMTP Server
  - System Data ➤ System Attributes(P.160-161) ➤ Common SMTP Attributes➤> SMTP Server IP Address
  - System Data ➤ System Attributes(P.160-161) ➤ Common SMTP Attributes ➤ SMTP Server Domain Name
- Input one of "Alarm Email Sending" table
  - Alarm Email Sending> E-Mail Address
  - Alarm Email Sending> Service for Alarms

Optional Programming

- None

Simple scenario with multiple email address

Mandatory Programming

- Input of SMTP Server
  - System Data ➤ System Attributes(P.160-161) ➤ Common SMTP Attributes > SMTP Server IP Address
  - System Data ➤ System Attributes(P.160-161) ➤ Common SMTP Attributes > SMTP Server Domain Name
- Input several "Alarm Email Sending" tables
  - Alarm Email Sending➤ E-Mail Address
  - Alarm Email Sending➤ Service for Alarms

Optional Programming

- None

External Control Relay triggering

Simple local system control (allocate one of External Control Relay port for Alarm Usage)

Mandatory Programming

- External Control Contacts(P.168) ➤ select "Alarm" usage
- External Control Contacts(P.168) ➤ control for : Alarm types

TNET system control (allocate one of External Control Relay port of TNET system for Alarm Usage)

Mandatory Programming

- T-NET Data ➤ T-NET Control Contact(P.334) ➤ select "Alarm" usage
- T-NET Data ➤ T-NET Control Contact(P.334) ➤ control for : Alarm types
- System Data ➤Alarm Relay TNET Access➤ relay control service No/Yes for each alarm to remote TNET system' external control contact port

Conditions

✓ Currently, does not support external call over Analog PSTN line.

Programming

Keyset Admin.

SYSTEM

- Alarm Attributes (PGM 163)

ZONE DATA

- Emergency Call Notify Station (PGM 439-Web only)
- DCOB Fault Notify Station (PGM 439-Web only)
- SIP Registration Fault Notify Station (PGM 439-Web only)

Web Admin.

SYSTEM DATA

- Alarm Attributes (PGM 163),
- Alarm System Notification,
- Alarm Relay TNET Access,
- Alarm To External Call,
- Alarm Email Sending,
- Alarm Paging

ZONE DATA

- Zone Attributes ➤Emergency Call Notify Station, Emergency Call Notify 2nd Station, DCOB Fault Notify Station, SIP Registration Fault Notify Station

Related Features

- Emergency Call
- Emergency Call Attendant Alert
- SIP Service

2.82.2 System Alarm Notification Stop

Description

Assigned stations receive the fault Alarm Signal, either a single tone burst repeated at 1-minute intervals or a continuous tone. The Alarm Signal may be terminated at the phone by dialing the Alarm Reset code or, if assigned, pressing the **{ALARM RESET}** button. To rearm the Alarm function, the alarm condition must be cleared and the Alarm signal terminated.

When enabled, the Emergency call or fault event causes an alarm ring at the Station. The Station can terminate the fault Alarm notification.

Station can have ‘Alarm Ring Coverage Button’ on flexible buttons for all or a specific type of alarm.Multiple Stations who have a specific ‘alarm coverage button’ is notified for the specific alarm ring. When the following desired ‘alarm type code’ is enabled, the station can view the alarm information and terminate the fault alarm notification.

A station can have multiple buttons of ‘alarm coverage’ according to alarm types.Alarm type code '0(all) is notified for all type of alarm ring.But an alarm coverage button for a specific alarm type goes first than 0(all) if there is.

Alarm Type code

Code No.	Alarm notification	Code No.	Alarm notification
0	All alarm	14	CABINET alarm (for UCP)
1	Emergency call	15	BAR full
2	DCOB Fault	16	IPCR fault
3	BATH	17	IPWATCH fail
4	DOOR-Bell	18	Temp License Expire
5	Alarm-Bell	19	License Expire
6	Station Overflow	20	License Overflow
7	CO Overflow	21	TNET License Expire
8	Alarm-Bell	22	TAPI fail
9	Station Overflow	23	Mitel PMS Link Down
10	VM FULL	24	UCS Server Link Down
11	WTIM/WITB Base fail	25	PUSH Server Link Down
12	WTIM/WTIB Chain fail	26	System Act on Slave
13	ISMDR full		

Operation

System

At detection of fault or Emergency Call, the Alarm signal is sent to assigned stations.

To stop alarm notification (At a station - who has the alarm ring)

1) Dial "alarm reset code”
   or, Press "stop" soft button
   or, Press "Alarm Coverage" button, and select digit '1(me)’ or ‘0(all)'

To stop alarm notification (At a CO called party – after answer the external alarm ring)

1) Dial "alarm reset code"

iPECS IP & LDP Phones

To assign a Flex button as an ***{ALARM RESET}*** *button to terminate the Alarm Signal*

- **[TRANS/PGM]** + **{FLEX}** + "*565" + **[HOLD/SAVE]**

To terminate an Alarm Signal, while idle by using ***{ALARM RESET}***

1) Dial the Flex Numbering Plan code "*565", confirmation tone is received and the Alarm Signal is terminated. If the alarm condition is cleared, the system will automatically rearm the alarm monitoring.
   Or, Press the **{ALARM RESET}** button.

To assign a Flex button as an ***{Alarm Coverage}*** *button to terminate the Alarm Signal*

- **[TRANS/PGM]** + **{FLEX}** + "*76" + ## + Alarm type code + **[HOLD/SAVE]**

To terminate an Alarm Signal, while idle by using ***{Alarm Coverage}***

1) Dial the Flex Numbering Plan code "*76##" and "alarm type code", confirmation tone is received and the Alarm Signal is terminated. If the alarm condition is cleared, the system will automatically rearm the alarm monitoring.
   Or, Press the **{Alarm coverage}** button.

External CO Party

To terminate an Alarm Signal, while hearing alarm announcement after answer the CO call

1) Dial the Flex Numbering Plan code "*565", confirmation tone is received and the Alarm Signal is terminated.

Conditions

- ✓ The station must terminate the Alarm signal at their phone. However, a Main Attendant can terminate the Alarm signal at the System Attendant.
- ✓ When no station is assigned to receive an Alarm, the System Attendant will receive the notification.
- ✓ For Digital Line and SIP registration faults, the station in the Zone associated with first Line of the BRI/E1/PRI interface or SIP channel receives the alarm signal.
- ✓ The notification process for Emergency calls follows the Zone from which the call was made and the System Attendant. The notified station can review the history of Emergency calls only from the Zone. The System Attendant can review Emergency Call history for all Zones.

Programming

Keyset Admin.

SYSTEM

- Alarm Attributes (PGM 163)

ZONE DATA

- Emergency Call Notify Station (PGM 439-Web only)
- DCOB Fault Notify Station (PGM 439-Web only)

- SIP Registration Fault Notify Station (PGM 439-Web only)

Web Admin.

SYSTEM DATA

- Alarm Attributes (PGM 163),
- Alarm System Notification,
- Alarm Relay TNET Access,
- Alarm To External Call,
- Alarm Email Sending,
- Alarm Paging

ZONE DATA

- Zone Attributes ➤Emergency Call Notify Station, Emergency Call Notify 2nd Station, DCOB Fault Notify Station, SIP Registration Fault Notify Station

Related Features

- Emergency Call
- Emergency Call Attendant Alert
- SIP Service

Hardware

2.82.3 System Alarm Announcement

Description

When an alarm ring is picked up by a internal station or external user, an audible message that alerts about the reason and related data is announced. This will help a user who received an alarm ring to do what and how to clear the status.

Operation

Assignment of announcement for an alarm type

1) Record system announcements by system attendant or upload wave files.
2) Assign the announcement # per an alarm type.

Alarm System Notification

Alarm Type	Internal Noti.Annc. No	External Noti.Annc. No	Page Noti.Annc. No
Alarm1	0, 1-200	0, 1-200	0, 1-200
Alarm2	0, 1-200	0, 1-200	0, 1-200
Emergency	0, 1-200	0, 1-200	0, 1-200
(others …)	0, 1-200	0, 1-200	0, 1-200

Announce a message on an internal alarm condition

1) An alarm condition is initiated.
2) The alarm rings to attendant, specified or alarm-ring coverage stations.
3) A user picks up the alarm ring.
4) Assigned announcement (Usage=On, Announcement #=recorded & valid system announcement #) is played.

Announce a message on an external alarm condition

1) An alarm condition is initiated.
2) The alarm rings to external party, specified by Alarm To External Call
3) A user picks up the alarm ring.
4) Assigned announcement (Usage=On, Announcement #=recorded & valid system announcement #) is played.

Conditions

- ✓ If the specified announcement assigned for an alarm type is not recorded then it will not be played.
- ✓ For emergency and bath alarm, the station number who initiated the alarm will be played after announcement.
- ✓ Log, notification and repeat scheme for alarm basic and failure will be considered later.

Hardware

2.82.4 Triggering of External Contact Relay Port for Alarm

Description

By association an alarm type with an external relay port, system can control an external circuit connected to the relay. The association will create a hardwired signaling externally when an alarm condition is initiated.

Operation

Association an alarm type with an external relay port

1) Assign Target system of T-NET including my system.

Alarm – External Relay Control

	Alarm Port1	Alarm Port2	Emergency Call	DCO Fault	. . .
My System	On/Off	On/Off	On/Off	On/Off	On/Off
CM	On/Off	On/Off	On/Off	On/Off	On/Off
LM 1	On/Off	On/Off	On/Off	On/Off	On/Off
LM 2	. . .				
LM 3					
. . .					

2) Set usage to 'Alarm' for external control contact port(s) and mark alarm types for association.

Programming# 168 (External Control Contacts)

Programming# 334 (T-NET Control Contact)

External Control Contact	Usage
First	(x) Unused (x) LBC (x) Door Open (x) External Control Device 1 (x) External Control Device 2 (O) Alarm [Mark] Alarm Port 1, Alarm Port 2 911, DCO, SIO, STA, CO,STA, CO, …
Second	. . .
Third	. . .
Fourth	. . .

Triggering a port of external relay on an alarm condition

1) An alarm condition is initiated.
2) The associated port of external relay is triggered and it activates an action on external device circuit connected.

Hardware

2.83 System Networking

2.83.1 Centralized Control T-NET

Description

In a Centralized Control T-NET (Transparent Network), a central system controls all remote modules and terminals providing transparent networked access to all the features and functions of the central system as well as the resources connected to the system.

Where the remote device is not directly reachable by the system, RTP packets must be relayed through a local VoIP channel. A remote device may not be reachable when WAN access for the device is through a firewall or NAPT server. In this case, the remote devices are assigned a zone to manage RTP traffic between other devices connected in the T-NET. The zone defines when an individual device requires use of the local VoIP channel. Zones are used to identify other group characteristics as outlined in section Remote Device Zone Management.

Remote sites may include an system operating in the local mode as a live back up to the remote central system. Under normal circumstances, the central system controls remote devices (gateway Modules and terminals) including any local system VoIP channels. However, should the WAN connection between the central system and the remote devices fail, the local system will assume the call server responsibility for the local devices. The local system thus provides local survivability and, based on configuration, may provide PSTN back-up service (Fail-over) for internal calls that normally route over the WAN.

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

Operation

System

Operation of Centralized Network is automatic when configured & defined.

Conditions

- ✓ In a Centralized Network, the maximum number of channels available is the maximum number of channels supported by the central system.
- ✓ In T-NET, Centralized Miscellaneous functions (Relay support, MOH, BGM, Alarms and External Page) are not supported but, all terminals in the T-NET can make and receive pages.
- ✓ 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.
- ✓ The local system will take over operation of registered devices if the central controlling system does not respond to three consecutive poll attempts over a 10-second period. The central system will gain control automatically upon return of the WAN connection.
- ✓ System can be installed behind a NAPT however, Fixed Nat – port forwarding is required for the host to be reachable by remote devices.
- ✓ Up to 15 local systems can operate in local mode as part of a T-NET if central system is eMG80/100.
- ✓ Up to 32 local systems can operate in local mode as part of a T-NET if central system is eMG800.

- ✓ Up to 100 local UCP modules can operate in local mode as part of a T-NET.
- ✓ You can set T-NET enable ON/OFF for using T-NET in local system.
- ✓ You must install a license and enable T-NET for all iPECS UCP modules that are part of the T-NET.
- ✓ Voice switching will be handled in local system if both caller and called device in same eMG system and both are TDM device without relay.
- ✓ eMG could be work as both central and local system.
- ✓ In case of eMG, Local VSF(VMIU/B) should be used when "RTP Relay Device Utilization" is set to "Separated (SRC to DEST).
- ✓ In case of UCP, Local VSF(UVMU/UVM) should be used when "RTP Relay Device Utilization" is set to "Separated (SRC to DEST).

Programming

Web Admin.

T-NET DATA

- T-NET Basic Attributes
- T-NET CM Attributes
- T-NET LM Attributes
- T-NET FoPSTN Attributes

Related Features

- Remote Device Zone Management

Hardware

- Remote system to support Local Survivability

2.83.2 Distributed Control Network

Description

In the Distributed Control Network, each iPECS system maintains control over the devicesregistered to it. The networked systems communicate employing QSig over ISDN Lines or H.450 over IP Lines for basic networking capabilities and the proprietary iPECS protocol for the advanced networking features. QSIG employs ISDN PRI channels only with support for ESTI standards ETS 300-237/238/256/257/260/261/361/362/363/364.

As a member of the network, you will have access to resources of othersystemsin the network such as outside Lines, Stations and Paging. A Network Numbering Plan determines the system call routing; your dialed digits are compared with the Numbering Plan Table for routing. Station to station calling, Call Transfers, Conferencing, Call Back, Call Waiting/Camp-on, Call Forward,and DNDoperation are transparent to the users although conditions noted below may apply.In addition, other features and functions as detailed in the following sections are available to you as part of the distributed network environment.

Operation

Operation of Distributed Networking is automatic when configured.

Conditions

- ✓ To use the networking features, the software lock-key installation is required. To obtain a lock-key, contact your local Ericsson-LG Enterprise representative with the system serial number.
- ✓ Each station must have a unique station number of up to eight (8) digits, which is based on their local system Station Numbering Plan.
- ✓ The Network Numbering Plan permits 251 entries, allowing up to 251 iPECS systems to be in a single distributed network environment.
- ✓ The alternative route employs a Speed Dial number to place a call and is not a Networked call. Thus, the Distributed Control Network features do not apply.

Programming

Keyset Admin.

SYSTEM

- Networking Basic Attribute (PGM 320)
- Networking CO Line Attribute (PGM 322)
- Network Numbering Plan Table (PGM 324)
- Network Feature Code Table (PGM 325)

Web Admin.

NETWORKING DATA

- Network Basic Attributes
- Network Supplementary Attributes
- Network CO Line Attributes
- Network Numbering Plan
- Network Feature Code Table

2.83.2.1 Net Intercom Call

Description

You can place calls to stations connected to other networked systems. Station calls across the network are always terminated in the Tone Ring Intercom Signaling mode. Station calls are placed in the normal manner by dialing the desired station number or pressing a **{Net-DSS}** button.

Operation

iPECS IP & LDP Phones

*To assign a Flex button as a **{NET DSS}** button*

- **[TRANS/PGM]** + {FLEX} + Station Number + **[HOLD/SAVE]**

To place an intercom call to a networked station

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial station number or press the **{NETDSS}** button.
3) Await answer.

SLT(Single Line Telephone)

To place an intercom call

1) Lift the handset to receive ICM dial tone.
2) Dial the station number.
3) Await answer.

Conditions

✓ A **{NET DSS}** button LED indicates the status of the associated station.

✓ The LCD will display the caller's Name if Calling Name Identification Presentation (CNIP) is enabled in the Networking parameters and the caller's Name is assigned in their local iPECS system.

✓ If no idle path is available over the Network, you will hear error tone.

✓ Network Intercom calls always signal in the Tone ring mode and are not subject to Intercom Caller Controlled ICM Signaling.

✓ If a network failure occurs, the system will place an outside call using the Alternate Speed Dial bin assigned in the Network Numbering Plan that corresponds to the called user. Note this is not a Distributed Network call and thus features associated with Networking such as Call Back, are not available.

✓ Network Intercom calls are subject to the CO Call Restriction Timer.

Programming

Keyset Admin.

SYSTEM

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

- Network Feature Code Table (PGM 325)

Web Admin.

NETWORKING DATA

- Net Basic Attributes
- Net CO Line Attributes
- Net Numbering Plan

Related Features

- Intercom (ICM) Call
- Speed Dial

Hardware

2.83.2.2 Net Call Transfer

Description

An active call, internal or outside, can be transferred to other stations across the Distributed Network. Calls can be transferred across the network employing screened or unscreened transfer in the same manner as a transfer within a system and is transparent to the user.

From a network perspective, the iPECS can use either the 'Join' or 'Re-route' signaling method to complete a call transfer over the network. The Networking Transfer Mode assigns the method that will be used and should be selected to match other systems connected in the private network.

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

Operation

iPECS IP & LDP Phones

*To assign a **{NET DSS}** button*

- **[TRANS/PGM]** + {FLEX} + Station Number + **[HOLD/SAVE]**

To transfer an active call with Screened Call Transfer

1) Press the **[TRANS/PGM]** button.
2) Dial the station to receive the transfer.
3) At answer, announce the call.
4) Hang-up to complete the transfer.

Or,

1) Press the **{NET DSS}** button for the desired station.
2) At answer, announce the call.
3) Hang-up to complete the transfer.

To transfer an active call with Unscreened Call Transfer

1) Press the **[TRANS/PGM]** button.
2) Dial the station to receive the transfer.
3) Hang-up to complete the transfer.

Or,

1) Press the **{NET DSS}** button for the desired station.
2) Hang-up to complete the transfer.

SLT(Single Line Telephone)

To transfer an active call with, Screened Call Transfer

1) Momentarily depress the hook-switch.
2) Dial the Station number to receive the transfer.
3) At answer, announce the call.
4) Hang-up to complete the transfer.

To transfer an active call with Unscreened Call Transfer

1) Momentarily depress the hook-switch.
2) Dial the Station number to receive the transfer.
3) Hang-up to complete the transfer.

Conditions

- ✓ A **{NET DSS}** button LED indicates the status of the associated station.
- ✓ iPECS eMG automatically determines the type of transfer, local or networked, based on the station number dialed and the Network Numbering Plan.
- ✓ If you press the **[TRANS/PGM]**or other buttons during a transfer, the transfer is cancelled. The original call is re-established if the **[TRANS/PGM]** button or original CO Line button is pressed.
- ✓ Network Call Transfer will recall to the transferring station if no response is received from the called system in the Net Transfer Recall Time.
- ✓ Networked systems are not designed to be non-blocking. Should no path be available to complete the transfer, error tone is provided to the caller.
- ✓ If the receiving station is busy, the transferring user will receive busy tone and may camp the call to the busy station. If the receiving station is in DND, the call is rejected, the caller receives fast-busy tone and cannot camp-on.
- ✓ If a network failure occurs, the system will place an outside call using the Alternate Speed Dial bin assigned in the Network Numbering Plan that corresponds to the called user. Note this is not a Distributed Network call and thus features associated with Networking such as Call Back, are not available.
- ✓ The LCD will display the caller's Name if Calling Name Identification Presentation (CNIP) is enabled in the Networking parameters and the caller's Name is assigned in their local iPECS system.

Programming

Keyset Admin.

SYSTEM

- Networking Basic Attribute (PGM 320)
- Networking CO Line Attribute (PGM 322)
- Network Numbering Plan Table (PGM 324)
- Network Feature Code Table (PGM 325)

Web Admin.

NETWORKING DATA

- Net Basic Attributes
- Net Supplementary Attributes➤Net Transfer Mode, Net Trans Recall timer
- Net CO Line Attributes
- Net Numbering Plan

Related Features

- ■ Call Transfer, Speed Dial

2.83.2.3 Net Call Identification Service

Description

When Calling Name Identification Presentation (CNIP) and Connected Name Identification Presentation (CONP) are enabled, the station name is provided as the caller identification received and displayed at a station receiving a call from a Networked station.

Operation

Operation of the Station Name Display is automatic when configured.

Conditions

✓ The Station Name must be defined in the system of the calling station.

Programming

Web Admin.

STATION DATA

- Station Name Display

NETWORKING DATA

- Net Basic Attributes➤Net CNIP Enable, Net CONP Enable

Related Features

- Net Intercom Call
- Net Call Transfer

Hardware

2.83.2.4 Network Call back (Call Completion)

Description

When you call a networked station that is busy, you can leave a Call Back request. When the busy station returns to idle, you will receive a Call Back from the system.

Upon answer, the iPECS systemwill place a call to the originally busy station. Network Call Back operates in the same manner as Call Back and is subject to the conditions associated with Call Back.

From a network perspective, iPECS systememploys H.450 Call Completion, Busy Subscriber (CCBS) protocol to provide call back capabilities. iPECS systemsupports the Connection Retain mode in the H.450 (CCBS) protocol

Operation

To activate a Network Call Back while receiving busy tone

1) Press the **[CALLBK]** button.
2) The call is cleared after a confirmation tone.
3) When the busy station goes to Idle, the originator receives a callback ring.
4) When the originator answers the callback ring, a new call will be activated to the calling station.

Conditions

- ✓ A stand-alone IP Phone that supports H.450 can activate the Call Completion feature.
- ✓ A station can leave or have only one callback message, a new request cancels the previous request and the new request is honored.
- ✓ When you call a networked station, if the station is in DND or does not answer, you will not be able to leave a Voice message using the **[Message/Call Back]** button. You can leave a Voice message only if the called station forwards calls to their Voice Mailbox.
- ✓ When you receive Call Back ring, the system dedicates a network path for your call. You must answer your Call Back ring prior to the expiration of the Network Transfer Recall Timer or, the Call Back is cancelled..
- ✓ There are two methods to provide Net Call Back, connection mode and connectionless mode based on the Net CC Retain Mode.

Programming

Keyset Admin.

SYSTEM

- Networking Basic Attribute (PGM 320)
- Networking CO Line Attribute (PGM 322)
- Network Numbering Plan Table (PGM 324)
- Network Feature Code Table (PGM 325)

Web Admin.

NETWORKING DATA

- Net Basic Attributes➤Net CC Retain Mode, Net Trans Recall timer

- Net CO Line Attributes
- Net Numbering Plan

Related Features

- Call Transfer, Station
- Net Call Transfer

Hardware

2.83.2.5 Net Call Camp-On (Call Offer)

Description

When calling to a networked station that is busy, you may camp a call on to the called station. The called station receives a camp-on tone and the **[HOLD/SAVE]** button LED will flash. Network Camp-on operation is the same as system Camp-on and is subject to the same conditions.

From a network perspective, iPECS system employs H.450 Call Offer protocol to provide camp-on capabilities.

Operation

iPECS IP & LDP Phones

To activate Call Offer

1) Dial a busy station number and receive busy tone.
2) Press the **[CAMP ON]** button or “*”.
3) The busy station receives off-hook muted ring.
4) The calling station hears a ring-back tone.

To answer the Call Offer

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

Or,

1) Hang-up and receive normal ring.
2) Lift the handset or press the **[SPEAKER]** button.

Conditions

✓ Call Offer is only applied to a station that is in talking state.
✓ If the busy party is in a conference or paging, Call offer is not available.
✓ The system does not support the path reservation mode defined in the QSIG specification.

Programming

Keyset Admin.

SYSTEM

- Networking Basic Attribute (PGM 320)
- Networking CO Line Attribute (PGM 322)
- Network Numbering Plan Table (PGM 324)
- Network Feature Code Table (PGM 325)

Web Admin.

NETWORKING DATA

- Net Numbering Plan

Related Features

- Call Waiting/Camp-On

Hardware

2.83.2.6 Net Conference

Description

You can join stationsand outside Lines from other networked systems to a conference call with other stations or CO lines.

Operation

iPECS IP & LDP Phones

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 you receive 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.
6) Press the **[CONF]** button to establish conference.

To place a conference on hold

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

To retrieve held conference

1) Lift handset and Press the flashing **[CONF]** button, all parties reconnected.

To drop held conference

1) Press the flashing **[CONF]** button on idle, all parties disconnected.

SLT(Single Line Telephone)

To establish an 3-party conference

1) Establish the first call.
2) Momentarily press the hook-switch, the connected party is placed on exclusive hold and you receive dial tone.
3) Place the second call.
4) When connected, momentarily press hook-switch twice in 2 seconds to connect all parties.

Conditions

- ✓ Multi-line conference time is not applied on 3-party Net Conference.
- ✓ A SIP phone cannot initiate a Net Conference.

Programming

Keyset Admin.

SYSTEM

- Networking Basic Attribute (PGM 320)
- Networking CO Line Attribute (PGM 322)
- Network Numbering Plan Table (PGM 324)
- Network Feature Code Table (PGM 325)

Web Admin.

NETWORKING DATA

- Net Basic Attributes
- Net Numbering Plan

Related Features

- Multi-Party Conference

2.83.2.7 Net Station Message Waiting

Description

When a called station does not answer or is in DND, you can activate a Station Message Wait indication to request a Call Back. A station may receive a Message Wait from any number of other stations in the system. The station receiving the Message Wait can return the calls using the **[MESSAGE/CALL BACK]** button.

Operation

iPECS IP & LDP Phones

To leave a Message Wait, while receiving ring back tone or no response on a call announce

1) Press the **[MESSAGE/CALL BACK]** button, confirmation tone received.
2) Hang up, Message Wait activated.

To leave a Message Wait, while receiving DND tone

1) Press the **[MESSAGE/CALL BACK]** button, confirmation tone received.
2) Hang-up, Message Wait activated.

To retrieve Station Messages Waiting

1) Press **[MESSAGE/CALL BACK]** button, the message contents summary as below or the Station Messages Waiting list will display.

ST CLVS VM FS MS 001 001 005 006 001 004

2) Dial “1” to select ST (Station Message Wait)
3) “1” = ST: Station Message Wait
4) “2” = CL: CLI Message Wait
5) “3” = VS: Integrated VM Message Wait
6) “4” = VM: external Voice Mail
7) “5” = FS: Feature server
8) “6” = MS: SMS message wait
9) Press the **[VOL UP]**/**[VOL DOWN]** button to scroll through the Station Messages.

MSG : 700100 700118

To return a call for the current Station Message

1) Press the **[HOLD/SAVE]** button.

To delete the first Message Wait from the list

1) Press the “*” button, the list is updated removing the message from the list.

To delete all waiting Messages

1) Press the**[SPEED]** button.
2) Press the “*” button twice.

SLT(Single Line Telephone)

To leave a Message Wait, while receiving ring back tone or no response on a call announce

1) Momentarily press the hook switch.
2) Dial "*556", Activate Message Wait/Call Back code.
3) Hang up, Message Wait is activated.

To leave a Message Wait, while receiving DND tone

1) Momentarily press the hook switch.
2) Dial "*556", Activate Message Wait/Call Back code.
3) Hang up, Message Wait activated.

To retrieve a Station Message Wait;

1) Dial "*557", Message Wait/Call Back Answer code.

Conditions

- ✓ The system employs a line from the Network CO Group to return a Station Message Wait call back.
- ✓ Only LIP and LDP Phones with an LCD can receive a Station Message Wait from a networked station.

Programming

Keyset Admin.

SYSTEM

- Networking Basic Attribute (PGM 320)
- Networking CO Line Attribute (PGM 322)
- Network Numbering Plan Table (PGM 324)
- Network Feature Code Table (PGM 325)

Web Admin.

NETWORKING DATA

- Net Basic Attributes
- Net CO Line Attributes
- Net Numbering Plan

Related Features

- Station Message Wait/Call Back
- Do Not Disturb (DND)

2.83.2.8 Net Call Forward

Description

A user can forward (send) their calls to another station, Station Group or a user's VMIB mailbox across the private network. The user may forward all calls immediately, if the user does not answer the call or if the station is busy. In addition, the user may select to activate 'Follow-me' forward, forwarding calls from a location other than the station. Calls forward in the same manner as a Call Forward within a system and is transparent to the user.

From a network perspective, the iPECS system can use either the 'Join' or 'Re-route' signaling method to complete a call forward over the network. The Networking Transfer Mode assigns the method that will be used and should be selected to match other systems connected in the private network.

- Join - an additional connecting path is needed to transfer the call to another station.
- Routing - a new connecting path is used to forward the call.

You select the type and condition under which calls will forward by entering a Call Forward code as follows:

Code 0	Remote Call Forward, forwards all calls to the station, except recalls, activated from a remote station, Call Forward, Follow-me.
Code 1	Unconditional, all calls to the station, except recalls, forward 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

iPECS IP & LDP Phones

To activate Call Forward, (Unconditional, Busy, No Answer, or Busy/No Answer)

1) Lift the handset or press the **[SPEAKER]** button to receive dial tone.
2) Press the **[FWD]** button or Soft key. For the LIP8002, the **[DND]** button is used to activate Forward.
3) Dial desired Call Forward code (“1”~“4”).
4) Dial the station or station group to receive calls.
   Or, Dial an outside Line access code (9, 8xx, 88xx) and the desired external phone number.
   Or, Press the **[SPEED]** button and dial the desired bin number.
5) Replace the handset, return to idle.

To deactivate Call forward

1) Press flashing **[FWD]** button, Call Forward will deactivate and the **[FWD]** button LED is off.

Or,For an iPECS IP Phone,

1) Lift the handset or press the **[SPEAKER]** button to receive dial-tone.
2) Press the **[FWD]** Soft button. For the LIP8002, the **[DND]** button is used to activate Forward.
3) Press the # dial pad button.

SLT(Single Line Telephone)

To activate Call Forward,(Unconditional, Busy, No Answer, or Busy/No Answer)

1) Lift the handset to receive dial tone.
2) Dial “*554”, Call Forward code.
3) Dial desired Call Forward code (“1”~“4”).
4) Dial the destination station, station group, or dial CO access code (9, 8xx, 88xx) and desired external phone number.
5) Momentarily press the hook-switch and receive confirmation tone.
6) Replace the handset to return to idle.

To deactivate the Call forward

1) Lift the handset and receive stutter dial-tone.
2) Dial “*554”, Call Forward code.
3) Dial “#” to cancel Call Forward.

Conditions

- ✓ To use Call Forward, a station must be permitted to forward calls in the system database.
- ✓ Authorization codes are required for ‘Follow-me’ Call Forward and may be required to activate Network Call Forward ‘Follow-Me’
- ✓ A station is separately allowed Call Forward and Off-Net Call Forward.
- ✓ The iPECS does not check the status of the receiving destination; once forward is activated, the receiving system is responsible for routing the call.
- ✓ Incoming analog CO lines must provide Disconnect Supervision and the Open Loop Detect timer must be set for proper operation.
- ✓ Station calls may automatically forward using Preset Call Forward or may forward to your integrated Voice Mailbox. If the Preset Call Forward timer is shorter than the Call Forward No-Answer timer, calls will not forward over the network.
- ✓ Network Call Forward ‘Follow-Me’ can only be deactivated at the original station.
- ✓ Network Call Forward Follow-Me is only available when employing H.323/H.450 networking. When employing Q sig, Call Forward ‘Follow-me’ is not available over the network.
- ✓ For proper operation of Call Forward Follow-Me, all systems in the network must employ the same type of Authorization codes.
- ✓ Call Forward Follow-me is not available between networked systems.

Programming

Keyset Admin.

SYSTEM

- Networking Basic Attribute (PGM 320)
- Networking CO Line Attribute (PGM 322)
- Network Numbering Plan Table (PGM 324)
- Network Feature Code Table (PGM 325)

Web Admin.

NETWORKING DATA

- Net Basic Attributes
- Net Supplementary Attributes
- Net CO Line Attributes
- Net Numbering Plan

Related Features

- Call Forward
- Authorization Codes (Password)
- DND (Do Not Disturb)
- DND - One Time DND
- Direct Inward System Access (DISA)
- Unsupervised Conference
- Dialing Restrictions
- Station Groups
- Station Speed Dial
- System Speed Dial
- Intercom Signaling Mode
- Call Forward, Preset

Hardware

2.83.2.9 CO Transit-In

Description

When a DID call is received by a networked system, the system will attempt normal DID call routing. The ICLID table is examined, followed by the MSN table, then the defined DID number conversion is completed.

The system does a look-up in the Flexible DID Table to identify the destination. If the number is a station in a networked system, the system routes the call to the destination over the appropriate network connection with CLI provided to the called station.

Operation

Operation of Distributed Networking is automatic when configured.

Conditions

- ✓ CO Transit-In calls are not subject to recall or time restrictions at the originating system.
- ✓ If no path is available over the network, the caller will receive busy tone.
- ✓ The system, which originally received the call, will generate two SMDR records for the call, the original incoming call and the transit-call to the destination station.

Programming

Keyset Admin.

SYSTEM

- Networking Basic Attribute (PGM 320)
- Networking CO Line Attribute (PGM 322)
- Network Numbering Plan Table (PGM 324)
- Network Feature Code Table (PGM 325)

Web Admin.

NETWORKING DATA

- Net Basic Attributes➤CNIP Enable, CONP Enable
- Net Supplementary Attributes
- Net CO Line Attributes
- Net Numbering Plan

Related Features

- Direct Inward Dialing (DID)
- ICLID Call Routing
- Multiple Subscriber Number (MSN)

Hardware

2.83.2.10 CO Transit-Out

Description

CO Transit-Out increases efficiency of Lines and reduces call costs by routing outgoing calls to the nearest appropriate point in a network of systems. At one system you can dial the appropriate Line access code for Lines in the remote system. Codes are assigned in the Network Numbering Plan at both systems and digit repeating passes the dialed digits to the remote system.

When you dial the code, the local system compares the digits with appropriate local Numbering Plan codes then to the Network Numbering Plan codes to determine routing. When the number is found in the Network Numbering Plan table, the call is routed to the designated networked system. Any digits received are sent to the remote system for proper routing.

Operation

Operation of Distributed Networking is automatic when configured.

Conditions

- ✓ To use CO Transit-Out, the networked system user must dial the CO transit-out code; the CO Transit-Out service is not activated by pressing a CO Line button.
- ✓ The Station COS of the networked system is applied for toll restriction.
- ✓ For CO Transit-Out, any numbering sequence code can be assigned in (Networking Data ➤ Net Numbering Plan ➤ Numbering Plan Code). If a conflict exists between the System and Network Numbering Plans, the System Numbering Plan has priority.
- ✓ The terminating PSTN lines must be assigned as Networked PSTN CO lines in the terminating system.
- ✓ For proper operation of CO Transit-Out, the Network CO Line type and System Usage must be assigned as PSTN.
- ✓ While not recommended, a call may be routed to the terminating system through an ancillary system when a direct network connection is not deployed between the originating and terminating systems.
- ✓ When enabled in the Net Numbering Plan, you must enter a valid Station or System Authorization code for access. The associated COS is implemented for the call.
- ✓ The system, which terminates the call, will generate two SMDR records for the call, the original incoming call from the station number and the transit-out call.
- ✓ The station receives busy tone when a networking path is not available during transit-out.
- ✓ For Transit-out, the Attendant of the calling system must be permitted access to the CO Lines.

Programming

Keyset Admin.

SYSTEM

- Networking Basic Attribute (PGM 320)
- Networking CO Line Attribute (PGM 322)
- Network Numbering Plan Table (PGM 324)
- Network Feature Code Table (PGM 325)

Web Admin.

NETWORKING DATA

- Net Basic Attributes
- Net CO Line Attributes
- Net Numbering Plan

Hardware

2.83.2.11 BLF Presentation

Description

When systems are networked, the status of stations in one system can be displayed at stations of other systems in the network. The LED of a {Net DSS} button provides the BLF (idle or busy) indication for the assigned station.

When only two systems are networked, BLF status is provided directly between the two systems with each serving as the others BLF Manager for the other. When more than two systems are in a network, the Ericsson-LG Enterprise BLF Manager application provides the BLF status across the network.

BLF Manager is a Windows® application that communicates with systems on the network as the BLF Gatekeeper. Networked systems send regular updates of station status to the BLF Manager. Periodically, the BLF Manager sends a UDP multi-cast message to all systems on the network.

The BLF Manager PC can reside in any location that has IP access to all systems in the network to receive and send both UDP and TCP packets.

Operation

Operation of the BLF Manager is automatic when configured.

Conditions

- ✓ The BLF Manager Gatekeeper is installed as a central function of all systems in a network and must be able to communicate using UDP and TCP over an IP network with the other networked systems.
- ✓ The system can provide Net DSS BLF status for all stations.
- ✓ The BLF Manager provides BLF for stations only, status for Lines is not provided over the network; CO BLF is not supported.
- ✓ The LED of a **{Net DSS}** button indicates BLF status of busy (LED On), idle (LED Off) or DND (LED flashing). Ring signals at a station, including recalls, are not registered across the network and will display as idle.
- ✓ Each system updates the station status to the BLF Manager at intervals of the BLF status duration.
- ✓ If the BLF Manager is not available on the network, BLF services can be activated between only two Systems. In this case, employ the Network Numbering Plan to configure IP address of the networked KSU LAN port as the Destination MPB IP address. Do not assign an address for the BLF Manager in the Networking Supplementary Attributes.

Programming

Keyset Admin.

SYSTEM

- TCP/UDP Port Assign for BLF (PGM 321-Button 2/3)
- BLF Manager IP Address (PGM 321-Button 4)
- Duration of BLF Status (PGM 321-Button 5)
- DEST SYSTEM IP Address (PGM 324-Button 6)
- DEST SYSTEM Port number (PGM 324-Button 7)

Web Admin.

NETWORKING DATA

- Net Basic Attributes➤CNIP Enable, CONP Enable
- Net Supplementary Attributes➤TCP/UDP Port for BLF,BLF Manager IP Address, Duration of BLF State
- Net CO Line Attributes
- Net Numbering Plan➤Dest System IP Address, Dest System Port Number

Hardware

- Windows PC running the BLF Manager application

2.83.2.12 Centralized Attendant Service(CAS)

Description

Calls can be routed over the network to the Attendants of another system to provide Centralized Attendant Service (CAS). A local System Attendant can route local Attendant calls to the Central Attendants by activating Attendant DND. The Centralized Attendants receive the call under the {Net DSS} button assigned for the local System Attendant.In addition,outside calls to networked systems can directly ring the Centralized Attendants.

Operation

Local Attendant iPECS IP or LDP Phone

*To assign **[ATD DND]** button*

- **[TRANS/PGM]** + {Flex Button} + **[TRANS/PGM]** + “83” + **[HOLD/SAVE]**

To route local Attendant calls to the Central Attendants

1) Press the **{ATD DND}** button.

Centralized Attendant iPECS IP or LDP Phone

To answer a remote Attendant call

1) Answer the call as normal, the **{NET DSS}** button for the remote Attendant flashes.

Conditions

- ✓ DID/DISA calls which route only to the Attendant destination and calls to the Attendant from a Station (dial “0”) are routed to the Centralized Attendant assigned as the Main attendant for the networked system.
- ✓ The Centralized Attendant should be assigned as a Main Attendant in a networked system and be assigned **{NET DSS}** buttons for the Attendants in the networked systems.

Programming

Keyset Admin.

STATION

- Attendant DND Button (PGM 115)

SYSTEM

- Attendant Assign (PGM 164)
- DID/DISA Destination (PGM 167)
- CO Ring Assignment (PGM 144)

Web Admin.

STATION DATA

- Flexible Buttons➤ Attendant DND Button

CO LINE DATA

- CO/IP Ring Assignment

SYSTEM DATA

- Attendant Assignment

- DID/DISA Destination

Hardware

- iPECS IP or LDP Phone

2.83.2.13 Centralized Voice Mail

Description

An external Voice Mail may be attached to a system in a network to provide centralized Voice Mail services to the stations of all systems in the network. Calls are passed to this central VM over the network to the Network VM Hunt Group assigned of the central iPECS system. The Station Group number must be assigned in the Network Numbering Plan code of each networked system to deliver calls to the central VM properly.

Operation

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

Conditions

- ✓ The centralized VM must be configured in the central system to which it is attached as a Network VM. The Voice Mail Dialing table and other characteristics of the Network VM are established in the central system.
- ✓ The destination must be defined as part of the network Numbering Plan configuration for each system that will use the centralized VM except for the central system.

Programming

Keyset Admin.

SYSTEM

- Network Numbering Plan Table (PGM 324-Button 6)
- Station Groups (PGM 190) – Ext AA/VM Group

Web Admin.

NETWORKING DATA

- Net Numbering Plan

STATION GROUP DATA

- Station Group Assignment
- Station Group Attributes

Hardware

2.83.2.14 Networked System Paging

Description

You may have access to the Page Zones of other systems in the network. A warning tone is provided, if assigned in the networked system, and then you are connected to the PageZone. The conditions such as Page Time-out of the networked system will apply to your page.

You can access and make page announcements to stations and external zones of a networked system by dialing apre-programmed number or with a Flex button just as if the page was within the samesystem.

Operation

iPECS IP or LDP Phone

To assign a Flex button to access a Page Zone in Networked system

- **[TRANS/PGM]** + **{Flex Button}** + Net Numbering Plan code for Page + **[HOLD/SAVE]**

To Page a Networked system Zone

1) Lift the handset and Dial the Network Paging code and Zone, or press a **{NET PAGE ZONE}** button.
2) If assigned, after the Page Warning Tone, make announcement.
3) Replace the handset to return to idle.

To queue for a page when busy is received

1) Press the[MESSAGE/CALL BACK] button.
2) Replace the handset, returning to idle.

SLT(Single Line Telephone)

To make a page

1) Lift the handset.
2) Dial the Network Paging code and Zone.
3) If assigned, after the Page Warning Tone, make announcement.
4) Replace the handset, to return to idle.

To queue for a page when busy tone is received

1) Dial “556”, the Call Back code.
2) Replace the handset returning to idle.

Conditions

- ✓ The conditions such as Page Access, Page Time-out, etc. apply to Network Paging.
- ✓ A Network Page code must be programmed in Networking Data ➤Net Numbering Plan in the calling system.
- ✓ A Network Page code and page destination must be programmed in Networking Data ➤ Net Feature Code Table in the called system.
- ✓ The Network Page destination can be Internal Page, External Page and All Call Page.
- ✓ You will hear error tone if there is no idle networking path or if the page destination is invalid.

✓ You can page within a networked system without lifting the handset.

Programming

Keyset Admin.

NETWORKING DATA

- Net Basic Attributes
- Net CO Line Attributes
- Net Numbering Plan
- Net Feature Code Table

Web Admin.

NETWORKING DATA

- Net Basic Attributes
- Net CO Line Attributes
- Net Numbering Plan
- Net Feature Code Table

Related Features

■ Internal/External & All Call Page

Hardware

2.84 System Scenario Call Routing

Description

In addition to other routing, the system can be configured to route incoming calls based on scenarios. Scenarios can activate automatically or a group of scenarios can be activated manually by the Attendant.

Scenarios include characteristics such as date and time, Line, Called and calling number, Tenancy group, etc. Scenario routing has priority over other routing destinations.

For example, if Line 1 is assigned to ring at station 100 but the Caller ID and other characteristics match a scenario that routes to a System announcement, the call routes to the announcement.

Scenario characteristics include:

- Caller ID
- Called Number
- Days and time (Start Day and End Day, weeks, start time and end time)
- Destination (Type and Value)
- Scenario Priority
- Scenario State
- Scenario Voice Mail box
- Scenario COS (Class of Service)
- Scenario DISA Active or Not
- Scenario Tenancy Group number
- Scenario Zone
- Scenario Start CO and End CO
- Scenario Group: this is used for scenario group by attendant
- Zone Holiday

Operation

Attendant

To activate a Scenario Group

1) Press **[DND]** button in system attendant.
2) Dial or select item 5.
3) Dial the Scenario Group number.
4) Press **[HOLD/SAVE]** button.

Programming

Keyset Admin.

SYSTEM

- System Call Routing (PGM251-Web only)
- Station ICM Group (PGM 125)
- CO tenancy group (PGM 141-B10)

Web Admin.

SYSTEM DATA

- System Call Routing

STATION DATA

- Station ICM Group

CO LINE DATA

- CO Tenancy Group

Related Features

- Day/Night/Timed/Scenario Ring Mode

Hardware

2.85 Traffic Analysis

Description

The iPECS monitors, stores and periodically or upon request outputs various traffic statisticscovering 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, i.e. blocking level too high, and/or
- Recommend system upgrades.

Attendants enable Periodic Reporting. Once enabled, the system continues to monitor andoutput the requested report until Periodic Report is disabled. On-demand reports selected bythe System Attendant are output only upon request. The Traffic Report is sent to the definedsystem RS-232 or TCP port.

System resources covered by Traffic Reports are:

- Attendant Traffic Report
- Call Summary Report
- Hourly Call Report
- H/W Unit Usage Summary Report
- CO Summary Report
- Hourly CO Report

Summary Traffic Reports cover one of five Analysis periods selected at time of print:

1) Today's peak activity hour (within 24 hours)
2) Yesterday's peak activity hour (24 hours prior to Today's activity)
3) Last hour activity
4) Today's total activity
5) Yesterday's total activity

Operation

System Attendant

To print the All Periodic Traffic Report

1) Press the **[TRANS/PGM]** button.
2) Dial "0122", the All Periodic report code.
3) Dial 4 digits (HH – HH).
4) Select Analysis Period ('1'-'5').
5) Press the **[HOLD/SAVE]** button.

To cancel the periodic All Summary Report

1) Press the **[TRANS/PGM]** button.
2) Dial "0123", the cancel All Summary report code.

3) Press the **[HOLD/SAVE]** button.

To print a traffic report

1) Press the **[TRANS/PGM]** button.
2) Dial the report code, “0121 & 0124” – “0129”.
3) 0121 All Summary Traffic Reports
4) 0124 Attendant Traffic Report
5) 0125 Call Summary Report
6) 0126 Hourly Call Report
7) 0127 Hardware Usage Summary Report
8) 0128 CO Summary Report
9) 0129 Hourly CO Report
10) Press the **[HOLD/SAVE]** button.

Conditions

✓ The Print All Summary Traffic Reports generates the Attendant, Call Summary and CO Summary Traffic Reports.

Programming

Keyset Admin.

SYSTEM

- RS-232 Port Settings (PGM 174)
- Print Port Selections (PGM 175)

Web Admin.

SYSTEM DATA

- RS-232 Port Settings
- Print Port Selections

Related Features

- SMDR Call Records

Hardware

- Device to capture reports

2.85.1 Traffic Analysis, Attendant

Description

The Attendant Traffic Report covers operational statistics for the Attendants. The report outputsperiodically or the Attendant requests output of the report for a defined Analysis period. Thefollowing is a sample report and description of the report fields.

```
Site Name : abc co
Report Type : Attendant Traffic Report- Today Peak
Date : 06/05/02 08:34
============================================================================
ATD   Meas   ---------------------Calls---------------------   ------------------Time------------   Speed  ATD
No    Hour   Total  Ans   Abdn  H     Abd   Held   Avail  Talk   Held   Noans  Ans    Type
100   13:00  104    82    22    3     0     18     10:12  14:21  01:23  00:52  00:23  Sys
101   13:00  90     72    15    1     0     11     12:43  30:12  00:54  00:23  00:21  Main
```

Field	Description
ATD No	Attendant Station Number
Meas Hour	(Measurement Hour) Hour data accumulation began
Calls Total	Total number of calls, except group & recalls, routed to the Attendant
Calls Ans	(Calls Answered) Calls answered during the Analysis period
Calls Abdn	(Calls Abandoned) Calls abandoned before answer by the Attendant, does not include calls abandoned while on hold.
Call H-Abdn	(Calls Abandoned from Hold) Calls abandoned while on hold
Calls Held	Number of calls placed on hold by the Attendant
Time Avail	(Time Available) Time attendant was available to handle new calls
Time Talk	Total time the Attendant was active on calls
Time Held	Time Attendant had calls on hold
Time NoAns	(Time No Answer) Average time calls were ringing or in queue for attendant before abandoned
Speed Ans	(Speed of Answer) Average time calls rang before answer by Attendant
ATD type	(Attendant Type) System or Main

Operation

System Attendant

To print the Attendant Traffic Report

1) Press the **[TRANS/PGM]** button.
2) Dial “0124”, the Attendant Traffic report code.
3) Select Analysis Period (‘1’-‘5’).
4) Press the **[HOLD/SAVE]** button.

Conditions

✓ The Peak Hour is the hour when the system has the highest total call volume.

Programming

Keyset Admin.

SYSTEM

- RS-232 Port Settings (PGM 174)
- Print Port Selections (PGM 175)

Web Admin.

SYSTEM DATA

- RS-232 Port Settings
- Print Port Selections

Related Features

- SMDR (Station Message Detail Recording)
- ACD Statistics Report

Hardware

- device to capture reports

2.85.2 Traffic Analysis, Call Reports

Description

Call activity statistics are provided in the Hourly Call and Call Summary Reports.

The Hourly Call Report covers hourly-completed call activity for the selected Analysis period.The report indicates the number of completed calls for each hour during the Analysis period asshown below.

Site Name : abc co
Report Type : Attendant Traffic Report- Today Peak
Date : 06/05/02 08:35
===

Anal Hour	# Calls Completed
15:00	0
…..	0
…..	
05:00	211
04:00	543
Totals Calls	754

Operation

System Attendant

To print the Hourly Call Report

1) Press the **[TRANS/PGM]** button.
2) Dial “0126”, the Hourly Call report code.
3) Press the **[HOLD/SAVE]** button.

Programming

Keyset Admin.

SYSTEM

- RS-232 Port Settings (PGM 174)
- Print Port Selections (PGM 175)

Web Admin.

SYSTEM DATA

- RS-232 Port Settings
- Print Port Selections

Related Features

- SMDR (Station Message Detail Recording)
- ACD Statistics Report

Hardware

- Device to capture reports

2.85.3 Traffic Analysis, H/W Usage

Description

The Hardware Usage report provides statistics for the system's special Hardware resourcessuch as the Voice Mail as shown in the following sample report.

Site Name : abc co
Report Type : Hardware Usage Report- Today Peak
Date : 06/05/02 08:34
==

Unit Type	Num Unit	Anal Hour	Total Req	Total Denied
VSF	4	09:00	19	0

Operation

System Attendant

To print the Hardware Usage Summary Report

1) Press the **[TRANS/PGM]** button.
2) Dial "0127", the H/W Usage Summary report code.
3) Select Analysis Period ('1'-'5').
4) Press the **[HOLD/SAVE]** button.

Programming

Keyset Admin.

SYSTEM

- RS-232 Port Settings (PGM 174)
- Print Port Selections (PGM 175)

Web Admin.

SYSTEM DATA

- RS-232 Port Settings
- Print Port Selections

Related Features

- SMDR (Station Message Detail Recording)
- ACD Statistics Report

Hardware

- Device to capture reports

2.85.4 Traffic Analysis, Line Reports

Description

The CO Traffic Summary and Hourly reports provide statistics on a summary or hourly basisfor allLine Group activity.

The following provides a sample report and description of the majorfields in the report.

Site Name : abc co
Report Type : CO Group Summary Report- Today Peak
Date : 09/08/02 08:34
==
Peak Hour for All CO 11:00

Grp No	Num COs	Anal Hour	Total Usage	Total Seize	Inc Seize	Out Seize	Grp Ovl	% ACB	% FAO
1	16	10:00	9	6	4	2	0	0	0
20	1	10:00	1	1	0	1	0	0	0

Field	Description
Grp No.	Line Group number
Anal Hour	(Analysis hour) hour during the analysis period with peak usage.
Total Usage	Total number of call attempts on Lines in the Group
Total Seize	Total number of times Lines in the group were used for any call
Inc Seize	(Incoming Seizures) Total number of incoming calls answered for Lines in the group.
Out Seize	(Outgoing Seizures) Total number of outgoing calls attempted on Lines in the group.
ACB	(All COs Busy) Percentage of the time that all Lines in the group were simultaneously busy.
FAO	(Failed Attempts Outgoing) Percentage of outgoing calls offered to the Lines in the group that were denied due to All Trunks Busy condition.

Operation

System Attendant

To print the CO Traffic Summary Report

1) Press the **[TRANS/PGM]** button.
2) Dial ‘0128’, the CO Traffic Summary report code.
3) Select Analysis Period (‘1’-‘5’).
4) Press the **[HOLD/SAVE]** button.

To print the CO Hourly Traffic Report

1) Press the **[TRANS/PGM]** button.
2) Dial ‘0129’, the CO Hourly Traffic report code.
3) Dial CO Group number.
4) Press the **[HOLD/SAVE]** button.

Programming

Keyset Admin.

SYSTEM

- RS-232 Port Settings (PGM 174)
- Print Port Selections (PGM 175)

Web Admin.

SYSTEM DATA

- RS-232 Port Settings
- Print Port Selections

Related Features

- SMDR (Station Message Detail Recording)
- ACD Statistics Report

Hardware

- Device to capture reports

2.86 Universal Night Answer (UNA)

Description

UNA (Universal Night Answer) allows you to be alerted over an external loud bell and answer defined outside calls by dialing the UNA code.

While primarily intended for alternate answering during the Night service mode, UNA will also function in all service modes.

Calls will appear on an appropriate **{LINE}**, a **{POOL}** or **{LOOP}** button. An External Control Contact can be assigned to activate an external Loud Bell to alert users of incoming calls.

The iPECS IP and LDP Phones may be assigned a Flex button as a **{UNA}** button.

Operation

iPECS IP & LDP Phones

*To assign a Flex button as a **{UNA}** button*

- **[TRANS/PGM]** + **{FLEX}** + "*567" + **[HOLD/SAVE]**

To access an incoming UNA call

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial "*567", Universal Night Answer code, the oldest UNA call is connected.

Or,

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the **{UNA}** button; the oldest UNA call is connected.

SLT(Single Line Telephone)

To access an incoming UNA call

1) Lift the handset.
2) Dial "*567", Universal Night Answer code, the oldest UNA call is connected.

Programming

Keyset Admin.

SYSTEM

- Universal Night Answer (PGM 141-Button 7)

Web Admin.

CO LINE DATA

- Common Attributes ➤Universal Answer

Related Features

- LBC (Loud Bell Control)

Hardware

2.87 Wake-Up Alarm, Enhanced

Description

If permitted in the system database, you can set up to five (5) Wake-up or alerting time. When a time is reached, the system will signal your station with an audible and visual signal. The alerting for each of the 5 Wake-up alarms will repeat based on the type selected when you assign the alarm.

The alarms will repeat as follows:

Repeat Type	Repeating
1	Alarm activates one time at the assigned date and time
2	Alarm repeats daily Monday through Friday
3	Alarm will repeat daily, Monday through Saturday
4	Alarm will repeat daily

A System Announcement can be assigned to play as a Wake-Up announcement when user lifts the handset in response to the alarm signal. Each station can be assigned a different Wake-Up announcement, which is recorded by the Attendant.

Operation

Attendant

To register a Wake-Up alarm

1) Press the **[TRANS/PGM]** button.
2) Dial “044”, the Attendant Set Wake-Up(Station Range) code.
3) Dial "049", the Attendant Set Wake-Up(Call Group Number) code.
4) In case of 044, Dial the station range to be alerted. If a single station is to receive the alarm, enter “*” as the second station number.
5) In case of 049, Dial the call group number(00 - 99).
6) Dial the desired 2-digit hour (24-hour mode), then 2-digit minute for alerting.
7) Enter Repeating type, “1” to “4”.
8) For the one-time alarm (repeating type 1), enter the date for the alarm as 2 digits for year, month and day of the month.
9) Press the **[HOLD/SAVE]** button.

To erase Wake-Up

1) Press the **[TRANS/PGM]** button.
2) Dial “045”, the Attendant Reset Wake-Up(Station Range) code.
3) Dial "040", the Attendant Reset Wake-Up(Call Group Number) code.
4) In case of 045, Dial the desired station range, for a single station, enter an “*” as the second station number.
5) In case of 040, Dial the call group number(00 - 99).
6) Press the **[HOLD/SAVE]** button.

iPECS IP & LDP Phones

To register wake-up time alarm from a Station, perform the following steps

1) Press the **[TRANS/PGM]** button.
2) Dial “41”, the code for setting the Wake-up Alarm.
3) Enter the Wake-up alarm index number, “1 to 5”.
4) Dial the desired 2-digit hour (24-hour mode), then 2-digit minute for alerting.
5) Enter Repeating type, “1” to “4”.
6) For the one-time alarm (repeating type 1), enter the date for the alarm as 2 digits for year, month and day of the month.
7) Press the **[HOLD/SAVE]** button.

To erase a wake-up time from a Station, perform the following steps

1) Press the **[TRANS/PGM]** button.
2) Dial “42”.
3) Enter Wake-Up alarm index number, “1” to “5”.
4) Press the **[HOLD/SAVE]** button.

To stop the alarm notification

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

SLT(Single Line Telephone)

To register Wake-Up

1) Lift the handset.
2) Dial “*561”, the SLT(Single Line Telephone) Programming code, confirmation tone is heard.
3) Dial “41”, the set Wake-up code.
4) Dial the 2-digit hour and 2-digit minute for alerting.
5) Press Hook-flash to save.

To stop the alarm notification

1) Lift the handset.

To erase Wake-Up

1) Lift the handset.
2) Dial “*561”, the SLT(Single Line Telephone) Programming code, confirmation tone is heard.
3) Dial “42”, the erase Wake-up code.
4) Press Hook-flash.

Conditions

- ✓ When registering or erasing an Alarm, iPECS IP and LDP phones can use the**[VOL UP]**/**[VOL DOWN]** buttons to scroll between the Wake-up alarms and “#” can be used to scroll through the repeating types.
- ✓ The iPECS IP and LDP phones will display an asterisk near the time to indicate a Wake-up alarm is set for the current day.

- ✓ If you have all five Wake-up Alarms set, the Attendant cannot set a Wake-up alarm for you. When the Attendant sets an alarm you it is assigned to the lowest Wake-up Alarm index number (1-5) available for the station. If no Alarm index is available, the Attendant setting is ignored.
- ✓ When the Attendant erases the Wake-up alarm, all five alarm indices are erased.
- ✓ Changing the Wake-up alarms from standard to enhanced, will cancel and erase all active Wake-up alarms.
- ✓ For the one time alarm, the year can be set from 2000 to 2031.
- ✓ The Call Group Number means Hotel Data – Call Group of Room.

Programming

Keyset Admin.

STATION

- Wake Up Announcement (PGM 114-Button 24-6)

SYSTEM

- Enhanced Wake-Up Usage (PGM 161-Button 24-23)

Web Admin.

STATION DATA

- VM Attributes ➤Wake-Up Announcement

SYSTEM DATA

- New 5 Wake Up Usage: ON

Related Features

- ■ Wake-up Alarm, Standard

2.88 Wake-up Alarm, Standard

Description

The standard Wake-up Alarm feature allows you or an Attendant to set a Wake-up alarm or desired time to be alerted.

When the time is reached, the system will signal your station with an audible and, for iPECS IP and LDP Phones, a visual signal is provided in the LCD.

A System Announcement can be assigned to play as a Wake-Up announcement when you lift the handset in response to the alarm signal.

Each station can be assigned a different Wake-Up announcement, which is recorded by the Attendant.

Operation

Attendant

To register a Wake-Up Alarm

1) Press the **[TRANS/PGM]** button and Dial “044”, the Attendant Set Wake Up code.
2) Dial “044”, the Attendant Set Wake-Up(Station Range) code.
3) Dial "049", the Attendant Set Wake-Up(Call Group Number) code.
4) In case of 044, Dial the station range, for a single station, enter an “*” in place of the second station number.
5) In case of 049, Dial the call group number(00 - 99).
6) Dial the 2-digit hour and 2-digit minute for alerting.
7) For a daily (repeating alarm), dial “#”.
8) Press the **[HOLD/SAVE]** button.

To erase Wake-Up

1) Press the **[TRANS/PGM]** button and Dial “045”, the Attendant Erase Wake-up code.
2) Dial “045”, the Attendant Reset Wake-Up(Station Range) code.
3) Dial "040", the Attendant Reset Wake-Up(Call Group Number) code.
4) In case of 045, Dial the desired station range, for a single station, enter an “*” in place of the second station number.
5) In case of 040, Dial the call group number(00 - 99).
6) Press the **[HOLD/SAVE]** button.

iPECS IP & LDP Phones

To register Wake-Up

1) Press the **[TRANS/PGM]** button and Dial “41”, the Set Wake-up code.
2) Dial the 2-digit hour and 2-digit minute for alerting.
3) For a daily (repeating alarm), dial “#” and Press **[HOLD/SAVE]** button.

To stop the alarm notification

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

To erase Wake-Up

1) Press the **[TRANS/PGM]** button and Dial “42”, the Erase Wake-up code.
2) Press the **[HOLD/SAVE]** button.

SLT(Single Line Telephone)

To register Wake-Up

1) Lift the handset and Dial “*561”, the SLT(Single Line Telephone) Programming code, confirmation tone is heard.
2) Dial “41”, the set Wake-up code.
3) Dial the 2-digit hour and 2-digit minute for alerting.
4) For a daily (repeating alarm), dial “#”.
5) Hook flash, conformation tone is provided.

To stop the alarm notification

1) Lift the handset.

To erase Wake-Up

1) Lift the handset and Dial “*561”, the SLT(Single Line Telephone) Programming code, confirmation tone is heard.
2) Dial “42”, the erase Wake-up code.
3) Hook flash, a conformation tone is provided.

Conditions

- ✓ In cases where a remote device is installed, the time display in the remote terminal is adjusted by the Device Zone assignment. Thus, the Wake-up alarm time is based on the time displayed on the phone and not the system time.
- ✓ When receiving a Wake up signal, lifting the handset will return MOH.
- ✓ The Wake-Up alarm signal is the station ring signal with a 30 seconds On and 90 seconds Off cycle 3 times. If no action is taken by you to terminate the alarm, the Attendant is notified with a display of the station number that did not respond.
- ✓ Time must be entered in the Military 24-hour format (hh:mm).
- ✓ The daily alarm will reset and repeat each day until erased (cancelled). The One-time alarm will reset and cancel automatically.
- ✓ A System announcement can be recorded and assigned to play when the user responds to the wake up alarm.
- ✓ The Call Group Number means Hotel Data – Call Group of Room.

Programming

Keyset Admin.

STATION

- Wake Up Announcement (PGM 114-Button 24-6)

Web Admin.

STATION DATA

- VM Attributes ➤Wake-Up Announcement

SYSTEM DATA

- New 5 Wake Up Usage: OFF

Related Features

- Enhanced Wake-up Alarm

Hardware

2.89 Web Call Back

Description

The Station User Web portal incorporates a Web Call Back feature. In the portal, you may be permitted to request the system to establish a telephone call between two telephone numbers. The call is subject to the dialing restrictions of the station number associated with the portal access. An SMDR record is generated with the station number that accessed the portal identified.

Operation

To activate web call back

1) Access your Station Web portal and select the Call Back menu (Set Enable in PGM 113 or Common Attributes -> Routing attributes -> Web Call Back Service).
2) Enter the “to” and “from” telephone numbers.
3) Press Call button. The page will update to indicate success or failure of the call.

Conditions

- ✓ Web Call Back must be enabled in your Station’s attribute assignments.
- ✓ You will receive an error message if any of the following occur:
  - Call Back is disabled
  - The outgoing call is restricted due to your COS
  - No outside Line is available or

Programming

Keyset Admin.

STATION DATA

- Web Call Back Service (PGM 113-Button 24-11)

Web Admin.

STATION DATA

- Common Attributes Web Client Service

Hardware

2.90 Zone Management

2.90.1 Holiday Assignment

Description

The Administrator can assign Holiday Zones (up to 40) and Vacation Zones (up to 5 for eachZone). Devices in a Zone follow the holiday and vacation assignments of the zone. When thedate of a Zone is set to Holiday or Vacation, the Zone will operate in the programmed ring mode.

Operation

Operation of Zone Holiday Assignment is automatic when programmed.

Conditions

- ✓ When the Tenancy group Ring mode for Flexible DID Conversion (PGM231) is programmed, the priority is higher than the Zone Holiday Assignment.
- ✓ Zone Holiday Assignment has a higher priority than the Attendant ring mode.
- ✓ It is recommended to use the same zone for each outside Line and Station of a Tenancy group.

Programming

Keyset Admin.

SYSTEM

- Zone Holiday Assignment (PGM 444)

Web Admin.

ZONE DATA

- Zone Holiday Assignment

CO LINE DATA

- CO/IP Ring Assignment

Hardware

2.90.2 Remote Devices

Description

Remote devices, in particular those not reachable by the system, are managed by groupingdevices with similar characteristics in a Zone.

Placing devices into Zones simplifiesmanagement allowing definition of common characteristics to the devices within the zone.

Zone attributes include:

- Nation code
- Language
- Page area
- Time-zone, GMT or system
- Holiday ring modes
- RTP relay
- RTP Relay group.

Operation

Zone operation is automatic once configured.

Conditions

✓ Zone attributes which may affect ringing, such as Holiday and Vacation settings, have lower priority than other ring assignments such as Tenancy group ring mode established for DID lines.

✓ Zone Attributes do not provide adjustment of time for DST.

✓ It is recommended to assign outside Lines and stations of a Tenant group in the same Device Zone.

✓ Wake-up time is based on the time displayed in the station's LCD.

Programming

Keyset Admin.

ZONE DATA

- Device Zone Number (PGM436-Web only)
- Device Zone Attributes (PGM 437-Web Only)
- Zone Access & Page Relay (PGM 438-Web only)
- Zone Attributes (PGM 439-Web only)
- Zone RTP Relay Group (PGM 440-Web only)
- Inter-zone Attributes (PGM 441-Web only)
- Zone Holiday Assignment (PGM 444)

Web Admin.

ZONE DATA

- Device Zone Number
- Device Zone Attributes
- Access & Page Relay

- Zone Attributes
- Zone RTP Relay Group
- Inter-Zone Attributes
- Zone Holiday Assignment

Hardware

2.91 Flexible Page Feature

2.91.1 Flexible Page Codes

Description

For What ?

1) Extended Page Codes
   - Number of Codes are 50
   - Access Code : prefix(numbering plan) & suffix(01 – 50)
2) Flexible Page Zone
   - Page Zone assignment (Internal / External / Internal & External / System Overall)
3) Various Source Type
   - Live Voice, Announcement Play, Silent Text
4) Emergency Attributes Equipped
   - Urgency Level to give override privilege between Pages
   - Override privilege : Override Page, Override All Calls
5) Authorization Option
   - Authorization (OFF/ON) before paging

Emergency Usage Oriented

1) Enabled Flexible Page Feature in Phone in Locked state

Operation

Flexible Page Feature (How To Access)

*Prefix + Suffix Code (ex. *7401, *7402, …, *7450): Dial Prefix and Suffix Code*

✓ Prefix Code :
- Numbering Plan(P.106-109) > Flexible Page Prefix (ex. *74)
- Available Pages are listed from "System Data > Flexible Page Feature"

✓ Suffix Code :
- Order# of Pages in "System Data > Flexible Page Feature"
- Two Digits : 01, 02, 03, …, 50

Prefix Dialing & View & Select One

1) Dial Prefix Code
2) Pages are listed from "System Data > Flexible Page Feature" list
3) Dial the Suffix or Select& [SAVE]
4) Page is initiated

*Page Stop/Cancel Code (ex. *7400, *74**)*

✓ To Stop/Cancel on-going Announce or Silent Text Page
- Stop/Cancel Page initiated by Me only :
  Flexible Page Prefix & '00' (ex. *7400)

- Stop/Cancel Page initiated by Anybody :
  Flexible Page Prefix & '**' (ex. *74**)

✓ Page Stop/Cancel Privilege
- Station Data > Terminal Attributes(112) > Allow to Stop - Flexible Paging :
  "Not Allow to Stop"
  "Allow to Stop Initiated by Me Only"
  "Allow to Stop Initiated by Any One"

One-Touch Flex Button for Flexible Page Feature

Web Admin

✓ Station Data > Flexible Buttons(P.115)
  Type : "Programming (Numbering Plan)"
  Input :
- To view page list, at first: Flexible Page Prefix Only (ex. *74)
- To initiate page immediately: Flexible Page Prefix & Suffix (ex. *7401)
- To Stop/Cancel on-going Paging: Flexible Page Prefix & '00' or '**' (ex. *7400, *74**)

Phone PGM

1) [PGM] + [Flex Button]
2) Dial Digits
   - To view page list, at first : Flexible Page Prefix Only (ex. *74)
   - To initiate page immediately : Flexible Page Prefix & Suffix (ex. *7401)
   - To Stop/Cancel on-going Paging : Flexible Page Prefix & '00' or '**' (ex. *7400, *74**)
3) [SAVE]

System Data > Flexible Page Feature

Mandatory Attributes

✓ To be available Page and be listed to user when access Prefix code
- Page Zone : should be available
- Action : should be available

System Data > Flexible Page Feature> Order

Used for 2-digits of Suffix Code of Flexible Page Numbering

✓ 01 – 50
  Example)
- Prefix(Flexible Numbering Plan in P.106-109) : *74
- Access Code of Flexible Page Table Order #1 = *7401
- Access Code of Flexible Page Table Order# 2 = *7402
- . . .
- Access Code of Flexible Page Table Order# 50 = *7450

System Data > Flexible Page Feature> Name

Subject (Identity) of the Page

✓ Paging display on paged station : “[name] FROM XXXX”

Example)

- Name = “Help Request”
- LCD on paged station : “ Help Request FROM XXXX “

✓ Description of a Page in page list when access by Prefix code

Example)

- (01) Classroom Emergency
- (02) Help Request
- (03) …
- SEND EXIT

Paging display on paged station : “[name] FROM XXXX”

Flexible Page Feature (Attributes)

Common Attributes

1) Page Zone (Mandatory Attribute)
   - page group #
   - 0 (default/unused)
   - INT(1-n) / INT_ALL / EXT1 / EXT2 / EXT_ALL / INT_EXT_ALL / SYS_ALL
2) Action (Mandatory Attribute)
   - Live Voice / Announce Play / Silent Message
3) Page Urgency Level
   - 0 is highest, 255 is lowest
   - Resolution for conflict of page zone# between on-going pages
   - Making page with higher level urgency overrides on-going page with lower level urgency
     < Exception 1 – Regardless of Urgency Level >
     Silent Message page ongoing is always overridden by new paging made by other user.
     < Exception 2 – Regardless of Urgency Level>
     Same page made by same user is protected.
     Different page made by same user is always overridden.
4) Override By Urgency
   - Emergency Oriented attribute
   - Override Paged Call Only / Override All Calls (disconnect talking calls and force to receive page)
5) Authorization
   - OFF / ON
   - Ask station password before paging

Live Voice Related Attributes

1) Hands free
   - OFF/ON
   - Ask handset Off-hook before paging

Announce Play Related Attributes

1) Annc. No
- System Announcement # recorded
- If not available or not recorded then page is not initiated

2) Page Repeat Count
- How many time to repeat the announce page
- Page … Page … Page …

3) Page Repeat Interval
- If "Page Repeat Count" is available
- Pause interval between every repeat of the announce page
- Page (interval) Page (interval) Page (interval)

4) Announce Repeat Count
- How many time to repeat play of announcement at one repeat of announce page
- Page(Play … Play … Play …) (Interval) Page (Play … Play … Play …) …

Silent Message Related Attributes

5) Silent Text No
- Index # of Silent Text Messages table in "System Data > Silent Text Messages"
- If not available or empty : Paging Only without SMS sending

6) SMS Only
- OFF : Paging and SMS
- ON : SMS Only

To Override Exceptional Cases

Exceptional Cases

1) Page Zone Busy
- Target Page Zone is already occupied by ongoing paging made by other user
  and
- The ongoing page is same or upper level of urgency
  and
- The ongoing page is 'Live Voice 'or 'Announce Play'

2) No-Member To Hear Page
- Internal Page Zone (P.118) : None of station member authorized in the Page Zone
  or
- All of station members are in use state that the page cannot override
  User is on a call and the page with "Override By Urgency = Override Paged Call Only"
  or user is on paged state and the page with same or lower urgency level

Live Voice

1) Retrial on Busy
- Busy Tone
- Display ". . . PAGE XX BUSY"
- User hang up and retry later by user

2) Retrial on No-Member
- Error Tone
- Display “NOT EXISTED PAGED STA”
- User hang up and retry later by user

Announce Play

1) Retrial on Busy
- Automatic retrial by system internally
- Retry in every 6 seconds, repeat retrial 30 times

2) Retrial on No-Member
- Automatic retrial by system internally
- Retry in every 15 seconds, repeat retrial 9 times

Silent Message

1) Retrial on Busy
- Automatic retrial by system internally
- Retry in every 6 seconds, repeat retrial 30 times

2) Retrial on No-Member
- Automatic retrial by system internally
- Retry in every 15 seconds, repeat retrial 9 times

Stop/Cancel Page

Live Voice

1) User hang up the page call

Announce Play

1) Permission per a station
- Station Data > Terminal Attributes(P.112) > Able to Stop - Flexible Paging= Not Allow to Stop
/ Allow to Stop Initiated by Me Only
/ Allow to Stop Initiated by Any One

2) Dial the Stop/Cancel Code or press One-Touch flex button that has the code
- To Stop/Cancel Page initiated by Me
: Flexible Page Prefix Code + 00 (ex. *7400)
- . To Stop/Cancel Page initiated by All (including Me)
: Flexible Page Prefix Code + ** (ex. *74**)

Silent Message

1) Permission per a station
- Station Data > Terminal Attributes(P.112) > Able to Stop - Flexible Paging= Not Allow to Stop
/ Allow to Stop Initiated by Me Only
/ Allow to Stop Initiated by Any One

2) Dial the Stop/Cancel Code or press One-Touch flex button that has the code
- To Stop/Cancel Page initiated by Me
: Flexible Page Prefix Code + 00 (ex. *7400)
- To Stop/Cancel Page initiated by All (including Me)
: Flexible Page Prefix Code + ** (ex. *74**)

3) Being paged station of Silent Message Paging
- state : being paged state
- LCD : "[name] FROM XXX" or "[subject] FROM XXX"
- Got SMS Message : [Message] Flashing ON
  If user press the [Message] button, will go to SMS received list menu directly

4) How long in Being paged state
- Maintain 3 minutes, if there is no interrupt by other paging or user's action

RING LED Flashing in Being Paged Station

'RING LED' is Flashing in Being Paged Station

✓ Flashing Rate for an incoming call

Call Log in Being Paged Station

Call Log option for page

1) Leave Call Log for Paged Station
2) Station Data > CLI Attributes(P.113) > Page Call Log
- None
- All
- Emergency (Including Urgency Level 0) Only
  Emergency Page Zone : SYS_ALL(XX) = 51(eMG80/100), 116(UCP/eMG800/vUCP)
  Or, a page with "Page Urgency Level = 0"

Flexible Page from Mobile Extension

Mobile Extension Table (PGM 236) :

1) PGM Auth = Enable
2) Input CLI Number, Or Telephone Number with "Tel Num As CLI Num = ON"

Supported Flex Page :

1) Live Voice / Announce Play / Silent Message
2) Authorization OFF/ON

Conditions

✓ To be able to make a Page Call from a Station :
- Station Data > Common Attributes(111) > Page Access = ON

✓ To be able to Get a Paging Call to a Station :
- Station Data > Internal Page Zone(118)

Programming

Web Admin.

SYSTEM ID & NUMBERING PLANS

- Flexible Numbering Plan (P.106-109) ➤Flexible Page Prefix

SYSTEM DATA

- Flexible Page Feature

- Silent Text Messages

Related Features

- Authorization Code
- Paging, Emergency Paging
- Page Zone

Hardware

2.91.2 Silent Text Messages

Description

Pre-defined silent text messages to serve by SMS sending when user conduct a Silent Paging. 60 messages in System Data > Silent Text Messages.

Operation

Usage of Subject and Contents

Subject

1) Normally, just dummy(not used)
2) Backup for 'Name' of Page
   - Used instead of 'Name' of the Flexible Page if it is empty
3) Backup for 'Contents'
   - Used instead of 'Contents' of a Silent Text message if it is empty

Contents

1) Used for SMS message

To send Silent Text by SMS to Page Group Member

Flexible Page Feature in System Data > Flexible page Feature

1) Page Zone
2) Action = Silent Message
3) Silent Text No = One of Text Message from "System Data > Silent Text Messages"
4) SMS Only
   - for Paging(Page Message)& SMS sending :
     SMS Only = OFF
     Silent Text is available(Silent Text No. = n for valid Silent Text Message)
   - for Paging(Page Message) Only :
     SMS Only = OFF
     Silent Text is unavailable(Silent Text No. = 0 or empty Silent Text Message)
   - for SMS sending Only :
     SMS Only = ON
     Silent Text is available(Silent Text No. = n for valid Silent Text Message)

To Stop/Cancel Silent Paging Status

Being paged station of Silent Message Paging

1) state : being paged state
2) LCD : "[name] FROM XXX" or "[subject] FROM XXX"

How long Maintain in Being paged state for a Silent Paged ?

1) Maintain being paged status during 3 minutes, if there is no interrupt by other paging or user's action

To Stop/Cancel Paging within 3 minutes

1) Dial "Flexible Page Prefix Code + 00" to Cancel/Stop a page initiated by 'Me' only (ex. *7400)

2) Dial “Flexible Page Prefix Code + **” to Cancel/Stop a page initiated by ‘All’ (ex. *74**)

3) Need Permission to Allow Cancel/Stop by above codes

- Station Data > Terminal Attributes(P.112) > Able to Stop - Flexible Paging
  Not Allow to Stop
  Allow to Stop Initiated by Me Only
  Allow to Stop Initiated by Any One

Conditions

✓ To be an available Message, Subject or Contents should be an available input

✓ To be able to make a Page Call from a Station :
- Station Data > Common Attributes(111) > Page Access = ON

✓ To be able to Get a Paging Call to a Station :
- Station Data > Internal Page Zone(118)

Programming

Web Admin.

SYSTEM ID & NUMBERING PLANS

- Flexible Numbering Plan (P.106-109) ➤Flexible Page Prefix

SYSTEM DATA

- Flexible Page Feature
- Silent Text Messages

Related Features

- Authorization Code
- Paging, Emergency Paging
- Page Zone

Hardware

3 Intercom

3.1 Barge In

Description

Barge in permits authorized extensions to intrude into other existing outside or intercom calls.

When you intrude on an existing call, a Conference is established with you (the supervisor) and the parties on the existing call.

Barge-in from your phone is assigned for one of three operating modes:

- (0) – Barge in is disabled
- (1) – Barge in Monitor mode, the intruding station can listen only
- (2) – Barge in Speech mode, the intruding station hears the conversation and may join and may join the conversation to speak to the other parties.

Operation

iPECS IP & LDP Phones with 3-soft buttons

To Barge-in in the Monitor mode

1) Call the busy station and receive busy tone.
2) Press the **[MONITOR]** soft button. You will need to use the right Navigation button to display the Monitor selection.
3) Warning tone is sent to the connected parties and you are connected to the call in a “listen-only”, monitor mode.

To Barge-in in the Speech mode

1) Call the busy station and receive busy tone.
2) Press the **[MONITOR]**Soft button. You will need to use the right Navigation button to display the Monitor selection.
3) Warning tone is sent to the connected parties and you are connected to the call in a “listen-only”, monitor mode.
4) If Barge-in Speech mode is assigned, press the **[JOIN]**Soft button to join the conversation. The parties receive intrusion tone and you connect to the call.
5) To terminate the call and return the all parties to idle, press the **[DROP]** soft button.

To exit Barge-in

1) Hang-up the phone.

Conditions

- ✓ This feature is only supported for iPECS IP and LDP Phone models with 3 soft buttons.
- ✓ This feature is only available for active calls; you cannot barge in on a ringing or held call.

- ✓ The Barge-In supervisor phone display will indicate the Id of the connected parties prior to the Barge-in operation.
- ✓ If 'Privacy' option is ON, then any person could not barge-in to the station.
- ✓ It's possible to get a call monitored and intruded while the call is being recorded into voicemail box when there is available MCIM channels.

Programming

Keyset Admin.

STATION

- Barge In Mode (PGM 113 Button 24-7)

Web Admin.

STATION DATA

- Common Attributes ➤Barge In Mode

Related Features

- Intrusion

Hardware

- iPECS IP or LDP Phones with 3-soft buttons

3.2 Direct Station Selection/Busy Lamp Field (DSS/BLF)

Description

You can assign Flex buttons on iPECS IP and LDP Phones, or DSS Consolesto provide one-button calling to another station and the button LED indicates the status of the station.

The **{DSS/BLF}** button can also be used for one-button transfer.

The LED will illuminate when the station is busy and flash when receiving a call. In addition, the DSS/BLF button will flash when the associated station calls you.

Operation

iPECS IP & LDP Phones

Operation of this feature is automatic for assigned Flex buttons.

*To assign a Flex button for **{DSS/BLF}***

- **[TRANS/PGM]** + {FLEX} + Station number + **[HOLD/SAVE]**

Conditions

- ✓ A station receiving ICM ringing is considered busy and the associated DSS/BLF LED on all other stations will flash at 30 ipm.
- ✓ A station receiving ICM ringing will receive visual indication by a flashing LED of the Flex button associated with the calling station.
- ✓ When a station receives a Camp-On the LED of a DSS/BLF button associated with the calling station will flash.
- ✓ The station is considered busy when in use, receiving ICM Ring at an iPECS IP or LDP Phone, or receiving any ring at a SLT(Single Line Telephone).

Related Features

- Intercom Call (ICM Call)
- Station User Programming & Codes

Hardware

- iPECS IP or LDP Phone

3.3 Intercom (ICM) Call

Description

A non-blocking ICM is available to all stations only in UCP system.

You may place an intercom callto other stations in the system by dialing applicable digits as defined in the Station NumberingPlan.

Operation

iPECS IP & LDP Phones

To place an intercom call

1) Lift the handset or press the **[SPEAKER]** button to receive ICM dial tone.
2) Dial station number or press the**{DSS/BLF}** button.
3) For ring-back tone, await answer or, for Intercom splash-tone, speak and await answer.

SLT(Single Line Telephone)

To place an intercom call

1) Lift the handset to receive ICM dial tone.
2) Dial the station number.
3) For ring-back tone, await answer or, for Intercom splash-tone, speak and await answer.

Conditions

- ✓ 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 received should a time-out occur.
- ✓ ICM Dial tone is removed after dialing the first digit.
- ✓ If the called station is busy, Intercom Busy tone is provided for the Busy Tone time then, error tone is sent by the system. You may disconnect from the call or activate a feature such as Message Wait/Callback prior to the time-out.
- ✓ With your iPECS IP and LDP Phone, you can place consecutive Intercom calls without the need to regain ICM dial tone (no need to hang-up) between calls; simply presses another **{DSS/BLF}** button.
- ✓ An Intercom call to a station in the HF answerback or Voice Announce mode (H 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

Keyset Admin.

STATION

- Prime Line (PGM 113-Button 7)

SYSTEM

- ICM Dial Tone Timer (PGM 181-Button 6)
- Inter Digit Timer (PGM 181-Button 7)

Web Admin.

STATION DATA

- Common Attributes ➤Prime Line

SYSTEM DATA

- System Timers ➤ICM Dial Tone Timer, Inter Digit Timer

Related Features

- Intercom Signaling Mode
- Speakerphone

Hardware

3.4 Intercom Call Hold

Description

While on an active Intercom (ICM) call, using aniPECS IP or LDP Phone you can place the call on hold. The heldstation will receive the assigned Music-on-Hold.

The call is placed on Exclusive Hold andrecalls you after expiration of the Exclusive Hold Recall Timer.

An **[ICM]** button must be assigned to the iPECS IP or LDP Phone.

Operation

iPECS IP & LDP Phones

*To assign an **[Intercom]** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]**+ “87” + **[HOLD/SAVE]**

To place an active ICM call on hold

1) Press the **[HOLD/SAVE]** button, the **[ICM]** button LED will flash at the exclusive hold rate. ICM dial tone is received.

To retrieve the held ICM call

1) Press the **[ICM]** button or the **{DSS/BLF}** button associated with the held station, the**[ICM]** button LED illuminate and the ICM call connected.

Conditions

✓ Multi-intercom call can be placed on hold at station. Press the [ICM], [HOLD] button or the {DSS/BLF} button associated with the held station, the [ICM] button LED illuminate and the ICM call connected by FIFO.

Related Features

- MOH (Music-On-Hold)
- Intercom Call (ICM Call)
- Exclusive Hold
- Hold Recall

Hardware

- iPECS IP or LDP Phone

3.5 Intercom Caller Controlled Signaling

Description

You can change the signaling mode of an Intercom (ICM) call from Tone ring to Voice announceor Voice announce to Tone ring.

Operation

iPECS IP & LDP Phones

To change the ICM Signaling mode

1) Place your intercom call as normal.
2) Dial “#”, the ICM Signaling mode will change from Voice announce to Tone ring or Tone ring to Voice announce.

SLT(Single Line Telephone)

To change the ICM Signaling mode

1) Place your intercom call as normal.
2) Dial “#”, ICM Signaling mode will change from Voice announce to Tone ring or Tone ring to Voice announce.

Conditions

- ✓ The ICM Signaling mode cannot be changed when an Intercom call is placed to a Linked Pair station. Such intercom calls always employ Tone signaling.
- ✓ If the signaling mode is changed, the call is not subject to Call Forward, No Answer.
- ✓ The signaling mode for a specific Intercom call can only be changed once and cannot be changed back to the original signaling mode.
- ✓ Changing the signaling mode does not affect privacy at the called station.

Programming

Keyset Admin.

STATION

- Caller Controlled ICM Signaling (PGM 111-Button 15)

Web Admin.

STATION DATA

- Common Attributes ➤Forced Hands-Free Answer

Related Features

- Intercom Signaling Mode, Linked Station Pairs

Hardware

- iPECS IP or LDP Phone to receive Voice announced calls

3.6 Intercom Lock-Out

Description

If you take no action after going off-hook for the Dial Tone timer or fail to dial anadditional digit within the Inter-digit timer, you will receive error tone for 30 seconds andbe placed out-of-service (locked-out).

The LED of associated **{DSS/BLF}** buttons as well asthe station's **[ICM]** button, if configured, flutter rapidly to indicate the out-of-service state.

For iPECS IP & LDP Phone users, if the **[SPEAKER]** is used, the station will receive error tone for 30second and then automatically return to idle and will not "Lock-out".

Operation

System

Operation of Intercom Lockout is automatic based on the Dial Tone & Inter-digit timers.

Conditions

- ✓ If the station is assigned Howler Tone, error tone is presented for 30 seconds followed by 30 seconds of Howler tone followed by lockout and silence.
- ✓ To release the lock-out, simply hang-up.

Programming

Keyset Admin.

STATION

- Howler Tone (PGM 111-Button 5)

SYSTEM

- ICM Dial Tone Timer (PGM 181-Button 6)
- Inter Digit Timer (PGM 181-Button 7)

Web Admin.

STATION DATA

- Terminal Attributes ➤Howler Tone

SYSTEM DATA

- System Timers ➤ICM Dial Tone Timer, Inter Digit Timer

Related Features

- Intercom Call (ICM Call)

3.7 Intercom Step Call

Description

When busy tone is received on a dialed Intercom call, you may place a call to anotherstation by dialing the last digit of the station number. The system replaces the last digit of thepreviously dialed busy station with the dialed digit and places an Intercom call to the newstation number.

Operation

iPECS IP & LDP Phones

To activate step call, while receiving busy on a dialed Intercom call

✓ Dial a digit other than the last digit of the busy station's intercom number.

Conditions

✓ If the user dials the last digit of the busy station, Camp-On will be activated.

✓ After receiving busy tone, if the user takes no action for the Busy Tone timer, 7seconds, the system will start the Intercom Lockout procedure.

Related Features

- Intercom Lock-Out
- Intercom Call (ICM Call)

Hardware

- iPECS IP or LDP Phone

3.8 Intercom Tenancy Group

Description

Stations in the iPECS are assigned to an Intercom Tenancy Group. Stations ina group are allowed or denied the ability to place intercom calls to stations in other groups ona group-by-group basis.

Each Intercom Tenancy Group is assigned an Attendant station. All dial "0" calls from a stationin the group are routed to the assigned Attendant. In addition, the assigned Attendant cancontrol the Day/Night service mode for stations in the group switching from Day to Night mode.

Each group is assigned a separate Auto Ring Mode Table (Weekly Time Table) to change the Ring and COS (Class of Service) modeautomatically during the day and night service mode. In addition, DID calls to the system canbe routed to a specified Tenancy group. By defining the group as the destination inthe Flexible DID Conversion table calls will follow the Tenancy Group Auto Ring Mode table (Weekly Time Table).

Operation

System

Operation of Intercom Tenancy Groups is automatic when programmed.

Conditions

- ✓ Intercom calls from a station to a denied access Intercom Tenancy Group will return error tone.
- ✓ Intercom Tenancy 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 Intercom Tenancy groups.
- ✓ The Attendant of a Tenancy Group can be any station in the system and it is not affected by Tenancy Group access.
- ✓ When the Attendant of an Intercom Tenancy Group sets Day/Night/Timed mode, it will affect only the assigned Tenancy Group.
- ✓ Calls to and from outside Lines are not affect by Intercom Tenancy. However; calls cannot be transferred between groups if access is not allowed between the groups.
- ✓ Intercom Tenancy Group 01 is the default or group 0 is the unassigned group.
- ✓ Outside Lines can be configure for use by a specific Tenancy group only, stations in other tenancy groups are not able to access the Line.

Programming

Keyset Admin.

STATION

- ICM Group Number (PGM 111-Button 17)
- ICM Tenancy Group (PGM 125)

CO LINE

- Tenancy Group (PGM141-Button 10)

TABLES

- Flexible DID Conversion Table (PGM 231)

- Weekly Time Table (PGM 233)

Web Admin.

STATION DATA

- Common Attributes ➤ ICM Tenancy Group Number
- Station ICM Tenancy Group

CO LINE DATA

- Common Attributes ➤ CO Tenancy Group

TABLES DATA

- Flexible DID Conversion
- Auto Ring Mode Table

Related Features

- Intercom Call (ICM Call)
- Outside Line Ring Assignment
- Flexible Numbering Plan
- Call Transfer
- Auto Service Mode Control

3.9 Intercom Transfer

Description

You can transfer an active Intercom call to other stations in iPECS system. Intercom callscan be transferred after announcing the call (screened) or without announcing the call(unscreened).

The Intercom station is placed on Exclusive Hold. The Transfer Recall Timer is initiated and, ifthis timer expires before the Intercom call is answered, the call will recall your stationuntil answered or abandoned.

Operation

iPECS IP & LDP Phones

To perform a Screened ICM transfer, while on an ICM call

1) Press the**[TRANS/PGM]** button and Dial the station to receive call.
2) At answer or Splash tone, announce call.
3) Hang-up, return to idle.

Or,

1) Press the**{DSS/BLF}** button for the desired station.
2) At answer or Splash tone, announce call.
3) Hang-up, return to idle.

While on an Intercom call, Unscreened call transfer

1) Press the**[TRANS/PGM]** button.
2) Dial Station to receive call.
3) Hang-up, return to idle.

Or,

1) Press the**{DSS/BLF}** button for the desired station.
2) Hang-up, return to idle.

SLT(Single Line Telephone)

To perform a Screened transfer of an active Intercom call

1) Momentarily depress the Hook-switch.
2) Dial Station to receive call.
3) At answer or Splash tone, announce call.
4) Hang-up, return to idle.

While on an Intercom call, Unscreened call transfer

1) Momentarily depress the Hook-switch.
2) Dial Station to receive call.
3) Hang-up, return to idle.

Conditions

- ✓ The **[ICM]** button, if configured, provides an appearance for the transferred station. The LED indicates status and pressing the button connects to the station.

- ✓ If the receiving station is busy, the transferring station may camp the call on to the busy station, see Camp-On.
- ✓ A station in DND or out-of-service cannot receive a transfer, and such attempts will result in error tone.

Related Features

- Exclusive Hold
- Hold Recall
- Call Waiting/Camp-On
- DND (Do Not Disturb)
- Call Transfer

Hardware

3.10 Message Wait/Call Back

Description

When user presses Message button, various message types that system supportsactivated or retrieved.

The message type is as below.

- Internal Call Back.
- Missed Call.
- Voice Mail
- SMS(Internal SMS)

The Admin value : Station Data – Terminal Attribute(112) - Message Wait Button, can choose the Message button working.

If the value is set to All Message, all messages are displayed as list type.If the value is set to one specific Message, the button is working only the message type.

3.10.1 Missed Call (CLI Message Wait)

Description

When your iPECS IP or LDP Phone with display receives an external call with Calling Line Identification (CLI) and the call is abandoned (disconnects before answer), the system will generate a CLI Message Wait (missed call) log with the Caller Identification, date and time. You may access this CLI log with the **[Message/Call Back]** button to review and, if desired, return the call.

You may assign a **{CLI MESSAGE WAIT}** button with direct access to the CLI Message Wait list. The button LED flashes at 30 ipm to indicate a CLI Message Wait.

Operation

iPECS IP & LDP Phones

*To assign a **{CLI MESSAGE WAIT}** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + 32 + **[HOLD/SAVE]**

To review CLI Messages

1) Press the **[MESSAGE/CALL BACK]** button, the LCD shows a summary of all types of messages waiting. The text of LCD including Soft buttons may be different according to Phone type and LCD size.

INTERNAL CALL BACK(000) MISSED CALL(002) VOICE MAIL(000) SMS(01)

2) Dial '2' to select Missed Call(CLI message). The LCD displays the first CLI message details including the Caller ID, time and number of missed calls from the caller.
3) The stations with 3 soft button, missed call list are displayed as below.

4) Users can choose each missed call, check detail information or call back to missed number.

18/04 05:50 ? STA 1001
19/04 13:20 ? Mr. James
20/04 14:10 ? 0314504629

5) The stations without 3 soft button, the first CLI message is displayed.

5551122323
21/11 09:10 CNT: 2

6) Use the Volume button or Navigation button to scroll through the missed call list.

Conditions

✓ Message – Missed Call is same as CALL LOG – Missed Call.

Related Features

■ Station Message Wait/Call Back

Hardware

■ iPECS IP or LDP Phone with display

3.10.2 Message Wait Reminder Tone

Description

In addition to the **[MESSAGE/CALL BACK]** button LED, iPECS IP and LDP Phones can be sent a notice as a periodic reminder tone or voice prompt to you of message waits in queue. This notice is sent to the station only while idle and is heard over the speaker.

Operation

System

When a user has voice mail, system can provides this as voice prompt instead of dial tone according to option based on station. But DECT phones do not support the above {Message Wait Notice} option.

- Disable: System provides normal dial tone when a user goes to off hook status.
- Tone: System provides warning tone instead of dial tone to give indication when a user goes to off hook status.
- Prompt: System provides message indication as voice prompt when a user goes to off hook status.

Conditions

- ✓ The interval between tones can be 00 to 60 minutes. A setting of 00 disables the reminder tone.
- ✓ The notice will continue until all messages have been retrieved.
- ✓ Tone will be provided as stutter dial-tone when the handset is lifted to indicate messages are waiting.
- ✓ Voice prompt can be provided when there is an idle VM channel. Otherwise, tone can be provided.

Programming

Keyset Admin.

STATION

- Message Wait Notice (PGM127-Button 20)

SYSTEM

- Message Reminder Tone Timer (PGM 181-Button 8)

Web Admin.

STATION DATA

- Station VM Attributes ➤ Message Wait Notice

SYSTEM DATA

- System Timers ➤MSG Wait Reminder Tone Timer

Related Features

- Station Message Wait/Call Back

Hardware

- iPECS IP or LDP Phone

3.10.3 Short Message Service (SMS)

Description

The Short Message Service (SMS) provides the ability to send and receive text messages to and from iPECS IP and LDP Phones equipped with a display as well as UCS Client and WLAN Phone.

The text is comprised of words, numbers, or an alphanumeric combination. Each short message can be up to 100 characters in length when Latin alphabets are used.

You send and receive messages from your iPECS IP or LDP Phone or the Station Portal of the iPECS system Web Home Page. For operation using the Station Portal Web page, see your System Administrator.

Operation

*To assign a Flex button as a **{SMS Sending}** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + 36 + **[HOLD/SAVE]**

To resend an existing SMS message

1) Press the **[TRANS/PGM]** button.
2) Dial “36”, the SMS send code.

SMS SENDING MODE RESEND-(1) EDIT-(2)

3) Dial “1” to resend a message.

DIAL STA DEST(XXX-XXX) SKIP-(*)

4) Dial the desired station range to receive the message or “*” to use the displayed station range.

STA 100-124 SEND-(SAVE) CANCEL-(#)

5) Press the **[HOLD/SAVE]** button to send the message or “#” to cancel.

To send a new message

1) Press the **[TRANS/PGM]** button.
2) Dial “36”, the SMS send code.

SMS SENDING MODE RESEND-(1) EDIT-(2)

3) Dial “2” to send a new message.

DIAL STA DEST(XXX-XXX)

4) Dial the desired station range to receive the message.

STA 100-124 SEND-(SAVE) CANCEL_(#)

5) Dial the message using two digits for each character as shown in the table below, use the **[REDIAL]**button to delete a character.

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

6) Press the **[HOLD/SAVE]** button to send and store the message or “#” to cancel.

To retrieve SMS Messages

1) Press **[MSG]** or **[CALL BACK]** button. The message contents summary will be shown as below.

```
INTERNAL CALL BACK(000)
MISSED CALL(002)
VOICE MAIL(000)
SMS(01)
```

2) Dial “4” to select SMS message wait.

3) The display shows the first line of received SMS messages.

```
MSG: [0] EMERGENCY CONFER
[1] LETS GO TO LUNCH
```

4) Use the **[VOL UP]**/**[VOL DOWN]** to scroll through messages.

```
MSG: [2] WE HAVE A MEETING A
[3] CONFERENCE AT 9 AM
```

5) Dial the message number (“01”~“10”) to view the entire message, use the **[VOL UP]**/**[VOL DOWN]** to view each line of the message.

```
MSG:> WE HAVE A MEETING AT
>7PM IF YOU CAN NOT MAKE IT
```

To delete SMS messages

1) Dial “#”.

```
DELETE MESSAGE?
YES : 1 NO : 2 ALL:3
```

2) Dial “1” to delete the current message, “2” to return to idle or “3” to delete all received SMS messages.

Conditions

- ✓ The system can store up to 10 received SMS messages and 1 sent SMS message for a station.
- ✓ Once the received message storage capacity is reached, additional incoming SMS messages are ignored.
- ✓ For operation with WLAN Phone or UCS Client, refer to the respective User Guide.

Hardware

- iPECS IP or LDP Phone with Display
- iPECS WLAN Phone
- iPECS UCS Client

3.10.4 Internal Call Back

Description

Internal Call Back is two types.

- **Leave Call Back number to called station** - when a called station does not answer or is in DND, by pressing Message button, user can leave his number to the called station to request a Call Back. When the called station goes to idle, press Message button, choose Internal Call Back, the calling station number is listed. By pressing Message button, call is served back to the calling station.
- **Serve Call Back Ring to calling station -** when a busy station is called, by pressing Message button, user may request to be placed in a queue to receive a Call Back. When the called station goes to idle, the system notify to the caller with Call Back ring. When you answer the Call Back ring, the previously busy station is start to ringing.

Operation

iPECS IP & LDP Phones

To leave a Internal Call Back, while receiving ring back tone or no response on a call announce

1) Press the **[MESSAGE/CALL BACK]** button, confirmation tone received.
2) Hang up, Message Wait activated.

To leave a Internal Call Back, while receiving DND tone

1) Press the **[MESSAGE/CALL BACK]** button, confirmation tone received.
2) Hang-up, Message Wait activated.

To leave a Call Back ring(queue for a station), while receiving busy

1) Press the **[MESSAGE/CALL BACK]** button, the user receives confirmation tone.
2) Hang up, return to idle.

To respond to a Call back recall, when the busy station is available the system calls back,

1) Lift the handset or press the **[SPEAKER]** button.
2) Previously busy station is called.

To retrieve Station Messages Waiting

1) Press **[MESSAGE/CALL BACK]** button, the message contents summary as below or the Station Messages Waiting list will display.

1. INTERNAL CALL BACK(003) 2. MISSED CALL(002) 3. VOICE MAIL(000) 4. SMS(01)

2) Choose 1 for Internal Call Back.

MSG : 100

To return a call for the current Station Message

1) Press the **[OK]** button or **[Message] [Call Back]** button..

To delete the first Message Wait from the list

1) Press the "*" button or **[DELETE]** soft button.

SLT(Single Line Telephone)

To leave a Message Wait, while receiving ring back tone or no response on a call announce

1) Momentarily press the hook switch.
2) Dial "*556", Activate Message Wait/Call Back code.
3) Hang up, Message Wait is activated.

To leave a Message Wait, while receiving DND tone

1) Momentarily press the hook switch.
2) Dial "*556", Activate Message Wait/Call Back code.
3) Hang up, Message Wait activated.

To retrieve a Station Message Wait

1) Dial "*557", Message Wait/Call Back Answer code.

To leave a Call Back (queue for a station), while receiving busy

1) Momentarily press the hook switch.
2) Dial "*556", Activate Message Wait/Call Back code.
3) Hang up, Call Back is activated.

To respond to a Call back recall, when the busy station is available the system calls back

1) Lift the handset.
2) Previously busy station is called.

Conditions

- ✓ A Message Wait/Call Back return call will always ring at the receiving station overriding the Intercom signaling mode selected.
- ✓ A station can leave only one callback request.
- ✓ The **[MESSAGE/CALL BACK]** button LED will continue to flash until all Message Wait and Call Back requests, including Voice Mails, have been serviced.
- ✓ If a station is attempting to leave a message and the system Message Wait queue is full, the station will receive ICM busy tone.
- ✓ A Message Wait remainder tone can be enabled to remind the user of messages waiting.
- ✓ A station in Call Forward can leave a message wait.
- ✓ A Message Wait indication is left at the originally called station even if the call is forwarded.
- ✓ A user of an iPECS IP or LDP Phone with Display may call back to stations that left messages in any desired order, or the normal ("oldest first") order.
- ✓ Placing an Intercom call to a station will cancel any existing Message Wait from that station.

Programming

Keyset Admin.

SYSTEM

- Message Reminder Tone Timer (PGM 181-Button 8)

Web Admin.

SYSTEM DATA

- System Timers ➤ MSG Wait Reminder Tone Timer

Related Features

- Message Wait Reminder Tone
- CLI Message Wait

Hardware

3.11 Paging

3.11.1 Internal/External & All Call Page

Description

If your station is permitted to access page facilities, you can connect and transmit voice announcements to any or all of the systems Internal/External Page zones. Stations are grouped into “zones” to receive pages to the zone. Stations not assigned to any zone will not receive pages including All Call pages.

A page warning tone, if assigned, will be provided to the Page Zone(s) prior to the audio connection. You may continue the page announcement for the specified Page Time-out after which you will be disconnected and the Page Zone(s) returned to idle.The default Page Zone dial codes are as follows:

Internal Page Zones	*301~*335(eMG80/100) / *301~*400(eMG800/UCP)
Internal All Call Page	*543
External Page Zones	*545
External Page Zones	*546 (Only UCP)
External All Call Page	*548
All Call Page	*549

Flexible buttons of iPECS IP and LDP Phones may be assigned to access a Page Zone as a **{PAGE ZONE}** button.

Operation

iPECS IP & LDP Phones

*To assign a Flex button as a **{PAGE ZONE}** button*

- **[TRANS/PGM]** + {FLEX} + Page Zone number + **[HOLD/SAVE]**

To make a page

1) Lift the handset.
2) Dial the desired paging code or press a **{PAGE ZONE}** button.
3) If assigned, after the Page Warning Tone, make announcement.
4) Replace the handset to return to idle.

To queue for a page when busy is received

1) Press the [MESSAGE/CALLBACK] button.
2) Replace the handset returning to idle.

SLT(Single Line Telephone)

To make a page

1) Lift the handset.
2) Dial the desired paging code.
3) If assigned, after the Page Warning Tone, make announcement.
4) Replace the handset, to return to idle.

To queue for a page when busy tone is received

1) Dial "*556", Activate Message Wait/Call Back code.
2) Replace the handset returning to idle.

Conditions

- ✓ Stations, which are denied access to paging, will receive error tone when any Page Access code is dialed.
- ✓ Stations dialing a Page Code will be queued when any of the other Internal or External Page zones are busy.
- ✓ DTMF signals from stations are transmitted through the systems External Page port after access.
- ✓ If an iPECS IP or LDP Phone user attempts to page using the speakerphone, pre-selection will be activated and the display shows 'Lift Handset for Page when 'Lift Handset for Page' is ON'. If 'Lift Handset for Page' is OFF then user can make page on speakerphone without lift handset.
- ✓ Stations in DND or busy will not receive Page announcements.
- ✓ Stations, which are not included in any Internal Page Zone, will not receive any page, including All Call.
- ✓ For external paging, an external amplifier and speaker(s) are required.
- ✓ The systems External Control Contacts may be assigned to activate when External Page is accessed. The contact can be used to activate the External page equipment.
- ✓ A station is permitted only one Page Zone queue request at a time. If a station attempts another Page Zone queue, only the later queue request is honored.
- ✓ When a busy Page Zone becomes idle, the system will select the oldest page queue, and signal the appropriate station. The signaled station will have an audible ring indicating the queue callback. The audible ringing will be a distinctive signal. The All Page Zone, while signaling a queued station, is considered busy. In addition, the All Page Zone is considered busy when any page zone is active.
- ✓ The queue recall is always in the tone ring mode regardless of the station's ICM signaling mode.
- ✓ The Call Back ring signals the station for 15 seconds, after which, the queue is canceled and the next station in the queue is signaled.
- ✓ If the waiting station is busy, and the Page zone becomes available the next idle station in the Page Queue list is signaled and the busy waiting station is placed at the bottom of Page Queue list. If there is no other available queued station in the Page Queue, the Page Queue for the busy station is canceled.
- ✓ When the waiting station goes to idle, and both a "Page Queue" and "CO Call back Queue" exist, the Page Queue is given priority.
- ✓ This feature isn't available for vUCP.

Programming

Keyset Admin.

STATION

- Page Access (PGM 111-Button 7)
- Internal Page Zone Access (PGM 118

SYSTEM

- Privacy Warning Tone (PGM 161-Button 4)

- External Control Contacts (PGM 168)
- Paging Timeout Timer (PGM 181-Button 9)

Web Admin.

STATION DATA

- Common Attributes ➤Page Access
- Internal Page Zone, Lift Handset for Page

SYSTEM DATA

- Common Attributes ➤Privacy Warning Tone
- External Control Contacts
- System Timers ➤Paging Timeout Timer

Related Features

- Meet Me Page Answer

Hardware

- External Amplifier & Speakers

3.11.2 Meet Me Page

Description

During a Page, you may request another user respond to the Page and “Meet” you by dialing the Meet-Me code. When the party dials the Meet-Me code, the page is terminated and an intercom call is connected between you and the party.Flexible buttons of iPECS IP and LDP Phones may be assigned as a **{Meet Me Page}** button.

Operation

iPECS IP & LDP Phones

*To assign a Flex button as a **{Meet Me Page}** button*

- **[TRANS/PGM]** + {FLEX} + “*544” + **[HOLD/SAVE]**

To answer a page with Meet Me Page

1) Lift the handset or press the **[SPEAKER]** button.
2) Dial “*544”, the Meet Me Page code or press the **{Meet Me Page}** button.

SLT(Single Line Telephone)

To answer a page with Meet Me Page

1) Lift the handset to receive intercom dial tone.
2) Dial “*544”, the Meet Me Page code.

Conditions

✓ A Meet Me Page must be answered within the Page Time-out timer.

✓ A station may answer a Meet Me Page from any station regardless of pickup or paging group assignments and page access permission.

✓ The paging party must remain off-hook until the paged party answers the Meet Me request. The initiator may press the Mute button to eliminate transmitting audio over the page circuit while waiting for the party to answer.

Programming

Keyset Admin.

SYSTEM

- Paging Timeout (PGM 181-Button 9)

Web Admin.

SYSTEM DATA

- System Timers ➤Paging Timeout Timer

Related Features

■ Internal/External & All Call Page

3.11.3 SOS Paging

Description

A System announcement can be recorded and then used a page announcement. A Flex button of an iPECS IP or LDP phone is required.

Operation

iPECS IP & LDP Phones

*To assign a Flex button as a **{SOS Page}** button*

- **[TRANS/PGM]** + **{FLEX}** + "Page code" + System announcement number + **[HOLD/SAVE]**

To activate SOS paging

1) Press the assigned **{SOS Paging}** flexible button.

Condition

✓ This feature can be only activated by pressing assigned flexible button in idle state.
✓ The System announcement for SOS Paging can be recorded only from an Attendant station.
✓ Paging zones includes Internal, External and All Call page zones.
✓ SOS paging is not restricted by the Paging timer. The entire SOS paging announcement will play over the Page zones until complete even if the Paging time out expires.

Programming

Keyset Admin.

STATION

- Page Access (PGM 111-Button 7)

SYSTEM

- Paging Timeout Timer (PGM 181-Button 9)

Web Admin.

STATION DATA

- Common Attributes ➤Page Access

Related Features

■ Internal/External & All Call Page
■ Integrated Voice Mail

Hardware

■ iPECS IP or LDP Phone

3.12 Push to Talk Paging

Description

Each iPECS IP, LDP and WLAN Phone can be assigned as a member of one or more of the system's nine Push-To-Talk (PTT) page groups.

As such a user, you may login or log-out of any one or all of the PTT groups to which your station is assigned.

Once logged in, you may place or receive one-way page announcements to/from other users who are logged in to the same PTT group.

To place a PTT page announcement, the user must press and hold the **{PTT}** Flex button.

An Attendant may log other stations in and out of PTT groups.

Operation

iPECS IP & LDP Phones

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

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + "99" + **[HOLD/SAVE]**

To log-in to a PTT group

1) Dial "#0", the PTT Log-in/out code.
2) Dial the desired PTT group number ("1"~"9" and "0" for all groups).

To log-out of the PTT group(s)

1) Dial "#0", the PTT Log-in/out code.
2) Dial "*".

To place a page to the active PTT group

1) Press and hold the **{PTT}** Flex button.
2) After confirmation tone, make your page announcement.

Attendant

To log other stations in to a PTT group

1) Press the **[TRANS/PGM]** button.
2) Dial "077", the Attendant PTT log-in/out code.
3) Dial the desired station range, for a single station dial the station number and '*'.
4) Dial the PTT group number ("1"~"9" and "0" for all groups).
5) Press the **[HOLD/SAVE]** button.

To log other stations out of a PTT group

1) Press the **[TRANS/PGM]** button.
2) Dial "077", the Attendant PTT log-in/out code.
3) Dial the desired station range, for a single station dial the station number and '*'.
4) Dial "*".
5) Press the **[HOLD/SAVE]** button.

Conditions

- ✓ Conditions associated with Internal/External & All Call Page apply to Push-to-Talk paging.
- ✓ To access PTT paging, the station must be permitted access to system paging and assigned to one or more PTT groups..
- ✓ 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.
- ✓ A station can only login to PTT groups to which it is assigned access.
- ✓ The station must have a **{PTT}** to place or receive PTT announcements. As a default, the iPECS WLAN phone is assigned a PTT button.
- ✓ The station may be assigned and logged in to the default active PTT group in the system database.

Programming

Keyset Admin.

NUMBERING

- PTT Log-in/out (PGM 109-Button 4)

STATION

- Page Access (PGM 111-Button 7)
- Default Active PTT Group (PGM 111-Button 16)
- PTT Group Access (PGM 119)

Web Admin.

SYSTEM ID & NUMBERING PLANS

- Flexible Numbering Plan ➤PTT Group Log-in/Log-out

STATION DATA

- Common Attributes ➤Page Access, Active PTT Group Number

Related Features

- Internal/External & All Call Page

Hardware

- iPECS IP or LDP Phone
- iPECS WLAN
- iPECS UCS Client

3.13 Emergency page

Description

If a station makes a Emergency page, the station can page to not only idle stations and external page ports but also talking stations.

When the Emergency page is starting, the talking stations disconnect the current call and receive the emergency page.

Except that the Emergency page can page to talking stations, the Emergency page is equal to All Call Page.

Flexible buttons may be assigned to make a emergency page as a**{Emergency Page}** button.

Operation

*To assign a Flex button as a **{Emergency Page}** button*

- **[TRANS/PGM]** + **{FLEX}** + *589 (Emergency Page code) + **[HOLD/SAVE]**

To make a page

1) Lift the handset.
2) Dial"*589",Emergency Page code or press a **{Emergency Page}** button.
3) If assigned, after the Page Warning Tone, make announcement.
4) Replace the handset to return to idle.

Programming

Web Admin.

STATION DATA

- Common Attributes ➤Page Access

SYSTEM DATA

- Common Attributes ➤Privacy Warning Tone
- External Control Contacts
- System Timers ➤Paging Timeout Timer

Hardware

4 Outside (CO/IP)lines

4.1 Auto Fault Detection and Recovery

Description

If the system detects a Line fault ona PRI Line or SIP Trunk,the Line/Trunkis placed in an Out-Of-Service state and temporarily placed it in the unused Line group. Upon recovery the system will automatically return the Line to in-service. Co line fault information is provided to NMS if it is connected.

Operation

System

Operation of Fault Detection and Recovery is automatic.

Attendant

To terminate the alarm signal

1) Dial "*565", the Alarm Rest code.

Conditions

- ✓ The "Unused" Line Group contains Lines that are not used or are temporally blocked.
- ✓ This feature is only available for E1, T1 an PRI Lines, and SIP trunks.

Programming

Keyset Admin.

CO/IP

- CO/IP Line Group (PGM 141-Button 1)

System Data

- DCOB Fault Notify (PGM163-Button 6)
- SIP Registration Fault Notify (PGM163-Button 7)

Web Admin.

CO LINE DATA

- Common Attributes ➤CO/IP Group

System Data

- Alarm Attributes ➤ Emergency Call Notify, DCOB Fault Notify, SIP Registration Fault Notify

Related Features

- ■ Outside Line Groups

Hardware

4.2 Caller ID Mode

Description

The Caller ID signals for analog CO Lines varies from region to region. To support these differences, each analog CO Line is assigned the type of Caller Id signal provided from the CO Line. The Caller Id signal type can be configured as:

- FSK
- DTAS FSK
- DTMF
- Russia-CID

Operation

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

Conditions

✓ Caller ID is provided from analog CO Lines after the first ring signal. To detect Caller ID, 'ICLID Ring Timer' should be set in UCP.

Programming

Keyset Admin.

CO/IP

- CO Additional Attribute - CID setting (PGM 148-Button 1)

Web Admin.

CO LINE DATA

- CID/CPN Attributes ➤ CID Mode
- Analog Attributes ➤ ICLID Ring Timer

Hardware

4.3 CO Line Flash

Description

Analog CO Lines recognize a brief open or ground connection, “Flash”, as a request for new dial tone. When used behind a PBX, a Flash is often used to activate a PBX feature or call transfer.

Operation

iPECS IP & LDP Phones

*To assign a **{Flash}** Flex button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “94” + **[HOLD/SAVE]**

While connected to an analog CO line

1) Press the **[FLASH]** button, the system generates a flash on the analog CO line.

SLT(Single Line Telephone)

While connected to an analog CO line

1) Momentarily depress the Hook-switch.
2) Dial “*551”, Flash Command To CO Line code, the system generates a Flash on the analog CO Line.

Conditions

- ✓ Stations may Flash on an analog CO Line defined for PABX operation and will experience COS (Class of Service) dialing restrictions after a PABX Trunk access code is dialed.
- ✓ During a Flash, the LED for the Line button will remain lit.
- ✓ A Flash may be stored as a part of a Station or System Speed Dial number.
- ✓ While connected to an internal call or dial tone, activating Flash will return internal dial tone.
- ✓ While making a Page Announcement, activating Flash will terminate the Page Announcement and return internal dial tone.
- ✓ The Flash function is not available on ISDN and IP Line calls.

Programming

Keyset Admin.

CO/IP

- CO Flash Timer (PGM 142-Button 12)

Web Admin.

CO LINE DATA

- Analog Attributes ➤Flash Timer

Related Features

- ■ Station Speed Dial, System Speed Dial

4.4 CO Line Release Guard Time

Description

To assure that the service provider's switching equipment has sufficient time to restore to the idle condition, the system will hold analog CO Lines in a busy state after release of theLine.

The time between the disconnect and when the system changes the CO Line status from busy to idle is the CO Line Release Guard time.

Operation

System

Operation of this feature is automatic.

Programming

Keyset Admin.

SYSTEM

- CO Release Guard Timer (PGM 180-Button 16)

Web Admin.

SYSTEM DATA

- System Timers ➤CO Release Guard Timer

Hardware

4.5 CO Line 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 various PBX and carrier systems.

Operation

System

Operation of Ring detect is automatic.

Conditions

- ✓ Ring On and Ring Off are assigned on a system basis.
- ✓ The CO Ring Detect is applied to analog CO Lines only.

Programming

Keyset Admin.

SYSTEM

- CO Ring OFF Timer (PGM 180-Button 17)
- CO Ring ON Timer (PGM 180-Button 18)

Web Admin.

SYSTEM DATA

- System Timers ➤CO Ring On Timer
- System Timers ➤CO Ring Off Timer

4.6 Dial Pulse Signaling

Description

An analog CO Line can be configured to send dial pulse signals to the central office. If programmed as a pulse CO line, the system will send open loop pulses at 10 pps with the assigned break/make ratio.

Operation

System

Operation of this feature is automatic when programmed.

Conditions

- ✓ The break/make ratio is system programmable as 60/40 or 66/33.
- ✓ You can change the signaling from pulse to DTMF by dialing '*' while on a Line assigned for pulse signaling.

Programming

Keyset Admin.

CO/IP

- CO Line Signal Type (PGM 141-Button 5)

SYSTEM

- Pulse Dial (Break/Make) Ratio (PGM 176)
- Pause Timer (PGM 181-Button 10)

Web Admin.

CO LINE DATA

- Common Attributes ➤Analog Attributes➤CO Line Signal

SYSTEM DATA

- Pulse Dial (Break/Make) Ratio
- System Timers ➤Pause Timer

Related Features

- Dial Pulse to Tone Switchover
- DTMF Signal Sending

Hardware

4.7 Direct Inward Dialing (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. Digits sent to the system are generally the last digits (3 or 4) dialed by the caller.

After collecting the digits from the carrier, iPECS converts the digits employing one of three DID digit conversion Types:

- **Type 0** - collect incoming digits based on the programmed digit count and convert the digits according to the DID conversion pattern, resulting in the DID destination number.
- **Type 1** - use the incoming DID digits as the destination number without converting.
- **Type 2** - Use the result of DID conversion Type 0 as an index to the Flexible DID Conversion Table and use the table information to route the call.

Operation

System

Operation of DID is automatic once configured.

Conditions

- ✓ If ICLID routing is assigned for the Line, the received Caller Id is compared to the ICLID Table for routing first. If Caller Id does not match an entry in the ICLID Table, the normal DID call processes are used.
- ✓ 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 are routed to the Attendant, a tone, Station group, or System announcement. When the Attendant receives such calls, the call is appropriately identified in the Attendant iPECS IP or LDP Phone display.
- ✓ For a station that is part of a non-pilot Station Hunt group, DID calls will follow the group hunt process if the Station is busy or does not answer the call. DID calls are subject to Group Call Pick-up and Directed Call Pick-up.
- ✓ If a System announcement is defined as the destination in the Flexible DID Conversion Table, a Caller Controlled Routing Table for the announcement can be defined. iPECS can be configured to drop (disconnect) the call after playing the recorded announcement.

Programming

Keyset Admin.

CO/IP

- CO Service Type (PGM140-Button1)
- ISDN DID Remove the Number of Digit (PGM 143-Button 5)
- DID Conversion Type (PGM 145-Button 2)

SYSTEM

- DID/DISA Busy Destination (PGM 167-Button 1)

- DID/DISA Error Destination (PGM 167-Button 2)
- DID/DISA No Answer Destination (PGM 167-Button 3)
- DID/DISA No Answer Timer (PGM 181-Button 2)

TABLES

- Customer Call Routing Tables (PGM 228)
- Flexible DID Conversion Table (PGM 231)

Web Admin.

SIP DATA

- SIP User ID Attributes

CO LINE DATA

- Common Attributes ➤ CO Line Type
- CID/CPN Attributes ➤ DID Remove Number
- DID Service Attributes ➤ DID Conversion Type

SYSTEM DATA

- DID/DISA Destination
- System Timers ➤ DID/DISA No Answer Timer

TABLES DATA

- CCR (Customer Call Routing) Table
- Flexible DID Conversion

Related Features

- Integrated Auto Attendant
- Directed Call Pick-Up
- Group Call Pick-Up
- IP Trunk
- IP Address Dialing

Hardware

- IP, BRI or PRI Line

4.8 Digit Conversion

Description

When you place a call, the digits you dialare compared with the Digit Conversion Table. If a match is found, the digits are converted as defined in the table. Within the table are 15 sub-tables with up to 200 entries of up to sixteen digits for each sub-table for eMG80/100. With eMG800/UCP, within the table are 32 sub-tables with up to 200 entries of up to 24 digits for each sub-table.

Different conversion can be assigned for the same dialed number based on the system service mode and LCR configuration. Conversion can also be limited to outside calls from a station or an outside Line. The latter may be used in networked environments where a system in the network employs Lines from another system to place a call.

Operation

Digit Conversion is applied automatically according to programming.

Conditions

- ✓ Each CO line can have a Digit Conversion Tables.
- ✓ There are two (2) special entries for configuring digit conversion:

3) X: Mask Digit, any digit is accepted
4) F: Ignore Digit

Programming

Keyset Admin.

SYSTEM

- Digit conversion Table (PGM 270)

Web Admin.

Station DATA

- Common Attribute ➤ Digit Conversion Table Index

Table DATA

- Digit conversion Table

Hardware

4.9 Direct Inward System Access (DISA)

Description

Each outside Line may be assigned for DISA service, which allows an incoming caller to gain access to the system resources and/or features.

The iPECS will answer the outside call and provide intercom dial tone or route toa System announcement where Caller Controlled Routing 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 the incoming DISA call, DND Warning tone is provided to indicate an Authorization Code must be entered.

Operation

System

Incoming call subject to DISA service

1) Recognize incoming call.
2) Answer call and connect caller to Intercom dial tone or System announcement.
3) Process call based on received digits/programming.

DISA Caller

To access the system's resources from an external party

1) Place call to DISA facility of the system.
2) At receipt of dial tone/announcement, dial as desired. If DND Warning tone is received, enter an Authorization Code to receive dial tone.

Conditions

- ✓ Each outside Line is separately assigned for DISA operation during Day, Night and/or Timed system operation mode. DISA operation is active only when the system is in the assigned operating mode(s).
- ✓ DISA callers can be routed to a System Attendant announcement in place of Intercom dial tone. The announcement can be associated with a CCR Table or assigned to disconnect after playback.
- ✓ A DISA caller can be required to enter an Authorization Code to access the system's 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 results in disconnect.
- ✓ DISA callers are subject to COS (Class of Service) dialing restrictions. If Authorization Codes are required and the code entered matches a Station Authorization Code, the station's COS (Class of Service) will apply. Otherwise, the assigned DISA COS (Class of Service) will apply. In both cases, the Line COS (Class of Service) for the outgoing Line will be active.
- ✓ The system will disconnect an outgoing DISA call if the Unsupervised Conference timer expires or disconnect supervision is received. A disconnect warning tone is provided 15 seconds prior to disconnect.
- ✓ If a DISA caller encounters a system All Lines Busy, busy tone is received for 5 seconds before ICM Dial tone is again presented and the DISA caller may try another call.

- ✓ LEDs associated with the DISA Line appearance will provide normal status indications at all stations except the Attendants. The LED for the Line at an Attendant will flutter at 240 ipm when busy.
- ✓ An iPECS IP or LDP Phone user can only receive a DISA call with an available DISA Line appearance button.

Programming

Keyset Admin.

CO/IP

- DISA Account Code (PGM 142-Button 5)
- DISA CO Access (PGM 142-Button 11)
- DISA Attributes (PGM 146)

SYSTEM

- DISA Retry Counter (PGM 160-Button 5)
- DISA COS (Class of Service) Assignment (PGM 166)
- DID/DISA Busy Destination (PGM 167-Button 1)
- DID/DISA Error Destination (PGM 167-Button 2)
- DID/DISA No Answer Timer (PGM 181-Button 2)
- Unsupervised Conference Timer (PGM 182-Button 5)

TABLES

- CCR Audio Text Tables (PGM 228)
- Weekly Time Table (PGM 233)

Web Admin.

CO LINE DATA

- Common Attributes ➤DISA Account Code, DISA CO Access
- DISA Service Attributes

SYSTEM DATA

- System Attributes ➤DISA Retry Count
- DISA COS (Class of Service)
- DID/DISA Destination
- System Timers ➤DID/DISA No Answer Timer, Unsupervised Conference Timer

TABLES DATA

- CCR Table
- Auto Ring Mode Table

Related Features

- Day/Night/Timed/Scenario Ring Mode
- Dialing Restrictions
- Authorization Codes (Password)
- Unsupervised Conference
- Integrated Auto Attendant/Voice Mail
- Auto Service Mode Control

4.10 DTMF Signaling

Description

Dual Tone Multi-Frequency (DTMF) signals are used with Lines assigned for DTMF signaling.

The duration of the DTMF signal can be adjusted from 40 to 990 milliseconds.

Operation

System

Operation of this feature is automatic when programmed.

Conditions

✓ The system mutes your voice transmission to reduce interference while sending DTMF tones.

Programming

Keyset Admin.

CO/IP

- CO Line Signal Type (PGM 141-Button 5)

SYSTEM

- Inter Digit Timer (PGM 181-Button 7)
- DTMF Duration Timer (PGM 182-Button 9)

Web Admin.

CO LINE DATA

- Analog Attributes ➤CO Line Signal

SYSTEM DATA

- System Timers ➤Inter-Digit Timer, DTMF Duration Timer

Related Features

- Dial Pulse Signaling
- Dial Pulse to Tone Switchover

Hardware

4.11 IP Address Dialing

Description

If allowed, you can place calls using an IP path. The system accepts user dialed digits as the IP address for the called party. When dialing an IP call, the asterisk, “*”, is used as the “dot” between bytes of the IP address.

Operation

iPECS IP & LDP Phones

To place an IP Call

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the**{IP Line Group}** button or dial IP Group access code.
3) Dial “xxx * xxx * xxx * xxx”, use “*” as the dot in the IP address.
4) Press “#” to complete dialing.

SLT(Single Line Telephone)

To place an IP Call

1) Lift the handset.
2) Dial the IP Group access code.
3) Dial ‘xxx “*” xxx “*” xxx “*” xxx‘, use “*” as the dot in the IP address.
4) Press “#” to complete dialing.

Programming

Keyset Admin.

STATION

- CO/IP Group Access (PGM 117)
- Direct IP Call (PGM 122-Button 1)

CO/IP

- CO/VOIP Mode (PGM141-Button11)

Web Admin.

STATION DATA

- CO/IP Group Access
- Station IP Attributes ➤Direct IP Call

CO LINE DATA

- VoIP Attributes

Hardware

- IP Line

4.12 IP Trunking

4.12.1 H.323 v4 Service

Description

When assigned to support H.323 protocol, VoIP channels provide protocol conversion between H.323 v4 and the iPECS protocol.

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 are supported employing H.450.1 ~ H.450.12 standards, which define the following supplementary services:

- H.450.1: Supplementary Services Framework
- H.450.2: Call Transfer
- H.450.3: Call Diversion
- H.450.4: Call Hold
- H.450.5: Call Park & Pick-up
- H.450.6: Call Waiting
- H.450.7: Message Waiting
- H.450.8: Name Identification
- H.450.9:Call Back
- H.450.10: Camp-On
- H.450.11: Intrusion
- H.450.12: Additional Information

Operation

System

Operation of H.323 Service is automatic.

Programming

Keyset Admin.

STATION

- Station IP Attributes (PGM 122)

BOARD

- H.323 VoIP Attributes (PGM 130)

CO/IP

- CO/IP Line Group (PGM 141-Button 1)
- CO VoIP Mode (PGM 141-Button 11)

Web Admin.

STATION DATA

- Station IP Attributes

BOARD BASEDDATA

- H.323 VoIP Attributes

CO LINE DATA

- Common Attributes ➤CO/IP Group
- VoIP Attributes ➤CO VoIP Mode

H.323 DATA

- H.323 Basic Attributes
- H.323 CO Group Attributes
- H.323 Incoming route table

Related Features

- System Networking
- SIP Trunk Service

Hardware

- IP Line

4.12.2 SIP Trunk Service

Description

When assigned to support SIP (Session Initiation Protocol), VoIP channels provide protocol conversion between SIP and the iPECS protocol. This permits the VoIP channel to connect to external SIP networks for call services. In addition, to the IETF RFC-3261 Session Initiation protocol draft standard, iPECS VoIP channels support other SIP related RFCs including:

RFC-2617	HTTP Authentication, Basic & Digest
RFC-3515	Refer Method
RFC-3264	Offer/Answer Model
RFC-3265	SIP Basic Call Flow Examples
RFC-3891	SIP “Replaces” Header

Using the SIP database assignments, the system will register and authenticate with the SIP proxy server permitting the system to interoperate employing SIP to establish, manage and terminate real-time voice sessions with external parties.

Operation

System

Operation of SIP Service is automatic.

Programming

Keyset Admin.

STATION

- SIP User ID Table (PGM111-Button 19)
- Station IP Attributes (PGM 122)

CO/IP

- CO/IP Line Group (PGM 141-Button 1)
- CO VoIP Mode (PGM 141-Button 11)

Web Admin.

STATION DATA

- Common Attributes ➤SIP User Table Index
- Station IP Attributes

SIP DATA

- SIP CO Attributes
- SIP User ID Attributes

CO LINE DATA

- Common Attributes ➤CO/IP Group
- VoIP Attributes ➤CO VoIP Mode

Related Features

- System Networking

- H.323 v4 Service

Hardware

- SIP Trunk
- VOIP SW (Soft VOIP channel) for UCP

4.12.3 VOIP SW (Soft VOIP channel) for UCP

Description

VOIP SW is virtual channel. It has no Hardware. It can be installed according to VOIPCL8 license in UCP100/600.

Operation

Operation of VOIP SW Line Service is automatic.

Conditions

✓ Out band DTMF and 2833 is supported, not support In band DTMF.
✓ Only G.711 codec is supported.
✓ Call progress tone is supported by recorded source.

Programming

Web Admin.

SYSTEM ID & NUMBERING PLANS

- Device Port Number Change ➤Virtual Registration➤ Device ID, CO, VOIU (SW), Max Port, Soft Channel

STATION DATA

- SIP User Table Index
- Station IP Attributes

CO LINE DATA

- Common Attributes ➤CO/IP Group
- VoIP Attributes ➤CO VoIP Mode

SIP DATA

- SIP User ID Attributes
- SIP CO Attributes➤ RTP Diversion Method, DTMF Type

Related Features

- SIP Trunk Service

Hardware

4.13 IP WAN Dialing after Answer

Description

The iPECS system permits sending and receiving DTMF signals after connecting to an external VoIP party.

The DTMF signal can be DTMF tone, Text String, or DTMF protocol based on the system programming.

Operation

System

Operation is automatic based on the system database.

Conditions

- ✓ The connected VoIP party must transmit DTMF digits in the mode selected in the system database otherwise; the DTMF digits will not be recognized.

Programming

Web Admin.

SIP DATA

- SIP CO Attributes➤ DTMF Type

Hardware

- ■ IP Line

4.14 ISDN (Integrated Service Digital Network)

Description

The iPECS system supports both BRI (Basic Rate Interface) and PRI (Primary Rate Interface) ISDN circuits.

The ETSI 30B+2D channel configuration is supported through switcheson the PRIU Interface board to select E1 digital Line operation or PRI ISDN operation.

4.14.1 Calling/Called Party Identification

Description

The iPECS system receives calling party identification in the ISDN call Set-up message, CLIP-Calling Line Identification Presentation.

The answering party identification, which may be different from the called party, is received in the ISDN connect message, COLP-Connected Line Identification Presentation. When provided, the LCD of iPECS IP and LDP Phones displays the identification, which is included in call records.

LINE RINGING CLI 03438502821

The system will also compare the Calling Line 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.

For an incoming call, the received identification may be sent to the selected serial port for output. The output will appear as below:

- Output at ring start:
  AA : BBBBBBBBCCC
- Output at ring stop (abandoned)
  AA : BBBBBBBBCCC : DDDDDDDDD(EEE)
- Output at answer
  AA : BBBBBBBBCCC : DDDDDDDDD(EEE) -> FFF
- Where:

5) AA – ISDN CO line number
6) BB…B –Calling/Connected Line Identification received
7) CCC – Called station
8) DD…D – Speed Dial name
9) EEE – Speed Dial bin number
10) FFF – Answering station

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, not reported, to the far-end user. Calling Line Identification Restriction and Connected Line Identification Restriction may be enabled in the system database or you may activate restrictions with a **{CLIR}** and **{COLR}** Flex buttons.

Operation

System

Operation of this feature is automatic.

iPECS IP & LDP Phones

*To program **{CLIR}** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “81” + **[HOLD/SAVE]**

*To program **{COLR}** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “82” + **[HOLD/SAVE]**

To activate CLIR or COLR, before placing or answering an ISDN call

✓ Press the **{CLIR}** or **{COLR}** Flex button.

Conditions

✓ This feature may not be available in the specific ISDN service area or may be a subscription service.

Programming

Keyset Admin.

STATION

- CLIP LCD Display (PGM 114-Button 1)
- COLP LCD Display (PGM 114-Button 2)

CO/IP

- COLP Table Index (PGM 143-Button 1)
- Type of Calling Number (PGM 143-Button 4)

ISDN/ICLID

- CLIP Print to Serial Port (PGM 200-Button 2)

ISDN LINE

- CLIP/COLP Table (PGM 201)

Web Admin.

STATION DATA

- CLI Attributes ➤CLIP Display, COLP Display

CO LINE DATA

- CID/CPN Attributes ➤ CLIP &COLP Table Index, Type of Number for Calling Party Info

ISDN LINE DATA

- ISDN Attributes ➤CLI Print to Serial
- CLIP/COLP Table

Hardware

■ ISDN Line

4.14.2 ISDN AOC (Advice of Charge)

Description

When ISDN Advice of Charge service is provided from the ISDN, the iPECS system will deliver charge information for display in the LCD of iPECS IP and LDP Phones 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

✓ AOC information, which is 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).

✓ This feature may not be available in the specific ISDN service area or may be a subscription service.

Programming

Keyset Admin.

SYSTEM

- SMDR attributes (PGM 177)

CO/IP

- Advice of Charge Type (PGM 143-Button 9)

Web Admin.

SYSTEM DATA

- SMDR Attributes

CO LINE DATA

- ISDN Attributes ➤Advice of Charge

Related Features

- SMDR (Station Message Detail Recording)
- Call Cost Display

Hardware

- ISDN Line

4.14.3 Keypad Facility

Description

The ISDN Keypad Facility Information Element (IE) may enable the user to activate certain ISDN services (e.g. Off-Net Forward). To access this facility, the station must be enabled and have a **{KEYPAD FACILITY}** Flex button. When activated, the digits dialed by a user are sent in the Keypad Facility IE instead of the Called Party Number IE.

Operation

iPECS IP & LDP Phones

*To program **{KEYPAD FACILITY}** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + "89" + **[HOLD/SAVE]**

To activate the keypad facility while on an ISDN line

1) Press the **{KEYPAD FACILITY}** button.
2) Dial desired digits; other actions disable the Keypad facility feature.

Conditions

- ✓ This feature can be activated or deactivated only after an ISDN Line is seized.
- ✓ Once activated, the system will continue to send dialed digits as Keypad Facility IE messages regardless of other ISDN messages. Thus, in the connected mode, DTMF tones are not sent to the connected party, only the keypad message is sent.
- ✓ If a Speed Dial bin includes the**[MESSAGE/CALLBACK]** button as the first entry, the remaining digits in the Speed Dial number are sent in a Keypad Facility IE.
- ✓ This feature may not be available in the specific ISDN or may be a subscription service.
- ✓ The system can handle only a single Call Reference; services that require handling of two simultaneous Call References, cannot be supported.

Programming

Keyset Admin.

STATION

- Flex Button Assignment (PGM 115)

Web Admin.

STATION DATA

- Flexible Buttons

Related Features

- Station Speed Dial, System Speed Dial

Hardware

- iPECS IP or LDP Phone, ISDN Line

4.14.4 Multiple Subscriber Number (MSN)

Description

Up to eight (8) Subscriber Numbers (telephone numbers) can be servedbya Basic Rate Interface allowing Direct-Dial-In call delivery.

This “Sub-Address” information is included as digits in the Called Party Number Information Element.When the MSN or Sub-address information is received, the system references the MSN Table for an index to the Flexible DID Table, which determines call routing.

If no index to Flexible DID Table is assigned then the call routes to all stations that have an idle **{MSN}** button corresponding to the MSN Telephone Number received or to a button for the outside Line. If none is available, the call routes employing the DID/DISA Destination (PGM 167).

The system will use the MSN number when a call is received using an MSN button on ISDN or SIP Lines.

Operation

System

Operation of the MSN feature is automatic when configured.

iPECS IP & LDP Phones

*To program an **{MSN}** button*

- **[TRANS/PGM]**+{FLEX}+**[TRANS/PGM]**+“85”+“#”’+MSN digits+**[HOLD/SAVE]**

Web admin

*To program an **{MSN}** button*

- [LOOP]’ + MSN digits + [SAVE]

Conditions

- ✓ If the Called Party Number from the ISDN does not match an MSN Table entry, the received digits are treated as defined for DID calls to determine an index to the Flexible DID Table.
- ✓ The MSN Table employs the outside Line entry as a quick look-up reference. If the Line number is programmed in the MSN Table, then only those entries with the Line number are searched, otherwise the entire MSN Table is searched for a match to the Called Party Number.
- ✓ When assigning an **{MSN}** button the “MSN digits” should match one of the MSN Table entries. To assign an MSN button on the phone, the MSN number must exist in the system MSN Table.
- ✓ An MSN loop button is overwritten by an individual CO line button, if it is exist. On the other hand, an MSN loop button is not overwritten by CO Group button.
- ✓ Multiple duplicate **{MSN LOOP}**buttons can be assigned on an iPECS IP or LDP Phone.
- ✓ If enabled in SMDR, the MSN number is output as the Account code for outgoing calls employing an MSN button.
- ✓ MSN Call Wait can be assigned to deliver the call to a busy station. If not assigned, the call is routed using the DID/DISA Destination (PGM 167) or is disconnected.

Programming

Keyset Admin.

STATION

- MSN Wait (PGM 114-Button 12)
- Flexible Button Assignment (PGM 115)

CO/IP

- CO/IP Attributes III (PGM 143)

ISDN/ICLID

- ISDN Attributes (PGM 200)
- MSN Table Attributes (PGM 202)

SYSTEM

- Print MSN on SMDR (PGM 177-Button 21)

Web Admin.

STATION DATA

- Common Attributes ➤MSN Wait
- Flexible Buttons

CO LINE DATA

- ISDN Attributes

ISDN LINE DATA

- ISDN Attributes
- MSN Table

SYSTEM DATA

- SMDR Attributes ➤Print MSN

Related Features

- Direct Inward Dial (DID)

Hardware

- ISDN or SIP Line

4.15 ISDN Supplementary Services

In many cases, the ISDN service provider offers enhanced services, to which a user may subscribe.

The iPECS system allows access to these ISDN “Supplementary Services” implemented under the ETSI regime as described below.

4.15.1 ISDN Call Deflection

Description

When the ISDN Supplementary Service “Call Deflection” is supported, you 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 receive the call to the ISDN.

The ISDN then sends incoming calls to the desired telephone number. In this case, the system does not set-up a Line to Line (Unsupervised Conference) connection for the call.

Operation

iPECS IP & LDP Phones

To activate ISDN Call Deflection to an external 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”, “#”).
4) Press **[SPEED]** button and desired bin number.
5) Replace the handset to return to idle.

To deactivate ISDN Call Deflection

✓ Press flashing **[FWD]** button, Call Forward will deactivate and the **[FWD]** button LED is off.

Conditions

✓ The ISDN must support the Call Deflection Supplementary Service as defined by the ETS300-202/206/207 standard protocol.

✓ ISDN lines that support Call Deflection must be assigned in the system database.

Programming

Keyset Admin.

CO/IP

- ISDN-SS CD/CR (PGM 143-Button 7)

Web Admin.

CO LINE DATA

- ISDN Line Attributes ➤ISDN-SS CD/CR

Related Features

- ISDN Supplementary Services

Hardware

- ISDN Line
- iPECS IP or LDP Phone

4.15.2 ISDN Malicious Call ID Request

Description

When the ISDN supports the Malicious Caller Id supplementary service, iPECS can request the Calling Line ID (CLID) from the ISDN. The CLID is saved and output in the SMDR call record. Malicious Caller Id is activated during an ISDN call and requires a **{MCID REQUEST}** Flex button to access the feature.

Operation

iPECS IP & LDP Phones

*To program an **{MCID REQUEST}** button*

- **[TRANS/PGM]** + {FLEX} + "*0" + **[HOLD/SAVE]**

To request MCID, while on an incoming ISDN call

1) Press the **{MCID REQUEST}**button; the CLID is output in the SMDR.

Conditions

✓ The ISDN must support the Malicious Caller Id Supplementary Service as defined by the EN300-130 standard protocol.

✓ If the MCID request is successful, the SMDR call record will include the CLID returned from the ISDN and the characters "MT". If the request fails, the characters "MF" are included in the record.

✓ If the system is programmed to save SMDR records, the MCID is stored in the system memory as part of the call record.

Programming

Keyset Admin.

CO/IP

- MCID Request (PGM 109-Button 1)

SYSTEM

- SMDR Save (PGM 177-Button 1)

Web Admin.

SYSTEM ID & NUMBERING PLANS

- MCID Request

SYSTEM DATA

- SMDR Attributes ➤ Save Enable

Related Features

- ISDN Supplementary Services
- SMDR (Station Message Detail Recording)

Hardware

- ISDN Line, iPECS IP or LDP Phone

4.16 Line Monitor

Description

When a Line is configured in the system and the system determines that the service provider connection is lost, the Line is placed in an Out-of-Service condition. In this state, the Line cannot be employed for an outgoing call but incoming calls operate normally.

Operation

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

Conditions

✓ LIK-LGCM can't support it. Only UCP-LGCM support it.

Programming

Keyset Admin.

CO/IP

- Line Monitoring (PGM148-Button 12)

Web Admin.

CO LINE DATA

- Analog Attributes ➤Analogue Line Monitor

Hardware

4.17 Multiple Calling Line ID

Description

Using an ISDN Line, sending your Id (telephone number) to the other party is referred to as Calling Line Id Presentation (CLIP) or Connected Line ID Presentation.

When you place or receive a call, the system constructs your Id and sends the information to the service provider. The Id consists of the Area Code + COLP/CLIP Table value + a Station CLI.

There are five Station CLIs associated with your station and each ISDN Line is assigned to use one of the five. The five CLIs are identified as “Station CLI 1” through “Station CLI 5”. As a default, the “Station CLI 1” is your station number. Each of the five Station CLIs can be assigned up to 16 digits.

In addition to the Station CLI, each Line is assigned a COLP/CLIP Table entry number (00~50). The corresponding Table value is a digit string of up to ten (10) digits. Also, each ISDN Line is assigned an Area code.

Note if the COLP/CLIP Table entry is set at 50, the “Station CLI 1” is always used and the Area code and Table value are not added to the “Station CLI 1”. That is only the “Station CL 1” is used as your Id.

Operation

Operation of this feature is automatic as configured.

Conditions

- ✓ Your station can be assigned to restrict sending an Id for the call.
- ✓ Your station can be configured to replace your Station CLI with the Station CLI of the Attendant.
- ✓ As a default, the Line is assigned to use the Station CLI (Station CLI 1) and ISDN Lines use the COLP/CLIP Table entry 50 or “Station CLI” in the Web Admin.
- ✓ The ISDN Line can be assigned to use different Table entries for the COLP and CLIP values.
- ✓ For Transit calls additional options are available to employ the CLI of the original forwarding party.
- ✓ The LCD of the station can display the CLI or COL Id if CLIP and COLP Display is enabled.
- ✓ Multiple Calling Line Ids applies to SIP trunks when the “From Id is assigned to use the Extension CLI.

Programming

Keyset Admin.

STATION

- Station CLI 1 (PGM 114-Button 6)
- Station CLI 2 (PGM 114-Button 24-1)
- Station CLI 3 (PGM 114-Button 24-2)
- Station CLI 4 (PGM 114-Button 24-3)
- Station CLI 5 (PGM 114-Button 24-4)

CO/IP

- CLI Type (PGM 143-Button 24-1)
- COLP Table Index (PGM 143-Button 1)
- CLIP Table Index (PGM 143-Button 2)

Web Admin.

STATION DATA

- CLI Attributes ➤Station CLI 1-5

CO LINE DATA

- CID/CPN Attributes ➤COLP & CLIP Table Index
- Type of Number for Calling Party Info

Hardware

4.18 Outside Line Groups

Description

The outside Lines, in the system can be placed together into groups for assigning access by stations and common access dial codes as well as to separate the type of Line (analog, digital or IP). There are 20 Line groups in eMG80/100 and there are 200 Line groups in eMG800/UCP.

Unused Lines are assigned to Group 21 in eMG80/100 and to Group 201 in eMG800/UCP. Private Lines are assigned to Group 00.

Conditions

- ✓ Unused outside Lines must be assigned to the appropriate group so that stations do not accidentally access these lines.
- ✓ Outside Lines in Groups 01 through 20 for eMG80/100 and Groups 01 through 200 for eMG800/UCP can be accessed individually by dialing the code “88” and the Line number.
- ✓ An outside Line in Group 00 can be accessed only with a Flex button assigned to the Line.
- ✓ The system will select an outside Line from a group based on the Round Robin or Last Choice method as configured in the system database.

Programming

Keyset Admin.

STATION

- CO PGM (PGM 112-Button 6)
- CO/IP Group Access (PGM 117)

CO/IP

- CO/IP Group (PGM 141-Button 1)

SYSTEM

- CO Line Choice (PGM 160-Button 4)

Web Admin.

STATION DATA

- Common Attributes ➤ CO PGM
- CO/IP Group Access

CO LINE DATA

- Common Attributes ➤ CO/IP Group

SYSTEM DATA

- System Attributes ➤CO Line Choice

Hardware

4.19 Outside Line Preset Forward

Description

Each outside Line can be assigned a Ring-No-Answer Preset Forward destination. An incoming call on the Line will be routed to the defined Forward destination. At expiration of the Preset Forward timer, the call forwards to the defined Preset Forward destination, which is an index to the ICLID Ring Assignment Table.

The destination can be a station or station group including an adjunct Voice Mail. When the call forwards to an adjunct Voice Mail group, a predefined Voice Mail ID (VMID) is sent to the VM system to identify the Mailbox to receive the call.

Operation

System

Operation of this feature is automatic.

Conditions

- ✓ Outside Line Preset Forward is disabled for calls initially routed to a station group.
- ✓ Outside Line Preset Forward will override Call Forward No-Answer at a station.
- ✓ Outside Line Preset Forward is disabled if the Preset Forward Timer is set to 0.
- ✓ The outside Line Preset Forward destination cannot be the integrated Voice Mail group.

Programming

Keyset Admin.

CO/IP

- Preset Forward timer (PGM 147-Button 1)
- Ring Table Index (PGM 147-Button 2)
- VMID Number (PGM 147-Button 3)

ISDN & ICLID

- ICLID Ring Assignment Table (PGM 204)

Web Admin.

CO LINE DATA

- CO/IP Preset FWD Attributes ➤Preset FWD Timer, Preset ICLID Ring Table Index, Preset VMID Digit

ISDN LINE DATA

- ICLID Ring Assignment

Related Features

- ■ Call Forward
- ■ ICLID Call Routing

Hardware

4.20 Outside Line 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 Lines. Separate ring assignments are made for Day, Night and Timed Ring operation mode. In addition, the audible signal at the station can be delayed by 1 to 9 ring cycles allowing other stations to answer the call first.

Operation

System

Operation of this feature is automatic.

Conditions

- ✓ Separate assignments are made for stations to ring in the Day, Night and/or Timed Ring mode.
- ✓ A busy station receives muted ring or Call Waiting tones as appropriate for the station's Off-hook ring assignment.
- ✓ The system Ring mode can be selected manually by the Attendant or automatically. In the Automatic mode, Day/Night selection is determined based on the Automatic Ring Mode Selection table.
- ✓ The LCD of the Attendant station will display Night and Timed Ring Mode and the **[DND]** button LED will flash.
- ✓ If an outside Line is not assigned to ring at any station, incoming calls on the outside line will ring the first available Attendant.

Programming

Keyset Admin.

CO/IP

- CO Ring Assignment (PGM 144)

TABLES

- Weekly Time Table (PGM 233)

Web Admin.

CO LINE DATA

- CO/IP Ring Assignment

TABLES DATA

- Auto Ring Mode Table

Related Features

- ■ Day/Night/Timed/Scenario Ring Mode
- ■ Auto Service Mode Control
- ■ Off-Hook Signaling

4.21 Private Line

Description

YouriPECS IP or LDP Phone can be assigned exclusive use of one or more outsideLines. These "Private" lines are assigned to a special Line Group (group 00). To use a Line from this group, your phone must be assigned a Flex button for the outside Line appearance.

Operation

iPECS IP & LDP Phones

To place an outgoing Private line call

1) Press the **{LINE}** button assigned as the Private line, dial tone is received from the Line.
2) Dial the desired number.

To answer an incoming Private line call

1) Press the flashing **{LINE}** button.

Conditions

- ✓ Private lines will not recall to the Attendant station.
- ✓ Private lines are not subject to Call Pick-Up except by an assigned Secretary with an appearance of the Private line.
- ✓ When put on hold, a Private line will recall the user after the appropriate Hold timer. The user will receive recall for the Hold Recall time, and, if still unanswered, the user will continue to receive recall for the Attendant Recall time. If the call remains unanswered, at expiration of the Attendant Recall time, the call will be disconnected and the Private line returned to idle.
- ✓ A Private line will follow Call Forwarding only if the receiving station has an appearance (**{LINE}** button) for the Private line. However, calls on a Private line can be forward to the Voice Mail permitting a voice mail to be recorded.
- ✓ IP channels cannot be assigned as Private lines.

Programming

Keyset Admin.

STATION

- Flex Button Assignment (PGM 115)

CO/IP

- CO/IP Line Group (PGM 141-Button 1)

Web Admin.

STATION DATA

- Flexible Buttons

CO LINE DATA

- CO/IP Group

Related Features

- Hold
- Call Forward
- Executive/Secretary Forward
- Call Forward, Preset

Hardware

- iPECS IP and LDP Phones

5 iPECS phone

5.1 Answering Machine Emulation

Description

When a call is sent to a Voice mailbox, the associated station can be assigned to notify you and allow you to screen the call. Two methods of notification and call screening are provided, Ring or Speaker mode.In the Ring mode, you are notified by flashing of the AME (Answering Machine Emulation) Flex button. You may press the Flex button to hear the caller as the voice message is stored. In the Speaker mode, when the call is sent to the Voice Mailbox, the caller's voice is automatically broadcast over the speaker of youriPECS IP or LDP Phone.

You may terminate the screening leaving the caller in voice mail to record a message, talk with the caller and record the conversation in the mailbox, or answer the call and disconnect the Voice Mail.YouriPECS IP or LDP Phone must be assigned with an AME Flex button for proper operation.

Operation

iPECS IP & LDP Phones

*To assign an **{AME}** button-Manual Mode:*

- **[TRANS/PGM]** + {FLEX} + "*564" + "0" + **[HOLD/SAVE]**

*To assign an **{AME}** button-Auto Mode:*

- **[TRANS/PGM]** + {FLEX} + "*564" + "1" + **[HOLD/SAVE]**

To screen a call in the Manual mode

1) Press the flashing **{AME}** button, the caller's voice is broadcast over the station speaker and stored in the Voice Mailbox.

To stop the voice broadcast and leave the caller in Voice Mail

1) Press the illuminated **[SPEAKER]** button.

To talk with the caller and record the conversation in Voice Mail

1) Press the illuminated **[MUTE]** button.

To answer the call and cancel the voice message

1) Press the illuminated **{AME}** button, the caller is connected and the Voice Mail disconnected.

Conditions

✓ AME is supported only on iPECS IP and LDP Phones and an **{AME}** Flex button must be assigned on the phone.

✓ If you answer a call using the **{AME}** button, the caller is connected in the normal manner, the Voice Mail is disconnected and, with the integrated Voice Mail, any message recorded by the caller is not stored.

Programming

Keyset Admin.

NUMBERING

- AME Feature (PGM 109-Button 2)

Web Admin.

SYSTEM ID & NUMBERING PLANS

- AME Feature

Related Features

- External Auto Attendant/Voice Mail
- Integrated Auto Attendant/Voice Mail

Hardware

- iPECS IP or LDP Phone

5.2 Auto Release of [SPEAKER]

Description

After completion of certain features, the **[SPEAKER]** button turns off automatically, returning the iPECS IP or LDP Phone to idle.

Operation

System

Auto Release of the [SPEAKER] button operation is automatic. For the supported features, refer to the conditions.

Conditions

- ✓ This feature applies to all User and Attendant Programming except Custom Message, outside Line Disable and Version Display. Auto Release of **[SPEAKER]** button also applies to features including Call Park, Call Back, Call Forward and Queuing.
- ✓ If, during Station User Programming, erroneous data is entered, error tone is received and the user must correct the error before the station will return to idle automatically.

Hardware

- iPECS IP or LDP Phone

5.3 Automatic Speaker Select

Description

The iPECS IP and LDP Phones programmed for Auto Speaker Select can access an outsideLine or an internal call by pressing the appropriate button without the need to lift the handset or press the**[SPEAKER]** button.

Audio from the call is sent to the speaker and the microphone is active, as if you pressed the **[SPEAKER]** button to answer the call. This is useful if you are receiving a call but want to place a call instead of answering the ringing call or if you multiple calls ringing at your phone, you can select the call to answer.

Operation

iPECS IP & LDP Phones

To access an internal or external system resource

1) Press a **{FLEX}** button for the Line or station.

Conditions

✓ For iPECS IP and LDP Phones not equipped/assigned with speakerphone, the user must lift the handset to be heard.

✓ Paging while on the speakerphone will cause feedback from the 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, the user must lift the handset within the predefined 5-second period or return to idle.

Programming

Keyset Admin.

STATION

- Auto Speaker Select (PGM 111-Button 1)

Web Admin.

STATION DATA

- Terminal Attributes ➤Auto Speaker Selection, Speaker/Headset

Hardware

■ iPECS IP or LDP Phone

5.4 Background Music (BGM)

Description

The iPECS IP and LDP Phones can receive audio, generally music, from an internal or external source while idle. Music from the source is received over the speaker and will be shut-off during ringing, pages, or while the station is off-hook.

Operation

iPECS IP & LDP Phones

To Receive Background Music

1) Press **[TRANS/PGM]** button.
2) Dial “73”, the BGM code.
3) Dial the two (2)-digit (00-37)BGM source.
4) 00 – no BGM
5) 01 – Music 1 (Internal)
6) 02 – Music 2 (External)
7) 03 – System Announcement MOH 1
8) 04 – SLT(Single Line Telephone) MOH 1
9) 05 – SLT(Single Line Telephone) MOH 2
10) 06 – SLT(Single Line Telephone) MOH 3
11) 07 – SLT(Single Line Telephone) MOH 4
12) 08 – SLT(Single Line Telephone) MOH 5
13) 09 – System Announcement MOH 2
14) 10 – System Announcement MOH 3
15) 11 – System Announcement MOH 4
16) 12 – System Announcement MOH 5

. . .

40) 36 – System Announcement MOH 29
41) 37 – System Announcement MOH 30
42) Press **[HOLD/SAVE]** to save your selection.

Attendant

To Transmit BGM through an External Page Port from an Attendant

1) Press **[TRANS/PGM]** button.
2) Dial “074” or “075”, the Attendant Station Program code for External Page Port 1, respectively.
3) Dial the two (2)-digit (00-37) BGM source.
4) 00 – no BGM
5) 01 – Music 1 (Internal)
6) 02 – Music 2 (External)
7) 03 – System Announcement MOH 1
8) 04 – SLT(Single Line Telephone) MOH 1

9) 05 – SLT(Single Line Telephone) MOH 2
10) 06 – SLT(Single Line Telephone) MOH 3
11) 07 – SLT(Single Line Telephone) MOH 4
12) 08 – SLT(Single Line Telephone) MOH 5
13) 09 – System Announcement MOH 2
14) 10 – System Announcement MOH 3
15) 11 – System Announcement MOH 4
16) 12 – System Announcement MOH 5

. . .

40) 36 – System Announcement MOH 29
41) 37 – System Announcement MOH 30
42) Press **[HOLD/SAVE]** to save your selection.

Conditions

- ✓ The speaker volume is adjustable at the station by using the **[VOL UP]/[VOL DOWN]**buttons.
- ✓ BGM is delayed 1 second after a return to idle state.
- ✓ Silence is provided if no BGM source is assigned.
- ✓ For remote devices, BGM must be locally provided; BGM is not sent from the eMG system to remote devices.
- ✓ System Announcement MOH 1 employs System announcement 201 and must be recorded properly. System announcement MOH2 ~ MOH30 may use any unused System announcement except for announcement 201 and 202. The announcement number must be defined in the Music Sources program (PGM 171).
- ✓ For SLT(Single Line Telephone) MOH, the SLT(Single Line Telephone) port must be connected properly and the station number of the SLT(Single Line Telephone) port used for MOH must be assigned as the SLT(Single Line Telephone) MOH port.

Programming

Keyset Admin.

SYSTEM

- BGM Type (PGM 171-Button 1)
- Internal MOH Type (PGM 171-Button 3)
- Assign SLTMOH Port (PGM 171-Button 4)
- VSF MOH2 (PGM 171-Button 5)
- VSF MOH3 (PGM 171-Button 6)
- VSF MOH4~30 (PGM 171-Button 7~33)

Web Admin.

SYSTEM DATA

- Music Sources ➤BGM Type, Internal Music Type
- Music Sources ➤SLTMOH 1-5
- Music Sources ➤VSF MOH2-30

Related Features

- MOH (Music-On-Hold)
- Internal/External & All Call Page

Hardware

- An external music source properly connected to the KSU or UCP, refer to the iPECS Hardwar Description and Installation Manual.

5.5 Call Log Display

Description

Using aniPECS Phone with 3-Soft buttons, you can view a log of incoming, outgoing and missed calls on the display. A Flex button must be assigned as a **{CALL LOG}** button, which allows easy access to the Call Log menu.

The call log is displayed according to the following admin.

Call Log usage

Most Phones show the call log. But the way to display the call log is different between System Serve and Phone self. This option is to select which way to display the call log.

The available phone type is LIP-8012D/8024D/8040L and LIP-9008/9010/9020/9030/9040 and 1010i/1020i/1030i/1040i/1050i.

The Call Log usage condition is as below:

- ✓ For color graphic phones (8050V/9070/9071,1080i), UCS client, and DECT phone: only 'Phone self' automatically regardless of this option.
- ✓ For other phone type except the above LIP and LDP phones : only 'System serve' automatically regardless of this option.

Call Log menu

This option select the call log menu for a call log type. The call log types have 4: Lost call, All call, Received call, and Dialed call.

If this option is ALL, display all calls. If this option is Type, display the call log 4 type and then select the desired type.

Answered By Group Member

One of the members can answer the call to the same ring group such as call coverage, linked pair, etc. At that time, the call log can set whether to leave the received call list or not by selecting Answered or Do-Not-Log.

But the call log is left in the received call list on the phone even though setting Do-Not-Log.

Internal Call Log

This option enables the call log to display 'Both (Incoming and Outgoing)' or 'Incoming only' or 'None'.

- **Both**: Incoming and outgoing call
- **Incoming only**: show Incoming call only
- **None**: don't leave the internal call as call log.

Operation

iPECS IP & LDP Phones

*To assign a **{CALL LOG}** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “57” + **[HOLD/SAVE]**

To access the Call Log menu

1) Press the **{CALL LOG}** button.

1. RECEIVED CALL
2. DIALED CALL
OK

2) Using the volume Up/Down keys, select the desired log (received, dialed, or lost).

123467890
24/07 14:51
BACKDELETEOK

Conditions

✓ The number of records in the Call Log is limited by the Call Log attribute in the System Data.

Programming

Web Admin.

STATION DATA

- CLI Attributes➤Call Log Usage, Call Log menu, Answered By Group Member, Internal Call Log

Hardware

- iPECS IP and LDP Phones with LCD

5.6 Call Profile Routing

Description

Call Profile Routing is an extension of the Station ICR feature. You can configure and assign up to ten (10) scenarios to each of the three (3) Call Profiles. Then you can activate a Call Profile to route your calls.

The system will attempt to route your incoming calls to the first destination that meets the scenario. If the destination does not answer in the Profile Timer, the call routes to the next destination.

Operation

iPECS IP & LDP Phones

To create a Call Profile Scenario

1) Press the **[TRANS/PGM]** button and dial “24”, the ICR menu code.
   Or Log in to the Station Web Portal.
2) Select the desired Scenario number (0 - 9).
3) Select the type of Caller ID (0 – 5):
4) Type 0 – Station CID
5) Type 1 - All Station
6) Type 2 – CO CID
7) Type 3 - All CO call
8) Type 4 - All call
9) Type 5 - N/A
10) Enter the Scenario Start and End Date (YYMMDD).
11) Select day-of-week using Flex button:
12) Flex button 1 (MON) – Flex button 7 (SUN)
13) Flex 8 button (Holiday)
14) Enter scenario Start Time and End time (HHMM).
15) Select Type of destination (0 – 6):
16) Type 0 – Station
17) Type 1 - Hunt group
18) Type 2 - CO Line Number
19) Type 3 - CO Group number
20) Type 4 - CO Loop
21) Type 5 - CO Transit
22) Type 6 - N/A
23) Select Scenario Priority (0 to 9) (0:highest priority).
24) Select Forward from Net call (0/1).
25) Select Call Profile Table Index (0-3) (0: not use CP).
26) Select Call Profile Timer (00–60 seconds).
27) Select Call Profile Apply Option: All call, Direct call (except CCR, DISA)

To activate ICR Call Forward

1) Lift the handset or press the **[SPEAKER]** button to receive a dial tone.
2) Press the [**FWD**] button.
3) Dial the desired Call Forward code (0-4).
4) Dial "*587", the Station ICR code.
5) Replace the handset, return to idle.

To activate/deactivate a Profile from your phone

1) Activate ICR Call Forward.
2) Press the **[TRANS/PGM]** button and dial "26", the Call Profile menu.
   Or Log in to the Station Web portal.
3) Select the Call Profile Table desired ("0": ICR or "1"-"3": Call Profile 1-3).

To change Call Profile while away from your phone

1) Call to iPECS System (DISA CO Line).
2) Dial "#" to access your Voice Mailbox.
3) Enter your Station Number and Authentication code.
4) Dial "7" to set call forward.
5) Enter "1" to set Station Call forward.
6) Enter "Station ICR Code".
7) Enter "Call Profile Number" (0:ICR Forward, 1-3: Call Profile 1-3).

Conditions

- ✓ Call Profile Routing is available on digital and IP Lines only and is applied to outside calls that are routed to your phone as a Transfer, DID or DISA call. Calls that are assigned to Ring to your phone are not subject Call Profile Forward.
- ✓ The conditions associated with Station ICR apply to Call Profile Routing.
- ✓ It is recommended hunt group is set last call profile destination.

Programming

Keyset Admin.

STATION

- Call Forward (PGM 111-Button 2)
- Station Call Forward No Answer Timer (PGM 123-Button 1)

SYSTEM

- System Call Forward No Answer Timer (PGM 181-Button 1)

Web Admin.

STATION DATA

- Common Attributes ➤Call Forward
- Station Timer ➤Station Forward No Answer Timer
- Station ICR Scenario ➤Call Profile Table Usage
- Station ICR Scenario ➤Caller ID, Time Condition

- Station ICR Scenario ➤Destination
- Station ICR Scenario ➤Scenario Priority
- Station ICR Scenario ➤Forwarding from NET Call
- Station ICR Scenario ➤Call Profile Table Index
- Station ICR Scenario ➤Call Profile Timer
- Station ICR Scenario ➤Call Profile Apply Option

SYSTEM DATA

- System Timers ➤Call Forward No Answer Timer

Related Features

- Call Forward
- Station ICR

Hardware

- iPECS IP or LDP Phone

5.7 Call Wait for Internal & External calls

Description

Internal/external calls can be waited without busy if there are available associated flexible buttons.

The indication of LED on button is blinking, the caller name/number is displayed on LCD, and Audible Muted(Off Hook) Ring (or Tone) will be heard .

Station (DSS) button can have Internal call waiting, CO (Grp.)/Loop button can have External call waiting, and specially 'Unified-LOOP(U-LOOP)' button can have both Internal and External calls waiting according to setting option; OFF, For external / Internal, For external, For Internal.

In case of external DID call, If you are busy on a call and you receive a DID call, the call will, as a default, follow the reroute assignments for the DID call destination (PGM 167). The system can be configured to queue the call to your station allowing you to end or hold your active call to answer the DID call. DID call will queue for the DID/DISA No-answer time then will route based on the reroute assignments.

The following table is overview of condition and operation for call waiting.

		1st condition	2nd condition	Max. waiting	Audible	LED	LCD
LIP/ LDP	Normal Incoming	P.113 Station Attributes {Call Wait= For External/Internal , For External, For Internal}	available empty U-LOOP buttons, or Station/CO(GRP) / (CO)LOOP buttons	U-LOOP(48), Other Button(as available Flexible Button)	Muted-Ring	**Green:** steady/held/recall **Amber:** Normal/Fast flashing Red: Normal/Fast flashing	Name /Number of Waiting Call for a few second, back to original
	Ring/ACD Group Call Queue Indication	P.191 Group Attribute Ring & ACD {Indication = ON}	With or Without Button	N/A			
	Auto Hold: Station Data > Common Attributes(111) > "Automatic Hold = OFF/ON" Auto Transfer: Station Data > Common Attributes(111) > "Auto Transfer By Button = OFF/ON"						
SLT	Normal Incoming	Call Wait = ON	Unconditionally	2 calls	Off Hook Signal Tone	NA	
	Camp-On	When busy after 2 normal calls		1 call			
	Pickup 2nd Incoming call, Switch between 2 picked-up Calls: Press [Hook Flash] 2 times continuously.						
	In case of pressing [Hook Flash] 1 time, it will make a new call and hang up, and then the call will be transferred to the new call or after connected to new call => Broker Call by 1 time of [Hook Flash] / Conference by Fast 2 times of [Hook Flash]						

Introduction of various types of call-queuing(incoming/making /talking/held call) buttons

1) DSS button
   - Assignment of a specific station number
   - Call-queuing from/to the specific station number
   - Presence of the station status
2) CO channel button
   - Assignment of a specific CO channel
   - Call-queuing for an external calling/receiving/held call via the specific CO channel
   - Presence of the CO channel status
3) CO-Group button
   - Assignment of a specific CO-group number
   - Call-queuing for an external calling/receiving/held call via the specific CO-Group
   - Presence of the queued call on the button
4) CO-LOOP button
   - Assignment of a special type 'LOOP' for non-specific/random/whole-range of CO channels
   - Call-queuing for an external calling/receiving/held call via the button
   - Presence of the queued call on the button
5) MSN-LOOP button
   - Assignment of a external DID telephone number
   - Call-routing of an incoming external call with called party number to stations who have MSN LOOP buttons with same DID telephone number as the incoming called party number
   - Call-queuing and presence of status of the incoming external call
   - Call-queuing and presence of status of an outgoing external call which was created on the button when make a call by pressing the button.
   - Multiple buttons assignment with same or different MSN telephone number
   - An option : MSN Wait = OFF/ON (second incoming MSN routing call allow = yes/no)
6) Unified-LOOP button
   - Assignment of coverage of my station number
   - Multiple buttons assignment
   - A button can have an receiving/making/held call for both type of internal and external call
   - DSS/CO/CO-Group/CO-LOOP/MSN-LOOP button goes first than Unified-LOOP button (An incoming call is presented on related DSS/CO/CO-Group/CO-LOOP/MSN-LOOP button at first, the call is presented on an Unified-LOOP button at second priority)

Do not need to enable "call-coverage-mode" feature for utilization of this button.

Operation

iPECS IP & LDP Phones-Keyset

*To assign a **{Call Wait}** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “34” + **[HOLD/SAVE]**

To activate/deactivate call wait

1) Press the **{Call Wait}** button.

2) Dial activate/deactivate code, "0: Off", "1:For External/Internal", "2:For External", or "3: For Internal" respectively.

*To assign a **{U-Loop}** button*

- [PGM] + flexible button + Call-coverage numbering plan + My station number.
  - Example) 671001 ('67' is call-coverage feature code, '1001' is my station number).

iPECS IP & LDP Phones-Web

To activate/deactivate call wait

✓ [Call Wait] = the following options (PGM113: Call Wait)
- **OFF**: Deactivate Call wait for all calls
- **For External/Internal**: Activate Call wait for External & Internal calls
- **For External**: Activate Call wait for External call
- **For Internal**: Activate call wait for Internal call

*To assign a **{U-Loop}** button*

- [Station Data] => [Flexible buttons] => select 'U-Loop' Type and click [SAVE] button.

Call Wait Flow

1) One Unified-LOOP(U-LOOP) button can keep one incoming/making/talking/held internal/external call.
2) An incoming call is acknowledged to be waited if there is an idle Unified-LOOP(U-LOOP) button. First call to a station is acknowledged even without an idle button. From second call, there need to be an idle U-LOOP or Station/CO(Grp.)/(CO)LOOP button to accept the incoming call without reject it to 'busy'.
3) To make a call, user can dial internal or external number with or without press of the button.
   If user presses a button to make a call then the button will be used for the call.
   If user does not press a button then an idle button is automatically seized for the call.
   - Make of internal call: [button] + dial station number
   - Make of external call: [button] + co access code + external number
4) Auto-Hold: a talking in a button is automatically sent to being held when press the other button. Station Data > Common Attributes(111) > Automatic Hold.
5) Auto-Transfer: can transfer a talking call to a new making call by press idle button to make a call (ex. Operation: user is talking a call => press an empty U-LOOP button and make a call => complete the transfer by on-hook). Station Data > Common Attributes(111) > Auto Transfer By Button.

Call Waite Indication

1) Audible Muted Ring (Continuous/One-Burst/Silence by PGM111 Station Data Programming >"Off-Hook Ring Type".
2) LCD Display
   - Display Caller Name & Number of Waiting Calls
   - Sequential Display : Talking Call => 1st waiting call (for 6 seconds) => 2nd waiting call (for 6 seconds) =>… => back to talking call if there is no more waiting call

3) LED Flashing
- Talking/Making : Green Steady
- Held : Green Normal-Flash
- Held Recall : Green Fast-Flash, or Red Fast-Flash to distinguish from 1st waiting' Red Normal-Flash
- Incoming Waiting :
  Single waiting : Amber Normal-Flash
  Multiple waiting : 1st waiting(Red Normal-Flash), from-2nd (Amber Normal-Flash)
- Differentiation of current LCD displayed Name &Number by LED: Amber Fast-Flash(for synchronization of Display and Flashing : Amber Fast-Flash button is displaying now).

SLT(Single Line Telephone)

1) For SLT, maximum 2 normal calls and additionally one camp-on call can be waited.
2) Off-Hook Signaling Tone is presented when call comes in during a talking.
3) Pick-up 2nd call during a talking state: Hook-Flash x 2 times of normal interval between Hook-Flashes. 1st Flash(transfer) + 2nd Flash(pickup waiting call or switch to the other call).
4) Switch calls between one talking and the other one waiting call: Hook-Flash x 2 times of normal interval between Hook-Flashes.
5) Camp-On Call : Can accept maximum 2 of normal incoming calls, and the 3rd call is returned to 'busy' and then caller can indicate 'Camp-On' by press '*'. The SLT(Single Line Telephone) user will hear Off-Hook signaling tone and pick up the camp-on call by 2 times of Hook-Flash on normal interval. User can switch between 3 calls (2 normal incoming and 1 camp-on call) by press 2 times of Hook-Flash.
6) Transfer a talking call : 1 time of Hook-Flash and then make a new call
7) Switch of calls between on transferred call and a connected call(made) : 1 time of Hook-Flash
8) Conference between SLT(Single Line Telephone), transferred and a connected call(made) call : 2 times of Hook-Flash on fast interval(within 2 or 3 seconds).
9) Recommended(by self-test) SLT(Single Line Telephone) Hook Flash Timer (P.182): SLT(Single Line Telephone) Hook Flash Bounce Timer(1*100ms) / SLT(Single Line Telephone) Maximum Hook Switch Flash Timer(5*100ms) / SLT(Single Line Telephone) Minimum Hook Flash Timer(7*10ms).

Conditions

- ✓ Other routing such as Preset Call Forward applies to DID calls even if DID Call Wait is active.
- ✓ External DID calls will follow the call routing defined in PGM 167 after the expiration of the DID/DISA no answer timer.
- ✓ An iPECS IP or LDP Phone must have an appearance button for the DID line.
- ✓ For external DID calls, Assigning the ICLID Timer, which enables ICLID routing, for a DID line, disables DID Call Wait.
- ✓ Combination with exist DSS/CO-Individual/CO-Group/CO-LOOP/MSN-LOOP buttons.

Priority.
- For internal call: DSS > Unified-LOOP.
- For external call: CO-Individual > MSN-LOOP > CO-Group > CO-LOOP > Unified-LOOP.

An incoming internal call.
- The call is acceptable if there is appropriate DSS button or an idle Unified-LOOP button.
- Via DSS button first if there is associated DSS button.

An incoming external call.
- The call is acceptable if there is an appropriate CO-Individual or idle MSN-LOOP or idle CO-Group or idle CO-LOOP or idle Unified-LOOP button.
- Via one of above buttons according to priority.

Making of internal call.
- In idle state, if dial a station number by dial pad.
  One of DSS button or Unified-LOOP button is seized automatically for the call.
- Seize DSS button and dial : the DSS button is seized.
- Seize Unified-LOOP button and dial.
  If there is DSS button, the DSS button is seized.
  Or, the pressed Unified-LOOP button is seized.

Making of internal call.
- In idle state, if dial CO-access-code and external number by dial pad.
  One of CO-Individual/CO-Group/CO-LOOP/Unified-LOOP button is seized automatically for the call.
- Seize CO-Individual/CO-Group/CO-LOOP button and dial : the CO-Individual/CO-Group/CO-LOOP button is seized.
- Seize Unified-LOOP button and dial.
  If there is CO-Individual/CO-Group/CO-LOOP button, the CO-Individual/CO-Group/CO-LOOP button is seized.
  Or, the pressed Unified-LOOP button is seized.

Auto-hold and auto-transfer are implemented also using previous DSS/CO-Individual/CO-Group/CO-LOOP/MSN-LOOP buttons.
- On internal/external talking state, the call is transferred to the right-next call of making by pressing of CO-Individual/CO-Group/CO-LOOP/MSN-LOOP/Unified-LOOP

Programming

Keyset Admin.

STATION
- Call Wait (PGM 113-Button 9)

CO/IP LINE
- ICLID Timer (PGM 142-Button 14)

SYSTEM
- DID/DISA Destination (PGM 167)
- DID/DISA No Answer Timer (PGM 181-Button 2)

Web Admin.

STATION DATA
- CLI Attributes ➤Call Wait

CO LINE DATA
- Analog Attributes ➤ICLID Ring Timer

SYSTEM DATA

- DID/DISA Destination
- System Timers ➤DID/DISA No Answer Timer

Related Features

- Direct Inward System Access (DISA)
- Direct Inward Dial (DID)
- ICLID Call Routing
- Intercom (ICM) Call

Hardware

- iPECS IP or LDP Phone

5.8 DND –One-Time DND

Description

While on a call, if you wish not to be interrupted by another call,pressing the DND button will activate One-time DND. For the duration of the active call your phone will be in DND. One-time DND can be activated while you are on a call and receiving ring for a new call. In this case, pressing the DND button will terminate the ringing and the call is routedbased on the following precedence:

- Previous or active Call Forward busy.
- Preset Call Forward busy.
- Station Call Coverage.
- Direct Transfer to Voice Mailbox.
- Return busy signal and disconnect.

When you complete your active call, DND is cancelled and the **[DND]** LED is extinguished.

Operation

iPECS IP & LDP Phones

To activate One-Time DND while ringing

1) Press the **[DND]** button, the **[DND]** LED lights, station goes to DND state.

System

Deactivation

1) When the station returns to idle, DND is disabled and the **[DND]** LED extinguishes.

Conditions

- ✓ Outside Line recalls will override One-Time DND.
- ✓ Your station must be permitted to activate DND to activate One0time DND.
- ✓ A station cannot reject a call from the Attendant using One-time DND.
- ✓ One-time DND is not available to the Attendant.
- ✓ One Time DND cancels existing Callback queues.

Programming

Keyset Admin.

STATION

- DND (PGM 111-Button 3)
- Call Forward (PGM 111-Button 2)
- Preset Call Forward (PGM 120)
- Direct Transfer to Mail box (PGM 120-Button 6)

SYSTEM

- System Call Forward No Answer Timer (PGM 181-Button 1)

Web Admin.

STATION DATA

- Common Attributes ➤DND, Call Forward
- Preset Call Forward➤ Transfer to Mail box

SYSTEM DATA

- System Timers ➤Call Forward No Answer Timer

Related Features

- Call Waiting/Camp-On
- DND (Do Not Disturb)
- Call Forward
- Call Forward, Preset
- Station Call Coverage

Hardware

- iPECS IP or LDP Phone

5.9 Flex Button Direct Speed Dial Assignment

Description

You may program a telephone number directly to a Flex button without the need to assign the number to a Speed Dial bin. In this case, the system automatically allocates the telephone number to the highest numbered unused Station Speed Dial bin.

Operation

iPECS IP & LDP Phones

To assign a telephone number to a Flex button

1) Press the **[TRANS/PGM]** button.
2) Press the desired Flex button.
3) Press the **[TEL NUM]**soft button.
4) Press the outside Line button or dial the Line access code.
5) Dial the telephone number.
6) Press the **[HOLD/SAVE]** button.
7) Dial the name to be associated with the number (optional).
8) Press the **[HOLD/SAVE]** button.

To place a call to the assigned telephone number

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the assigned Flex button.

Conditions

- ✓ When a Flex button is assigned with a telephone number, the system will allocate the number to the highest available Station Speed Dial bin. If no bin is available, you receive error tone when attempting to assign the telephone number.
- ✓ The telephone number may include any of the special Speed Dial instructions (**[MESSAGE/CALLBACK]**, **[FLASH]**, display security, etc.).

Related Features

- ■ Station Speed Dial

Hardware

- ■ iPECS IP and LDP Phone

5.10 Flexible LED Flash Rates

Description

The flash rates used with the various buttons on the iPECS IP and LDP Phones can be adjusted on a system wide basis to meet your needs. Up to 37 different functions can be assigned any one of 15 different flash rates.

Operation

System

System implements flash rates automatically based on database entries.

Conditions

✓ Available Flash rates and functions, which can be assigned, are given in iPECS Administration & Maintenance Manual.

Programming

Keyset Admin.

SYSTEM

- LED Flashing Rates (PGM 170-Button 1~36)

Web Admin.

SYSTEM DATA

- LED Flashing Rates

Hardware

■ iPECS IP or LDP Phone

5.11 Group Listening

Description

The iPECS IP and LDP Phones have a built-in speaker. If allowed, you may employ the speaker to monitor a call while using the handset to converse with the outside party.

This enables a group of people in the room to listen to both parties in the conversation.

Operation

iPECS IP & LDP Phones

While on a call using the handset

✓ Press the **[SPEAKER]** button, speaker activates, the speakerphone microphone will be muted while the handset is off-hook.

Conditions

✓ 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.

✓ With Group Listen active, pressing the **[MUTE]** button will mute the handset microphone. The connected party is still heard over both the handset receiver and the speaker.

✓ If full speakerphone operation is desired and permitted with Group Listen active, simply place the handset on-hook.

Programming

Keyset Admin.

STATION

- Group Listening (PGM 113-Button 3)

Web Admin.

STATION DATA

- Common Attributes ➤Group Listening

Related Features

- Speakerphone
- Mute

Hardware

- iPECS IP or LDP Phone

5.12 Intercom Signaling Mode

Description

You can set the signaling you receive when youriPECS IP and LDP Phonereceives an incoming ICM calls while the station is idle. There are three signaling modes available.

H - Call announce with Handsfree answerback

When an ICM call is received, youhear a brief splash of tone followed by the caller's voice. You may respond to the caller without the need to "Lift the handset" or "press the **[SPEAKER]** button".

P - Call announce with Privacy

When an ICM call is received, youreceivea brief splash of tone followed by the caller's voice. To respond you must lift the handset or press the **[SPEAKER]** button.

T - Tone ringing

An ICM call will cause your iPECS IP or LDP Phone to provide audible ICM ring tone. You must lift the handset or press the**[SPEAKER]** to answer the call.

Operation

iPECS IP & LDP Phones

To change ICM Signaling Mode

1) Press **[TRANS/PGM]** button, the **[SPEAKER]** button LED lights steady.
2) Dial the Station User Program code "13", confirmation tone is received.
3) Dial the desired ICM Signaling Mode code ("1" for H, "2" for T or "3" for P).
4) Press the **[HOLD/SAVE]** button.

Conditions

- ✓ Message Wait, Callback, Call Forward and Attendant Override will ring in the tone mode, regardless of ICM Signaling Mode you select.
- ✓ The ICM signaling Mode Selection does not affect Page announcements.
- ✓ The default ICM Signaling mode is Tone ring and the active mode is stored in battery-protected memory.
- ✓ A SLT(Single Line Telephone) always functions in the Tone ring mode.

Related Features

- Intercom Call (ICM Call)
- Paging
- Message Wait/Call Back
- Call Forward
- DND Override
- Intrusion

Hardware

- iPECS IP or LDP Phone

5.13 Microphone Mute

Description

You can turn off the microphone of your iPECS IP or LDP Phone, which will stop audio transmission from the handset, speakerphone or headset microphone, “Mic Mute”.

Operation

iPECS IP & LDP Phones

To Mute the Microphone

1) Press the **[MUTE]** button, the **[MUTE]** button LED is on and the microphone (Handset, Speakerphone, Headset) is muted; the connected party receives silence.

To activate the microphone

1) Press the illuminated **[MUTE]** button, the **[MUTE]** button LED is off, and the microphone is activated, transmitting audio to the connected party.

Conditions

- ✓ Changing from speakerphone to handset or vice versa during a mute condition will returns the microphone to an active status; your voice is transmitted to the connected party.
- ✓ Returning to idle or placing another outside Line or intercom call will change the mute status to its normal (active microphone) condition.

Programming

Keyset Admin.

STATION

- Headset Ring (PGM 111-Button 8)
- Speakerphone/Headset Selection (PGM 111-Button 9)

Web Admin.

STATION DATA

- Terminal Attributes ➤Headset Ring, Speaker phone, E&MIC Headset

Related Features

- Speakerphone
- Group Listening
- Headset Compatibility

Hardware

- iPECS IP or LDP Phone

5.14 Off-Hook Signaling

Description

When you receive a call while you are on an active call, you will receive the assigned Off-hook signals. For ICM calls, a Camp-On, Voice-Over Announcement or Off-hook ring signal may be received. For outside calls, you will receive a muted ring or a single tone burst. This signal is delivered to the iPECS IP or LDP Phone speaker.SLT(Single Line Telephone)s will only receive camp-on tones.

Operation

System

Operation of Off-hook ring signals is automatically controlled.

Conditions

- ✓ While using the speakerphone, a Camp-On tone is provided over the speaker in place of the assigned Off-hook ring Signal.
- ✓ Activating DND or One-Time DND places the station in DND, terminating any Off-hook signaling.
- ✓ Off-hook ring signals terminate when the call forwards or is answered, or abandoned.
- ✓ A station, which is receiving Off-hook ring signals, will receive normal ring upon return to idle status.

Programming

Keyset Admin.

CO/IP

- CO Line Ring Assignment (PGM 144)

SYSTEM

- Off-Hook Ring Type (PGM 161-Button 1)

Web Admin.

CO LINE DATA

- CO/IP Ring Assignment

STATION DATA

- Common Attributes ➤Off-Hook Ring Type

SYSTEM DATA

- System Attributes ➤Off-Hook Ring Type

Related Features

- Call Waiting/Camp-On, Outside Line Ring Assignment
- DND (Do Not Disturb), DND –One-Time DND

Hardware

- iPECS IP or LDP Phone

5.15 On-Hook Dialing

Description

Using your iPECS IP or LDP Phones equipped with a Speakerphone, you can place as well as receive calls while the handset is on-hook. With Automatic Speaker Select, you need not press the speaker button to initiate a call, simply dial the number or press a button to activate the Speakerphone is automatically.

Operation

iPECS IP & LDP Phones

To activate On-Hook Dialing

1) Place desired call (dial station ICM number, select a Line and dial, etc.) the **[SPEAKER]** button LED lights and the call is processed.

Conditions

- ✓ When you place a call in using On-hook dialing, you activate the Speakerphone and, you must press the illuminated **[SPEAKER]** button to return to idle.
- ✓ When the speakerphone is used, the microphone is active unless the [**MUTE**] button is pressed and the [**MUTE**] button LED is On.

Programming

Keyset Admin.

STATION

- Auto Speaker Selection (PGM 111-Button 1)

Web Admin.

STATION DATA

- Terminal Attributes ➤Auto Speaker Selection

Related Features

- Mute
- Speakerphone
- Automatic Speaker Select
- Headset Compatibility
- Group Listening

Hardware

- iPECS IP or LDP Phone

5.16 Outside Line Name Display

Description

When you receive or place and outside call, the Line number is displayed in the LCD. If the Line is assigned a name and Name display is enabled, the Line name is displayed instead of the number.

Operation

iPECS IP & LDP Phones

To display the Line name

✓ Press the **{LINE}** button, or dial a Line/Group access code, LCD displays the Line name.

Conditions

✓ Each outside Line is assigned a name of up to 12 characters in the system database.

Programming

Keyset Admin.

CO/IP

- CO Line Name Display (PGM 142-Button 1)
- CO Line Name Assign (PGM 142-Button 2)

Web Admin.

CO LINE DATA

- Common Attributes ➤CO Name Assign, CO/IP Name Display

Hardware

■ iPECS IP or LDP Phone w/Display

5.17 Prime Line Immediate/Delayed

Description

When you go to an off-hook state, the system normally provides ICM dial tone. If desired, a station can be assigned to access a different pre-assigned system resource (Prime Line).

The Prime Line can be any of the Idle Line Settings:

- Seize an outside Line
- Call another station
- Activate a Flex button feature
- Call to pre-assigned Hunt group
- Call to pre-assigned station Speed
- Call to pre-assigned system Speed

Prime Line access can be defined as immediate or delayed. When assigned immediate, upon an off-hook event, the system provides access to the Prime Line.

With Delayed Prime Line, you receive normal Intercom dial tone for the Prime Line Delay timer and after the delay, the Prime Line is accessed. This allows you to make a selection overriding the Prime Line.

Operation

iPECS IP & LDP Phones

To access the station's Prime Line

✓ Lift the handset or press **[SPEAKER]** button and take no action, Prime Line as assigned will be accessed.

Conditions

✓ Any of the station's Flex buttons may be assigned as the Prime Line. When you lift the handset or press the **[SPEAKER]** button, the system will act as if the Flex button had pre-selected the button prior to going off-hook.

✓ Selection of another Flex button or Feature button just prior to an off-hook event will override the Prime Line assignment.

✓ A SLT(Single Line Telephone) can be assigned an Idle Line Selection with a Speed Bin Number.

✓ When Delayed Prime Line is set, you must wait, taking no action until the Prime Line is accessed. You will receive ICM dial tone during this period and may dial any valid numbering plan digit(s) or select a Flex button or feature button.

✓ If the Prime Line Delay Timer is greater than Dial tone time-out, the Delayed Prime Line will not activate. It will be necessary to reduce the delay timer or extend the Dial tone time-out.

Programming

Keyset Admin.

STATION

- Prime Line setting (PGM 113-Button 7)
- Idle Line Selection (PGM 121)

SYSTEM

- Prime Line Delay Timer (PGM 182-Button 6)

Web Admin.

STATION DATA

- Common Attributes ➤Prime Line
  Idle Line Selection

SYSTEM DATA

- System Timers ➤Prime Line Delay Timer

Related Features

- Speakerphone
- Intercom Call (ICM Call)
- Station Flexible Buttons

Hardware

- iPECS IP or LDP Phone

5.18 Ring Tone Selection and Download

Description

You can select one of Ring tones so that the iPECS IP Phone ring can be distinguished from other nearby phones. Eight tones are stored in the iPECS IP Phone permanent memory. The first four tones are fixed and the 5th through 8th Ring tone can be downloaded from a library of 10 tones stored in the system's protected memory.After downloading the tone from the system memory, it can be selected as the Differential Ring Tone.

If using an LDP Phone, you can select one of 15 Ring tones stored in the phone.

Operation

iPECS IP Phone

To download a Ring Tone from the system memory

1) Press the **[TRANS/PGM]** button.
2) Dial 15, Ring Tone Download code.
3) Dial the desired Ring tone location, "5"~"8".
4) Dial digits "00" through "10" to hear the Ring Tones.
5) Press the **[HOLD/SAVE]** button.

To select the downloaded Ring Tone

1) Press the **[TRANS/PGM]** button.
2) Dial 11 for Intercom Ring tones, 12 for Line Ring tones.
3) Dial the digit ("5"~"8") to select the Ring Tone.
4) Press the **[HOLD/SAVE]** button.

LDP Phone

To select the Ring Tone

1) Press the **[TRANS/PGM]** button.
2) Dial 11 for Intercom Ring tones, 12 for Line Ring tones.
3) Dial the digit ("1"~"15") to select the Ring Tone.
4) Press the **[HOLD/SAVE]** button.

Conditions

✓ The downloadable ring tone files are stored in system memory as *.wav files with a maximum length of 4 seconds.

Related Features

- Station User Programming & Codes
- Differential Ring

Hardware

- iPECS IP or LDP Phone

5.19 Save Number Redial (SNR)

Description

The last dialed number on anoutside call may be stored (up to 48 digits) in a buffer for future redial. This number is saved in memory until you request a new number be stored. Numbers dialed for subsequent calls do not affect the Save Number buffer.

Operation

iPECS IP & LDP Phones

To save a dialed number, after dialing the number on an outside call

1) After dialing and before hanging up, press the **[SPEED]** button twice.

To save a dialed number using the LIP-8000/8000E series

1) After dialing and before hanging up, press the **[RIGHT NAVIGATION]** button.
2) Locate and press the **[HOLD/SAVE]** soft button.

To dial a saved number

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the **[SPEED]** button and Dial “#”.

To dial a saved number using the LIP-8000/8000Eseries

1) Press the **[DIR]** soft button.
2) Press the **[SPEED]** soft button and Dial “#”.

Conditions

- ✓ The saved number can be a maximum of 32 digits.
- ✓ Dialing the saved number will automatically seize the outside Line that was used for the original call. If the Line is busy, a Line from the same group will be selected and the saved number dialed. If all Lines from the group are busy, you will receive the All Lines busy tone and may queue.
- ✓ If you press the **[HOLD/SAVE]** button after seizing a Line without dialing, the Save Number Redial buffer will be erased.
- ✓ If there is no **{LINE}** button, the call will be presented on a **{POOL}**, or **{LOOP}** button.
- ✓ Save Number Redial is protected from power failure.
- ✓ Manually dialing a Flash during an outside Line will cause only those digits after the Flash to be stored and re-dialed as the Save Number Redial.

Related Features

- Station Speed Dial, System Speed Dial
- LNR (Last Number Redial), CO Line Flash

Hardware

- iPECS IP or LDP Phone

5.20 Serial DSS/BLF Console

Description

The iPECS IP & LDP Phones can be connected to a DSS/BLF Console to provide additional Flexible button for your use. This can be particularly useful for an Attendant or Secretarial answering position.

The Console is attached to the phone through a serial cable connection and thus does not use another station port on the system. For more information about DSS models except the below, ask your dealer.

- **LIP-8012DSS:**12 button Console with paper button designation label.
- **LIP-8012LSS:**12 button Console with LCD for button designation label.
- **LIP-8048DSS:**48 button Console with paper button designation label.
- **LIP-8040LSS:**40 button Console with LCD with Triple color LED.
- **LIP-9012DSS:**12 flexible buttons with tri-color LED. The DSS button kit uses a paper label for the button designations.
- **LIP-9024DSS:**24 flexible buttons with tri-color LED. The DSS button kit uses a paper label for the button designations.
- **LIP-9024LSS:**12 flexible buttons with tri-color LED. The LSS button kit uses an LCD for the button labels and has two flexible button pages to represent 24 total flexible buttons. The two lower buttons control the page view. The page button LED will flash when there is new activity under a button on the page. You can press the page button at any time to change the flexible button page view.
- **LIP-9048DSS:**48 flexible buttons with tri-color LED. The DSS button kit uses a paper label for the button designations.
- **LDP-7048DSS:**48 button Console with paper button designation label.
- **LDP-9048DSS:**48 button Console with paper button designation label.
- **LDP-9248DSS:**48 button Console with paper button designation label.
- **1048idss:**48 button Console with paper button designation label.
- **1048ilss:** 24 flexible buttons & page buttons. The LSS button kit uses an LCD for the button labels and has two flexible button pages to represent 48 total flexible buttons.

The DSS models employ a paper label strip to designate the function of each button. The LSS model employs an LCD to designate the button function.

The designation for a button on the LSS model employs standard designations such as STA xxx for a **{DSS/BLF}** button but can be modified in the system database to meet your needs.

With some restrictions you may equip your phone with up to four (4) Serial Consoles. For installation details, refer to the Hardware Description and Installation Manual or Quick Installation guide of Each phone.

Operation

iPECS IP & LDP Phone

Flex buttons on the Serial Console operate the same as the buttons on the phone

Conditions

- ✓ The maximum number of Serial Consoles supported in an eMG80/100 system is 100 and no more than 60 can be the LSS model and in eMG800/UCP system is 500 and no more than 200 can be the LSS Model.
- ✓ The iPECS IP Phone model LIP-8004 does not support a Serial DSS/BLF Console.
- ✓ The LIP-8048DSS must always be powered by external adaptor available from Ericsson-LG Enterprise.
- ✓ An iPECS IP Phone can only provide power to two Serial Consoles, the remaining Consoles must be powered by the external adaptor
- ✓ Linked pair secondary station cannot have a Serial Console.
- ✓ When configuring Flex buttons from the phone, the Serial Console Flex buttons are not used, use the **[VOL UP]**/**[VOL DOWN]** buttons to scroll. For example, when you press the **[VOL UP]** button the Flex buttons on the phone will represent the first buttons on the first Console. Note that even though a Console may have only 12 buttons, the system allocates 48. When programming Flexible button labels for an LSS, the1st Console starts from button 49, the 2nd starts from button 97, the 3rd Console starts from button 145, 4th starts from button 193.
- ✓ As a default, the 48 button Console is assigned with button Map 1 to 4 depending on the position of the Console. The 12 button Consoles are not assigned a map.
- ✓ If DSS configuration is changed and the phone restarts, programmed DSS data is initialized from the changed DSS.
- ✓ While idle, the **[VOL UP]**/**[VOL DWN]** buttons control the contrast of the LCD of both the iPECS IP Phone and the LSS model serial.
- ✓ Your station must be enabled to employ a Serial Console, if not enabled, any connected console will not function.
- ✓ The LIP Phones can also be associated with the standardLIP-8040LSS-40 button Console with LCD designations. A maximum of nine (9) such consoles can be associated with a station.
- ✓ LDP-7048DSS-48 button Console with paper button designation labels may be used with the LDP 7000 series phones. LDP-9048DSS-48 button Console with paper button designation labels may be used with an LDP 9000 series phone.
- ✓ EHS headset module connection way is same as serial DSS.
- ✓ If different type DSS is connected as before, then previous DSS configuration that is changed is deleted.
  ex 1 : 1048ilss + 1048ilss ➔ 1048ilss + 1048idss
  ex 2 : 1048ilss + 1048ilss ➔ 1048ilss+ ESHA headset module

Programming

Keyset Admin.

STATION

- Serial DSS Usage (PGM 111 –Button21)
- Flex Button Assign (PGM 115)
- Serial LSS Label Edit (PGM 129)

Web Admin.

STATION DATA

- Terminal Attributes ➤Serial DSS Usage
- Flexible Buttons➤ LSS Label

Hardware

- iPECS IP & LDP Phone with Serial DSS/BLF Console

5.21 Silent Text Message

Description

Silent Text Messaging is used to respond to aVoice Over call without disconnecting the existing call. Silent Text Messages are sent by pressing a pre-programmed message button or the**[DND]** button.

Operation

iPECS IP & LDP Phones

*To program a **{TEXT MESSAGE}** button*

- **[TRANS/PGM]** + **{FLEX}** + **[TRANS/PGM]** + “51” + Display Message code (“00”-“20”) + **[HOLD/SAVE]**

To respond to an Off-Hook Voice Over call with a Text message

1) Upon receiving the announcement, press the desired **{TEXT MESSAGE}** or the**[DND]** button, the text message is sent to the display of the calling party and the OHVO call is terminated.

Conditions

- ✓ Silent Text Messaging sends the DND display message, “STA IN DO NOT DISTURB” displays in the LCD of the calling Station.
- ✓ When Silent Text Messaging is used to respond to an OHVO call, the existing call will not be disconnected.
- ✓ When a SLT(Single Line Telephone) places the Voice Over call, the receiving station will receive the “NO DISPLAY AVAILABLE” message and error tone to indicate Silent Text Messaging is not available to the SLT(Single Line Telephone).

Programming

Keyset Admin.

STATION

- Voice Over (PGM 113-Button 6)

Web Admin.

STATION DATA

- Common Attributes ➤Voice Over

Related Features

- Call Waiting/Camp-On
- Group Listening
- Voice Over
- Pre-defined & Custom Text Display Messages

Hardware

- iPECS IP or & LDP Phone

5.22 Speakerphone

Description

Your iPECS IP or LDP Phone is equipped with speakerphone circuitry enabling the telephone to be used hands-free in two-way conversations.

Operation

iPECS IP & LDP Phones

To activate the speakerphone

1) Press the **[SPEAKER]** button, **[SPEAKER]** LED lights steady.

To switch from Handset to Speakerphone

1) Press the **[SPEAKER]** button, **[SPEAKER]** LED lights steady.
2) Replace Handset, Speakerphone activated.

To terminate a speakerphone call

1) Press the **[SPEAKER]** button, **[SPEAKER]** LED extinguishes.

Conditions

- ✓ If Automatic Speaker Select is enabled for the station, pressing a DSS, outside Line or Speed Dial button will automatically activate the speakerphone.
- ✓ The **[MUTE]** button LED indicates the status of the Microphone, when lit the Microphone is inactive.
- ✓ 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 is active. In order to activate the Speakerphone Microphone, the Handset must be placed on-hook.
- ✓ Each iPECS IP and LDP Phone equipped with Speakerphone is allowed/denied Speakerphone operation in the system database.
- ✓ 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.
- ✓ The LIP-8002 phones are not equipped for speakerphone operation but do support On-hook dialing.

Programming

Keyset Admin.

STATION

- Headset Ring (PGM 111-Button 8)
- Speakerphone/Headset (PGM 111-Button 9)

Web Admin.

STATION DATA

- Terminal Attributes ➤Headset Ring, Speakerphone, E&MIC Headset

Related Features

- Mute
- Group Listening
- Automatic Speaker Select
- Headset Compatibility

Hardware

- iPECS IP or LDP Phone

5.23 Station Flexible Buttons

Description

The iPECS IP and LDP Phones incorporate a field of "Flex" buttons as well as the fixed feature buttons.The Flex buttons are assigned in the system database to access features, functions and resources of the system and Telephone number. Specifically, Flex buttons can be assigned as:

- An Empty button has no system database assignment.
- A **{DSS/BLF}** button places One-touch ICM call to a designated station and display status of the station.
- A Flex Numbering Plan button activates the feature associated with the assigned digits from the Flexible Numbering Plan.
- A Speed Dial bin button accesses and dials the number from the assigned Speed Dial bin.
- A Pool button accesses an outside Line from the Line group designated for outgoing calls. A Pool button is used primarily to provide an outgoing call appearance when a direct Line appearance is not available. It may be used as the last choice for an incoming outside call. The Pool button LED provides the status for the duration of the call only.
- A Loop button provides an appearance for incoming any outside call when a direct Line appearance button is not available. The Loop button LED provides the status for the duration of the call only.
- A Station User Program Code button, access or activate the special features available with the Station User Program Codes, see section 5.26.
- An outside Line Appearance button provides access to the individual Line assigned to the Flex button. The Line button LED provides the status of the outside Line.
- Telephone number including CO access code.
- Number string.

With the exception of Line buttons, you can be assign the feature or function of the button from your iPECS IP or LDP Phone.. If allowed in the database, you can also assign or reassign Line buttons.

Operation

iPECS IP & LDP Phones

To assign a Flex button at the station

1) Press the **[TRANS/PGM]** button.
2) Press the desired Flex button.
3) Dial the digits from the Flexible Numbering Plan, or Any number.
4) Press the **[HOLD/SAVE]** button.

Or,

1) Press the **[TRANS/PGM]** button.
2) Press the desired Flex button.
3) Press the **[TRANS/PGM]** button.
4) Dial the digits from the Station User Program Code or Fixed Number Plan.
5) Press the **[HOLD/SAVE]** button.

Conditions

- ✓ The **{LOOP}** and **{POOL}** buttons provide a status indication for the call as long as the station has supervisory control.
- ✓ If there is no available **{LINE}**,**{LOOP}** or **{POOL}** button for a call, the station will not ring for an incoming outside call; transfers to the station immediately recall the initiator and recalls are routed directly to the Attendant.
- ✓ A station may have multiple **{LOOP}** and **{POOL}** buttons.
- ✓ When all outside Lines in a group are busy, any **{POOL}** buttons associate with the group will show busy, otherwise, **{POOL}** buttons will only display the status of an activate call.
- ✓ **{POOL}** buttons will access a Line from the designated group using the Round Robin or Last Used method as assigned in the database.
- ✓ The priority for the appearance of an outside call transfer is first a direct Line appearance **({LINE}** button), if not available a **{LOOP}** appearance is employed, if not available a **{POOL}** appearance is used. If there is no appearance available, the transferring station recalls immediately.
- ✓ Station User Program Codes are defined in section 5.25.

Programming

Keyset Admin.

NUMBERING

- Flexible Numbering Plan (PGM 105~109)

STATION

- Flexible Button Assignment (PGM 115)

Web Admin.

SYSTEM ID & NUMBERING PLANS

- Flexible Numbering Plan

STATION DATA

- Flexible Buttons

Related Features

- Flexible Numbering Plan
- Station User Programming & Codes

Hardware

- iPECS IP or LDP Phone

5.24 Station Individual Call Routing (ICR)

Description

Station ICR is an extension of Call Forward that permits you to establish scenarios to route your incoming calls. Each of the ten scenarios defines rules to route incoming calls based on Time, Day of week, Date and Caller ID to your desired destination.

Each scenario is assigned a priority of 0 to 9. When you receive an incoming call, the System will search the ICR scenarios, then route the call according to the destination in the highest priority matching scenario. If no matches are found the call normal routing assigned for the call.

Scenarios may be established from the Station Web portal, contact your Ericsson-LG Enterprise representative for details, or scenarios can be configured from an iPECS IP or LDP Phone.

Operation

iPECS IP & LDP Phones

To create a scenario

1) Press the **[TRANS/PGM]** button and dial "24", the ICR menu.
   Or Log in to the Station Web portal.
2) Select the desired Scenario (0 - 9).
3) Select the type of Caller ID (0 – 5):
4) Type 0 – Station CID
5) Type 1 - All Station
6) Type 2 – CO CID
7) Type 3 - All CO call
8) Type 4 - All call
9) Type 5 - N/A
10) Enter the Scenario Start and End Date (YYMMDD).
11) Select the Day-of-week using the Flex buttons:
12) Flex button 1 is Monday – Flex button 7 is Sunday
13) Flex 8 button is for holidays
14) Enter scenario Start and End time in 24-hour format (HHMM).
15) Select Type of destination (0 – 6):
16) Type 0 – Station
17) Type 1 - Hunt group
18) Type 2 - CO Line Number
19) Type 3 - CO Group number
20) Type 4 - CO Loop
21) Type 5 - CO Transit
22) Type 6 - N/A
23) Enter the specific Station, Hunt group, Line number, etc.
24) Select Scenario Priority (0 to 9).

25) Press the **[HOLD/SAVE]** button to store the scenario.

To activate ICR Call Forward

1) Lift the handset or press the **[SPEAKER]** button to receive a dial tone.
2) Press the **[FWD]** button.
3) Dial the desired Call Forward code (0-4).
4) Dial “587”, the Station ICR code.
5) Replace the handset, return to idle.

Conditions

- ✓ Any station allowed access may define scenarios in the station’s Web portal.
- ✓ The conditions as listed under Call Forward apply.

Programming

Keyset Admin.

STATION

- Call Forward (PGM 111-Button 2)
- Station Call Forward No Answer Timer (PGM 123-Button 1)

SYSTEM

- System Call Forward No Answer Timer (PGM 181-Button 1)

Web Admin.

STATION DATA

- Common Attributes➤Call Forward
- Station Timer ➤Station Call Forward No Answer Timer

SYSTEM DATA

- System Timers ➤Call Forward No Answer Timer

Related Features

- Call Forward

Hardware

- iPECS IP or LDP Phone

5.25 Station User Programming& Codes

Description

You can program an array of functions and features, access status information and assign special features codes to Flex buttons of your iPECS IP or LDP Phone. The Station User Program Codes used for these purposes are fixed as listed below.

Code	Description	Code	Description
11X	Intercom Differential Ring	82	{COLR} Button Assignment
12X	CO Line Differential Ring	83	{ATD DND} button assignment
13	Intercom Answer Mode (1: HF/2: TONE/3: PV)	84	{Account Code} Button Assignment
14X	Call Coverage Attribute Setting	85	{LOOP} Button Assignment
15X	Station Ring Download	86	{ATD Intrusion} Button Assignment
21	Knock Down Station COS	87	{INTERCOM} Button Assignment
22	Restore Station COS	88	{Camp-on} Button Assignment
23	Walking COS	89	{Send Keypad Facility IE} Button Assignment
30	VM Mobile Notify	8#	{OHVO} Button Assignment
31	Message Retrieve Method	8*1	DID Restiction Button Assignment
32	Message Retrieve Example	8*2	DISA Restiction Button Assignment
		8*3	Bomb Threat History Button Assignment
33	User Authorization Code Registration	8*5	Headset Button Assignment
34	DID Call Wait	8*6XYZA	Toggle Ring Mode Button Assignment X,Y,Z,A : 1(Day), 2(Night), 3(Timed), 4(Auto)
35	Message Wait in Executive/Secretary pair		
36	Send SMS Message	90	{SPEED} Button Assignment
37	Register Mobile Extension	91	{CONFERENCE} button assignment
38	Make Mobile Extension active	92	{CALLBACK} button assignment
39	Register Mobile Extension CLI	93	{DND} button assignment
41	Set Wake-Up Time	94	{FLASH} button assignment
42	Wake-Up Time Disable	95	{MUTE} button assignment
51XX	Custom/Pre-select Message Display (XX = 00-20)	96	{MONITOR} button assignment
52	Register Custom Message (Message 00)	97	{REDIAL} button assignment
53	Active Conference Room	98	{CALL FORWARD} button assignment
54	Deactive Conference Room	99	{PTT} button assignment
551	CONF Group Monitor	9*XX	{In-Room Indication} button assignment (XX = 01-10)
552	CONF Group Active	*6XXX	Set Forced Fwd Dest (XXX : hunt group no.)
553	CONF Group Deactive	*7XXX	Forced FWD To Dest (XXX : hunt group no.)
57	{Call Log Display} Button Assignment	*8	Register Bluetooth
61	Headset/Speakerphone Mode	*9	Bluetooth Usage
62	Change Ring Mode	*0	Hotdesk Login
71	LCD Display Mode	**	Hotdesk Logout
72	Version Display		
73	Background Music		
74	Station User Name Registration		
75	Display Phone IP Address		
76	Change Phone IP Address		
77	Display Phone MAC Address		
78	Network Config		
79	Display Phone Version		
7#	Forced Release Button Assignment		
80	{Record} Button Assignment - With Voice Mail		
81	{CLIR} Button Assignment		

In addition, a Station User Program Menu display is provided by the iPECS IP and LDP Phone display to assist you during programming 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. The codes are also used to assign a function or feature to a Flex button.The Station User Program Menu displaysare illustrated below.

First top-level Menu selection

[1] RING/NEWKEYSET [2] COS

Under selection [1] Ring, select 1~5 as below

[1] STA RING TYPE [2] CO RING TYPE

[3] ANSWER MODE [4] CALL COVER ATTR

[5] STA RING DOWNLOAD [9] EAR-MIC HEADSET

[0] ENBLOCK MODE [9] STA RING TYPE

Under selection [2] COS (Class of Service), select 1~3 as below

[1] COS DOWN [2] COS RESTORE

[3] WALKING COS [4] ICR SCENARIO

[5] LIP KEYSET STAT [6] CALL PROFILE

Next top-level Menu selection

[3] MSGRET/MOBILE-EXT [4] WAKE UP TIME

Under selection [3] MESSAGE RETRIEVE, select 1~8 as below

[1] MESSAGE RETRIEVE METHOD [2] MESSAGE RETRIEVE EXAMPLE

[3] AUTH REGISTER [4] DID DISA CALL WAIT

[5] CHOICE EXEC/SEC MSG [6] SEND BRIEF SMS MESSAGE

[7] REG MOBILE-EXT [8] ACTIVE MOBILE-EXT

[9] REGISTER MOBILE-CLI
[0] VM MOBILE NOTIFY

Under selection [4] WAKE UP TIME select 1~2 as below

[1] SET WAKE UP TIME
[2] WAKE UP DISABLE

Next top-level Menu selection

[5] MESSAGE
[6] HEADSET

Under selection [5] MESSAGE, select 1~2 as below

[1] SET PRE : CUST MSG
[2] PGM CUSTOM MSG

[3] ACTIVE CONF-ROOM
[4] DEACTIVE CONF-ROOM

[5] CONF-GROUP
[6] USER GREETING

Under selection [6] HEADSET select 1~2 as below

[1] HEADSET OR SPK MODE
[2] HEADSET RING MODE

Next top-level Menu selection

[7] SUPPLEMENTARY
[*] SYSTEM

Under selection [7] SUPPLEMENTARY, select 0~9 as below

[1] LCD DISPLAY LANGUAGE
[2] MFIM VERSION

[3] BGM
[4] REGISTER STA NAME

[5] DISP PHONE IP ADDR
[6] CHANGE PHONE IP ADDR

[7] DISPLAY MAC ADDR
[8] NETWORK CONFIG

[9] DISP PHONE VERSION
[*] DISP ADD-ON PACKAGE

Under selection [*] SYSTEM, access Admin

[#] ENTER ADMIN [6] SET FORCED FWD DEST

[7] FORCED FWD TO DEST [8] REGISTER BLUETOOTH

[9] BLUETOOTH USAGE [0] HOTDESK LOGIN

[*] HOTDESK LOGOUT [#] ENTER ADMIN

Operation

iPECS IP & LDP Phones

To assign a Station User Program Code to a Flex button

1) Press the **[TRANS/PGM]** button, the Station User Program Menu is displayed.
2) Press the desired Flex button.
3) Dial the desired Station User Program Code and additional inputs that may be required.
4) Press the **[HOLD/SAVE]** button.

To activate a Station User Program Code feature or function

1) Press the **[TRANS/PGM]** button, the Station User Program Menu is displayed.
2) If desired, use the **[VOL UP]/[VOL DOWN]** to display the desired menu item, or dial the desired Station User Program Code with additional inputs as required.

Related Features

- Station Flexible Buttons
- Temporary Station COS (Class of Service)/Lock
- Walking COS (Class of Service)
- Station Message Wait/Call Back
- CLI Message Wait
- Wake-Up Alarm
- Pre-defined & Custom Text Display Messages
- Headset Compatibility
- Attendant Station Program Codes

Hardware

- iPECS IP or LDP Phone w/Display

5.26 Two-Way Record

Description

With your iPECS IP or LDP Phoneyou can record any active conversation to your Voice Mail mailbox or to the hard disk drive of an iPECS UCS Client.

All calls including incoming, outgoing, internal, external, conference, conference rooms and conference group calls can be recorded. A **{RECORD}** button must be assigned to access this feature.

When the recording destination is an iPECS IPCR server, recording can be paused(Secret)Flex button andthe Recording Announcement can be activated using configured Flexible buttons.

Operation

iPECS IP & LDP Phones

*To assign a flexible button as a **{RECORD}** button*

- **[TRANS/PGM]** + **{FLEX}** + **[TRANS/PGM]** + “80” + recording destination (VM Group, UCS Client, or recording server) + **[HOLD/SAVE]**

*To assign a **{Secret}** button*

- **[TRANS/PGM]** + **{FLEX}** + **[TRANS/PGM]** + “80” + recording destination (IPCR server group) + “1” + **[HOLD/SAVE]**

*To assign a **{Announce}** button*

- **[TRANS/PGM]** + **{FLEX}** + **[TRANS/PGM]** + “80” + recording destination (IPCR server group) +”2” + **[HOLD/SAVE]**

To activate Two-Way Record while on a call

1) Press the **{RECORD}** button, record warning tone heard, and recording starts.

To mute Two-Way Record while on a call

1) Press the **{Secret}** button, recording and tones stop.
2) Press the **{Secret}** button to restart recording.

To stop Two-Way Record while on a call

1) Press the **{RECORD}** again.
   Or, Hang-up, return to idle.

To manage the recordings

1) Use the procedures for the voice Mail or as outlined in UCS Client User Guide, the iPECS IPCR User Guide or the third party guide, as appropriate.

Conditions

- ✓ The **{RECORD}** button LED will flash at 240 ipm while recording.
- ✓ This feature is available when using the integrated VM, the Feature Server or using an external AA/VM, which employs the SMDI communications mode. When an external AA/VM system employs the in-band (DTMF) mode, Two-Way Record is not available.

- ✓ With the Feature Server or integrated VM, internal calls can be recorded as well as external calls.
- ✓ Use of this feature when the Two-Way Recording Warn Tone is disabled may be interpreted as a violation of federal, state, or local laws, and an invasion of privacy. Check applicable laws in your area before recording calls using this feature.
- ✓ If a destination for the recording is not defined for the **{RECORD}** button, the Call Record Destination defined in PGM 112 is employed.
- ✓ On-demand recording using a Voice Mailbox or hard drive records the call from the point in time when the **{Record}** button is pressed. When using the IPCR server, the entire call will be record as long as the **{Record}** button is pressed before the call is terminated.
- ✓ The UCS Client can record one call at a time and must be idle. While recording, if the UCS Client places or receives a call, recording terminates.
- ✓ When call recording begins, you and the connected party will receive a Call Recording confirmation tone and the tone may repeat at intervals.
- ✓ The UCS Client used for recording must be operating in a 'Local' mode not remote.
- ✓ A third party SIP Voice Mail or recording server may be used to record calls. In this case, the Recording Destination should be assigned with the Station group of the SIP server.
- ✓ When you activate call recording to an iPECS IPCR the entire call is recorded as long as the call is active when you activate recording. It would be expected that other SIP recording servers would provide similar On-demand recording.
- ✓ The IPCR Agent Table in the iPECS UCP must be configured when using the IPCR application or a third-party SIP recording server.
- ✓ Call Tracking Number는 "tracking=xxxxx" xxxxx is tracking number, unique number for a call until it's released. It's followed as the specification of "iPECS SIP Interface Spec for VM_UMS_V1.2.docx".
- ✓ If there is no MCIM Channel, phone will display "NO MCIM CHANNEL".
- ✓ MCIM Link channels reserved for linking MCIM. MCIM link channels used one in case of two more MCIM boards. But it's not used when it's two way recording.

Programming

Keyset Admin.

STATION

- Two-way Record Privilege: ON (PGM 112-Button 11)
- Call Record Destination (PGM 112-Button 21)
- Auto Call Record (PGM 112-Button 20)

SYSTEM

- Two-way Record Warning Tone (PGM 161-Button 19)

Web Admin.

STATION DATA

- VM Attributes ➤Two Way Record, Automatic Talk Recording Option, Automatic Talk Recording Destination

CO LINE DATA

- Common Attributes ➤Auto Call Recording Destination

SYSTEM DATA

- System Attributes ➤Record Warning Tone

Related Features

- External Auto Attendant/Voice Mail
- Integrated Auto Attendant/Voice Mail
- Conference
- Conference Group
- Multi-Party Voice Conference
- Conference Room
- Auto Call Recording

Hardware

- iPECS IP or LDP Phone
- Integrated VM, IPCR, Feature Server or External SMDI based AA/VM system
- UCS Client Softphones

5.27 Voice Over

Description

Using an iPECS IP or LDP Phone while you are off-hook on a call (internal or outside), you may receive a voice announcement through the handset receiver with the existing call.

The Off-Hook Voice Over (OHVO) is muted so as not to interfere with the existing conversation. You may respond to theVoice Over calling party using Camp-On response or may use Silent Text Messaging to respond.

Operation

iPECS IP & LDP Phones

Placing a Voice Over (OHVO) while receiving busy

1) Dial “#” or press a pre-programmed **{OHVO}** button.
2) After splash tone, begin announcement.

Responding to a Voice Over announcement

1) Use Camp-On response procedure or Silent Text Messaging or One-Time DND.

SLT(Single Line Telephone)

Placing a Voice Over (OHVO) while receiving busy

1) Dial “#”.
2) After splash tone, begin announcement.

Responding to a Voice Over announcement

1) Use Camp-On response procedure.

Conditions

- ✓ When you respond to a Voice Over via Camp-On, all conditions and options available to Camp-On apply.
- ✓ Voice Over may be used to notify you of a transferred call (Intercom or outside) by announcing the call then releasing to complete the transfer.
- ✓ When a call is transferred with Voice Over, you will receive muted ringing after the transfer is complete.
- ✓ The Pre-defined or Custom Text Display Message feature may be used in response to a Voice Over announcement and the conditions associated with these features apply.
- ✓ Both the originating and receiving station must be programmed to allow OHVO calls.
- ✓ When Silent Text Messaging is used to respond to a Voice over call, the existing call will not be disconnected.
- ✓ If you are in conference, using the Speakerphone or Group Listen, Voice Over is not available. Camp-On will be activated and a Camp-On tone sent to your phone.

Programming

Keyset Admin.

STATION

- Voice Over (PGM 113-Button 6)

Web Admin.

STATION DATA

- Common Attributes ➤Voice Over

Related Features

- Call Waiting/Camp-On
- Group Listening
- Silent Text Message
- Pre-defined & Custom Text Display Messages

Hardware

- iPECS IP or LDP Phone to receive Voice Over

5.28 Phone Lock/Unlock

Description

For What ?

1) Protect a phone from normal call features and allow only making of emergent call features.
2) Personal protect from unauthorized touch. Owner can protect personal phone from unauthorized visitor during absence.
3) Hot-Desk login/logout from/to a locked dummy station

Supported Features in 'Locked' state

1) Unlock : clear protection and return to normal mode
2) Make a pre-programmed flexible page call
3) Make an emergency call to external emergency center
4) Answer, Hold & Retrieve an incoming call (Can't transfer the call)

Protected Features in 'Locked' state

1) Protected Features in 'Locked' state
2) Almost all of call features, numbering, buttons supported in normal mode
3) With protected action, Phone goes to auto idle and there will not be error tone or other response

Operation

Phone Lock/Unlock Access

Numbering

1) Flexible Numbering Plan (P.106-109) > Phone Lock/Unlock (ex. 47*)

One-Touch Button with Locked/Unlocked Indication by LED Flashing

1) Steady RED (Locked) / Flash OFF (Unlocked)
2) [Web]
   Station Data > Flexible Buttons(P.115)
   Type = "Programming (Numbering Plan)"
   Input = Phone Lock/Unlock code digits (ex. 47*)
3) [Phone]
   [PGM] + [Flex Button] + Phone Lock/Unlock code digits + [SAVE]

Soft-Menu in Locked mode

UNLOCK PAGE CALL

Automatic Phone Lock

1) Terminal Attributes (P.112) > Phone Auto Lock Usage, Phone Auto Lock Timer

Phone Lock/Unlock Process

Lock

1) from Unlocked (normal mode)

2) Phone Lock/Unlock Code(or Button)
3) Authorization (Station Password Only)
4) Changed To Locked

Unlock

1) from Locked
2) Phone Lock/Unlock Code(or Button)
3) Authorization (Station Number & Password)
4) Changed To Unlocked

Phone Lock/Unlock for Hot-Desk

Hot-Desk Login - Using Unlock

1) from a Locked (Dummy) Station
2) Phone Lock/Unlock Code(or Button)
3) Authorization (Station Number & Password of a Hot-Desk)
4) Login To a Hot-Desk

Hot-Desk Logout – Using Lock

1) from a Hot-Desk Station
2) Phone Lock/Unlock Code(or Button)
3) Authorization (Station Password of the Hot-Desk Station)
4) Option 1 - Station Data > Terminal Attributes (P.112) > PHONE Lock Action for Hot-Desk =“Just Lock Phone “ : Locked Hot-Desk (remain in the Hot-Desk Station in Locked)
   Or, Option 2 : Station Data > Terminal Attributes (P.112) > PHONE Lock Action for Hot-Desk =“Logout from Hot-Desk “ : Logout To Locked (Dummy) Station

Supported Functions in Locked mode

Unlock

1) Phone Lock/Unlock Numbering Code
2) Or, One-Touch Flex Button
3) Or, Soft-Menu “UNLOCK” Button

Page

1) Flexible Page Numbering Code
2) Or, One-Touch Flex Button
3) Or, Soft-Menu “PAGE” Button

Call

1) Dial Emergency Code. Or, Soft-Menu “CALL” Button

Forced Lock/Unlock by Administrator

Station Data > Terminal Attributes(P.112) > [PHONE Lock Status(Force to Set)]

1) Unlock : force to Unlock
2) Lock : force Lock

Automatic Phone Lock

Station Data > Terminal Attributes(P.112) > PHONE Auto Lock Usage

1) Do Not Apply : do not apply auto lock
2) After Duration of Idle : if there is no action by user during "PHONE Auto Lock Timer"
3) After Duration of Idle : if there is no action by user during "PHONE Auto Lock Timer"

Station Data > Terminal Attributes(P.112) > PHONE Auto Lock Timer

1) 0 - 1430 (minutes)
2) Lock is implemented in <Timer minutes + Extra Seconds (00 - 59 seconds)>
   Example)
   PHONE Auto Lock Timer = 3 (minutes)
   Phone will go to locked in (3 minutes + n seconds; n = 00-59)

Phone Lock/Unlock Supported Phone Type

Supported Phones

1) LIP/LDP Desk Top Phones Only
2) Except UCS, WIT, DECT, IPATD

Conditions

Supported Phones

✓ LIP/LDP Desk Top Phones Only
  Except UCS, WIT, DECT, IPATD, SLT(Single Line Telephone) types

Programming

Web Admin.

SYSTEM ID & NUMBERING PLANS

- Flexible Numbering Plan (P.106-109) ➤ Phone Lock/Unlock

STATION DATA

- Terminal Attributes(112) ➤ Common Attributes ➤
- *** PHONE Lock Status(Force to Set),
- PHONE Auto Lock Usage,
- PHONE Auto Lock Timer,
- PHONE Lock Action for Hot Desk

Related Features

- Authorization Code
- Paging, Emergency Paging
- Emergency Call

Hardware

- iPECS IP or LDP Phone (Desk Top Phones)

6 Attendants

6.1 Active CPU (UCP) Display for UCP

Description

When the system is equipped for CPU Redundancy, the System Attendant can determine andchange the active and standby CPU.

System Attendant

To assign a Flex button for Active CPU Display

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + "078" + **[HOLD/SAVE]**

To view the active UCP

1) Press the **[TRANS/PGM]** button.
2) Dial "078" the Active CPU Display code, the active CPU is displayed.

Or,

1) Press the **{Active CPU Display}** Flex button to display the active CPU.

To Change Standby UCP to Active UCP

1) Press the **[TRANS/PGM]** button.
2) Dial "078" the Active CPU Display code. Or, press the **{Active CPU Display}** Flex button.
3) Dial "#" to change the standby iPECS UCP module to active.

Conditions

- ✓ The system must be equipped with and properly programmed in Redundancy Data for Redundant Processor operation and for the Attendant to display or change the active UCP.
- ✓ The System Attendant can assign a Flex button to indicate when the back-up iPECS UCP module is active. When assigned, the LED of the Flex button will turn ON indicating the back-up UCP is active.

Programming

REDUNDANCY DATA

- Redundancy Attributes ➤CPU Redundancy Usage

Related Features

- ■ Redundant System Processor

Hardware

6.2 Alternate Attendant

Description

This feature allows an Alternate answer point while the Attendant station is in an unavailablemode. When in the unavailable mode, the next available Attendant in the Attendant group willreceive Attendant calls and recalls.

Operation

Attendant

*To assign a flexible button to activate **{ALT ATD}** button*

- **[TRANS/PGM]** + {FLEX} + "*562" + **[HOLD/SAVE]**

To toggle the Attendant Unavailable feature

1) Dial "*562", the Attendant Unavailable code or press **{ALT ATD}** button.

Conditions

- ✓ Alternate Attendant activates the DND feature at the Attendant station and affects all calls to the Attendant station.
- ✓ A Flex button can be assigned to activate Alternate Attendant. The **{ALTATD}** button LED indicates the status of the Alternate Attendant feature, ON indicates the Attendant is unavailable.
- ✓ A station, which is receiving calls forwarded from the System Attendant, cannot use the Alternate Attendant feature.
- ✓ All except for one Attendant can activate Alternate Attendant. When the last Attendant at tempts to activate this feature, error tone is received.
- ✓ An Attendant forwarded to an unavailable Attendant is also considered to be in the unavailable Attendant mode.

Programming

Keyset Admin.

SYSTEM ID & NUMBERING PLAN

- Flexible Numbering Plan (PGM 106-19 Attendant Unavailable)

Web Admin.

SYSTEM ID & NUMBERING PLAN

- Flexible Numbering Plan➤ Attendant Unavailable

6.3 Attendant Call & Call Queuing

Description

Any station can call the Attendant by dialing the Attendant Call code “0”. When an Attendant call experiences a busy, 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 “0”, the Attendant Call Code.

Condition

- ✓ Call routing order follows the order of entry in the Attendant Assignments program.
- ✓ The Attendant is informed of a queued Attendant call by the **[HOLD/SAVE]** button LED flashing. The calling intercom party will receive ring-back tone or MOH, as configured in the system.
- ✓ Calls to the Attendant’s station intercom number are sent to the station dialed as with any intercom call.
- ✓ When an Attendant calls another busy Attendant by dialing the station number, busy tone is received and camp-on is available.

Programming

Keyset Admin.

SYSTEM

- Attendant Assignment (PGM 164)
- MOH Type (PGM 171-Button 2)
- Attendant Call Queuing Ring back Tone (PGM 160-Button 1

Web Admin.

SYSTEM DATA

- Attendant Assignment
- Music Sources ➤MOH Type
- System Attributes ➤Attendant Call Queued Ring Back Tone

Related Features

- Attendant Positions
- Intercom Call (ICM Call)

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.4 Attendant Positions

Description

The iPECS eMG80/100 system hasthe capacity for 4 Attendants (1 System Attendant / 3 Main Attendants), iPECS eMG800 System has the capacity for 5 Attendants (1 System Attendant / 4 Main Attendants), and iPECS UCP system hasthe capacity for 50 Attendants (1 System Attendant/49 Main Attendants).

Attendants have enhanced call and control capabilities and are designated to receive dial '0' calls from other users.

There are 2 different types of Attendant as follows,

- System Attendant - The first station in the Attendant Group. In addition to Main Attendant features, the System Attendant is commonly used for System Attendant Programming, Admin Programming and Day/Night/Timed Mode control.
- Main Attendant – Attendant group stations other than the first station of the Attendant Group.

Condition

- ✓ As a default, the first Attendant (System Attendant) is assigned as Station 100 (logical number) for eMG80/100 and Station 1000 for eMG800, and others are not assigned. In the eMG, the first digital phone port is designated as the System Attendant and is Station number 100 for eMG80/100 and 1000 for eMG800.
- ✓ As a default, the first Attendant (System Attendant) is assigned as Station 1000 (logical number), and others are not assigned. In the UCP, the first digital phone port is designated as the System Attendant and is Station number 1000.
- ✓ Attendant calls (dialing "0") and recalls are routed to first available attendant.

Programming

Keyset Admin.

SYSTEM

- System Attendant Assignment (PGM 164-Button 1)
- Main Attendants Assignment (PGM 164-Button 2~4)

Web Admin.

SYSTEM DATA

- Attendant Assignment ➤System Attendant, Main Attendant

Hardware

- ■ iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.5 Attendant Recall

Description

Unanswered or abandoned outside calls that remain unanswered for the Hold or Transfer Hold timer, as appropriate, will recall the station placing the call on hold. If the call remains unanswered for the assigned Recall time, the first available Attendant will also receive recall.

The Attendant and station will receive the recall signal for the Attendant Recall Timer period after which the system will disconnect and return the outsideLine to idle.

Operation

System

Attendant recall operation is automatic.

Programming

Keyset Admin.

SYSTEM

- Attendant Recall Timer (PGM 180-Button 1)
- ICM Hold Recall Timer (PGM 180-Button 5)

Web Admin.

SYSTEM DATA

- System Timers ➤Attendant Recall Timer, I-Hold Recall Timer

Related Features

- Hold
- Call Transfer

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.6 Attendant Station User Program Codes

Description

Using the Attendant Station User Program Codes, the Attendant can print SMDR and Traffic reports on-demand, assign Authorization Codes, control certain user features, record System announcements, enable/disable Auto Service Mode Control, etc.

Items are available using the Program Code directly or scrolling the multi-level display menu. The following indicates the menu displays, including the digit for selecting the item, the item description and further required entries. The various levels of the display menu are indicated by indentation. For additional information, refer to Appendix of the iPECS Administration& Programming Manual.

Note also, some Program Codes are only available to the System Attendant or stations allowed access to the Attendant Program code “0”.

[1] PRINT

SMDR

- [1] PRINT SMDR STA BASE (station range input)
- [2] DELETE STATION BASE (station range input)
- [3] DISPLY CALL CHARGE
- [4] ABORT PRINTING
- [5] PRINT LOST CALL
- [6] DELETE LOST CALL

TRAFFIC - System attendant only

- [1] PRINT ALL SUMMARY (enter Analysis time & type)
- [2] PRINT ALL PERIDICLLY (enter Analysis time, type & Print time)
- [3] ABORT PERIDIC PRINTING
- [4] PRINT ATD TRAFFIC (enter Analysis time & type)
- [5] PRINT CALL SUMMARY
- [6] PRINT CALL HOURLY
- [7] PRINT H/W USAGE (enter Analysis time & type)
- [8] PRINT CO SUMMARY (enter Analysis time & type)
- [9] PRINT CO HOURLY (enter CO group number)

[2] COS

- [1] SET ICM ONLY MODE (enter station range)
- [2] RESTORE COS (enter station range)

[3] AUTHORIZATION - System Attendant only

- [1] REGISTER AUTHORIZATION (enter station)
- [2] ERASE AUTHORIZATION (enter station)

[4] DATE AND TIME

- [1] SYSTEM DATE TIME SET (System Attendant only)

- [2] LCD DATE MODE CHAGE (System Attendant only)
- [3] LCD TIME MODE CHAGE (System Attendant only)
- [4] ATD SET WAKE UP TIME (enter station range)
- [5] ATD WAKE UP DISABLE (enter station range)
- [6] USE PX TIME/DATE (Network time/date (0-2))
- [7] ATD HOTDESK LOGIN
- [8] ATD HOTDESK LOGOUT

[5] MESSAGE

- [1] SET PRE:CUSTOM MESSAGE (enter station range &Message (00~20))
- [2] DND/FWD/MESSAGE CANCEL (enter station range)
- [3] CUSTOM DISP MESSAGE REG (enter Message (11~20))
- [4] Monitor Conference Room
- [5] Delete Conference Room

[6] REC System ANNCEMENT - enter VSFslotsequence number

[7] SUPPLEMENTARY

- [1] REG STATION NAME (enter station number)
- [2] ISOLATE CO FAULT
- [3] AUTO D/N/T PGM (select enable/disable)
- [4] EXT PORT#1 BGM EN/DI (select 0-2)
- [5] EXT PORT#2 BGM EN/DI (select 0-2)
- [6] LCD Display Language (select language)
- [7] PTT Login / Logout, station range (enter station range)
- [8] Display CPU REDUNDANCY STATE
- [9] FONT SET (enter station range)
- [0] CONTRAST SET (enter station range)

[8] EMERGENCY HISTORY

[9] USB UPGRADE

[#] WHTU SUBSCRIBE

Operation

Attendant

To activate an Attendant Station User Program Code feature or function

1) Press the **[TRANS/PGM]** button, the Station Program Menu is displayed.
2) Dial “0” to access the Attendant Station Program codes.

3) Enter desired code or use the **[VOL UP]**/**[VOL DWN]** button to display the desired menu item and enter the desired code.
4) Dial additional inputs, if required.

Programming

Keyset Admin.

SYSTEM

- System Attendant (PGM 164-Button 1)
- Main Attendants Assignment (PGM 164-Button 2~4)

Web Admin.

SYSTEM DATA

- Attendant Assignment ➤System Attendant, Main Attendant

Related Features

- SMDR (Station Message Detail Recording)
- Traffic Analysis
- Temporary Station COS (Class of Service)/Lock
- Authorization Codes (Password)
- System Clock Set
- Pre-defined & Custom Text Display Messages
- Integrated Auto Attendant/Voice Mail
- Dial-by-Name
- Auto Service Mode Control
- Back Ground Music
- Station User Programming & Codes

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.7 Cabinet Alarm for UCP

Description

The Cabinet Alarm alerts the Attendant of a fault in the fans or power supplies in a cabinet. Thealarm notification displays the cabinet number.

The cause of the fault, cabinet fan, PSU or PSUfan is shown on the Cabinet Attributes Web Admin page. The alert may beterminated at the Attendant's phone by dialing the Alarm Reset code or, if assigned, pressingthe **{ALARM RESET}** button.

Operation

When an alarm condition occurs, the Attendant phone will display

CABINET XXX STS FAULT

System Attendant

*To assign a Flex button as an **{ALARM RESET}** button to terminate the Alarm Signal*

- **[TRANS/PGM]** + {FLEX} + "*565" + **[HOLD/SAVE]**

To terminate an Alarm Signal while idle

1) Dial the Alarm Reset code, "*565", confirmation tone is received and the display returns to normal. Or, Press the **{ALARM RESET}** button.

Conditions

- ✓ The PSU serial port must be connected to the gateway or UCP Module identified in System Data ➤ Cabinet Attribute.
- ✓ If the serial monitoring cable of the PSU is connected to a WTIM, refer to the Description and Installation Manual for proper DIP-switch settings; DIP-switch 3 of the WTIM must be set to the OFF position.
- ✓ The notification identifies the cause of the alarm as FAN1 or FAN2 for the Main Cabinet fans, PSU1 or PSU2 for a power supply fault or PSU1 FAN or PSU2 FAN.

Programming

Web Admin.

SYSTEM DATA

- Cabinet Attributes (197) ➤Cabinet Status Check

Related Features

- Alarm Signal/Door Bell
- Emergency Call Attendant Alert

6.8 Call Forward, Attendant

Description

As with other stations, Attendants can forward calls to other stations in the system. Calls may be forwarded unconditionally, on busy or no answer.

Operation

Attendant

To activate Call Forward

1) Lift the handset or press the **[SPEAKER]** button to receive intercom dial tone.
2) Press the **[FWD]** button.
3) Dial '0' ~ '5', the Call Forward type.
4) Dial the station number to receive forward calls.

To deactivate Call Forward

1) While idle, press the **[FWD]** button.

Or,

1) While Off-hook, press the **[FWD]** button.
2) Dial '#'.

Condition

- ✓ The Conditions of the Call Forward feature apply.
- ✓ If the System Attendant activates Unconditional Call Forward, the receiving station will receive Attendant calls and recall ring. In addition, if the receiving station is an iPECS IP or LDP Multi-button Phone, the receiving station will be able to activate features normally reserved for a Main Attendant.
- ✓ The system requires at least one Attendant be available at all times. The last available Attendant cannot activate Call Forward to an Attendant that has activated the Alternate Attendant feature.

Programming

Keyset Admin.

STATION

- Call Forward (PGM 111-Button 2)
- Station Call Forward No Answer Timer (PGM 123-Button 1)

SYSTEM

- System Attendant (PGM 164-Button 1)
- Main Attendants Assignment (PGM 164-Button 2~4)

Web Admin.

STATION DATA

- Common Attributes➤Call Forward
- Station Timers ➤Station Forward No Answer Timer

SYSTEM DATA

- Attendant Assignment ➤System Attendant, Main Attendant

Related Features

- Call Forward
- Alternate Attendant

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.9 Call Forward, Off-Net

Description

The System Attendant can forward incoming outside calls to a remote "Off-Net" location. Calls forward via a Speed Dial bin. When a call is received, the system will automatically place a call using the Speed Dial number, dialing the number and connecting the incoming call in an Unsupervised conference.

Operation

System Attendant

To activate Off-Net Call Forward

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the **[FWD]** button.
3) Dial '5', the Off-Net Forward feature code.
4) Dial Line access code for the Line to be forward.
5) Dial the Speed Dial bin to use to place the outgoing (Off-Net) call, the LED of any Off Net forwarded **{LINE}** button at the Attendant stations will flutter at a rate of 240 ipm while active.

To deactivate Off-Net Call Forward

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the **[FWD]** button.
3) Dial "5", Off-Net Forward code.
4) Dial Line access Code.
5) Dial '#'.

Conditions

- ✓ The System Attendant can forward any Line, Line Group or all outside calls using the appropriate Line access code. In addition, a Flex button assigned to a Line or group may be employed in place of the Line access code to forward calls.
- ✓ The conditions of the Unsupervised conference feature apply.
- ✓ The conditions of the Call Forward Off-Net feature apply.
- ✓ When enabled, calls that forward Off-net will receive the Off net forward prompt.
- ✓ When enabled, calls that forward Off-net will receive the DTMF repeat tone.
- ✓ To utilize Off-net forward of incoming calls on analog CO lines to another analog CO line, a valid Open Loop Detect Timer should be assigned by programming for the analog CO lines. This is to prevent CO line lock up in case the analog CO lines do not provide proper disconnect signals for the system to detect.
- ✓ To enable this feature, Day/Night/Timed COS of DISA COS (PGM 166) must be set on the available value to allow the CO line. (ex. The value is 1)

Programming

Keyset Admin.

STATION

- Call Forward (PGM 111-Button 2)
- Off-Net Forward (PGM 111-Button 14)

SYSTEM

- Off-Net Prompt Usage (PGM 160-Button 11)
- DTMF Repeat Tone Usage (PGM 160-Button 15)
- System Attendant Assignment (PGM 164-Button 1)
- DISA COS (PGM 166)
- Unsupervised Conference Timer (PGM 182-Button 5)
- Open Loop Detect Timer (PGM 142-Button 13)

Web Admin.

STATION DATA

- Common Attributes ➤Call Forward, Off-Net Forward

SYSTEM DATA

- System Attributes ➤Off-Net Prompt Usage, Repeat DTMF Tone
- Attendant Assignment ➤System Attendant
- System Timers ➤Unsupervised Conference Timer
- DISA COS ➤ Day/Night/Timed COS

CO LINE DATA

- Analog Attributes ➤Open Loop Detect Timer

Related Features

- Call Forward
- Unsupervised Conference

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.10 Day/Night/Timed/Scenario Ring Mode

Description

The system clock automatically controls the Ring Mode. Ring assignments are applied based on the time of day and day of week. Four modes of ring (Ring Assignments) are provided Day, Night,Timed and Scenario.

The Attendant controls the system Ring Service mode changing from Auto Service Mode to Day, Night, Timed or Scenario service mode. Based on the service mode selected, different ring assignments, COS (Class of Service) and answering privileges are invoked for the system users.

Operation

System Attendant

To change Day/Timed/Night Ring Mode manually

1) Press the **[DND]** button.
2) Dial “1”~”5” to select the desired mode (1: Day mode, 2: Night mode, 3: Timed mode, 4: Auto Service mode, 5: Scenario mode).
3) Press the **[HOLD/SAVE]** button.

To set Day/Timed/Night Ring/Scenario Mode automatically (Auto Service Mode Control)

1) Press the**[TRANS/PGM]** button.
2) Dial the Attendant Station Program code “073” to set Auto Mode control.

Conditions

- ✓ Only Attendants can change Day/Timed/Night/Scenario Ring Mode for the system manually and program the Auto Ring Mode Selection Table (Weekly Time table).
- ✓ Stations receive incoming ring for outside Lines based on the database assignments and the system mode (Day/Night/Timed/Scenario) when the call arrives.
- ✓ When the Auto Ring Selection Table is programmed, the ring, COS (Class of Service) and Line access mode are changed automatically based on the times assigned in the table.
- ✓ The System Attendant always has manual control of the system mode by enabling/disabling the Auto Service Mode Control.
- ✓ To employ the Scenario mode, scenarios must be assigned in the System Call Routing program.

Programming

Keyset Admin.

CO/IP

- CO Line Ring Assignment (PGM 144)

SYSTEM

- External Control Contact (PGM 168)

TABLES

- Weekly Time Table (PGM 233)
- System Call Routing (PGM 251) Web only

Web Admin.

CO LINE DATA

- CO/IP Ring Assignment

SYSTEM DATA

- External Control Contacts
- System Call Routing

TABLES DATA

- Auto Ring Mode Table

Related Features

- System Clock Set
- Outside Line Ring Assignment
- LBC (Loud Bell Control)
- Dialing Restrictions
- Auto Service Mode Control
- System Call Routing

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.11 DSS/BLF Consoles

Description

Two types of DSS/BLF Consoles are available and can be used with an Attendant station Serial Consoles and standard Consoles. For details on the Serial console operation, refer to section 'Serial DSS/BLF Console'.

The system allows an unlimited number of standard DSS/BLF Consoles to be installed in the system. Up to three (3) standard DSS Consoles may be associated with a station connected to the system.

However, the maximum of DSS consoles may be different according to the models of phone. Each button on the console can be assigned as an outside Line, DSS/BLF or feature button.

For ease of programming, each console is assigned to a button "map" in the system database.

The Administrator or user may then change individual Flex buttons as desired. Operation of the DSS/BLF Console Flex buttons is the same as Flex buttons on the iPECS IP and LDP Phones.

Conditions

- ✓ Each DSS Console is assigned to operate in connection with a specific station.
- ✓ There is no limit to the number of DSS Consoles in a system beyond the basic system capacities. The DSS Console employs a station number in the system.
- ✓ Assignments for the DSS button maps are detailed in the iPECS Administration & Programming manual. Note that the individual button assignments can be changed as with iPECS IP and LDP Phone Flex buttons.

Table DSS/BLF button maps

MAP 1	
Flex 1	Atd Override
Flex 2	All Call Page
Flex 3	Call Park 01
Flex 4-	Station Group 1
Flex 5	Camp-On
Flex 6	Int All Call Page
Flex 7	Call Park 02
Flex 8	Station Group 2
Flex 9	Monitor key (Release)
Flex 10	Ext All Call Page
Flex 11	Call park 03
Flex 12	Station Group 3
Flex 13~48	Station 100~135
MAP 2	
Flex 1 ~ 48	Station 136~183
MAP 3	
Flex 1 ~ 48	Station 184~231

Programming

Keyset Admin.

STATION

- Station Type (PGM 110-Button 1)
- DSS Map (PGM 110-Button 2)
- Flexible Button Assignment (PGM 115)

Web Admin.

STATION DATA

- Station Type ➤ DSS Map
- Flexible Buttons

Related Features

- Attendant Positions
- Station Flexible Buttons

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.12 Disable Outgoing Line Access

Description

The System Attendant can place outsideLines out-of service, disabling outgoing calls on the Line. This is normally done should an undetected fault in the carrier or system circuit interrupt service over theLine. Incoming calls continue to be processed normally.

Operation

System Attendant

To disable/enable Outgoing Line access (toggle)

1) Press the **[TRANS/PGM]** button.
2) Dial "072", the Attendant Station Program code.
3) Press the **{Line}** button of the line(s) to be disabled, confirmation tone is heard and the status for the selected Line(s) is changed.

Conditions

- ✓ The System Attendant may disable a busy Line. The feature will take effect after the desired Line goes to idle.
- ✓ Once the Line is disabled, all Attendant appearance for the disabled Line will flutter at 240 ipm, other stations will indicate the Line as busy and the LED is On.
- ✓ The outside Line status is stored in battery-protected memory in case of a power failure.
- ✓ Multiple Lines may be enabled/disabled without redialing the Attendant Station Program code. Confirmation tone is heard after each Line is enabled/disabled.
- ✓ When the system detects a fault on a Line, the Line is disabled for outgoing access automatically.
- ✓ Incoming calls on a disabled Line will continue to operate normally.

Programming

Keyset Admin.

SYSTEM

- Attendant Assignment (PGM 164-Button 1)

Web Admin.

SYSTEM DATA

- Attendant Assignment

Related Features

- ■ Attendant Positions

Hardware

- ■ iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.13 DND Override

Description

A station in the DND generally cannot receive an incoming call. The Attendant and the Secretary station of an Executive/Secretary pair however may override the DND status to signal the station of an awaiting call. A Flex button assigned for Intrusion is required.

Operation

Attendant

*To assign a flexible button as a **{ATD INTRUSION}** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + “86” + **[HOLD/SAVE]**

To activate DND Override while receiving DND tone

1) Press the **{ATD INTRUSION}** button, the call signals at DND station.

Conditions

✓ An Attendant may use Override to transfer an outside call to a station in DND.

Programming

Keyset Admin.

SYSTEM

- Attendant Assignment (PGM 164)

Web Admin.

SYSTEM DATA

- Attendant Assignment

Related Features

- Intrusion
- DND (Do Not Disturb)
- Executive/Secretary Forward

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.14 Emergency Call Attendant Alert

Description

When a station places a call to an Emergency number, the Attendant receives an emergency call alert.

The alert includes an alert tone and display of the emergency call information, which continues until the Attendant resets the alert. The information includes the calling station number, time and date.

The system stores the most recent emergency calls (up to 16). The Attendant can review the history at any time.A Flex button on the Attendant phone can be used to reset the alert and terminate the alert tone.

Operation

System

Operation of the alert is automatic.

Attendant

*To assign a Flex button as an **{ALARM RESET}** button to terminate the alert*

- **[TRANS/PGM]** + {FLEX} + "*565" + **[HOLD/SAVE]**

*To assign a Flex button as an **{Emergency Call History}** button*

- **[TRANS/PGM]** + {FLEX} +**[TRANS/PGM]**+ "08" + **[HOLD/SAVE]**

To terminate an alert signal while idle

1) Dial the Alarm reset Code, *565, confirmation tone is received and the Alert terminates.
   Or, Press the **{ALARM RESET}** button.

To view the Emergency Call History

1) Lift the handset or press the **[SPEAKER]** button.
2) Press the **[TRANS/PGM]** button.
3) Dial "08", the Emergency Log code. The display shows the first emergency call logged in the history.

EMERGENCY STA NO XXXX MM/DD HH:MM (xx)

4) Press the **[VOL UP]** and **[VOL DOWN]** buttons to scroll through the call history.
5) After viewing a call history record, the record is marked as "READ" as below.

EMERGENCY STA NO XXXX MM/DD HH:MM READ (xx)

6) If a record remains unread, the Attendant will receive alert tone upon returning to idle as notification that a record has not been reviewed.

Programming

Keyset Admin.

STATION

- Emergency CO(PGM 112, Button 18)

TABLES

- Emergency Service Call Table (PGM 226)

SYSTEM

- EMR CALL ATD NOTIFY (PGM 161, Button 24-5)

Web Admin.

STATION DATA

- Common Attributes ➤Emergency CO or Group

SYSTEM DATA

- System Attributes ➤Emergency Call ATD Notify

TABLES DATA

- Emergency Code Table

Related Features

- Alarm Signal/Door Bell
- Emergency Call E-911 (caller location) Support
- Emergency Call

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.15 Feature Cancel

Description

The System Attendant can cancel features such as DND, Call Forwarding and Pre-defined or Custom Messages that are active at other stations.

Operation

To deactivate DND/Call Forward/Pre-selected Message for other stations

1) Press the **[TRANS/PGM]** button.
2) Dial “052”, Attendant Station Program code.
3) Dial the desired station range or, for one station, enter the station number and “*”.
4) Press the **[HOLD/SAVE]** button, confirmation tone is heard and Attendant station returns to idle status.

Programming

Keyset Admin.

SYSTEM

- Attendant Assignment (PGM 164-Button 1)

Web Admin.

SYSTEM DATA

- Attendant Assignment

Related Features

- Call Forward
- DND (Do Not Disturb)
- Attendant Positions
- Pre-defined & Custom Text Display Messages

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.16 IP-Attendant

Description

The 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 displays of call, user and system status. IP Attendant incorporates the PC Mic. and Speaker for audio so that a separate desktop phone is not required.For further information on IP-Attendant, refer to the IP-Attendant Installation and User Guide.

Operation

System Attendant

Operation of the IP-Attendant is provided in the IP-Attendant Installation and User Guide.

Conditions

✓ IP-Attendant requires installation of a system Lock-key (license).

Programming

Keyset Admin.

SYSTEM

- Attendant Assignment (PGM 164)
- Station User Login (PGM 443) Web only

Web Admin.

SYSTEM DATA

- Attendant Assignment

DEVICE LOGIN

- Station User Login

Hardware

■ iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.17 Intrusion

Description

An Attendant can intrude upon an active station conversation. When the Attendant intrudes an Intrusion Tone is provided, if assigned, and a conference is established between the Attendant, station, and the connected party.Intrusion can only be activated using an **{ATD INTRUSION}** button.

Operation

Attendant

*To assign an **{ATD INTRUSION}** button*

- **[TRANS/PGM]** + {FLEX} + **[TRANS/PGM]** + "86" + **[HOLD/SAVE]**

To activate attendant intrusion while receiving busy on an Intercom call

1) Press **{ATD INTRUSION}** button, Intrusion Warning Tone is provided to the busy station and the Attendant join the conference to the call.

Conditions

✓ If an Attendant or Secretary of an Executive/Secretary pair presses the **{ATD INTRUSION}** button while receiving DND tone, the system will activate DND Override.
✓ The conditions of the Conference feature apply.
✓ If 'Privacy' option is ON, then Attendant can't intrude to the station.

Programming

Keyset Admin.

STATION

- Override Privilege (PGM 113-Button 4)

SYSTEM

- Automatic Privacy (PGM 161-Button 3)
- Privacy Warning Tone (PGM 161-Button 4)

Web Admin.

STATION DATA

- Common Attributes ➤Override Privilege

SYSTEM DATA

- System Attributes ➤Automatic Privacy, Privacy Warning Tone

Hardware

■ iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.18 LCD Display Format Control

Description

The System Attendant can select the format of the time and date provided in the LCD of all iPECS IP and LDP Phones in the system.

The System Attendant can select (toggle between) two formats for both time and date. The formats are:

- **Date:** Month/day/year or Year/month/date
- **Time:** 12 hour or 24 hour (military)

Operation

System Attendant

To change the LCD Date Format (toggle)

1) Press the **[TRANS/PGM]** button.
2) Dial "042", the Date Display Format program code.

To change the LCD Time Format (toggle)

1) Press the **[TRANS/PGM]** button.
2) Dial "043", the Time Display Format program code.

Programming

Keyset Admin.

SYSTEM

- Attendant Assignment (PGM 164-Button 1)
- LCD Date Display Mode (PGM 169-Button 1)
- LCD Time Display Mode (PGM 169-Button 2)

Web Admin.

SYSTEM DATA

- Attendant Assignment
- LCD Display Mode➤LCD Date
- LCD Display Mode➤Time Display Mode

Related Features

- Attendant Positions

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

6.19 System Clock Set

Description

The System Attendant can set the system Time/Date.

Operation

System Attendant

To set the system clock

1) Press the **[TRANS/PGM]** button.
2) Dial "041", the Attendant Station Program code.
3) Dial six (6) digits for the Date (MM/DD/YY) or **[HOLD/SAVE]** to skip the Date.
4) Dial four (4) digits for the Time (HH/MM) or **[HOLD/SAVE]** to skip the Time setup.
5) Press the **[HOLD/SAVE]** button, confirmation tone is heard and Attendant station returns to idle status.

To synchronize the system clock to the ISDN

1) Press the **[TRANS/PGM]** button.
2) Dial "046", the Attendant Station Program code.
3) Dial "1" to synchronize the Date and Time to the ISDN.
4) Press the **[HOLD/SAVE]** button, confirmation tone is heard and Attendant station returns to idle status.

Conditions

✓ The time is entered as a 24-hour format.

✓ If an NTP (Network Time Protocol server is assigned, the system will check the time every ten (10) minutes. If the system time is more than 10 seconds off, the system time is synchronized with the NTP server. This will override the Attendant setting.

Programming

Keyset Admin.

SYSTEM

- Network Time/Date (PGM 161-Button 12)
- System Attendant (PGM 164-Button 1)
- System Time (PGM 178-Button 1)
- System Date (PGM 178-Button 2)
- DST Enable (PGM 178 – Button 3)
- DST Start & End Time (Web only)
- Network Time/Date (PGM 195 – Button 1)
- NTP Sever address (Web only)
- Std system time, local Time Zone (Web only)

Web Admin.

SYSTEM DATA

- System Attributes ➤Network Time & Date
- Attendant Assignment
- System Date &Time

Related Features

- LCR (Least Cost Routing)
- SMDR (Station Message Detail Recording)
- Auto Service Mode Control
- Automatic System Time Synchronization
- Automatic System Daylight Savings Time
- Day/Night/Timed/Scenario Ring Mode

Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

7 SLT(Single Line Telephone)

7.1 Broker Call

Description

Broker Call allows a SLT(Single Line Telephone) user to engage in two (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:** the second call is originated by the SLT(Single Line Telephone)
- **Camped On Broker Call:** the second call camped on the SLT(Single Line Telephone).

Operation

SLT(Single Line Telephone)

To activate a Transfer Broker Call

1) Make or receive an intercom or external call.
2) Momentarily press the hook-switch, intercom dial tone received and active call is placed in Exclusive hold state.
3) Place second call.
4) To alternate between calls momentarily press the hook-switch.

To activate a Camp-On Broker Call

1) Make or receive an intercom or external call.
2) Receive a Call Waiting/Camp-On tone.
3) Momentarily press the hook-Switch, intercom dial tone received and the active call is placed on Exclusive Hold.
4) Dial the Camp-On Answer code “*621”, Camped on call is connected.

To alternate between the calls

1) Momentarily press the hook-switch.
2) Dial the Camp-On Answer code “*621”.

Conditions

- ✓ After a hook-switch flash, if the call results in an error, busy, no answer or an abnormal state, the SLT(Single Line Telephone) user may momentarily press hook-switch to retrieve the held call.
- ✓ During a Transfer Broker Call, if the SLT(Single Line Telephone) user goes on-hook, the Broker Call parties are connected completing a Call Transfer.
- ✓ During a Transfer Broker Call, if the active caller disconnects from the SLT(Single Line Telephone) user, the held party, if another station, is connected to the SLT(Single Line Telephone).If the held party is an outside call, the SLT(Single Line Telephone) user receives error tone and may go on-hook to receive recall and retrieve the held call.

- ✓ During a Camp-On Broker Call, if the SLT(Single Line Telephone) user goes on-hook, the active call is disconnected and the held call recalls to the SLT(Single Line Telephone).
- ✓ During a Camp-On Broker Call, if the active party disconnects from the SLT(Single Line Telephone), the SLT(Single Line Telephone) user receives error tone. The SLT(Single Line Telephone) user may momentarily press the hook-switch to retrieve the held party or go on-hook and receive recall.
- ✓ If the SLT(Single Line Telephone) user presses the hook-switch twice in less than 2 seconds, a 3-way conference is established.
- ✓ If after a hook-switch flash, the user takes no action for the dial tone timer, the SLT(Single Line Telephone) will receive error tone.
- ✓ If the SLT(Single Line Telephone) goes to an on-hook state with a call on hold, the SLT(Single Line Telephone) will receive recall ring automatically.
- ✓ The SLT(Single Line Telephone) Flash mode must be set to Transfer.

Programming

Keyset Admin.

STATION

- SLT(Single Line Telephone) Flash Mode (PGM 113-Button 24-8)

Web Admin.

STATION DATA

- Terminal Attributes ➤SLT(Single Line Telephone) Flash Mode

Related Features

- Message Wait/Call Back
- Call Waiting/Camp-On
- Exclusive Hold
- Call Transfer
- Conference

7.2 Howler Tone

Description

When a SLT(Single Line Telephone) station goes off-hook and does not initiate dialing within the Dial tone timer, delays dialing between digits in excess of the inter-digit time or stays off-hook at the completion of activating a feature or program, the station will receive howler tone as an error indication and the call attempt will be abandoned. In order to complete the call, the user must return to the on-hook state and restart the call.

Operation

System

The system will deliver howler tone automatically, as required.

Conditions

- ✓ Howler Tone is sent after a period, of about 30 seconds of error tone.
- ✓ Lock-out occurs when howler tone starts.

Programming

Keyset Admin.

STATION

- Howling Tone To Station (PGM 111-Button 5)

Web Admin.

STATION DATA

- Terminal Attributes ➤Howler Tone

Related Features

- ■ Intercom Lock-Out

Hardware

7.3 SLT(Single Line Telephone) Flash Mode

Description

When the SLT(Single Line Telephone) user momentarily presses the Hook-switch, a Flash is generated. The operation of the Flash function can be configured to for the following operation:

- **0: Flash Transfer** - Flash detected, outside Line is held and dial tone is received.
- **1: Flash Drop** - Flash detected and outside Line is disconnected.
- **2: Flash Ignore** - Flash detected but Ignored.
- **3: Hold Release** - Flash detected, the outside Line is held and the dial tone is received. If SLT(Single Line Telephone) user goes on-hook, the held Line is disconnected.

Programming

Keyset Admin.

STATION

- SLT(Single Line Telephone) Flash Mode (PGM 113-Button 24-8)

Web Admin.

STATION DATA

- Terminal Attributes ➤SLT(Single Line Telephone) Flash Mode

Hardware

7.4 SLT(Single Line Telephone)Message Wait Indication

Description

All SLT(Single Line Telephone) devices will receive a "Stutter" dial tone as an audible indication of a Message Waiting. In addition, industry standard Message Waiting telephones may be connected to the system. Software will cause the lamp to flash when a messaging is waiting.

Operation

System

The system switches the 90 VDC lamp On and Off for assigned SLT(Single Line Telephone)s indicating a Message Wait.

Conditions

- ✓ The system switches a 90 VDC supply On and Off to flash the SLT(Single Line Telephone)'s neon Message lamp.
- ✓ Although the SLT(Single Line Telephone) Battery Feed is removed during the 90 VDC On cycle, the system will recognize a SLT(Single Line Telephone) Off-hook event.
- ✓ The SLT(Single Line Telephone) must incorporate a 90 VDC neon lamp that is connected directly across the tip and ring of the voice network.

Programming

Keyset Admin.

STATION

- Station Type (PGM 110-Button 1)

Web Admin.

STATION DATA

- Station Type

Related Features

- Message Wait/Call Back

Hardware

- SLT(Single Line Telephone) with 90 VDC Neon lamp

7.5 SLT(Single Line Telephone)Name Entry

Description

A SLT(Single Line Telephone) user has the capability to program a name so that a calling user with an LCD can see the name instead of the station number.

Operation

SLT(Single Line Telephone)

To register the name at the SLT(Single Line Telephone)

1) Lift the handset.
2) Dial "*561", the SLT(Single Line Telephone) Programming code, confirmation tone is heard.
3) Dial "74", the SLT(Single Line Telephone) Name Program Code.
4) Enter name, refer to Station Speed Dial, Alphanumeric Chart.
5) Momentarily depress the hook-switch, receive confirmation tone.

To delete the name at the SLT(Single Line Telephone)

1) Lift the handset.
2) Dial "*561", the SLT(Single Line Telephone) Programming code, confirmation tone is heard.
3) Dial "74", the SLT(Single Line Telephone) Name Program Code.
4) Momentarily depress the hook-switch, receive confirmation tone.

Programming

Web Admin.

STATION DATA

- Station Name Display

Related Features

- Dial-by-Name
- Station Speed Dial

Hardware

7.6 Transfer CLI to SLT(Single Line Telephone)

Description

A SLT(Single Line Telephone) phone may be receive the Caller Id of the transferred outside call in place of the station number transferring the call.

The transferred call Caller Id is sent after the transferring station hangs up to complete the transfer.

Operation

The Caller Id is automatically sent to the SLT(Single Line Telephone) when configured.

Conditions

✓ For proper operation, use of the Original CLI should be assigned to the Transfer CLI.

Programming

Keyset Admin.

STATION

- Transfer CLI to SLT(Single Line Telephone) (PGM 114-Button 18)

Web Admin.

STATION DATA

- Terminal Attributes ➤Send SLT(Single Line Telephone) CLI Info
- CLI Attributes ➤FAST CLI For Transfer Call

Hardware

■ SLT(Single Line Telephone) w/display

8 SIP extension

8.1 SIP Phone Support

Description

The iPECSsystemincorporates basic SIP Extension service connecting SIP phones through a VoIP channel using IETF's standard RFC-3261 SIP protocols. SIP phones, including the Ericsson-LG Enterprise SIP phones, register for service with or without authentication. After registration, the SIP phone can place and receive calls, place calls on hold, transfer calls and will have access to the features of the system through the Flexible Numbering plan using the procedures applicable to a SLT(Single Line Telephone). Some of the features supported are listed below.

In addition to the Numbering Plan features,for the Ericsson-LG Enterprise SIP phones, the Call Forward and DND features can be synchronized using proprietary methods so that when you activate the feature at the SIP phone, the system is properly informed and can activate the feature properly within the system. When Feature sync is not enabled or a 3rd party SIP phone is employed, thephone will activate the internal Forward or DND feature and the system cannot properly inform status to others or control routing.

- Account Code Entry
- Attendant Call
- Broker Call, the SIP phone must support Call Wait
- Call Forward, requires Feature Sync to employ eMG80/100 Call Fwd feature.
- Call Park using Unscreened (Blind) Transfer
- Call Pick-Up, Group & Directed
- Call Waiting/Camp-on, the SIP phone must support Call Wait
- Conference Group
- Conference Room
- Do-Not-Disturb (DND), the SIP phones may have internal DND
- Door Open
- DSS/BLF, supported by Ericsson-LG Enterprise SIP phones
- Emergency Page
- Emergency Calling
- Integrated Voice Mail
- Last Number Redial
- Paging

5) Internal/External Page
6) Meet Me Page
7) All Call Page

- Station ICR
- Station & System Speed Dial

- Universal Night Answer
- Walking COS (Class of Service)
- Wake-Up Alarm

Operation

SIP Phone

✓ SIP phone operation generally follows the SLT(Single Line Telephone) procedures. Refer to the SIP phone user guide for operation of features of the phone.

System

Operation of SIP phone provisioning is automatic when configured.

Ericsson-LG Enterprise SIP phone

To activate One-Time Call Forward while receiving ring with Feature Sync active

1) Press the **[Call Forward]** button.
2) Dial destination number, ringing stops and the call Forwards.

To activate One-Time DND while receiving ring with Feature Sync active

1) Press the **[DND]** button, the ringing call is rejected.

Conditions

✓ A lock-key (License) is required to use a 3rd party SIP phone, contact your local Ericsson-LG Enterprise representative.

✓ SIP phones must be configured in accordance with the manufacturer's instruction and have an appearance of the station assigned in the system.

✓ When using Feature Sync, the Station must be allowed Call Forward and DND for proper operation.

✓ In addition to the feature set supported for 3rd party SIP phones, the Ericsson-LG Enterprise SIP phones support Voice Over, Intrusion, Forced Hands-free ICM.

✓ SIP phones provide for an End-of-dialing entry much like a mobile phone. Generally a Send or the '#' button is employed at the end of dialing so there is no delay in sending digits.

✓ With most SIP phones, a second Station appearance can be assigned to a button. In this case, a virtual MAC is used in the system configuration.

✓ Ericsson-LG Enterprise and most 3rd party SIP phones support 3-party conference through a built-in mixer. For operation refer to the SIP phone user guide. When not supported by the SIP phone, the phone will have access to the Conference Room and Group features for conferencing.

✓ Ericsson-LG Enterprise takes no responsibility of the compatibility of any 3rd party SIP phone with the system and any XX party SIP phone with the system. It is the user's responsibility to assure compatible operation.

✓ For SIP Phones, the UCP must have available MCIU or MCIM channels to serve the features listed below.

2) Conference Room
3) Conference Group

4) Voice Over
5) Intrusion
6) Automatic/On-demand Two-Way Recording

- ✓ Configuring authentication will require the SIP phone provide authentication at registration and during the initiation of a call.
- ✓ SIP phones require an available VoIP DSP channel, built-in the UCP or from a VOIB/VOIM, to relay Page announcements, BGM and MOH packets.
- ✓ Third party SIP phones may activate Call Forward or DND locally. In this case, the system is not informed of the Call Forward or DND status of the phone. Call Forward and DND employing the UCP can be accomplished using the SLT(Single Line Telephone) operation. With Feature Sync active and the station allowed Call Forward and DND, Ericsson-LG Enterprise SIP phone will notify the iPECS system of the feature status.
- ✓ A system VoIP channel is required to support local SIP phone features. A VoIP channel is required to serve below features.

7) DSP for generation of Busy/Error/Confirm/Ring-Back/Hold/ Page/Warning/ OHVA/Intrusion/Dial tones from system to SIP Phone.
8) Relay of Music On Hold from system to SIP Phone.
9) Relay of Paging from/to SIP Phone.
10) Voice RTP Packet Relay between private LAN and public WAN, local and remote, NAT resolution.

Note that since a VoIP channel is required for a SIP Trunk, an additional channel is not required by the SIP Phone to use the SIP Trunk.

Programming

Keyset Admin.

SIP DATA

- SIP Attributes (PGM 210) Web only
- SIP Phone Attributes (PGM 211) Web only
- SIP Phone Provisioning (PGM 212) Web only

Web Admin.

SIP DATA

- SIP Common Attributes
- SIP Phone Attributes
- SIP Phone Provisioning

Hardware

8.2 SIP Phone Provisioning, Ericsson-LG Enterprise Phones

Description

The Provisioning feature automatically configures the Ericsson-LG Enterprise SIP phones from the eMG database Provisioning Tables.

When the SIP phone powers up, it will download the configuration file from the system and apply the configuration to the phone.

A table may include the MAC address of a specific phone, in which case the table is only available to the SIP phone with the assigned MAC.

When no MAC address is assigned, the table is employed by all SIP phones of the type (model) entered as the “Conf Type”.

If the Provisioning Tables are not used, the system must be configured with the SIP Phone attributes as with any SIP phone and the SIP phone must also be configured separately to match. Characteristics that can be configured from the Provisioning Tables include:

- Registration Timer
- SIP Phone’s local port
- Proxy Port
- Transport type
- Codec priority
- Network Time Server and DST
- Default Volume levels
- Digit Map.

For more information on these parameters refer to the SIP phone Installer Guide available from your local Ericsson-LG Enterprise representative.

In addition, Feature Sync can be enabled so that when the SIP phone activates DND or Call Forward, the information is sent to the system and the system will activate the appropriate feature. This also permits the SIP phone to activate One-time Call Forward and One-time DND while receiving ring.

Operation

System

Operation of SIP phone provisioning is automatic when configured.

Conditions

- ✓ Provisioning is only available for Ericsson-LG Enterprise SIP desktop phone.
- ✓ The download employs TFTP as the file transfer protocol.

Programming

Web Admin.

SIP DATA

- SIP Common Attributes
- SIP Phone Attributes
- SIP Phone Provisioning

Hardware

- IP8800 & IP8800E series phone

8.3 SIP Virtual Mobile Extension

Description

Virtual Mobile Extension (VMEX) is a proprietary regional FMC (Fixed Mobile Convergence) implementation, which combines mobile service with SIP Call services and the iPECS system functionality.

Calls from the VMEX are directed from the Mobile Switching office to the carrier's SIP Call server and delivered to the iPECS system for routing.

The iPECS system routes the call internally or externally or activates a feature based on the digits received in the SIP "Invite" message.

When the iPECS system receives a call for the VMEX, the call is directed to the VMEX over a SIP trunk to the carrier's SIP Call server, which routes the call to the Mobile switching office to the VMEX.

Support for SIP Subscribe and Notify provide status updates to other devices within the iPECS system.

Operation

System

Operation of VMEX is automatic when configured.

Conditions

- ✓ VMEX service is regional specific and functions only where the carrier service is implemented in accordance with the iPECS system functionality.
- ✓ The VMEX station must be virtually registered as a VMEX SIP station and employs SIP for signaling.
- ✓ A VoIP channel is required to support proper operation of the VMEX as with any SIP Extension.

Programming

Web Admin.

SYSTEM ID & NUMBERING PLAN

- Device Port Number Change➤ Virtual Registration

SIP DATA

- VMEX Station Data
- VMEX Connection Table

TABLES DATA

- Mobile Extension Table

Hardware

9 iPECS UCS Client

9.1 iPECS UCS Premium & Standard Client Support

Description

iPECS system incorporates iPECS UCS Standard Client (PC & Mobile) service based on the built-In UC functionality. All features of the Client are available to the UCS Standard Client including one-to-one collaborationand video calling, Conference manager, Presence, and Instant Messaging.

The iPECS system supports iPECS UCS Premium Client (PC & Mobile) service with iPECS UCS Server installed separately. The addition of the UCS Server provides for six (6) party collaboration and videocalling as well as Lync integration.Please refer to the iPECS UCS Client and UCS Server manuals for the detailed information about the features and operations.

Presence for 3G call in Mobile UCS

Mobile UCS Client sent a busy message to system when it's calling of 3G call.The mobile UCS Client will be displayed the busy presence for another UCS Client member.

9.2 Call Control with UCS Client

Description

UCS Client can call control with my favorite station number (Destination Station number : DSN). There are two kinds of way. One is No voice(NV) Way and the other with voice(WV) Way

NV Method is that voice and call are happened in DSN. UCS Client cannot speak with each other. WV Method is that DSN and UCS Client can talk alternate respectively.

Operation

System

<u>Operation of Call Control is automatic when UCS Client is logged in</u>

1) UCS Client should be registered in ADM 446.
2) User should set DSN password in UCP web ADM 227.
3) UCS Client user must set DSN and password(PWD) in setting before log in.
4) If UCS Client logged in successfully with DSN and PWD, UCS Client can control call of DSN. NV can control simultaneously with DSN. WV can control DSN with IP Bridge mode.
5) If user wants to change the DSN, system moves automatically it from the member of previous P.G.

Conditions

- ✓ UCP makes automatically Personal Group(P.G) when UCS client log in first time with DSN. Master is DSN and Member is UCS-Client by default.

- ✓ DSN and PWD is optional for WV. If it's not added in WV, UCS Client is used stand-alone.
- ✓ If you want to change the Master and Member, you should set it in ADM 260 before log in. (This is modified after P2 1.1.8 or P1 1.0Di version. Master is UCS client and member is DSN in previous version.)
- ✓ If DSN is already the member of other P.G, user cannot assign DSN for NV.
- ✓ If UCS Client is already the Master of other P.G, user cannot assign DSN for NV.

Programming

Web Admin.

Station Password

- Web ADM 227 ➤Set DSN Password

UCS Client Registration

- Web ADM 446

Personal Group setting

- Web ADM 260,261

9.3 Mobile Native Voice Support for UCS Mobile client

Description

UCS mobile is an extension itself but UCS mobile client in a native voice mode has an extension of the associated station.

UCS mobile client with a native voice can control 6 features like MEX On/Off, Voice Mail forward On/Off and ACD On/Off by using a http. If UCS mobile client uses this feature, it must be activated on the associated station.

In case of an inbound call, a caller makes a call to a internal station, which is associated with UCS mobile client then a ring starts to the station and the mobile client. Then if UCS mobile client get the call, the call will be connected through a Native Call (3G/LTE).While conversation, UCS mobile user can search his or her UCS Mobile Client phonebook and transfer the call to an internal user.

In case of an outbound call, a mobile client has two options : One is a call through mode and the other is a native call mode.A called party will receive the CID of the station in case of a call through mode, otherwise a calling party will receive the CID of mobile phone number.

Operation

To activate or deactivate features

1) User can select each menu in the UCS client window.
2) Then the selected feature should be activated or deactivated on the associated station.

To transfer a call

1) While conversation using UCS client, user can search the desired station to transfer in the mobile client phonebook.
2) Transfer a call by selecting a transfer menu.

Conditions

✓ Mobile client with native voice mode can send the associated station number and password when registered. Then a personal group is configured by the system automatically to support the presence of the associated station.

✓ Otherwise, a system administrator should configure a personal group consisting of mobile client and the associated station, a mater station. And also the Linked Pair Mode attribute of PGM261should be set to ON.

Programming

Web Admin.

Station Group Data

- Personal Group (PGM 260)
- Personal Group Attributes (PGM261)

Table Data

- Mobile Extension Table (PGM236)

9.4 User base license &Single log-in vs Multi log-in

Description

From UCS v6, license is much simplified by supporting “User base license”

- Per User License (Basic, Advanced, Power User).
- One User License for UCS Standard and UCS Premium. For UCS Premium, simply add “Premium Server license”.
- Single log-in : UCS user able to log-in on to one client at a time
- Multi log-in : UCS User able to log-in on to multiple devices at the same time

Operation

System

<u>*Add/Delete User with User base License*</u>

User base license should be preprogrammed before adding user.If it’s not added, 1 is omitted.
If it’s device license, 2 is omitted.

<u>*<Single Log-in>Basic UC client configuration*</u>

1) Step 1: License management

2) Step 2: PGM446

3) Step 3: Set call control mode in PC UC Client setting

<u><Single Log-in>Advance UC client configuration for single login</u>

1) Step 1: License management

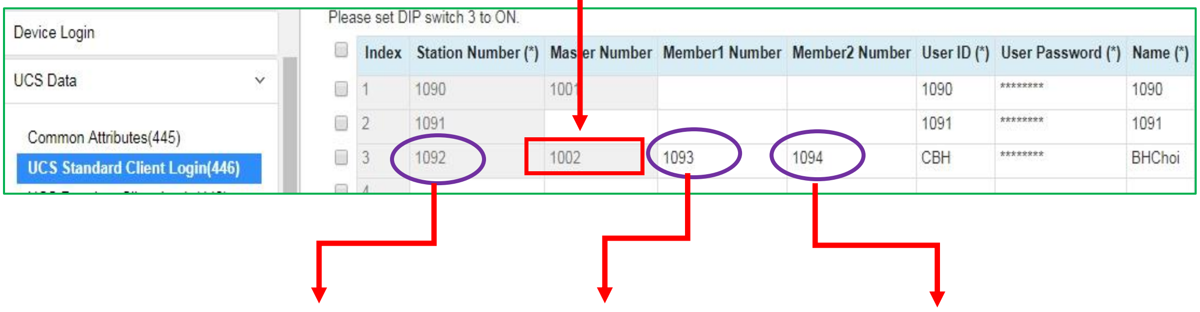
2) UC client (User of station can use 3 device PC, Mobile, Tablet with single)

3) Note: only a device could be used at a time

<u><Multi Log-in>Power UC client configuration (Case 1: LIP master + 3 UCS client)</u>

1) Step 1: License management

2) Step 2: PGM446



1092 > PC UC Client 1093 > Mobile UC Client 1094 > Tablet UC Client

3) Personal group will be made automatically (master 1002, member: 1092, 1093, 1094)

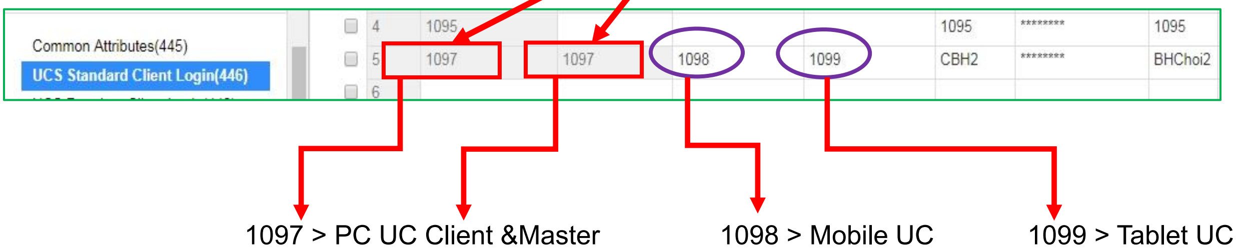
4) Virtual 3 UCS client will be registered (1092, 1093, 1094)

5) Step 3: UC client (User of station number 1002 can use simultaneously 3 device PC, Mobile, Tablet with one user id CBH)

<Multi Log-in>Power UC client configuration (Case 2: UCS client == Master)

1) Step 1: License management

2) Step 2: PGM446



3) Personal group will be made automatically (master 1097, member: 1098, 1099)

4) Virtual 3 UCS client will be registered (1097, 1098, 1099)

Conditions

- ✓ User and device base license works independently of each other. Device means device base license that is used in previous version. (Desktop standard, Desktop premium, Mobile)
- ✓ If you want to change license, you should delete it in PGM 446 and then you change user base license. You should add user after change the license.

Programming

Web Admin.

Station Password

- Web ADM 227 ➤Set DSN Password

UCS Client Registration

- Web ADM 446

Personal Group setting

- Web ADM 260,261

9.5 Push notification for UCS Mobile

Description

New way to deliver communication events (call, IM, etc) to UCS mobile clients

- **Background:** TCP "Keepalive" API deprecation from iOS 10 UCS Mobile unable to stay connected with call server as background mode
- **Solution:** Applying event push notification to awake UCS mobile and then able to start to communicate
- **UCS Standard :** Push notification service embedded in the call server
- **UCS Premium :** Push notification service is supported by UCS Premium Server

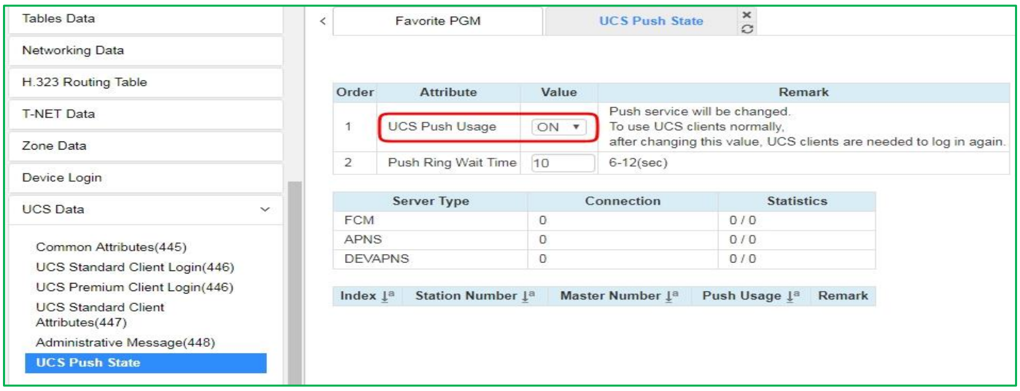
Operation

Push notification for UCS Mobile – System

✓ **Android:** https://fcm.googleapis.com/fcm/send

✓ **iOS:** https://api.push.apple.com

✓ **Port:**443 / Protocol : HTTP/2(HTTPS)



✓ DNS should be used to use Push service

Conditions

✓ iOS (TCP protocol should be used to use push/iOS 10 features) : UDP should be used if Mobile network is not connected because Apple does not support TCP keep alive System could not know status of UC client without TCP keep alive in TCP protocol.

✓ Android (Both TCP and UDP could be used)

Hardware

10 Security

10.1 Password in SMB

Password setting field

SMB has many password setting fields for Security.

- ID/Password for WEB log-in.
- Installation Wizard, WEB > Maintenance > User Management.
- Password for Keyset Administration, Remote Access and CID setting in PGM162.
- ID/Password for PPTP in System Data.
- ID/Password for PPP in PGM205.
- Station and System Authorization Code in PGM227.
- MAC, Password for Remote Device Registration in PGM442.
- ID/Password for Station User Login in PGM443.
- User ID/Password for UCS Client Login in PGM446.

10.2 Strong Password

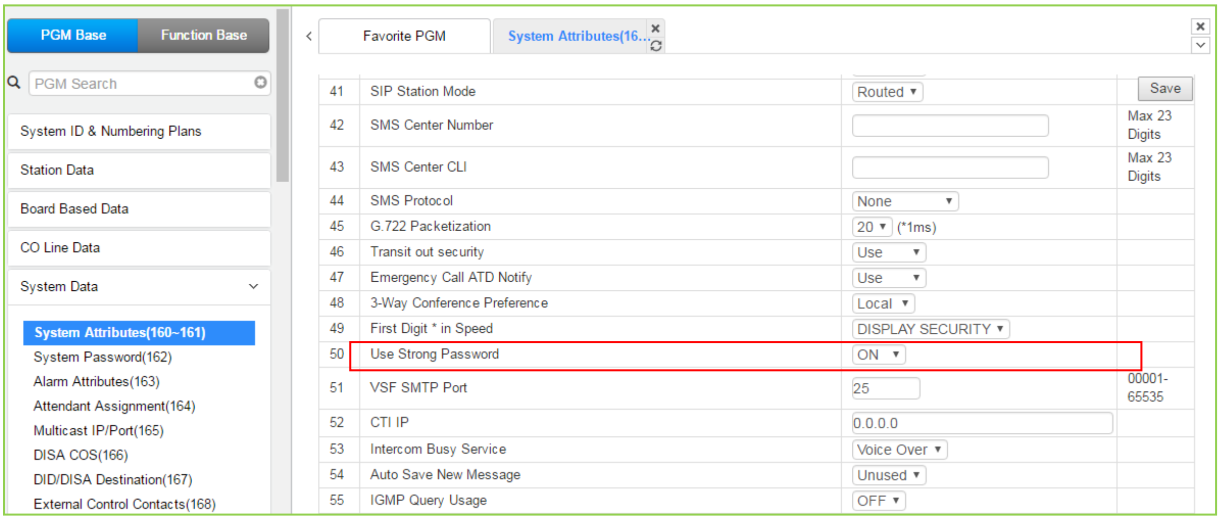
Password rule

SMB suggest to apply [Strong password] for WEB login, System password and Station User Login. [Strong password] has the rule consisting of length and combinations of alphabet, number and special character.

- Strong password usage option is set to ON by default.
  - System Attributes(PGM160-161) : [50] Use Strong Password - ON/OFF.
- Applied fields.
  - Keyset Admin, Remote Access and CID password in System Password(PGM162)
  - Station User Login in PGM443, Maintenance account password in Install Wizard
- Rule
  - Strong password consists of alphabet, number and special character
  - With 2 combinations: minimum length 10 or With 3 combinations: minimum length 8
  - Maximum length 12(*only available for WEB Password in Install Wizard and User Management)
  - Consecutive 3 numbers, alphabets are not allowed : 123(X), abc(X), aBc(OK)

Strong Password Enable

Strong password enabled by setting [Use Strong Password] to ON in PGM160-161.



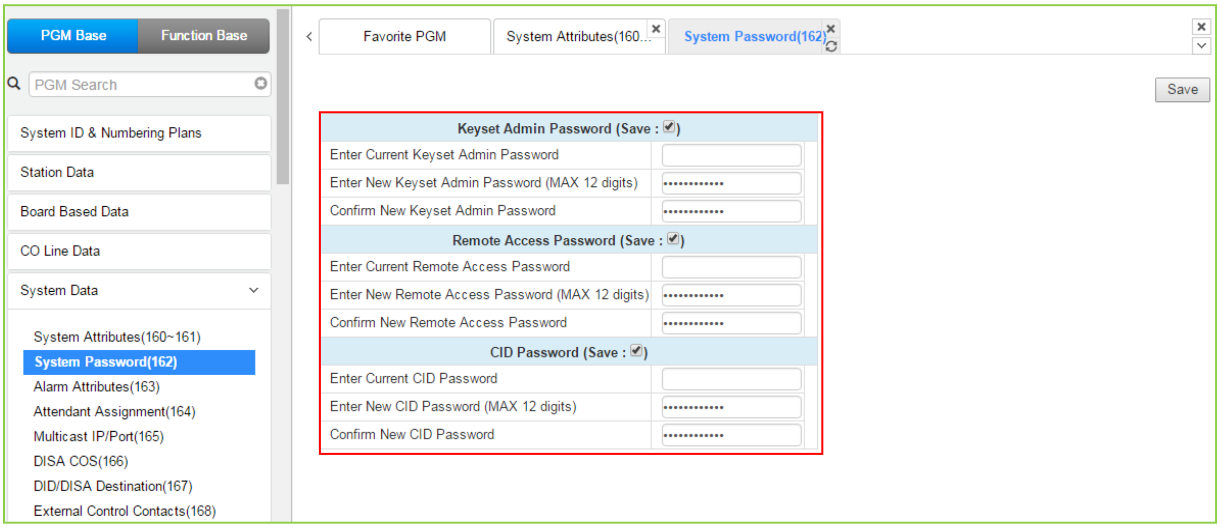
Fields of Strong password application

- System Password in PGM162.
  - Keyset Admin password:
    Case: When user try to enter admin using keyset by pressing * #
    Password: number, * and # (minimum length 8, max length 12)
  - Remote Access password:
    Case when user try to connect TCP/IP telnet connection to System
    Password: apply basic rule (max length 12)
  - CID password
    Case: when user try to admin CID/CLI related PGM fields like PGM113, 133, 151 and etc.
    Password: apply basic rule (max length 12)

- Station User Login in PGM443.
  - Station User Login
    Case: when user set ID/Password for SIP, IPCR and etc.
    Password: apply basic rule (max length 12)
- User Management / Maintenance account password in Install Wizard
  - User Management in Maintenance
    Case: when user set ID/Password for IPCR, UCS client and etc.
    Password : apply basic rule (max length 16)

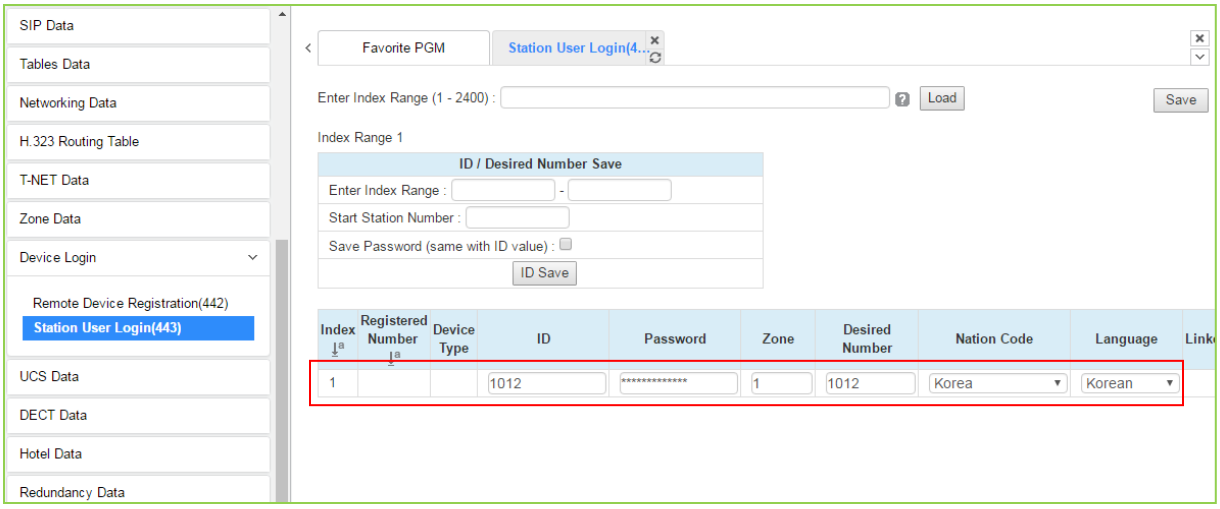
Strong password apply field

- System Password in PGM162.

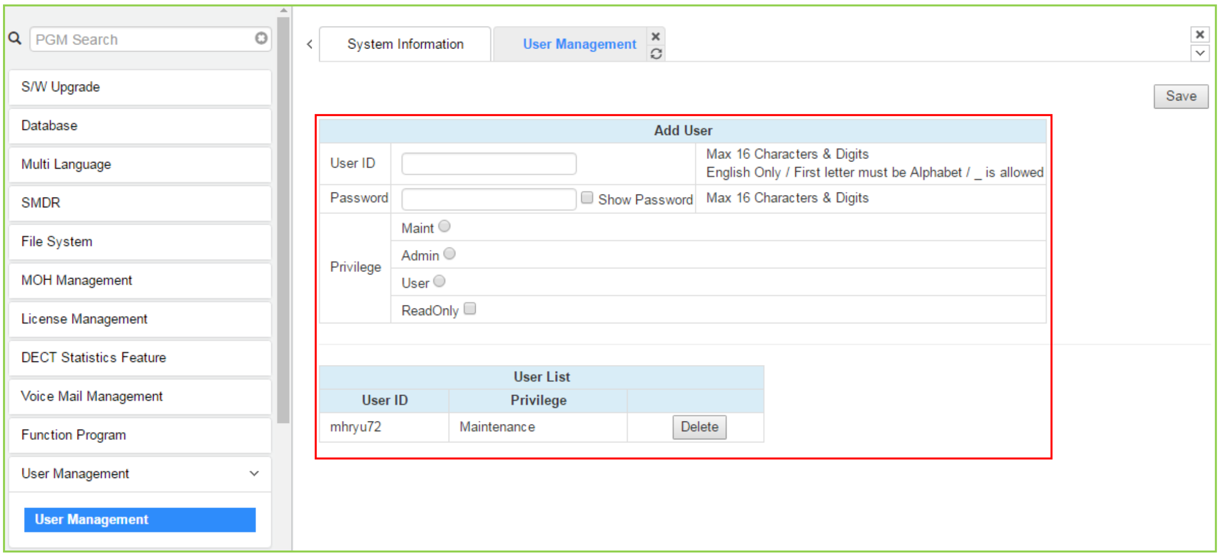


- **Keyset Admin password**: when user try to enter admin using keyset by pressing * #.
- **Remote Access password**: when user try to connect TCP/IP telnet connection to System.
- **CID password**: when user try to admin CID/CLI related PGM fields like PGM113, 133, 151 and etc.

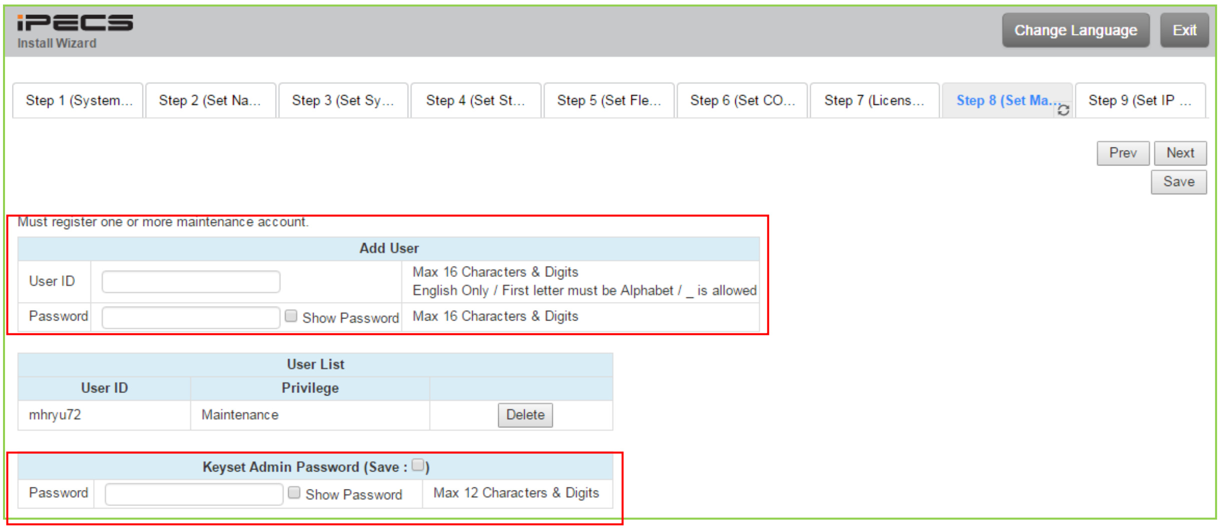
- Station User Login PGM443.



- User Management : Maintenance > User Management.



- Install Wizard : Set Maintenance password.

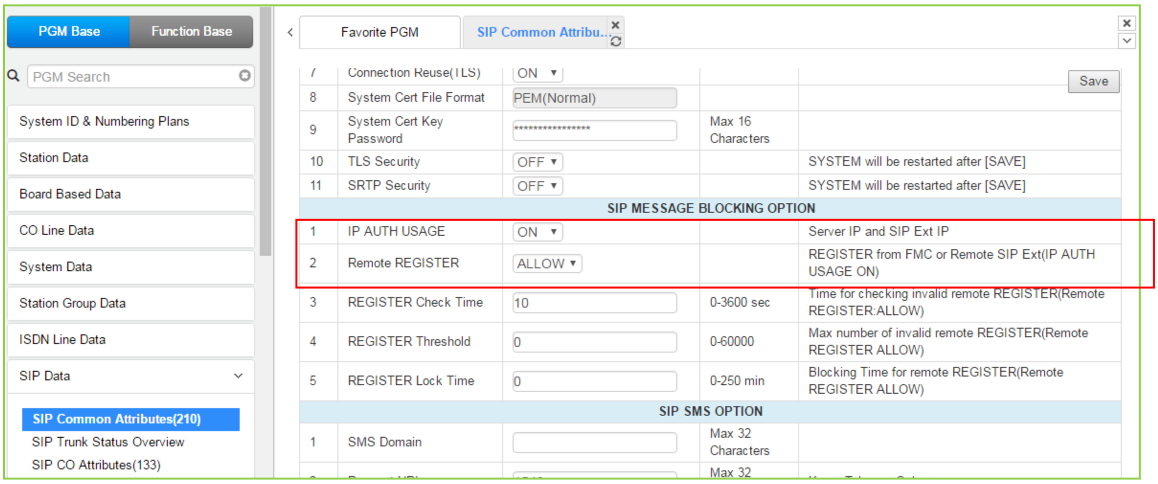


10.3 Security option for SIP

SMB provide Authentication and Register mode to protect unauthorized SIP call or register.

- IP AUTH USAGE option in SIP Common Attributes(PGM210).
  - With setting ON, system accept SIP message under proper condition.
- Remote REGISTER option in SIP common Attributes(PGM210).
  - It can protect Remote site REGISTER message by setting it ON or allow by setting it OFF.
- REGISTER threshold setting in SIP common Attributes(PGM210).
  - Time, Threshold and Blocking time with IP AUTH USAGE [ON] and Remote REGISTER [ALLOW].
- Require Authenticate via [407 Authentication] ON in SIP Phone Attributes(PGM211).
  - With setting ON, 401(Unauthorized)/407(Proxy Authentication Required) are used for SIP INVITE or REGISTER.

AUTH Usage, Remote Register option



When IP AUTH USAGE is "ON", system only respond to below cases.

- SIP message from.
  - Proxy Server IP or Domain in SIP CO Attributes(PGM133).
  - Registered SIP Extension.
  - Local Network : In the Same Subnet with System IP or in the System IP Range of PGM102.

Remote REGISTER option work with IP AUTH USAGE "ON".

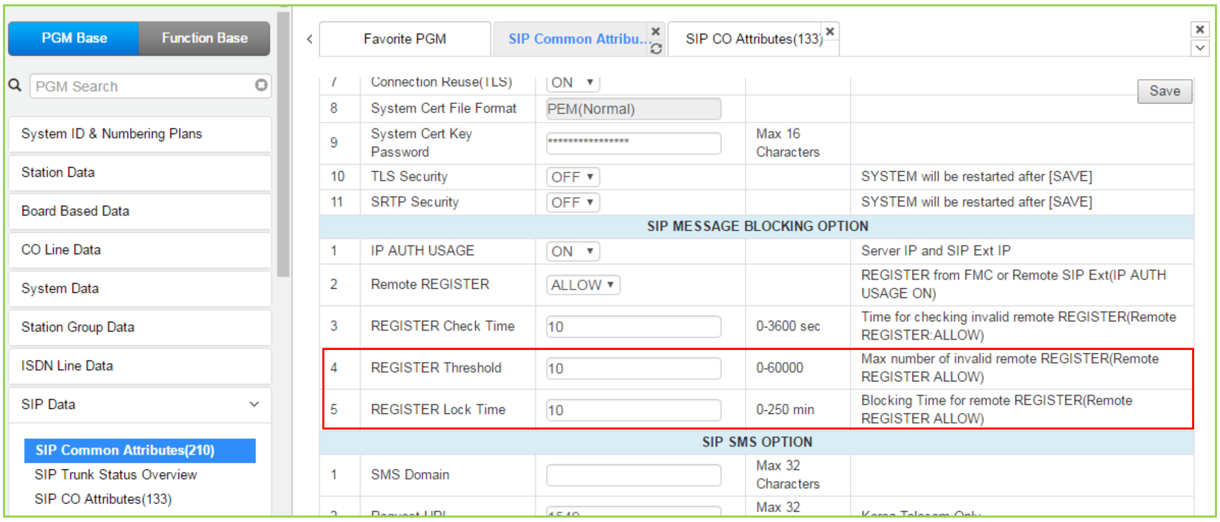
- [ALLOW] : can handle REGISTER message from Remote Site.
- [DENY] : ignore REGISTER message from Remote Site without any response.

REGISTER threshold setting

System can set time duration, trial times and blocking time of REGISTER from Remote Site.

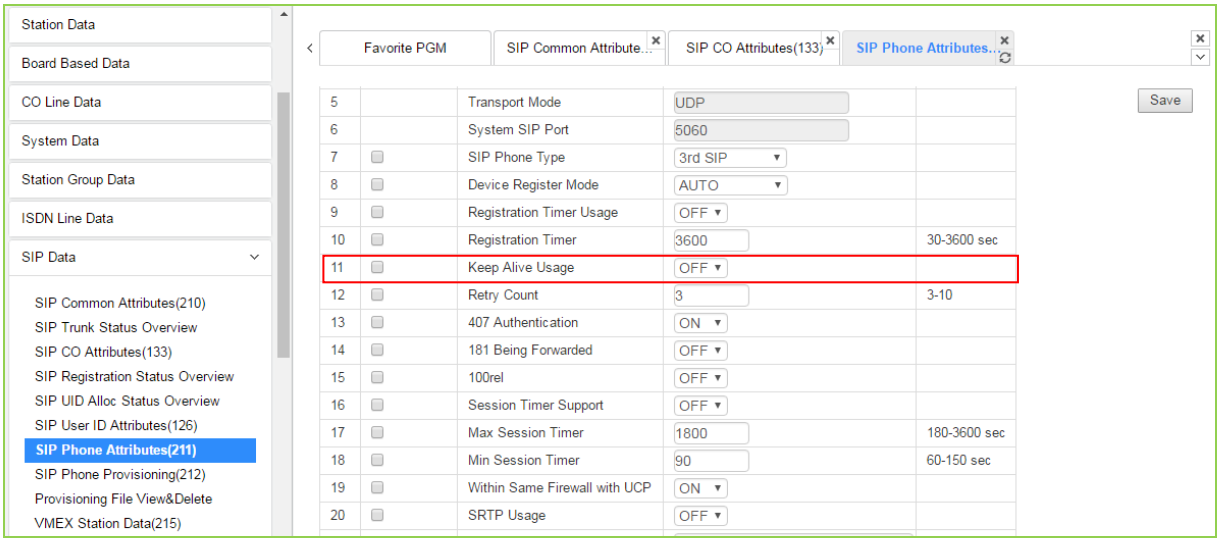
These three options work with IP AUTH USAGE [ON] and Remote Register [ALLOW].

In this case, if there are 10 times Invalid REGISTER from Remote Site for 10 seconds, system will ignore REGISTER from Remote site 10 minutes.



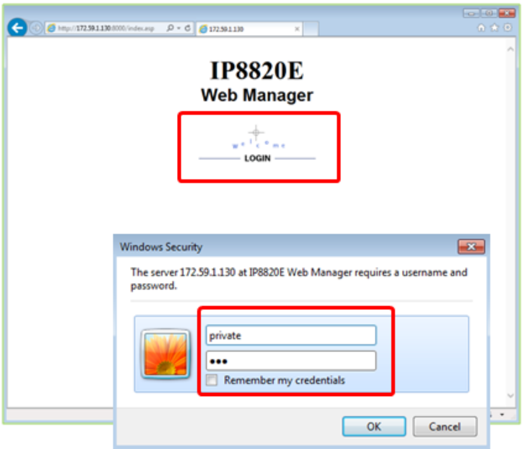
Require Authenticate

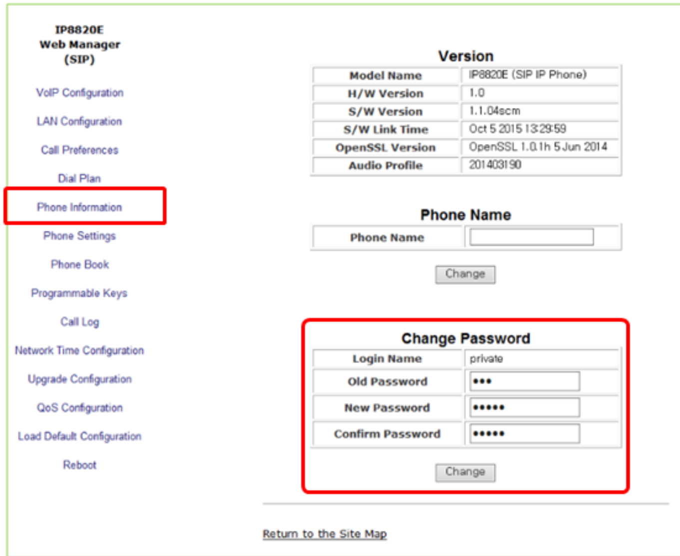
With setting ON, 401(Unauthorized)/407(Proxy Authentication Required) are used for SIP INVITE or REGISTER.



10.4 Security of SIP Phone (ELG)

Change WEB Password of SIP Phone





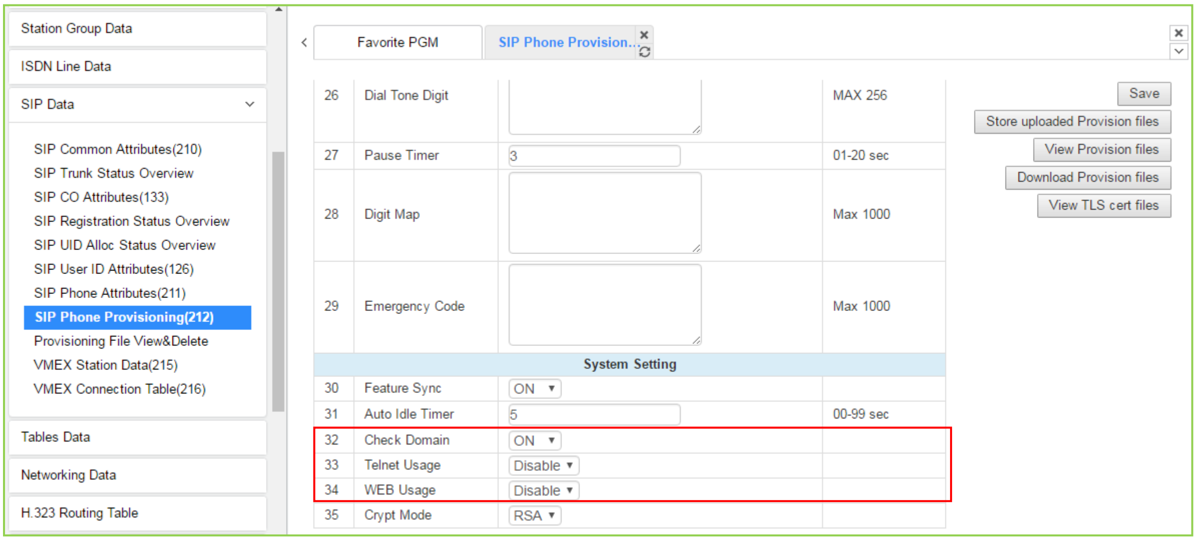
- **IP8800E SIP Phone WEB access:** http://172.59.1.130:8000 (port:8000).
- **Default ID/Password:** private/lip.

SIP Phone WEB log in password must be changed to block hacker's attack.

ELG SIP Phone Provisioning

System can control Telnet, WEB usage of ELG SIP Phone via provisioning.

Also incoming call from registered SIP phone can be checked with Check Domain option.



- **Check Domain (ON):** accept incoming call from only registered phone.
- **Telnet Usage (Disable):** Telnet will be blocked at SIP phone.
- **WEB Usage (Disable):** WEB will be blocked at SIP phone.

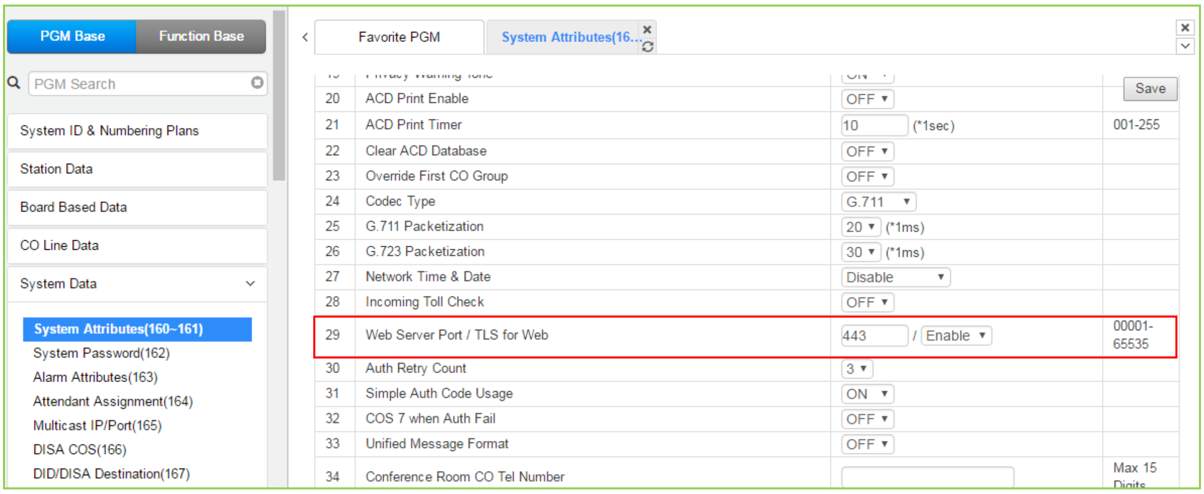
10.5 Security of WEB

TLS Encryption

SMB provide Encryption mode for WEB ID/Password and WEB Admin.

Web Server Port/TLS for WEB option in System Attributes(PGM160-161).

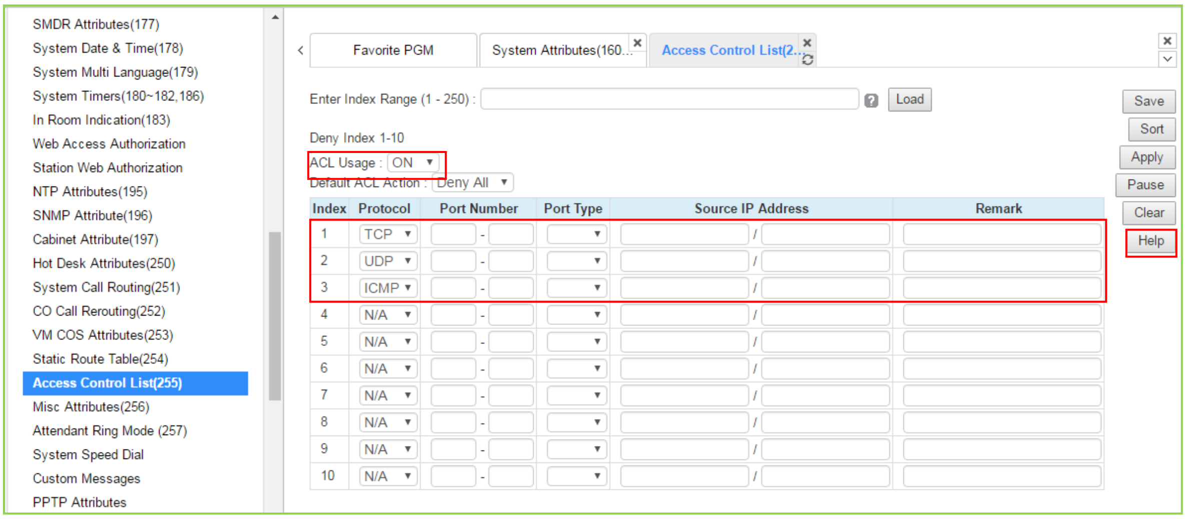
User can set Server port of WEB, also enable or disable TLS option.



10.6 Access Control

Access Control List

System can allow or deny packets by setting Protocol, Port #/source/destination and Source IP address to prevent unwanted network traffic in Access Control List(PGM255).



- **Exception:** ACL has no effect on SIP signal packet.
- **[Help] button :** Explain the meaning of each field - See the next page.

< ACL Usage >

Enable or disable the ACL function. The ACL in here is used to filter the inbound packet traffic. The inbound packets fromthe specified source address will be permitted or rejected according to "Default ACL Action" plan.

< Default ACL Action >

- Deny All: The only packets with in-range of the source address and protocol in ACL will be permitted. Other packets with different protocol or source address will be rejected.
- Allow All: Vice versa of "Deny All".
- Deny List: The protocol packets with in-range of the source address in ACL will be rejected. Other same protocol packets with out-of-range of the source address in ACL will be permitted. Filtering is not applied to the different protocol type or out of port range. So, other protocols not in list will be permitted. It operates same manner as “Allow All”.
- Allow List: The protocol packets with in-range of the source address in ACL will be permitted. Other same protocol packets with out-of-range of the source address in ACL will be permitted. Filtering is not applied to the different protocol type or out of port range. So, other protocols not in list will be permitted.

< Sort Button >

Sort ACL list.

< Apply Button >

Apply the current ACL rules to network stack.
There is a possibility of failed web connection with carelessly defined ACL rule and you can't proceed any web I/Fafter applying. Use the "acl stop" command at maintenance mode using the RS-232 connection to recover a faulty webconnection.

< Stop Button >

Delete all ACL rules from network stack.

< Clear Button >

Clear ACL list.

< Save Button >

Save the changes to DB file.

< ACL Setting >

1) Protocol.
   - It specify the destination protocol to apply.
2) Target Port.
   - Here two fields can be activated when the protocol is selected to TCP or UDP and specifythe destination port numbers.
   - Two input fields are for the start and end port numbers.
   - You can input the string "all" or "*" at one of two fields if you want to set all port range.
   - To specify the one port number range, you can input the number at any one field.
   - The valid range of input number is from 0 to 65535.
3) Source IP Address.
   - Here two fields are to input the source address. Two input combination can be done such as:

- Case 1: IP-address/subnet-mask
  First field is to input the IP address and next field is to the subnet mask.
  The blank of second field means one host network address.
  And the input of the string "all" or "*" at one of two fields means any address.
  The input of CIDR address in subnet mask field is available.
  So input number CIDR 24 will be expanded to 255.255.255.0.
- Case 2: start-IP-address/end-IP-address
  Two input fields can be filled with start and end address.

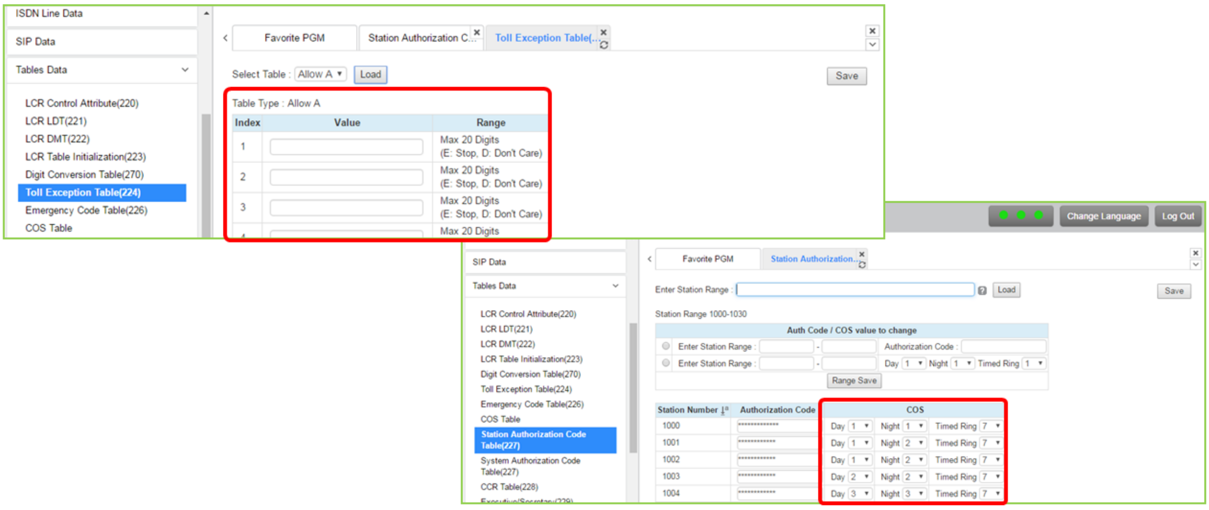
10.7 COS & Toll Control

COS and Toll Control

Control Class of Service for each Station and set Toll table to prevent unwanted call.

Allow or Deny entry can be set for COS level via Toll Exception Table in PGM224.

Control COS in Station Authorization Code Table(PGM227) or Station COS(PGM116).



10.8 Suggestion for Security

Suggested setting fields

For system security, refer to below table.

	Admin Field	PGM Code	Attribute
WEB Admin	System Data	System Setting(PGM160-161)	Set [Use Strong Password] to [ON]
		System Password (162)	Set Keyset Admin, Remote Access and CID password
		Web Access Authorization	Set proper authority to access admin field for 'User' and 'Admin' level
		System Attributes(160~161)	Enable TLS option : Web Server Port / TLS for Web
SIP Trunk	SIP Data	SIP Common Attributes(210)	Set [IP AUTH USAGE] to [ON]
		SIP CO Attributes(133)	Set [Invite Acceptance] to [Domain Only]
		Service Provider	Check IP address of Customer VOIP gateway along with ID & Password
SIP Extension	SIP Data	System Setting(PGM160-161)	Set [Use Strong Password] to [ON]
		Station User Login(443)	Set [User Password] as "Strong Password"
		SIP Phone Attributes(211)	Set [407 Authentication] to [ON] - registration and every invoke of call will be required authentication with SIP id & password
		SIP Common Attributes(210)	Set [IP AUTH USAGE] to [ON] - compare the real call signaling IP to registration IP of SIP phone
TLS for SIP (If need)	SIP Data	SIP Common Attributes(210)	Signal TLS OPTIONS : TLS Version, Crypt Mode, TLS Security and etc
		SIP CO Attributes(133)	Connection Mode - TLS, Proxy Server TLS Port
ACL (If need)	System Data	Access Control List (255)	IP of all packets will be allowed or denied

- Maintenance > Trace > HTTP Log View : save 100 records of WEB log-in history.
- Maintenance > Trace > WEB Access Log : save 300 records of program change history.

Recommend to use firewall

The best way to secure a system is to install a firewall in front of the system and to secure that as the first line of defense then to secure the routers then to secure the system.

A firewall will prevent the systems form being discovered on the internet limiting risk of any hacking attempts.

Thanks for purchasing iPECS system!