

# **iPECS**

## **iPECS eMG80**

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# **Feature Description & Operation Manual**

Please read this manual carefully before operating System.

Retain it for future reference.

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## ***Revision History***

<b>Issue</b>	<b>Date</b>	<b>SW</b>	<b>DESCRIPTION OF CHANGES</b>
1.0	Sep., 2013	1.0A	Initial Release

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# 1. INTRODUCTION

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## 1.1 Manual Application

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This document provides detailed information covering description and operation of the numerous features available in the iPECS eMG80 Release 1 system software. The document is written assuming the system employs the default-numbering plan for North America.

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## 1.2 Manual Organization

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

- 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
  - Section 8 SIP Extension
- 

## 1.3 Feature Information

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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 phones (LDP digital and LIP IP phones) and SLTs. In some cases a Flexible button on the Ericsson-LG 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 eMG80 Administration and Programming manual. Regarding WEB admin, you directly access to the web admin and check the programming.
  - **Reference:** lists related topical information to aid in understanding the feature.
  - **Hardware:** lists hardware required for proper feature operation.
-

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## ***1.4 Buttons & Term Variations***

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The eMG80 supports proprietary digital and IP phones. The proprietary models include the LDP-7000 series, LDP-9000 series, the LIP-7000 series and the LIP-8000/8000E 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 eMG80 as used in the respective manual such as "VSF" (used for voice mail) and "CO/IP lines" for outside Lines.

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## 2. SYSTEM

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### 2.1 8-Digit Station Numbering Plan

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#### Description

The system supports Station Numbering Plans from two to eight digits. The Station number consists of a Prefix and suffix called Add Digits. A Prefix digit string can be any four digits that do not conflict with the system Flexible Numbering Plan used for feature codes. This fixed digit string is added to the Add Digits string, which can be 2 to 4 digits in length, to uniquely identify each Station.

#### Operation

##### System

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

#### Conditions

1. 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 3<sup>rd</sup> party software to support the defined Station Numbering Plan.
2. The 8-digit numbering is configured in the 8-Digit Table (PGM 238).

#### Programming

<b>System ID</b>	Prefix Usage (PGM 100)
	8-Digit Table (PGM 238)

#### Related Features

- Station Numbering Plan

#### Hardware

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## 2.2 Account Code

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### Description

You may allow tracking of specific calls by entering a non-verified variable length (up to 12 digits) identifier for a call. The identifier or “Account Code” is output as part of the Station Message Detail Record (SMDR) for the call as discussed in Section 2.70. 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,

Press the {ONE-TOUCH ACCOUNT CODE} button.

### SLT

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**

1. When entering an Account Code during a call, DTMF digits are transmitted to the connected party.
2. A maximum of 1000 **{ONE-TOUCH ACCOUNT CODE}** buttons may be assigned in the system.
3. 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.
4. 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**

<b>NUMBERING</b>	Flexible Numbering Plan (PGM 106-Button 8)
<b>STATION</b>	Forced SMDR Account Code (PGM 113-Button 24-21)

### **Related Features**

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

### **Hardware**



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## 2.3 Alarm Signal/Door Bell

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### Description

The system can be configured to recognize the status of an external contact (normally open or closed). The system will signal assigned iPECS IP and LDP Phones when the contact activates. This capability 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-minute intervals or a continuous tone. The Alarm Signal may be terminated at the phone by dialing the Alarm Stop code or, if assigned, pressing the **{ALARM STOP}** button. To rearm the Alarm 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 the external 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 STOP}** 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 STOP}** button.

### Conditions

1. The Alarm contacts must be "dry", no voltage or current source connected.
2. Only iPECS IP and LDP Phones may be assigned to receive Alarm/Door Bell signals.
3. The LCD of phones assigned to receive Alarm/Door Bell signals will show "ALARM" or "DOOR BELL" as appropriate.
4. The Alarm Stop code is also employed to terminate alarm signals from digital and IP trunk alarms.

### Programming

<b>SYSTEM</b>	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)
<b>STATION</b>	Alarm/Door Bell Attribute (PGM 113-Button 8)

### **Related Features**

- Door Open

### **Hardware**

- iPECS IP or LDP Phone
- Alarm or Doorbell input contact, refer to the eMG80 Hardware Description & Installation Manual to install.

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## **2.4 Authorization Codes (Password)**

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### **Description**

Authorization Codes provide a means to control access to Voice Mail, Off Premise Call Forward, Walking COS (Class of Service), or DISA and may be required for outgoing outside Line or LCR access based on configuration of the eMG80 system database. When you dial an Authorization Code that matches an Authorization 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 Code is specifically related to a given station and intended for a single user. The System Authorization Codes are intended for use by any station in the system.

The Station Authorization Codes includes the associated station number and the assigned code. 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 employed using entry of table index to provide a unique identification of the entry.

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

### **Operation**

#### ***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**

##### *To assign a Station Authorization code:*

1. Lift the handset.
2. Dial SLT 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**

1. 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.
2. 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.
3. A user may enter an Authorization Code from any station to place an outside call using Walking COS (Class of Service).
4. The system provides memory for up to 500 Authorization codes, 140 for station codes and the remaining 360 are system codes.
5. 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 Retry Count.
6. If the Check Password option is enabled in the LCR database, when dialed digits

match the LDT table digits, the system will send second dial tone to request that you input a valid Authorization code.

7. An Authorization code may include any dial pad digit except "\*" and "#".
8. 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.
9. 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).
10. 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**

<b>STATION</b>	Station Account (PGM 112-Button 19) CO/IP Group Authorization (PGM 141-Button 8) DISA Authorization Code (PGM 142-Button 5)
<b>SYSTEM</b>	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)
<b>TABLES</b>	LCR Check Password (PGM 221-Button 6) Station Authorization Code Table (PGM 227)

## **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**

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## **2.5 Auto Call Recording**

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### **Description**

Your iPECS IP or LDP Phone can be configured to automatically record calls to a Voice Mailbox or the hard disk drive of an iPECS Phontage or UCS Client soft phone. 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 the Phontage or UCS Client, recordings are managed directly by the Phontage or UCS Client with the ability to listen to, delete or send the recording to others via e-mail.

### **Operation**

Recording of calls is automatic when assigned.

To manage the recordings, use the procedures outlined in the Phontage or UCS Client User Guide.

### **Conditions**

1. The Phontage or UCS Client can record one call at a time and must be idle. While recording, if the Phontage or UCS Client places or receives a call, recording terminates.
2. When call recording begins, you and the connected party will receive a Call Recording confirmation tone.
3. The Phontage or UCS Client used for recording must be operating in a 'Local' mode not remote.

### **Programming**

<b>STATION</b>	Two-way Record Privilege (PGM 112-Btn11)
	Auto Call Recording (PGM 112-Button 20)
	Call Recording Destination (PGM 112-Button 21)

### **Related Features**

### **Hardware**

- iPECS IP or LDP Phone, or PC with Phontage or UCS Client

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## **2.6 Auto Call Release**

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### **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 a pre-determined time.

### **Operation**

#### **System**

##### Auto Call Release of Intercom calls:

1. If you place an intercom call and the called station does not answer in the Intercom Call Release Time, the call is terminated and you will receive error tone.

##### Auto Call Release of outside calls:

1. If you seize an idle Line and do not dial within the Automatic Release Time, the call is terminated and you will receive error tone.

### **Conditions**

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

### **Programming**

<b>STATION</b>	Howling Tone to Station (PGM 111-Button 5)
<b>SYSTEM</b>	CO Call Automatic Release Timer (PGM 180-Button 12)
	STA Call Automatic Release Timer (PGM 182-Button 4)

### **Related Features**

- Howler Tone

### **Hardware**

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## **2.7 Automatic Call Distribution (ACD)**

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### **Description**

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

Calls route to an ACD group directly or otherwise including via call transfer, CCR and ICLID 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**

1. The conditions of System Announcements and ICLID Call Routing apply.
2. 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.
3. 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.
4. The caller may correct an entry by dialing '\*' to erase all digits entered then dial the correct entry.

#### **Programming**

<b>STATION GROUPS</b>	ACD Group (PGM 191) ICLID Usage (PGM191-Button 21)
<b>TABLES</b>	ICLID Route Table (PGM 203)



## ICLID Ring Assignment (204)

### 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 {GROUP STATUS} button at the Supervisor

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

To output ACD Statistics Report

1. Lift the handset or press the **[SPEAKER]** button,
2. Press the **{GROUP 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 **{GROUP 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**

1. 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.
2. 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**

- |                       |   |
|-----------------------|---|
| <b>NUMBERING PLAN</b> | ACD Group Supervisor Status (PGM 107-Button 7)                  |
| <b>STATION GROUPS</b> | ACD Group (PGM 191)<br>ACD Group Supervisor (PGM 191-Button 18) |

**Related Features**

- Station Groups

**Hardware**

## **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 calls manually. The Agent receives a brief tone (Zap Tone), if assigned, and then the call is connected to the agent.

#### **Operation**

##### **System**

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

#### **Conditions**

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

#### **Programming**

<b>STATION GROUPS</b>	ACD Group (PGM 191)
	Zap Tone (PGM 191-Button 24-1)

#### **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-UP-END} button:

[TRANS/PGM] + {FLEX} + “585” + ACD Group Number + [HOLD/SAVE]

To manually deactivate the Wrap-Up state:

1. Dial “585” the Wrap-Up-End code.
- Or,
- Press the {WRAP-UP-END} flexible button.

#### Agent SLT

To manually deactivate the Wrap-Up state:

1. Dial “585” the Wrap-Up-End code.

#### Conditions

1. During Wrap-Up, the assigned {WRAP-UP-END} 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

<b>NUMBERING PLAN</b>	ACD Agent Wrap-Up-End (PGM 109-Button 9)
<b>STATION GROUPS</b>	ACD Group (PGM 191)
	ACD Wrap-up Timer (PGM 191-Button 9)

#### Related Features

#### Hardware

### 2.7.3.3 Agent Help Request

#### Description

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

#### Operation

##### *Agent iPECS IP & LDP Phone*

To assign **{HELP}** buttons:

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

To request Supervisor assistance while on an active ACD call,

1. Press the **{HELP}** button.

##### **SLT**

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

1. Agents receive error tone to indicate there is no active Supervisor.
2. Using the **{HELP}** button during a call, no signals are sent on the active conversation and the connected party is unaware of the request for assistance.

#### Programming

<b>NUMBERING PLAN</b>	ACD Group Help (PGM 107-Button 5)
<b>STATION GROUPS</b>	ACD Group (PGM 191)
	ACD Supervisor (PGM 191-Button 18)

#### Related Features

- Station Groups
- Supervisor Help Response

#### Hardware

### 2.7.3.4 Agent Id Login/Logout

#### Description

Stations or 4-digit Agent Ids 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**

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**

1. The system outputs ACD events including Login/Logout messages when Discovery Manager Print (Output ACD Event Message) is enabled.
2. The Agent Id can be any 4-digit number (0000-9999). iPECS does not verify the Agent Id, other than requiring that four digits be entered.
3. Any Agent Id may be used to login except at a Hot Desk Station where a Station Authorization code is required.

**Programming**

<b>NUMBERING PLAN</b>	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)
<b>STATION GROUPS</b>	ACD Group (PGM 191)
<b>SYSTEM</b>	ACD Manager Print (Output ACD Event Message) (PGM 160-Button 13)

Unified Message Format (PGM 161-Button 18)

**Related Features**

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

**Hardware**

**2.7.3.5 Agent On/Off Duty w/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 ON/OFF DUTY (GROUP)}** 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 ON/OFF DUTY}** button.
  2. Dial the ACD group number unless assigned in the button

To assign an **{ACD 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 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.
- Or,
- Press **{ACD ON/OFF DUTY ALL GROUPS}** button.

**SLT**

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

1. 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.
2. The **{ACD 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.
3. 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.
4. Only one **{ACD ON/OFF-DUTY}** button with the Group number can be assigned at a station.
5. The default value of the ACD DND Wrap-Up Timer is 10 seconds.

## Programming

<b>NUMBERING PLAN</b>	ACD Agent On/Off Duty (PGM 107-Button 2) Agent ON/OFF duty in all hunt group (PGM 109-Button 18)
<b>STATION</b>	Auto ACD DND Reason (PGM 113-Button 15)
<b>STATION GROUPS</b>	ACD Group (PGM 191) ACD DND Wrap-up Timer (PGM 191-Button 20)

## 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 or Wrap-up status. In addition, an active Agent can view the queued call status for an ACD group while on a call using a **{Display Call Queue}** Flex button. The Call Queue display indicates the ACD group number, the number of calls in queue and the longest queue time.

## Operation

### *Agent iPECS IP & LDP Phone*

To assign a {DISPLAY CALL QUEUE} 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 {DISPLAY CALL QUEUE} 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 {DISPLAY CALL QUEUE} button, the queue status displays for several seconds then returns to the normal call display.

## Conditions

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

## Programming

<b>NUMBERING PLAN</b>	Display ACD Calls In Queue Status (PGM 107-Button 6)
<b>STATION GROUPS</b>	ACD Group (PGM 191)

## 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 secondary announcements are available with control timers. The first announcement is played after 1<sup>st</sup> control timer. The 2<sup>nd</sup> control timer determines the period between the first and second announcement. The second announcement can be replayed at defined intervals. If desired, the primary 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 to the ACD group. The CIQ announcement (“Your Call is number xx in Queue”) is played to queued callers after the 1<sup>st</sup> announcement is played and again after each time the 2<sup>nd</sup> announcement is played. Internal callers using an iPECS IP or LDP phone with Display also receive the “You are 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 (“01”-“70”)
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 (“01”-“70”).
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**

1. ACD group announcements are recorded in the integrated Auto Attendant/Voice Mail. Up to seventy (70) Announcements are available for group announcement recordings. Announcements 71 and 72 are reserved for MOH and Multi Language selection announcements, respectively.
2. 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)
3. To define a guaranteed announcement, assign 0 seconds to the 1st announcement control timer.
4. The sequence number for the voice storage device is given in the Administration and Programming Manual System Overview Web page.

## **Programming**

<b>STATION GROUPS</b>	ACD Group (PGM 191)
	ACD ICLID Routing (PGM 191-Button 21)
	CIQ Mention (PGM191-Button 24-21)

## **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-Answer timer is placed in a “No response” unavailable status, called ACD DND. In this state, the Agent is Off-duty and will not receive ACD calls from the group. If the Auto DND Reason code is “0”, after the ACD DND 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”, the Agent On/Off Duty Code.
  2. Dial the ACD group number.
- Or,
1. Press the **{ACD ON/OFF-DUTY}** button.
  2. Dial the ACD group number unless assigned in the button.

#### SLT

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

1. The system will output an Agent Available message with a Reason code when an Agent changes availability status.
2. 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

<b>NUMBERING PLAN</b>	Agent On/Off Duty (PGM107-Button 2)
<b>STATION</b>	Auto ACD DND Reason (PGM 113-Button 15)
<b>STATION GROUPS</b>	ACD Group (PGM 191)
	ACD No Answer Timer (PGM 191-Button 24-22)

### Related Features

- Agent On/Off Duty w/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 music on hold may be sent to the caller. The caller may dial a digit at any time while queued to exit the queue, except during a Guaranteed Announcement. The dialed digit is compared to digits defined in the ACD Group CIQ Route Table. If a match is found, the call is routed to the defined destination (Station, Station Group, etc.). If a match is not found, external callers receive an error 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**

1. 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 2.7.6.
2. Dialing during ring back tone is not recognized by the system.

### **Programming**

<b>STATION GROUPS</b>	ACD Group (PGM 191) ACD CIQ Route Table (PGM 191-Button 23)
-----------------------	--

### **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 that an assigned System announcement plays over a defined Page zone, if the number of calls in 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 the number of Calls in Queue falls below the threshold. The system allows different treatment and announcements for up to three different Calls-In-Queue thresholds.

### **Operation**

#### **System**

When the number of Calls-In-Queue changes, up or down the iPECS:

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 ("01"- "70")
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 ("01"- "70")
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**

1. 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.
2. The System has seventy (70) announcements employed for Auto Attendant, ACD groups, CCR, Page Alerts, etc. Two announcements are reserved for MOH and multi-language support, announcements 71 and 72 respectively.

## **Programming**

<b>STATION GROUPS</b>	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)

## **Related Features**

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

## **Hardware**

## **2.7.8 Event Messages**

### **Description**

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

### **Operation**

#### **System**

When enabled, Event messages are sent automatically

### **Conditions**

### **Programming**

<b>STATION GROUPS</b>	ACD Group (PGM 191)
<b>SYSTEM</b>	ACD Pack (PGM 175-Flex button 1-Button 9) ACD Manager Print (Output ACD Event Message) (PGM 160-Button 13)

### **Related Features**

- Station Groups

### **Hardware**

## **2.7.9 Group Mail Box**

### **Description**

A Voice mailbox can be associated with a Station Group. Calls to the group are assigned to overflow and 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 Station Group number as the station number and the defined Group Mailbox Password. Messages can 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} + 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**

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

### **Programming**

<b>STATION GROUPS</b>	Station Group Attributes (PGM 191)
	Mailbox Message Wait Station (PGM 191-Button 24-2)
	Mailbox Password (PGM 191-Button 24-3)

### **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 hunt group is assigned a name of up to 12 characters. The name is employed as the called party display for internal callers and in the output of group statistics.

### **Operation**

#### ***System***

*Operation of this feature is automatic when programmed.*

### **Conditions**

### **Programming**

<b>STATION GROUPS</b>	Station Group Attributes (PGM 191)
	ACD Group Name (PGM 191-Button 22)

### **Related Features**

- Station Groups

### **Hardware**

## **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 all agents are busy, the call is queued to the group or routes to the Alternate Destination. The queued 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 tone and, 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 idle the longest.

If there are no Agents active in the group when a call arrives, the call will receive the “No Member” treatment. Intercom calls receive re-order and external callers are routed to an Attendant.

### **Operation**

#### **System**

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

#### **Conditions**

1. The Alternate ACD Destination can be a group or a station but cannot be a member of the ACD group.
2. 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.
3. If an agent becomes available during an announcement, except a guaranteed announcement, the call is passed immediately to the agent.

### **Programming**

<b>STATION GROUPS</b>	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)

### **Related Features**

- Automatic Call Distribution

### **Hardware**

## 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 for training purposes or to assist the Agent. When used, the Supervisor is connected to the call with the microphone muted, the **[MUTE]** button LED is On. If ACD Warning Tone is enabled, a Warning Tone is provided to the Agent's call indicating the Supervisor has entered the call. The Supervisor 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 **{AGENT MONITOR}** button to activate the Agent Call Monitor feature.

#### Operation

##### *Supervisor iPECS IP & LDP Phone*

To assign an **{AGENT 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 **{AGENT 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

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

#### Conditions

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

#### Programming

<b>NUMBERING PLAN</b>	ACD Supervisor Monitor (PGM 107-Button 8)
<b>STATION GROUPS</b>	ACD Groups (PGM 191)
	ACD Group Supervisor (PGM 191-Button 18)
	ACD Warning Tone (PGM 191-Button 12)

#### 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 **{GROUP STATUS}** button.

#### Operation

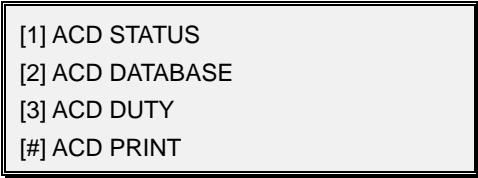
##### *Supervisor iPECS IP & LDP Phones*

To assign a **{GROUP STATUS}** button at the Supervisor

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

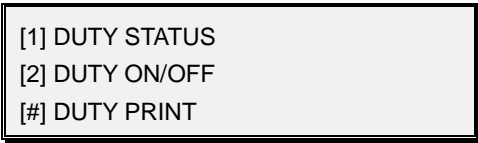
To view/control Agent Status

1. Lift the handset or press the [SPEAKER] button,
2. Press the **{GROUP STATUS}** button,



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

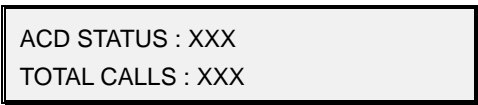
3. Dial "3", the Agent Duty code, the LCD menu will update:



[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**

1. 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**

<b>NUMBERING PLAN</b>	ACD Group Status (PGM 107-Button 7)
<b>STATION GROUPS</b>	ACD Group (PGM 191) ACD Group Supervisor (PGM 191-Button 18)

**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 is assigned. 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 and
- Agents in service and available

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



## Operation

### *Supervisor iPECS IP & LDP Phone*

To assign a {GROUP STATUS} button at the Supervisor

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

To display the Group status when the {GROUP 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,
2. Press the flashing {GROUP STATUS} button,
  3. Dial "1" to see the Group Status display.

## Conditions

1. 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.
2. The {GROUP 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

<b>NUMBERING PLAN</b>	ACD Group Status Code (PGM 107-Button 7)
<b>STATION GROUPS</b>	ACD Group (PGM 191)
	ACD Group Supervisor (PGM 191-Button 18)
	Supervisor Timer (PGM 191-Button 14)
	Supervisor Queued Call Count (PGM191-Button 15)

## 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 in real-time. A **{GROUP 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 **{GROUP 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 **{GROUP 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

1. 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

<b>NUMBERING PLAN</b>	ACD Group Status (PGM 107-Button 7)
<b>STATION GROUPS</b>	ACD Group (PGM 191)
	ACD Group Supervisor (PGM 191-Button 18)

## 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 Group Help Code, default 574 or **{HELP}** 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 **{HELP}** button

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

To request Supervisor assistance,

1. Press the **{HELP}** button.

##### **SLT**

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

1. The Agent Help request is sent to the lowest numbered available Supervisor station for the associated ACD Group.
2. If no Supervisor is available, the Agent receives error tone.
3. A Warning Tone, if enabled, will be sent to the Agent as the Supervisor is connected to the conversation.
4. Only one active Help request is allowed at a time, a second help request will cancel any active help request.

## Programming

<b>NUMBERING PLAN</b>	ACD Group Help Code (PGM 107-Button 5)
<b>STATION GROUPS</b>	ACD Groups (PGM 191)
	ACD Group Supervisor (PGM 191-Button 18)
	ACD Warning Tone (PGM 191-Button 12)

## 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 {LOGIN} or {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 **{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 **{LOGOUT}** button.
2. Dial supervisor Id code (“0000”-“9999”)

**Conditions**

1. The system will output a Login/Logout message when a Supervisor logs-in or out.
2. The Supervisor Id can be any 4-digit number (“0000”-“9999”).
3. The system does not verify the Supervisor Id codes, other than requiring four digits are entered.
4. Any Supervisor Id may be used to login through stations not assigned as a Hot Desk.
5. 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**

<b>NUMBERING PLAN</b>	Supervisor Login (PGM 107-Button 3) Supervisor Logout (PGM 107-Button 4)
<b>STATION GROUPS</b>	ACD Groups (PGM191) ACD Group Supervisor (PGM 191-Button 18)

**Related Features**

- Automatic Call Distribution
- Hot Desk

**Hardware**

- iPECS IP or LDP Phone

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## ***2.8 Automatic Pause Insertion***

---

### **Description**

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

- 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**

1. An automatically inserted pause is not counted as a digit in a Speed Dial number.
2. 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**

**SYSTEM**                      Pause Timer (PGM 181-Button 10)

### **Related Features**

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

### **Hardware**

---

## 2.9 Automatic Privacy

---

### Description

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

### Operation

#### *iPECS IP & LDP Phones*

##### To intrude into a call when Privacy is disabled

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

### Conditions

1. With Automatic Privacy disabled, privacy is still assured on all intercom and conference calls.
2. 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.
3. Only one station can intrude on an active call.
4. An intrusion tone can be provided to the call indicating another station has accessed the Line.
5. If either internal party presses another **{LINE}**, a **{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

<b>STATION</b>	Override Privilege (PGM 113-Button 4)
<b>SYSTEM</b>	Automatic Privacy (PGM 161-Button 3)
	Privacy Warning Tone (PGM 161-Button 4)

### Related Features

- Multi-Party Voice Conference
- Station Flexible Buttons

### Hardware

- iPECS IP or LDP Phone

---

## **2.10 Auto Service Mode Control**

---

### **Description**

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

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.

### **Conditions**

#### **Programming**

<b>STATION</b>	Station COS (Class of Service) (PGM 116)
<b>CO/IP</b>	CO/IP CO Ring Assignment (PGM 144) CO Line COS (Class of Service) (PGM 141-Button 2)
<b>SYSTEM</b>	DISA COS (Class of Service) (PGM 166) External Control Contact (PGM 168) PABX Access Codes (PGM 172) LD Digit Count (PGM 177-Button 3) LD Digit Code (PGM 177-Button 19)
<b>TABLES</b>	Toll Restriction Tables (PGM 224) Authorization Codes Table (PGM 227) Weekly Time Table (PGM 233)

#### **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

#### **Hardware**



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## **2.11 Automatic System Daylight Saving Time**

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### **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 terminals and 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**

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

### **Programming**

<b>SYSTEM</b>	System Time (PGM 178-Button 1)
	System Date (PGM 178-Button 2)
	DST Enable (PGM 178-Button 3)
	DST Start & End Time (Web only)

### **Related Features**

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

### **Hardware**

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## **2.12 Automatic Time Synchronization**

---

### **Description**

When enabled, the system automatically determines and sets the time of day employing Network Time Protocol (NTP) or ISDN time messages. When using NTP, the system, 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 server automatically. If the time deviates more than two (2) seconds, the system clock is adjusted to match the NTP server.

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

### **Operation**

#### **System**

Operation of this feature is automatic.

### **Conditions**

1. NTP packets are expected over UDP port 123. Assure this port is open and available.
2. A secondary NTP server address can be defined should the first server not respond.
3. 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**

<b>SYSTEM</b>	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)

### **Related Features**

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

### **Hardware**

---

## **2.13 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**

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

### **Programming**

<b>STATION</b>	CO Call Time Tone (PGM 112-Button 1)
<b>SYSTEM</b>	Elapsed Call Tone Timer (PGM 180-Button 19)

### **Related Features**

- Outside Call Time Restriction

### **Hardware**

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## **2.14 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 Call Forward code as follows:

- Code 0: Remote Call Forward, forwards all calls to the station, except recalls, activated from a remote station, Call Forward, Follow-me.
- Code 1: Unconditional, all calls to the station, except recalls, are forwarded internally or externally immediately upon receipt.
- Code 2: Busy, if the station is busy, forwards all calls, except recalls, to the selected station.
- Code 3: No Answer, forwards all calls, except recalls, to the selected station when the station does not answer within the No Answer timer.
- Code 4: Busy/No Answer, forwards calls if the selected station is busy or does not answer within the No Answer timer.
- 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.
- Code 7: Off Net Busy, forwards all calls, except recalls, to the selected station when station is busy, only SLT.
- 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.
- 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.

### **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,

1. Dial an outside Line access code (9, 8xx, 88xx) and the desired external phone number.

2. Press the **[HOLD/SAVE]** button to save.  
Or,
3. Press the **[SPEED]** button and dial the desired bin number.
4. 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
6. Dial an outside Line access code (“9”, “8xx”, “88xx”) and the desired external phone number.
7. Press the **[HOLD/SAVE]** button to save.  
Or
8. Press the **[SPEED]** button and dial the desired bin number
9. 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,
3. Press the # dial pad button.

**SLT**

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.
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**

1. A station receiving a forwarded call can transfer the call to the forwarding station.
2. A station, denied the use of Call Forward, will receive error tone in response to attempts to activate Call Forward.
3. A forwarded intercom call will signal the receiving station in the Tone Signaling mode, regardless of the Intercom Signaling Mode at the station.
4. You cannot forward calls to a station in DND and, if attempted, error tone is returned.
5. 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.
6. When Call Forward is active, you can make outgoing calls (internal or external) but cannot activate a Call back or Queue request.
7. 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.
8. For outside calls, manually activated Call Forward will override any Preset Call Forward assignment.
9. 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.
10. Call Forward status is maintained in the system’s non-volatile memory for protection from power outage.
11. A station in a Station Hunt 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 but routes hunt calls to the station based.

12. 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.
13. 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.
14. If the Speed Dial number used in Off-Net Call Forward contains a Flash, only digits prior to the Flash are dialed.
15. 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.
16. 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.
17. 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**

<b>STATION</b>	Call Forward (PGM 111-Button 2) Station Call Forward No Answer Timer (PGM 123-Button 1)
<b>SYSTEM</b>	System Call Forward No Answer Timer (PGM 181-Button 1)
<b>STATION GROUPS</b>	Member Forward (PGM 191-Button 14-Circ/Term Group, 24-23-UCD/ACD Group, or 12-RingGroup)

## **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

## **Hardware**

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## **2.15 Call Forward, Preset**

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### **Description**

With Call Forward, Preset, your calls forward to a pre-determined destination assigned in 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:

- Unconditional, all 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 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 Call Forward, Preset.

### **Operation**

#### **System**

Operation of Preset Call Forward is automatic.

### **Conditions**

1. A station receiving a forwarded call can transfer the call to the forwarding station.
2. Manual Forward has higher priority than Preset Forward and overrides any Preset Forward setting.
3. 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.
4. 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.
5. Preset call forward status is not shown in the LCD of the Station.
6. A station in a Station Hunt 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 routes



hunt calls to the station based on the system database.

7. 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**

<b>STATION</b>	Preset Call Forward (PGM 120) Station Call Forward No Answer Timer (PGM 123-Button 1) Direct Transfer Mail Box Destination (PGM 120-Button 6)
<b>STATION GROUPS</b>	Member Forward (PGM 191 Button-14-Circ/Term Group, 24-23-UCD/ACD Group, or 12-Ring Group)
<b>SYSTEM</b>	System Call Forward No Answer Timer (PGM 181-Button 1)

## **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

## **Hardware**

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## **2.16 Call Park**

---

### **Description**

You may place an active outside call in a special holding location called a Park Orbit for easy access from any station in the system. There are 19 Park Orbits allowing multiple calls to be parked simultaneously.

### **Operation**

#### ***iPECS IP & LDP Phones***

##### *To park an active external call*

1. Press the **[TRANS/PGM]** button.
2. Dial the Park Orbit number (601~619).
3. Return to idle.

##### *To retrieve a parked call*

1. Lift the handset or press the **[SPEAKER]** button,
2. Dial the Park Orbit number (601~619).

#### **SLT**

##### *To park an active external call*

1. Momentarily press the hook-switch.
2. Dial the Park Orbit number (601~619).
3. Return to idle.

##### *To retrieve a parked call*

1. Lift the handset.
2. Dial the Park Orbit number (601~619).

### **Conditions**

1. If the selected Park Orbit returns a busy signal, the user may simply dial another Park Orbit without the need to disconnect.
2. Intercom calls cannot be placed in a Park Orbit.
3. 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.
4. A Parked call will indicate busy at all appearances.

### **Programming**

#### **SYSTEM**

Call Park Recall Timer (PGM 180-Button 2)

### **Related Features**

- Hold Recall
- Attendant Recall

### **Hardware**

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## 2.17 Call Pick-Up

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### 2.17.1 Directed Call Pick-Up

#### Description

You may answer (“Pick-Up”) incoming and transferred intercom and outside calls ringing at 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 **{DIRECTED CALL PICK-UP}** button.

#### Operation

##### *iPECS IP & LDP Phones*

To assign a **{DIRECTED CALL PICK-UP}** button

[TRANS/PGM] + {FLEX} + “7” + [HOLD/SAVE]

To Pick-up a call ringing at another station

1. Lift the handset or press the **[SPEAKER]** button.
  2. Dial “7”, 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 **{DIRECTED CALL PICK-UP}** button.
  3. Dial the ringing station’s intercom number.

##### **SLT**

To Pick-up a call ringing at another station

1. Lift the handset
2. Dial “7”, the Directed Call Pick-up code.
3. Dial the ringing station’s number.

#### Conditions

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

## Programming

### SYSTEM

Ringling Line Preference Priority (PGM 173)

## Related Features

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

## Hardware

### 2.17.2 Group Call Pick-Up

#### Description

You can answer (“Pick-Up”) incoming and transferred intercom and outside calls ringing at another 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} + “566” + [HOLD/SAVE]

To Pick-up a call ringing at another station

1. Lift the handset or press **[SPEAKER]** button.
  2. Dial “566”, the Group Call Pick-up feature code.
- Or,
1. Press the **{GROUP CALL PICK-UP}** button.

##### **SLT**

To Pick-up a call ringing at another station

1. Lift the handset
2. Dial “566”, the Group Call Pick-up code.

## Conditions

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

## Programming

<b>STATION GROUPS</b>	Station Group (PGM 190-Button 1)
<b>SYSTEM</b>	Ringing Line Preference Priority (PGM 173)

## Related Features

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

## Hardware

### 2.17.3 Pick Up Group Call Pick-Up

#### Description

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

iPECS IP and LDP Phone users may assign a Flex button as a **{PICK UP GROUP CALL PICK-UP}** button.

#### Operation

##### *iPECS IP & LDP Phones*

To assign a **{PICK UP GROUP CALL 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 Pick-Up Group Call Pick-up feature code.  
Or,
3. Press the **{PICK UP GROUP CALL PICK-UP}** button.

**SLT**

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

1. Lift the handset
2. Dial “588”, the Pick-Up Group Call Pick-up code.

**Conditions**

1. Pick-Up Group Call 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.
2. To pick-up an outside call, you must have an idle appearance button available.
3. When several calls are ringing simultaneously, Call Pick-up will connect the oldest highest priority call (Ringing Line Preference, PGM 173).
4. Queue callback and Private Line calls are not subject to Call Pick-up and such attempts receive error tone.
5. Hands free announced intercom calls cannot be picked up by another station. Only ringing intercom calls are subject to Call Pick-up.
6. When a station belongs to multiple Pick-up Groups, calls to the group with the lowest number are answered first.
7. A Station must have “Pick up by Button” enabled to use a Flex button for Pick-up Group Call Pick-up.

**Programming**

<b>STATION GROUPS</b>	Pick Up Group (PGM 192-Button 1)
<b>STATION</b>	Pick up by Button(PGM 114-Button 20)
<b>SYSTEM</b>	Ringing Line Preference Priority (PGM 173)

**Related Features**

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

**Hardware**

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## **2.18 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**

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

### **Programming**

**SYSTEM**                      CO Call Rerouting (PGM252-Web only)

### **Related Features**

### **Hardware**

---

## **2.19 Call Transfer**

---

### **2.19.1 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***

###### *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**

1. For this feature, at least one of the two outside Lines (transferred or receiving) must provide detection of disconnect supervision and lost loop condition.
2. Calls on Lines using digital or VoIP provide "Answer Supervision", thus the iPECS system can provide Transfer Recall.



3. 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**

<b>STATION</b>	Off Net Forward (PGM 111-Button 14)
<b>SYSTEM</b>	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)

## **Related Features**

- Hold Recall
- Call Transfer, Station
- Unsupervised Conference

## **Hardware**

## **2.19.2 Call Transfer, Station**

### **Description**

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

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

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**

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**

1. The transferring station may camp a call on to a busy station.
2. 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.
3. To prevent Toll abuse, outside Lines without an active call (either incoming or dialed digits on outgoing) cannot be transferred.
4. For outgoing calls on analog Lines, the system will monitor the Line for dial tone to prevent Toll abuse.

**Programming**

<b>STATION</b>	No Touch Answer (PGM 111-Button 6)
<b>SYSTEM</b>	Transfer Recall Timer (PGM 180-Button 7)

**Related Features**

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

**Hardware**

### **2.19.3 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 Inbound 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 (01-70).
3. Hang-up to complete the transfer.

##### ***SLT***

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**

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

#### **Programming**

#### **Related Features**

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

#### **Hardware**

- iPECS IP or LDP Phone

## **2.19.4 Call Transfer, Voice Mail**

### **Description**

If permitted, you may transfer outside calls directly to another station's Voice Mailbox, either the integrated or an external 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**

2. 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**

<b>STATION</b>	Direct Transfer to Mail Box Destination (PGM 120-Button 6)
<b>SYSTEM</b>	Transfer Recall Timer (PGM 180-Button 7)

### **Related Features**

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

### **Hardware**

- iPECS IP or LDP Phone

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## **2.20 Call Waiting/Camp-On**

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### **Description**

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

After receiving a busy signal, the calling station camps on to the called station. The called station 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, or
- 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***

##### *To activate a Camp-On while receiving Intercom busy tone*

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

### **Conditions**

1. The user may only Camp-On to a station in the busy mode. A user may not Camp-On to a station in DND, in a conference, receiving a Page, etc.
2. The Camp-On procedure is also employed by an Attendant or Secretary of an Executive/Secretary pair to activate DND Override.
3. A Camp-On tone is sent each time the calling user presses the “\*” button.
4. 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.

### **Programming**

#### **Related Features**

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

### **Hardware**

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## **2.21 Conference**

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### **2.21.1 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 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 initiates 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 considered no-answer using the No-answer timer set when the group was created.

#### **Operation**

*To create a conference group from the iPECS User Portal:*

1. Access Station Program in the Web Admin.
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.

## **Operation**

### ***IPECS phone & LDP Phones***

#### *To assign a {GRP MONITOR} button*

**[TRANS/PGM] + {FLEX} + [TRANS/PGM] + "55" + [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 Phontage or 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:*

4. Press the **[TRANS/PGM]** button.
5. Dial "55", the Monitor Conference Group code.
6. 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**

1. The conditions associated with Multi-Party Voice Conference apply.
2. Members using the iPECS Phontage or UCS client Soft phone may view the state of other members in the conference.
3. A maximum of twenty (20) Conference Groups can be created for user by any system user and up to twenty-five (25) users can create a Conference Group.
4. The user must have access to the Station Web Admin with the proper password (Authentication code).
5. If there are insufficient Conference channels available when initiating the conference group call, the initiator receives error tone.
6. Two-way record can be used to record the Conference Group call.

### **Programming**

#### **Related Features**

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

### **Hardware**



## 2.21.2 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 w 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 101		
<b>BACK</b>	<b>ADD</b>	<b>DELETE</b>

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

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

### Programming

### Related Features

- Multi-Party Voice Conference

### Hardware

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

## **2.21.3 Conference Room**

### **Description**

In addition to ad-hoc Multi-party conferencing, you may open a Conference Room. Other internal and external parties are invited to the conference and can join the conference without further action by you, the Conference Room Supervisor. A Conference Room can be password protected so that only parties that enter 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.

A maximum of 13 parties can join a Conference Room and two such conferences can be active simultaneously.

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 ("1"~"9").

##### To delete a Conference Room

1. Press the **[TRANS/PGM]** button.
2. Dial "055".
3. Dial Room number ("1"~"9").

#### ***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 ("1"~"9").
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 (1~9)
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**

To open a Conference Room

1. Lift the handset.
2. Dial "561", SLT Programming code.
3. Dial "53", the Create Conference Room code.
4. Dial the desired Conference Room number ("1"~"9").
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 Programming code.
3. Dial "54", the close Conference Room code.
4. Dial the Conference Room number ("1"~"9").
5. Dial the Conference Room password.
6. Momentarily press the hook-switch, receive confirmation tone.

**Conditions**

1. Once established, a Conference Room will remain opened until the Room is closed.
2. iPECS Phontage and UCS Client Soft phones may also create, delete and join a Conference Room. For operation, refer to the iPECS Phontage or UCS Client User Guide.

**Programming**

**Related Features**

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

**Hardware**

## **2.21.4 Multi-Party Conference**

### **Description**

You can join multiple internal and external parties in a single call as a conference. You can join a maximum of thirteen (13) parties in a conference with a maximum of two such conferences simultaneously. There is no limit on the number of 3-party conferences.

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. Press the flashing **[CONF]** button, all parties reconnected.

### **SLT**

##### To establish an ad-hoc a 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, repeat steps 2 and 3 above to add additional parties.
5. When completed adding parties, momentarily press hook-switch twice in 2 seconds to connect all parties.

### **Conditions**

1. The **[CONF]** button remains illuminated at the initiators phone for the duration of the conference.
2. If the system receives a disconnect signal and no internal parties remain in the conference, the conference 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.

3. The normal Hold Recall process is applied to a conference on hold using the Unsupervised Conference recall Timer for recall timing.
4. If while setting up a conference, system error tone is received, the initiator must press the **[CONF]** button to regain Intercom dial tone; an SLT user must hook-flash.
5. A station that is busy, in DND or other non-idle state cannot be added to a conference.

## **Programming**

### **Related Features**

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

## **Hardware**

### **2.21.5 Unsupervised Conference**

#### **Description**

Using your iPECS IP or LDP Phone, you may establish a conference with external parties and exit the conference while 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 prior to expiration of the timer, a Warning Tone is provided and at expiration, the call disconnects.

If enabled, either party in an Unsupervised Conference can extend the Unsupervised Conference 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 on the dialed digit multiple of the Timer. For example, if the Unsupervised Conference timer is 5 minutes and the user dials the digit 4, the timer will extend to 20 minutes (4 multiplied by 5 minutes).

#### **Operation**

##### ***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 from a connected party

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

**Conditions**

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

**Programming**

<b>NUMBERING</b>	Unsupervised Conference Timer Extension (PGM 109-Button 3)
<b>STATION</b>	Off Net Forward (PGM 111-Button 14)
<b>SYSTEM</b>	CO to CO UC Time Extension (PGM 160-Button 12) Unsupervised Conference Timer (PGM 182-Button 5)

**Related Features**

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

**Hardware**

- iPECS IP or LDP Phone

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## **2.22    *Customer Site Name***

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### **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*

### **Conditions**

### **Programming**

<b>SYSTEM ID</b>	Site Name (PGM 100-Button 2)
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### **Related Features**

### **Hardware**

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## **2.23     *Data Line Security***

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### **Description**

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

### **Operation**

#### **System**

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

### **Conditions**

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

### **Programming**

**STATION**                                   Data Line Security (PGM 111-Button 4)

### **Related Features**

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

### **Hardware**





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## 2.25 *Delayed Auto Attendant*

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### 2.25.1 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 '1'.

#### **Operation**

##### **System**

Operation of this feature is automatic when assigned

#### **Conditions**

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

#### **Programming**

<b>CO/IP</b>	DISA Answer Timer (PGM 142-Button 24)
	DISA Delay Timer (PGM 142-Button 22)

#### **Related Features**

- DISA Service Attribute
- DID Service Attribute

#### **Hardware**

## **2.25.2 Normal Line Ring Assignment**

### **Description**

An incoming outside call on a “Normal” outside Line can ring to the integrated Auto Attendant either immediately upon detection or after a delay of up to 30 seconds. This allows other stations assigned immediate 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” outside Line is a standard non-DID/DISA line.

### **Operation**

#### **System**

Operation of this feature is automatic when assigned

### **Conditions**

1. 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.
2. If no delay is entered, the call will ring to the Auto Attendant immediately.
3. To assign Delayed Auto Attendant ring, at least one Station or Station Group must be assigned immediate ring.
4. Ring is assigned to a System announcement (01-70) or the AA Ring Time can be set. In the former case, the call is immediately routed to the announcement. In the latter case, the call will notify Stations assigned ring for the duration of the AA Ring time, then the call is routed to System announcement number ‘01’.

### **Programming**

**CO/IP**

CO Ring Assignment (PGM 144)

### **Related Features**

- Line Ring Assignment

### **Hardware**

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## **2.26     *Diagnostics & Maintenance***

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### **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 using a Web browser established over IP networks or a PPP connection over ISDN. Routines that can be accessed include trace functions at the device level, commands for diagnostics, database and software upload and download, and maintenance, and tools for manipulation at the OS level.

### **Operation**

### **Conditions**

### **Programming**

### **Related Features**

### **Hardware**

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## 2.27 Dial-by-Name

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### Description

A name, up to 16 characters, may be assigned to each Station and System Speed dial. In addition, each station may be assigned a 12-character name. When Names are assigned, if using an iPECS IP or LDP phone, you may place an intercom call to another station or select a Station or System Speed dial using the Name.

You select from one of three Dial-by-Name directories and enter characters employing 2 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 point using the **[VOL UP]/[VOL DOWN]** button and selecting a name to call. The number associated with a selected name is displayed by using the **[TRANS/PGM]** button.

### Operation

#### *iPECS IP & LDP Phones*

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

1. Press the **[SPEED]** button twice.
2. Dial the desired directory, I: Intercom, U: User Speed Dial, S: System Speed Dial.
3. Search the directory using the **[VOL UP]/[VOL DOWN]** button or by entering characters. See Station Speed Dial for character entry procedure.
4. Press the **[HOLD/SAVE]** button to place the call.

##### To use Dial by Name using the LIP-8000 series soft buttons

1. Press the **[DIR]** soft button.
2. Dial the desired directory, 1: User Speed Dial, 2: System Speed Dial 3: Intercom.
3. Search the directory using the Navigation buttons or by entering two digits for each character, refer to Station Speed Dial for the Character Entry Chart.
4. Press the **[HOLD/SAVE]** button to place the call.

##### To toggle between the name and number display

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

##### To program your Station User Name

1. Press the **[TRANS/PGM]** button.
2. Dial "071", the Attendant User Name Program code.
3. Dial the Station number.
4. Dial the name, up to 12 characters; refer to Station Speed Dial for The Character Entry Chart.
5. Press **[HOLD/SAVE]**.



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## **2.28     *Dial Pulse to Tone Switchover***

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### **Description**

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

### **Operation**

To switch from pulse to DTMF while on a pulse Line

1. Dial “\*”, the signaling changes to DTMF.

### **Conditions**

1. This command is only recognized for analog pulse dial CO lines.
2. Dial pulse to tone switchover is not available in the Redial features.
3. When “\*” is the first digit of a dialed number, the Display Security feature is enabled and not Pulse to tone switchover. Otherwise, when the system encounters an “\*” in a Speed Dial digit string, dialing will pause then the remaining digit are sent using DTMF signaling.
4. For VoIP calls, pulse dialing is not available thus; switchover is not required or supported.

### **Programming**

<b>CO/IP</b>	CO Line Type (PGM 141-Button 4)
	CO Line Signal Type (PGM 141-Button 5)

### **Related Features**

- Speed Dial
- Display Security

### **Hardware**

## **2.29 Dialing Restrictions**

### **2.29.1 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. The dialing 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 after answering a call will be allowed the dialing privileges assigned.

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

<b>Station COS (Class of Service)</b>	<b>Dialing Restriction</b>
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.

<b>Station COS (Class of Service)</b>	<b>Dialing Restriction</b>
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 the interaction between the Station COS (Class of Service) and CO line COS (Class of Service) assignments.

<b>Station COS (Class of Service)</b>	<b>Restriction</b>				
	<b>CO COS 1</b>	<b>CO COS 2</b>	<b>CO COS 3</b>	<b>CO COS 4</b>	<b>CO COS 5</b>
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 the

system database. When dialed as the first digit(s), these codes signal the system to apply the appropriate COS (Class of Service). If not dialed, the call is treated as an internal PBX call and dialing is not restricted.

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 appropriate Exception Table. Based on the Allow and Deny entries, the system applies the following rules to allow or deny the call.

Rule 1	If a table has no entries, no restrictions are applied.
Rule 2	If there are only Deny entries, restrictions are provided as Deny only.
Rule 3	If there are only Allow entries, restrictions are provided as Allow only.
Rule 4	If there are both Allow and Deny entries, the Deny entries are searched. If the dialed number matches a Deny entry, 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**

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

### **Programming**

<b>STATION</b>	Station COS (Class of Service) (PGM 116)
<b>CO/IP</b>	CO Line COS (Class of Service) (PGM 141-Button 2)
<b>SYSTEM</b>	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)
<b>TABLES</b>	Toll Restriction Tables (PGM 224)

### **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**

## **2.29.2 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, which includes Timed, and Night service modes. The service mode is generally controlled by the System 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*

### **Conditions**

### **Programming**

<b>STATION</b>	Station COS (Class of Service) (PGM 116)
<b>CO/IP</b>	CO Line COS (Class of Service) (PGM 141-Button 2)
<b>SYSTEM</b>	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)
<b>TABLES</b>	Toll Restriction Tables (PGM 224)
	Authorization Codes Table (PGM 227)

## **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**

### **2.29.3 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, temporarily preventing unauthorized outside calls from the station, i.e. "lock the station". The Station can still place 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***

###### *To activate Temporary COS (Class of Service)*

1. Momentarily press the hook-switch.
2. Dial "561", the SLT 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 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**

1. 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**

<b>STATION</b>	Station COS (Class of Service) (PGM 116)
<b>CO/IP</b>	CO Line COS (Class of Service) (PGM 141-Button 2)
<b>SYSTEM</b>	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)
<b>TABLES</b>	Toll Restriction Tables (PGM 224)
	Authorization Codes Table (PGM 227)

**Related Features**

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

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

## Hardware

### 2.29.4 Walking COS (Class of Service)

#### Description

You may temporarily override the toll restriction of a Station to make outside toll calls from a normally toll restricted Station. You must input an Authorization Code in order to activate Walking COS (Class of Service) and are subject to the assigned COS (Class of Service) of the Authorization Code. The COS (Class of Service) associated with the 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**

###### To activate Walking COS (Class of Service)

1. Momentarily press the hook-switch.
2. Dial “561”, the SLT 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

1. 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.

2. 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).
3. 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**

<b>STATION</b>	Station COS (Class of Service) (PGM 116)
<b>CO/IP</b>	CO Line COS (Class of Service) (PGM 141-Button 2)
<b>SYSTEM</b>	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)
<b>TABLES</b>	Toll Restriction Tables (PGM 224)
	Authorization Codes Table (PGM 227)

## **Related Features**

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

## **Hardware**

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## **2.30     *Differential Ring***

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### **Description**

Differential Ring allows any one of 14 different audible Ring signals to be assigned to iPECS IP and 15 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 the speaker. Different selections are assigned for Intercom and outside calls.

Eight different tones are stored in the iPECS IP Phones. Four of these tones are permanent while the other four are downloaded from the 10 ring-tones in the iPECS system. Note the system ring-tones may be replaced with any eight (8) second \*.wav file through the iPECS Web Maintenance.

In addition, each outside Line can be configured to provide one of twelve different ring tones so that you can determine which outside Line is ringing.

### **Operation**

#### ***iPECS IP Phones***

##### *To select the desired ring tone*

1. Press the **[TRANS/PGM]** button.
2. Dial "1" for Ring Selection.
3. Dial "1" for Intercom or "2" for outside Line ring
4. Dial Ring Tone selection "1"- "8", the ring tone is received.

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

1. Press the **[TRANS/PGM]** button.
2. Dial 1 for Ring selection.
3. Dial 5 for Ring tone download.
4. Dial the Ring tone storage bin "5"- "8".
5. Dial the Ring tone selection, "0"- "9", the tone is received.
6. Press the **[HOLD/SAVE]** button.

#### ***LDP Phones***

##### *To select the desired ring tone*

1. Press the **[TRANS/PGM]** button.
2. Dial "1" for Ring Selection.
3. Dial "1" for Intercom or "2" for outside Line ring
4. Dial Ring Tone selection "1"- "15", the ring tone is received.



### **Conditions**

1. To employ one of the system Ring tones, it must first be downloaded to a Ring tone storage bin in the iPECS IP.
2. The iPECS Phontage and UCS Client Soft phones do not have access to the system Ring tones. In the iPECS Phontage or UCS Client phones multiple tones are available and you may load an \*.wav file for use as a Ring tone.
3. Any or all of the ten ring-tones stored in the system can be replaced via the iPECS Web Maintenance.
4. In case of LDP phone, 15 Ring tones can be served from the memory in the phone.

### **Programming**

### **Related Features**

### **Hardware**

- iPECS IP or LDP Phone



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## 2.32 Door Open

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### Description

The iPECS hardware is equipped with relays that activate External Control Contacts. The contacts can be assigned to one of several functions including a Door Open Contact. When used 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 to activate 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” or “#\*2”) + [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”.
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

1. The eMG80 has two External Control Contacts, one each located on the Miscellaneous connector of the KSU and EKSU.
2. The Door Open feature dial code is based on the assigned External Contact as below:
  - External Contact 1 = “#\*1”
  - External Contact 2 = “#\*2”
3. The contacts are rated at 1 amp, 24 VDC.
4. A station will receive error tone if not allowed access to the Door Open feature (PGM 113-Button 24-2)

### Programming

<b>SYSTEM</b>	External Contact Control (PGM 168) Door Open Timer (PGM 181-Button 5)
<b>STATION</b>	Door Open (PGM 113-Button 24-2)

### Related Features

- LBC (Loud Bell Control)

### Hardware

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

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## 2.33 Emergency Call Caller-Location Support

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### Description

Emergency Call - Caller Location Support integrates the iPECS to a PBX ANI Link unit, such as from Tone Commander, to provide Caller Id and location information to an emergency center. When an Emergency call is placed from a Station, an assigned emergency 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 emergency center over a connected CAMA trunk. Caller Id and location information from the ANI Link unit database is sent to the emergency center in the proper format and the Station is connected to the 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 to the 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 is removed.

### Operation

Operation of this feature is automatic when assigned

### Programming

<b>STATION</b>	Emergency CO/Group (PGM 112-Button 18) Proctor Monitor (PGM 113-Button 22)
<b>CO/IP</b>	Proctor On/Off (PGM 141-Button 12)
<b>TABLES</b>	Emergency Service Call Table (PGM 226)

### Conditions

1. 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.
2. The ANI Link unit must be properly connected to an SLT port, an analog CO Line port and on the same LAN as the system KSU LAN port.
3. 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.
4. 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.
5. 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.
6. During a power failure, if the ANI Link unit Power Fail contacts, normally open, are connected to a system SLT 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.

7. SMDR output will identify any call made via the Emergency call feature, including the success or failure of such calls.

### **Related Features**

- Emergency Call
- Emergency Call Attendant Alert

### **Hardware**

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

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## 2.34 *Emergency Call*

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### **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**

1. 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.
2. 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".

### **Programming**

<b>STATION</b>	Emergency CO Group (PGM 112-Button 18)
<b>TABLES</b>	Emergency Service Call Table (PGM 226)

### **Related Features**

### **Hardware**

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## **2.35     *Executive/Secretary Forward***

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### **Description**

iPECS IP and LDP Phones can be assigned as Executive/Secretary pairs. By activating DND, the Executive also activates Unconditional Call Forward to the Secretary, which will forward Executive calls to the Secretary. With the “CO Call to Secretary” option enabled, all outside calls to the Executive forward to the Secretary regardless of the Executive’s station status. In addition, if the Secretary is in DND, Executive calls sent to the Secretary route back to the Executive 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 higher grade can call lower grade Executives overriding the Executive/Secretary forward.

With the “ICM Call to Secretary” option (PGM229-Button 5) enabled, all intercom calls to the Executive (except for calls from higher or same grade executive) forward to the Secretary regardless of the Executive’s station status.

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

### **Operation**

#### ***iPECS IP & LDP Phones***

##### To activate/deactivate Executive/Secretary Forward from the Executive

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

### **Conditions**

1. The eMG80 supports 36 Executive/Secretary pairs.
2. An Executive may have multiple Secretaries and a Secretary may have multiple Executives. Each forms a separate Executive/Secretary pair.
3. If the Secretary is busy when a call is received for the Executive, the caller will receive busy tone.
4. The Secretary may override the DND status of the Executive to Camp-On and transfer calls to the Executive.
5. 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.
6. If an Executive has multiple Secretaries, calls will automatically route to the Executive’s first idle Secretary.
7. The Executive may use Call Forward to send calls to stations other than the Secretary.
8. 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.
9. The Executive Grade can be assigned only for Country Code “82”, Korea.
10. The Executive can call to another executive who is same Group in ADM 229 – B7 even

though they are different grade.

## **Programming**

<b>STATION</b>	DND (PGM 111-Button 3) Left Message Exec (PGM 113-Button 10)
<b>TABLES</b>	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)

## **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



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## 2.36 External Auto Attendant/Voice Mail

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### 2.36.1 AA/VM Group

#### Description

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

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

#### Operation

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

#### Conditions

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

#### Programming

<b>STATION GROUPS</b>	Station Group Assignment (PGM 190) Station Group Attributes (PGM 191)
<b>TABLES</b>	Voice Mail Dialing Table (PGM 234)
<b>DIP SWITCH</b>	Virtual Dip Switch (PGM 453-Button 2)

#### Related Features

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

#### Hardware

- External AA/VM system

## **2.36.2 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 port, the system will send DTMF signals informing the AA/VM of the characteristics of the call. DTMF digit strings are assigned to various functions allowing the AA/VM to respond appropriately to the call. These definitions are entered in the Voice Mail Dialing Table (PGM 234).

### **Operation**

#### **System**

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

### **Conditions**

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

### **Programming**

<b>STATION GROUPS</b>	Station Group Assignment (PGM 190) Station Group Attributes (PGM 191)
<b>TABLES</b>	Voice Mail Dialing Table (PGM 234)
<b>DIP SWITCH</b>	Virtual Dip Switch (PGM 453-Button 2)

### **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

### 2.36.3 SMDI (Simplified Message Desk Interface) AA/VM Signaling

#### Description

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

#### SMDI Protocol

There are three types of SMDI messages. Within each message is an “Action Code”, which defines the function or required action of the AA/VM system. Fields within the messages also define the called/calling station and station status. The various message types and definition of the fields are shown in the chart below.

Type I message

cr lf **MD** ggg mmmm a xxxxxxxx sp yyyyyyyy sp cr lf^Y

Type II message

cr lf **MD** ggg mmmm a xxxxxxxx sp sp cr lf^Y

Type III message

cr lf **MD** ggg mmmm a sp yyyyyyyy sp cr lf^Y

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 various SMDI action codes.

**Table 2.36.3-1 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

**Table 2.36.3-1 SMDI Messages**

Action Code	Reason	Purpose	Message Type	SMDI Message MD 001 0001-
E	Error, invalid number	Error	II	E xxxxx yyyy
H	Two-way Record	record	II	H xxxxx yyyy
I	DND	DND	II	I xxxxx yyyy
N	No Answer	No Answer	II	N xxxxx yyyy
R	Direct ring to VM group	AA	III	R xxxxx yyyy

## Operation

### System

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

### Conditions

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

## Programming

<b>STATION GROUPS</b>	Station Group Assignment (PGM 190) Station Group Attributes (PGM 191)
<b>TABLES</b>	Voice Mail Dialing Table (PGM 234)
<b>DIP SWITCH</b>	Virtual Dip Switch (PGM 453-Button 2)

## 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

## **2.36.4 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 4-digit number (“0001” to “9999”). An entry of 0000 or blank sends the Station number to the VM system.

### **Operation**

#### **System**

Operation of this feature is automatic when programmed.

### **Conditions**

1. Voice Mailbox Translation applies to iPECS UMS and External In-band and SMDI Voice mail systems.

### **Programming**

<b>STATION</b>	Voice Mail Id (PGM 113-Button 14)
<b>STATION GROUPS</b>	Station Group Assignment (PGM 190) Station Group Attributes (PGM 191)
<b>TABLES</b>	Voice Mail Dialing Table (PGM 234)

### **Related Features**

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

### **Hardware**

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## 2.37 Flexible Numbering Plan

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### Description

Your access to the iPECS system resources and features is accomplished through feature codes or Flexible buttons on your iPECS IP or LDP Phone. The Administrator selects from one of the eight (8) different standard numbering plans and, if desired, assigns codes for individual functions in the Flexible Numbering Plan. The feature codes are defined in the system's Flexible Numbering Plan; refer to the *iPECS Admin and Programming Manual*.

### Operation

#### System

System implements feature activation based on the Flexible Numbering Plan.

### Conditions

1. Feature codes can be one to three digits in length.
2. During programming, conflicts in the Numbering Plan are not allowed. The existing non-conflicting Numbering Plan is used until correctly updated.

### Programming

<b>SYSTEM ID</b>	System Id (PGM 100-Button 4)
<b>NUMBERING</b>	Flexible Station Numbering Plan (PGM 105) Flexible Numbering Plan (PGM 106 – 109)

### Related Features

### Hardware

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## 2.38 *Forced Disconnect Intrusion*

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### 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*

[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 [TRANS/PGM] button and dial "56#", the Flexible Numbering Plan 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 [TRANS/PGM] button and dial "56#", the Flexible Numbering Plan code.
3. The call on the busy Station is terminated, the station receives warning tone and you are connected to the station.

### **SLT**

#### 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#", the Flexible Numbering Plan code.
5. The call on the busy Line is terminated, the connected station receives error tone and you are connected to the Line.

### Conditions

1. If the station was talking to other station or CO line, the call is disconnected when the station receives forced Intrusion call.
2. A Forced Intrusion call is not possible to a station in DND state.
3. A Forced Intrusion call is not possible to an Attendant.
4. Any station with a Forced Intrusion button can activate this feature.

## Programming

## Related Features

## Hardware

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## **2.39 Green Power Save**

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### **Description**

The system can disable the power to LDP Phones and SLTs (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**

1. When power is removed from an LDP Phone or SLT, the phone cannot place or receive calls.
2. 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.
3. 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**

<b>Green mode</b>	Green Mode Activation (PGM 500)
	Green Mode Time (PGM 501)

### **Related Features**

### **Hardware**



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## 2.40 Headset Compatibility

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### Description

You can use an industry standard headset with iPECS IP and LDP Phones in place of or in addition to the handset. The station is programmed for Headset operation.

In the Headset mode, pressing the **[SPEAKER]** button will send audio to the Headset instead of the speakerphone. In addition, when in the Headset mode, ring signals can be delivered to the 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 between Headset 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,
4. 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

1. The Intercom Signaling Mode can be set in the Headset mode as with the Speakerphone mode.
2. The station always receives Page announcements over the speaker of iPECS IP and LDP Phones.
3. 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.
4. 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**

<b>STATION</b>	/Headset Ring Mode (PGM 111-Button 8)
	Speakerphone/Headset (PGM 111-Button 9)

### **Related Features**

- Speakerphone
- Paging

### **Hardware**

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## **2.41 Hold**

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### **2.41.1 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*

1. Press the desired feature button or **{LINE}**; the active call is placed in the Preferred Hold state.

#### **Conditions**

1. Outside Lines placed on hold with Automatic Hold are placed in the assigned Preferred Hold.
2. There is no limit on the number of calls that can be placed on hold using Automatic Hold.

#### **Programming**

<b>STATION</b>	Automatic Hold (PGM 112-Button 2)
<b>SYSTEM</b>	Hold Preference (PGM 160-Button 7)

#### **Related Features**

- Hold Preference
- Hold Recall

#### **Hardware**

- iPECS IP or LDP Phone

## **2.41.2 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 the assigned 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**

1. A transferred call is placed in the Exclusive hold state at the receiving station regardless of the assigned Hold Preference.
2. Hold Preference is not available to an SLT.

### **Programming**

**SYSTEM**                      Hold Preference (PGM 160-Button 7)

### **Related Features**

- Call Transfer, Outside Line
- Exclusive Hold
- System Hold

### **Hardware**

- iPECS IP or LDP Phone

### **2.41.3 Hold Recall**

#### **Description**

When you place an outside call on hold, a hold timer is activated. If the timer expires, the held call will recall your station for the I-Hold Recall time. If the call remains unanswered, the Attendant 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**

1. Separate timers are assigned for the various types of hold: System, Exclusive, Transfer, Call Park, etc.
2. 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.
3. If the specific Hold timer is set to zero, recall is disabled.
4. 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**

<b>SYSTEM</b>	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)

#### **Related Features**

- Call Transfer, Line
- Exclusive Hold
- System Hold

#### **Hardware**

## **2.41.4 Exclusive Hold**

### **Description**

Outside lines may be placed in a waiting state such that other stations in the system are unable to access the line. Only the station placing the line on Exclusive hold can access the held line.

If the call remains on hold at expiration of the Exclusive Hold Recall Timer, normal Hold Recall will 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,
3. Dial "8#", the Held Call Access code.
  4. Dial the line number.

#### **SLT**

##### To place a call on Exclusive Hold

1. Momentarily press the hook-switch.

##### To access a call on Exclusive Hold from the holding SLT

1. Lift the handset.
2. Dial "8#", the Held Call Access code.
3. Dial the line number.

### **Conditions**

1. 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.
2. All Transferred outside calls are placed on Exclusive Hold for the receiving station.
3. The LED of **{LOOP}** and **{POOL}** buttons will display the active outside line status.

### **Programming**

#### **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)

### **Related Features**

- Call Transfer, Line
- Hold Preference
- Hold Recall
- System Hold

### **Hardware**

## **2.41.5 System Hold**

### **Description**

Outside Lines may be placed in a waiting state such that other stations in the system are able to access the Line. Stations must have access to the Line in the system database to access the held call.

If the call remains on hold at expiration of the System Hold Recall Timer, normal Hold Recall will 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#", the Held Call Access code.
3. Dial the line number.

To place a call on Exclusive Hold

1. Momentarily press the hook-switch.
2. Dial "560", the System Hold feature code.

To access a call from System Hold

1. Lift the handset.
2. Dial "8#", the Held Call Access code.
3. Dial the line number.

**Conditions**

1. 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.
2. 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.
3. The LED of **{LOOP}** and **{POOL}** buttons will display the active outside Line status.

**Programming**

**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)

**Related Features**

- Call Transfer, Outside Line
- Hold Preference
- Hold Recall
- Exclusive Hold

**Hardware**



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## **2.42 Hot Desk**

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### **Description**

The iPECS IP and LDP Phones can be assigned as Hot Desk phones allowing users (Agents) to login with the iPECS. The Hot Desk will become active and will take on the attributes defined for the Agent's Station number. When the Agent logs off, the Hot Desk phone becomes inactive and the Agent's calls are forward to the user-entered destination. A different Agent may then login through 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.

### **Conditions**

1. Hot desk can be used only for IP phone in eMG80 system
2. An active (logged in) Agent can login to another inactive Hot Desk. However, this will log the Agent off the previously active Hot Desk.
3. 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.
4. The Flex button map of the Hot Desk station is fixed and does not take on the configuration associated with the Agent's station.
5. 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
6. 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.

7. To identify the station number, only Station Authorization codes may be used for Hot Desk login.
8. An ACD Agent must use the Agent Login procedure as defined under the ACD feature description.

### **Programming**

<b>STATION</b>	Hot Desk Station (PGM 112-Button 13)
<b>TABLES</b>	Hot Desk Attributes (PGM 250-Button 1~3) Auto Logout Timer (PGM 250-Button 3)

### **Related Features**

- Station Groups
- Automatic Call Distribution
- Call Forward

### **Hardware**

- iPECS IP Phone

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## **2.43 ICLID Call Routing**

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### **Description**

The system can employ ICLID (Incoming Calling Line Id) to determine the routing of incoming external 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 a match 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**

1. 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.
2. 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.).
3. The ICLID received from the Line must be a telephone number to match an ICLID Route Table entry.
4. If ICLID routing is enabled for a DID Line, DID Call Wait is disabled.

#### **Programming**

<b>CO/IP</b>	ICLID Ring Timer (PGM 142-Button 14)
<b>ISDN/ICLID</b>	ICLID Route Table Attributes (PGM 203) ICLID Ring Assignment Table (PGM 204)
<b>STATION GROUPS</b>	ACD ICLID Use (PGM 191-Button 21)
<b>TABLES</b>	Flexible DID conversion (PGM 231)

#### **Related Features**

- Outside Line Ring Assignment
- Automatic Call Distribution
- Direct Inward Dial (DID)

#### **Hardware**

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## 2.44 *In-Room Indication*

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### 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 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

1. The In-Room Indication button operates only while the Supervisor's phone is idle.
2. If the In-Room button on a station other than the Supervisor's is pressed, error tone is heard.
3. 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

SYSTEM

In-Room Indication (PGM 183)

### Related Features

### Hardware

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## **2.45     ***Integrated Auto Attendant/Voice Mail*****

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### **2.45.1   **Integrated AA/VM Overview****

#### **Description**

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, four (4) access channels and 1 hour of storage. The storage can be increased to 16 hours with the addition of the Memory Expansion Unit.

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 Gateway with four (4) access channels and 15 hours of storage is available for the eMG80.

#### **Operation**

##### ***System***

*Operation of the integrated AA/VM is automatic based on configuration.*

#### **Conditions**

#### **Programming**

#### **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 MEMU, AA/VM 15 hour Memory Expansion Unit
- EMG80 VVMU, 2<sup>nd</sup> VSF Gateway

## **2.45.2 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 70 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.
2. Dial "06", the System Announcement Record code.
3. Dial the sequence number of the voice storage unit.
4. Dial the Announcement number (01-70).
5. With Multi Language support, enter the Language number (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 message.

##### *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 (01-70).
5. With Multi Language support, enter the language number (1~6), the current announcement is played followed by the "Press # to record" prompt.
6. Dial "#".
7. Press the **[SPEED]** button during playback to erase message.

## **System**

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

## **Conditions**

1. There are no individual time limits on an Auto Attendant announcement.
2. The external caller may experience ring-back tone before the system answers and play the System announcement.
3. The System Attendant must "Save" a recording before returning to the on-hook state, otherwise, the existing recording is used.
4. To record or delete an Auto Attendant message, all of the AA/VM channels must be in the idle state.
5. The external caller may dial at any time during an Auto Attendant announcement and must dial prior to the expiration of the CCR Analysis timer.
6. If the external caller dials an invalid selection or station, the system will play the "Invalid Entry" prompt and allow re-entry. The DISA Retry Counter determines the number of times the caller may retry entries.
7. If the external caller dials more than a single digit, the call is routed based on the Station Numbering Plan.
8. Calls answered by an Auto Attendant (CCR) Announcement are interactive DISA calls and are subject to conditions of a DISA call.
9. The "\*" digit is reserved in the Audio/Text Tables to repeat the current or previous Auto Attendant announcement.
10. The "#" digit is reserved for callers to access their Voice Mailbox remotely.
11. A CCR Announcement may be programmed to disconnect after playing.
12. The Auto Attendant Announcement feature is supported for DISA and DID calls.
13. 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 one of the gateways.

## **Programming**

<b>STATION</b>	VSF Access (PGM 113-Button 2)
<b>CO/IP</b>	CO/IP Ring Assignment (PGM 144) DID Service Attributes (PGM 145)
<b>STATION GROUPS</b>	Station Groups (PGM 190) Station Group Attributes (PGM 191)
<b>SYSTEM</b>	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)
<b>TABLES</b>	Customer Call Routing Tables (PGM 228) Flexible DID Conversion Table (PGM 231)

## **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**

### **2.45.2.1 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**

### 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:
8. "for [subscriber name], press 1"  
"for [subscriber name], press 2"  
...  
"for [subscriber name], press 9"
9. When the caller enters the desired digit, the system routes the caller to that station

## **Conditions**

1. 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**

<b>STATION</b>	VSF Access (PGM 113-Button 2)
	Company Directory First Name (PGM 127-Button 5)
	Company Directory Last Name (PGM 127-Button 6)

## **Related Features**

- CCR
- Integrated Auto Attendant

## **Hardware**

## **2.45.2.2 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 (001-072).
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.

### **Conditions**

### **Programming**

<b>TABLES</b>	Customer Call Routing Tables (PGM 228)
	Flexible DID Conversion Table (PGM 231)

### **Related Features**

- Integrated Auto Attendant
- Attendant

### **Hardware**

## **2.45.3 Integrated Voice Mail**

### **2.45.3.1 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**

1. There are no limits on the number of mailboxes that can be marked as administrators.
2. The broadcast message is sent to all mailboxes in the same tenancy group and is always retrieved first regardless message retrieve option.
3. A station can delete broadcast message without confirming receipt.
4. Only administrator can remove a recorded broadcast message.

#### **Programming**

##### **TABLES**

Customer Call Routing Tables (PGM 228)

Flexible DID Conversion Table (PGM 231)

#### **Related Features**

- Integrated Auto Attendant/ Voice Mail

#### **Hardware**

### **2.45.3.2 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**

1. There is no limit to the number of mailboxes that can be assigned as Announce only.

#### **Programming**

<b>Station</b>	Announce Only Mailbox (PGM 127-button 3)
	Announce Only Option (PGM 127-button 4)

#### **Related Features**

- Integrated Auto Attendant/ Voice Mail

#### **Hardware**

### **2.45.3.3 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**

1. 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.
2. The ability to activate Unconditional Call Forward from your Mailbox is only available when the Mailbox is accessed from a remote location.

#### **Programming**

#### **Related Features**

- Message Storage
- Message Retrieval
- Remote Message Retrieval
- Message Retrieval Options

#### **Hardware**

### **2.45.3.4 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**

1. The maximum number of distribution lists you can create is five (5) and up to 25 mailboxes can be assigned in a distribution list.

#### **Programming**

#### **Related Features**

- Integrated Auto Attendant/Voice Mail
- Send a Message
- Distribution Lists

#### **Hardware**

### **2.45.3.5 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**

1. 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.
2. The E-mail address for the System and the station are defined under the Web Admin.
3. To attach the Voice message to the E-mail notification, the Attach Message option (PGM 111 – Button 24) must be enabled.
4. 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**

<b>STATION</b>	VSF MESSAGE – SMTP Mail Server Address (PGM113 – Button 18)
	VSF MESSAGE – Receiver Mail Address (PGM113 – Button 19)
	VSF MESSAGE – SMTP Mail Server ID (PGM113 – Button 23)
	VSF MESSAGE – SMTP Mail Server P/W(PGM113 – Button24-1)
	VSF MESSAGE – Add Message
	VSF MESSAGE – Delete Message After Add (PGM113 – Button24-5)
	VSF MESSAGE – SMTP Security (PGM113 – Button24-12)
	VSF MESSAGE – SMTP Port (PGM113 – Button24-13)
	VSF MESSAGE – Sender Mail Address (PGM113 – Button24-14)
<b>BOARD</b>	VSF Sender Mail Address (PGM 132-Button 6)
<b>SYSTEM</b>	VSF SMTP Port (PGM113 – Button24-13)

#### **Related Features**

- Integrated Auto Attendant/Voice Mail

#### **Hardware**

### **2.45.3.6 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**

1. You must be permitted to mark a message as Private as part of the VM COS

#### **Programming**

**SYSTEM**

VM COS Attributes (PGM 253)

#### **Related Features**

- Send a Message
- Distribution Lists

#### **Hardware**



### **2.45.3.7 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**

1. You must be permitted to mark a message for Delivery Confirmation as part of the VM COS.

#### **Programming**

**SYSTEM**

VM COS Attributes (PGM 253)

#### **Related Features**

- Send a Message

#### **Hardware**

### **2.45.3.8 Message Cascade**

#### **Description**

Message cascading copies messages left for the originating mailbox to another mailbox. Once the message is copied into the other mailbox, notification events can take place based on setting for that mailbox settings. This is useful to differentiate the notification for Urgent messages.

Your Mailbox can be cascade based on one of the following selections.

- *Disable* - Disable this function.
- *Copy-Immediately* – Copy (Cascading) begins immediately, allowing storage of duplicate messages in several mailboxes.
- *Copy-Urgent Only* – Copy (Cascades) only those messages marked urgent.
- *Move-Immediately* – Move (Cascading) begins immediately, allowing storage of duplicate messages in several mailboxes.
- *Move-Urgent Only* – Move (Cascades) only those messages marked urgent.

#### **Operation**

*Operation of a Cascade Mailbox is automatic once configured*

#### **Conditions**

1. A mailbox can only have 1 cascade mailbox destination.
2. Mailbox messages are retained in each mailbox.

#### **Programming**

<b>Station</b>	Cascade Mailbox (PGM 127-button 7)
	Cascade Type (PGM 127-button 8)

#### **Related Features**

- Integrated Auto Attendant/Voice Mail

#### **Hardware**

### 2.45.3.9 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.

**SLT**

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.

**Conditions**

1. 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.

2. If the dialed number is not recognized, the "Invalid Entry" prompt is played. After the second invalid entry, you are disconnected.
3. 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.
4. 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.
5. 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**

<b>STATION</b>	Retrieve Message Order (PGM 113-Button 13)
	Auto save new message (PGM 161-Button 24-11)

## **Related Features**

- Message Retrieval Options
- Remote Message Retrieval
- Multiple Voice Mailbox Support

## **Hardware**

### **2.45.3.10 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

<b><u>Digit</u></b>	<b><u>Function</u></b>
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

<b><u>Digit</u></b>	<b><u>Function</u></b>
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**

1. 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**

1. 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.
2. When the "call sender" option results in an external call, dialing restrictions will be applied based on your Station COS (Class of Service).
3. 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.

4. 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**

### **SYSTEM**

Auto Save New Message (PGM 161-Button 24-11)

## **Related Features**

- Message Retrieval
- Remote Message Retrieval
- Voice Mailbox Settings
- Class of Service

## **Hardware**

### **2.45.3.11 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. If VM back up is assigned, the back-up station (Phontage or UCS Client) is also notified and the message stored on the hard drive of the Soft phone PC. Other notification including sending an e-mail with the Voice Mail can be configured for your integrated Voice Mailbox.

The integrated Voice Mail application provides 1 hour and 4 access channels and can be expanded to 16 hours.

#### **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**

1. 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.
2. If all the AA/VM channels are in use, Ring Back tone is provided until a channel is available.
3. All stations including an SLT can leave and receive voice messages.
4. Individual User Greetings and Voice Mails are protected from loss of AC power.
5. 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**

<b>STATION</b>	VSF Back-up Station (PGM 112-Button 23)
	VSF Access (PGM 113-Button 2)
	Direct Transfer to Mailbox (PGM 120-Button 6)
	Station Call Forward Timer (PGM 123)
<b>SYSTEM</b>	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)

## **Related Features**

- Call Forward
- Station Message Wait/Call Back
- Integrated AA/VM Overview
- Call Transfer, Voice Mail

## **Hardware**

### **2.45.3.12 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**

1. 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.
2. If notification is disabled, any existing notification will be terminated after the initial notification call.
3. 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.
4. The Mobile telephone number to notify is assigned in PGM 236.
5. 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.
6. The Retry counter is incremented after the system accesses the outside Line for notification.
7. The Retry count is from 1 to 9; the retry interval is from 1 to 3 minutes.
8. 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.
9. 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.
10. 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**

### **TABLES**

Hunt Call Enable (PGM 236-Button 6)  
VSF Notify (PGM 236-Button 7)

**SYSTEM**

Notify Retry Count (PGM 236-Button 8)  
Notify Retry Interval (PGM 236-Button 9)  
VM MEX Notify over LCO (PGM160-Button24-13)  
Notify Play Delay (PGM181-Btn18)

**Related Features**

- Mobile Extension
- Message Retrieval
- Remote Message Retrieval
- Message Retrieval Options

**Hardware**

**2.45.3.13 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, 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
- Page Zone
- Voice Mail
- System Announcement
- Company Directory

In addition, the system will monitor digits for a station number. If the user dials a station number, the system will transfer the caller to the station.

If this option set to OFF, CCR table is not worked, and next following digits are available, and follow system DISA call flow.

## **Operation**

### ***Outside 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 caller routes to the CCR destination or to the DID/DISA destination if the CCR destination is unavailable.

## **Conditions**

1. The caller can dial "\*" or "#" to replay your greeting.
2. 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 (PGM 167)
3. 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 PGM 167 DID/DISA Destination.
4. The Personal CCR option is available to outside callers only.
5. 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.
6. No retry count is applied to Personal CCR; the caller can attempt to enter a matching digit once.
7. The priority of Personal User Greeting is as followings.
  - Personal CCR
  - Outgoing Mailbox Destination
  - User Greeting DISA

## **Programming**

<b>STATION</b>	Station CCR Table (PGM 128)
<b>SYSTEM</b>	DID/DISA Destination (PGM 167)
	CCR Interval Digit Timer (PGM 180-Button 13)

## **Related Features**

- Integrated Auto Attendant
- Call Forward
- DISA

## **Hardware**

### **2.45.3.14 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. Follow local access procedures.

#### **Conditions**

1. The conditions associated with Message Retrieval and Message Retrieval Options apply.
2. The conditions associated with DISA apply.

#### **Programming**

#### **Related Features**

- Message Retrieval Options
- Integrated Auto Attendant
- Message Retrieval

#### **Hardware**

### **2.45.3.15 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

1. At the main Voice Mail prompt, dial “4” and follow the voice instructions to record and send you message.

#### **Conditions**

#### **Programming**

#### **Related Features**

- Integrated Auto Attendant/Voice Mail
- Distribution Lists
- Mark a Message as Private
- Mark a message for Delivery Confirmation

#### **Hardware**

### **2.45.3.16 Voice Mail Back-up Station**

#### **Description**

An iPECS Soft phone (Phontage and UCS Client) user receives notification of new messages for assigned stations. The Soft phone will indicate the total messages for the assigned stations as well as the Soft phone. The Soft -phone user can download the messages for other stations to the hard drive of the Soft phone PC and, using the Soft phone application, manage the messages on the hard drive. If enabled, the Soft phone user may delete voice messages from the Voice Mailbox of the other assigned stations.

#### **Operation**

##### **System**

System automatically notifies the back-up station of new messages

##### **Phontage or UCS Client**

See Phontage or UCS Client Guide

#### **Conditions**

1. Voice messages are stored in the integrated Voice Mail and must be expressly deleted. Deleting messages from the soft-phone hard drive does not delete the message from the Voice Mail memory.
2. It is not possible to backup Voice Mail messages to a Phontage PC when Phontage is connected to the system remotely in R/NAPT mode.

#### **Programming**

<b>STATION</b>	VSF Back-up Delete (PGM 112-Button 22)
	VSF Back-up Station (PGM 112-Button 23)
	VSF Back-up prompt (PGM 112-Button 24)

#### **Related Features**

- Integrated Auto Attendant/Voice Mail

#### **Hardware**

### **2.45.3.17 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**

1. The default Class of Service for all mailboxes is 1.

#### **Programming**

**SYSTEM** VM COS Attributes (PGM 253)

#### **Related Features**

- Integrated Auto Attendant/Voice Mail

#### **Hardware**

### **2.45.3.18 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**

1. If you are accessing your Voice Mail remotely, you must begin dialing within the CCR Analysis time, if not the call is released.
2. If the dialed number is not recognized, the “Invalid Entry” prompt is played.
3. 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.

#### **Programming**

#### **Related Features**

- Message Storage
- Message Retrieval
- Remote Message Retrieval
- Message Retrieval Options

#### **Hardware**



## **2.45.4 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***

##### *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**

1. For station status, items from “Listed message x” to “COS x” are not include unless active.

**Programming**

**STATION**                                   VSF Access (PGM 113-Button 2)

**Related Features**

- Integrated Auto Attendant/Voice Mail

**Hardware**

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## **2.46 IP Bridge Mode**

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### **Description**

Phontage and 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 the iPECS IP or LDP Phone by 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 in the Bridged mode.
2. Lift the handset of the linked iPECS IP or LDP Phone to receive audio.

### **Conditions**

1. The Soft phone must be used local to the bridged iPECS IP or LDP Phone.
2. Bridged operation must be selected at the Soft phone.
3. The Soft phone must be linked to the associated iPECS IP or LDP Phone.
4. Bridging is not available from the iPECS IP or Phone to the Soft phone.

### **Programming**

### **Related Features**

- Linked Station Pairs

### **Hardware**

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## **2.47 IP System DECT**

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### **Description**

The iPECS supports office building mobility employing Digital Enhanced Cordless Technology (DECT). Ericsson-LG's Multi-channel DECT Base stations connect to the Wireless Interface board of the system to establish cells. With the DECT handsets, you can roam and maintain an uninterrupted communications link to iPECS features and resources.

For further information on installation and operation of the IP System DECT solution, refer to the eMG80 and DECT Installation Manuals.

### **Operation**

DECT operation is automatic when configured.

### **Conditions**

1. The maximum number of DECT phones (desktop or portable handsets) is 48.

### **Programming**

<b>DECT</b>	DECT Registration (PGM 491-Web only)
	DECT Attributes (PGM 491)

### **Related Features**

### **Hardware**

- WTIB4
- GDC-600B/600BE Base stations
- GDC-450H/500H handsets

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## **2.48 IP FAX Relay, T.38 Support**

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### **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.

### **Conditions**

### **Programming**

### **Related Features**

### **Hardware**

- SLT port with FAX machine connected

---

## **2.49 LBC (Loud Bell Control)**

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### **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**

1. Two contacts are available, one each appearing in the Miscellaneous contacts of the KSU and the EKSU.
2. 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.
3. The contacts are rated at 1 amp, 24 VDC.
4. 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.

### **Programming**

<b>CO/IP</b>	Universal Night Answer (PGM 141-Button 7)
<b>SYSTEM</b>	External Night Ring (PGM 160-Button 6)
	External Contact Control (PGM 168)

### **Related Features**

- Universal Night Answer
- Door Open

### **Hardware**

- External Control Contact connected to an external loud bell.

---

## 2.50 LCR (Least Cost Routing)

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### Description

The LCR Tables are employed to define appropriate routing for outgoing calls based on the dialed number. Generally, the LCR Tables are structured to define the Least Cost Route for Long Distance calls.

Your dialed digits are compared to table entries and modified based on time of day, day of week, and assigned routes. There are four LCR Tables: LCR Control Attributes, LCR Leading Digit Tables, LCR Digit Modification and LCR Initialization Tables.

LCR Access modes are assigned in the LCR Control Attributes Table. These modes define the manner in which you access the LCR function. LCR may be disabled or one or more of the three access modes can be allowed to access LCR. The basic modes are:

1. 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 Modification Table).

2. 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 are compared to the Loop LCR codes in the LDT; the system will seize an outside Line from the assigned Line Group and sends digits from the DMT.

3. Direct CO LCR

If you select a **{LINE}** or **{LINE GROUP}** button, the Direct CO LCR mode is accessed. If the dialed number matches a Direct LCR code in the LDT and the seized Line belongs to the assigned Direct CO LCR group, the system will send digits modified based on the DMT.

### Operation

#### System

Operation of LCR is automatic based on assignments in the system's LCR Tables

### Conditions

1. There are a total of 6 LCR access mode combinations that can be defined as below:
  - 1) LCR Access Mode 00 (M00) – LCR is disabled.
  - 2) LCR Access Mode 01 (M01) – Loop LCR access is active.
  - 3) LCR Access Mode 02 (M02) – Loop and Internal LCR access are active.
  - 4) LCR Access Mode 11 (M11) – Direct CO and Loop access are active.
  - 5) LCR Access Mode 12 (M12) – Direct CO, Loop and Internal LCR access are

active, seize Co line according to the LCR attributes.

6) LCR Access Mode 13 (M13) – Direct CO, Loop and Internal LCR access are active, seize CO line according to the station attributes.

2. Multiple Leading Digit Tables can be configured for each station and outside Line.
3. 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.
4. 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.
5. For Direct CO LCR, the number of digits for the LDT Table should be programmed considering the dial-tone time-out of the network.
6. 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.
7. If the LCR CO group is not assigned, the system will not seize an outside Line and will make internal call.
8. If the LCR CO group is assigned as the unused group, the system it will seize an outside Line according to the station attributes.

## **Programming**

<b>TABLES</b>	LCR Control Assignment (PGM 220)
	LCR LDT Table (PGM 221)
	LCR DMT Table (PGM 222)
	LCR Table Initialization (PGM 223)

## **Related Features**

- Outside Line Access
- Outside Line Groups
- ISDN (Integrated Service Digital Network)
- Station Flexible Buttons

## **Hardware**



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## **2.51     *Linked Station Pairs***

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### **Description**

A station can be logically linked to a primary station so that the two stations function as a single station. When linked, the two stations effectively act as a single station with the station attributes of the primary station. The status of one station is reflected in the status of the other and features activated at one are active at the other. All internal or external calls to a linked pair station will ring both stations.

All features available to the primary station are available and controllable by the secondary station, one station may activate Call Forward and the other may cancel the forward. The displays of the linked stations will display the status of the linked pair. When one is busy, the display 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**

1. Support for Linked Pairs is available with the iPECS IP Phones, Phontage and UCS client Soft phones and WIT 400HE WiFi phones only.
2. Intercom calls to the Linked stations always signal in the Tone ring mode and cannot be changed using the Caller Controlled ICM Signaling feature.
3. 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.
4. 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.
5. 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.
6. 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, Phontage, or SLT 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.
7. The IP Attendant application can be linked as the secondary phone only.
8. A SIP Phone cannot be part of a linked pair.

9. 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**

### **SYSTEM**

Linked Station Pairs Table (PGM 124)

## **Related Features**

- Intercom Caller Controlled ICM Signaling

## **Hardware**

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## 2.52 LNR (Last Number Redial)

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### Description

The last number dialed is stored (up to 48 digits) 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.

For iPECS IP and LDP Phones with displays, the last 10 numbers are stored in the LNR buffer. The user may view the ten numbers using the [VOL UP]/[VOL DOWN] button to scroll and select the number to dial from the list.

### Operation

#### LDP Phone

To use Last Number Redial using the [Redial] button:

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

#### iPECS IP Phone

To use Last Number Redial using the LIP-8000 series phone soft buttons

1. Press the [DIR] Soft button.
2. Press the [SPEED] Soft button.
3. Dial “\*”.
4. Use the Navigation button to select the desired number.
5. Press the [HOLD/SAVE] button to place the call.

#### SLT

To use Last Number Redial

1. Lift the handset.
2. Dial “552”, the Last Number Redial code.

### Conditions

1. For iPECS IP and LDP Phones with display, the redial buffer will store duplicate numbers unless dialed consecutively.
2. 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.
3. Using Last Number Redial will cancel Automatic Called Number Redial if active.
4. Manually dialing a Flash during an outgoing call will cause only those digits dialed after the Flash to be stored in the LNR buffer.

## **Programming**

### **Related Features**

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

## **Hardware**

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## **2.53     *Mobile Extension***

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### **Description**

Your mobile phone may be registered to a station allowing the mobile phone to place and receive calls through the system. DID calls are sent to your iPECS IP or LDP Phone and your registered mobile phone simultaneously. If the mobile phone is paired with a Hunt group station, Hunt 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 external calls 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, the system matches the Calling Line ID (CLI) to the mobile phone and provides you with system dial tone.

You may be permitted to register and activate a mobile phone for the Mobile Extension feature.

### **Operation**

#### ***iPECS IP LDP Phones***

##### *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.  
Or  
Dial a Park code to Park the call  
Or  
Dial a Page zone to place an Internal/External or All Call page within the iPECS

system.

Or

Dial a Conference Room to transfer the external party to a Conference Room.

**Note**

The mobile may reconnect to the call by dialing “#”.

**System**

*Incoming DID calls are sent to active mobile phones automatically.*

**Conditions**

1. When the mobile phone places an external call through iPECS, the CLI of the corresponding station is used.
2. The Mobile Extension features are not supported on analog CO Lines as answer and disconnect supervision is not reliable on these Line types.
3. Message Wait and Callback cannot be activated to a mobile phone.
4. The Mobile Extension feature is not supported over a distributed networked environment.
5. When an incoming ISDN DID call is received, the system will access an ISDN line and place a call to the mobile phone. Thus, an ISDN line must be available for the system to notify the mobile user of the incoming call.
6. Hold and Transfer Recalls to the mobile phone are sent to mobile phone and the associated station.
7. ACD, Circular and Terminal Hunt Group calls can be routed to the active Mobile Extension.

**Programming**

**SYSTEM**

Mobile Extension Table (PGM 236)

**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

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## **2.54 MOH (Music-On-Hold)**

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### **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. 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 ports can be assigned as SLT MOH ports and connected to an audio source and
- Up to three System announcements can be assigned, recorded and used for MOH, one announcement, number 71, is dedicated for MOH and two 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 71, 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**

1. The External music source should only be connected to one of the inputs.
2. The System announcements can be externally recorded and uploaded to the system.

3. The Internal source uses an \*.wav format and any 8-second recording can be uploaded to the system in place of the internal source.
4. Separate MOH announcements should be recorded for each Language when multiple languages are supported.

### **Programming**

<b>CO LINE</b>	MOH (PGM 142-Button 6)
<b>SYSTEM</b>	MOH Type (PGM 171-Button 2)

### **Related Features**

- Hold
- Back Ground Music
- Multiple Language Support

### **Hardware**

- External Music source connected to KSU music source input.



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## 2.55 Multiple Language Support

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### Description

With the integrated Auto Attendant and Voice Mail, iPECS can support six (6) languages simultaneously. Prompts in the desired languages are loaded into the integrated AA/VM memory along with the Language Selection prompts. To assure the proper language is employed, the Language Selection prompts are played when an incoming call is assigned to be answered by a DID, DISA, System Announcement or Station Hunt Group announcement. The Language Selection prompts are played in multiple phrases, one in each of the equipped languages, with a request for the caller to input a digit to select the appropriate language. The system then employs the defined announcement (DID, DISA, etc.) recorded for the selected language.

### Operation

#### System

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

To record a 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, 72.
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

1. The Language Selection prompts must be recorded as announcement number 72.
2. Multi-language support is only available using the integrated Auto Attendant.
3. Separate announcements must be recorded by the Attendant for each language supported.

#### Programming

**SYSTEM**

Multi-language (PGM 179)

#### Related Features

- Integrated Auto Attendant/Voice Mail
- Direct Inward Dial (DID)
- Station Groups
- Announcements
- Direct Inward System Access (DISA)

**Hardware**

- MEMU recommended

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## 2.56 Multiple Voice Mailbox Support

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### Description

A station can access any Voice Mailbox by dialing the VM group, the mailbox number and password. If you use an iPECS IP and LDP Phone, one or more Flex buttons of your phone can be configured to access a specific mailbox 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

1. A Flex button can be assigned to access the Mailbox for a Station Group or outside Line Voice Mailbox.
2. The **{VMAILBOX}** button LED will flash indicating new messages have been received in the associated mailbox.
3. A Flex button cannot be assigned to access the mailbox of a networked station.
4. 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.

### Programming

#### Related Features

- External Auto Attendant/Voice Mail
- Integrated Voice Mail

#### Hardware

- iPECS IP or LDP Phone
- External or Integrated Voice Mail

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## **2.57 Network Security & Priority**

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### **Description**

Each IP device that connects to the eMG80 supports several security and priority protocols. IP devices incorporate a Web server, which will deliver the Module Web Admin pages 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**

1. For Web Admin, the password is encrypted using the Ericsson-LG Java Virtual Encryption plug-in. A Java Virtual Machine (MS or Sun) must be installed in the user's PC to support password encryption.
2. Security and priority characteristics can be set for all devices, local or remote.
3. Devices must be on the same LAN to support SRTP.
4. The iPECS implementation of IPSec employs a proprietary Key exchange protocol from the MPB to the iPECS device.
5. IEEE 802.1 VLAN priority and ID are set at each device using the device Web interface. Diffserv, IPSEC and SRTP treatments may also be set in the device via the local Web interface. For the eMG80 MPB, VLANs are configured via the maintenance interface of the RS-232 port.

### **Programming**

<b>BOARD</b>	RTP Security (PGM 132-Button 4)
<b>SYSTEM</b>	Web Password Encryption (PGM161-Btn15)

### **Related Features**

### **Hardware**

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## 2.58 Outside Call Redirect

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### 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

To assign a Flex button for {DSS/BLF}

[TRANS/PGM] + {FLEX} + [TRANS/PGM] + destination + [HOLD/SAVE]

#### *iPECS IP & LDP Phones*

To activate Call Redirect while receiving an outside call:

#### To reroute a call,

1. Press the **{Call Redirect}** button; the call is redirected to the defined destination.

### Conditions

1. Conditions associated with Call Forward apply.
2. Only the call ringing at the time Call redirect is activated will be redirected. The next incoming outside call is routed as normal.

### Programming

### Related Features

### Hardware

- iPECS IP or LDP Phone

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## 2.59 Outside Call Time Restriction

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### 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

1. 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.
2. 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

<b>STATION</b>	CO/IP Call Time Restriction (PGM 112-Button 3) Cut Off Timer (PGM 123-Button 2)
<b>CO/IP</b>	CO Cut Off Timer (PGM 142-Button 21)

### Related Features

- Call Duration Warning Tone Timer

### Hardware

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## 2.60 Outside Line Access

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### 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} + 8 + 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**

##### 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

1. 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 “1<sup>st</sup> CO/IP Group Override” is enabled.

2. 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.
3. 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.
4. 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.
5. When an analog Line is seized, the system will monitor the Line for dial tone.
6. The system selects Lines from a group using the Round-robin or Last-choice method as assigned in the system database.
7. 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.
8. iPECS IP and LDP Phone users may be assigned exclusive use of a Line as a Private Line.

## **Programming**

<b>STATION</b>	Individual CO Access (PGM 112-Button 4) CO Line Programming (PGM112-Button 6) Ringing Line Preference (PGM112-Button7) Prefer CO/Group (PGM112-Button 14th) CO/IP Group Access (PGM 117)
<b>CO/IP</b>	CO/IP Group (PGM 141-Button 1) CO Ring Assignment (PGM 144)

## **Related Features**

- Outside Line Groups
- Line Ring Assignment
- Ringing Line Preference
- Private Line

## **Hardware**



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## 2.61 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**

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

1. Momentarily press the hook-switch.
2. Dial "556", the Callback feature code.

##### To cancel the queue from the queued station

1. Lift the handset.
2. Dial "556", the Callback feature 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

1. An outside Line can have any number of simultaneous queue requests.
2. 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.
3. A Queue recall will always signal the station with tone ring, ignoring the station's assigned Intercom Signaling mode.
4. Queue recall will bypass a busy station, and place the station at the bottom of the queue list.

5. Queue recall will signal a station for 15 seconds, after which, the station is removed from the queue; the queue is cancelled.

### **Programming**

**STATION**                      CO/IP Queuing (PGM 112-Button 5)

### **Related Features**

- Outside Line Access

### **Hardware**

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## **2.62    *Personal Groups***

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### **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-Up, Call Forward and 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**

1. Unlike the 'Linked Pair stations, the stations in Personal Group with the Linked option enabled maintain a separate station database and station number

### **Programming**

<b>SYSTEM</b>	Personal Group Assignment (PGM 260)
	Personal Group Attributes (PGM 261)

### **Related Features**

- SMDR
- Call Coverage
- Wake-Up
- Message Wait/Call Back
- Linked Station Pairs

### **Hardware**

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## 2.63 Pre-Defined & Custom Text Display Messages

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### 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 the message you selected.

There are ten Pre-defined messages (01-10), ten system-wide Custom messages 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 messages allow for auxiliary information such as a time, date or number.

System level Custom Messages are entered from the Attendant or Administrator's phone or 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 may include 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 DATE DD: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**

To activate a Display Message

1. Lift the handset.
2. Dial “561”, the SLT 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 Feature Cancel code.

To enter the User Custom Message (00)

1. Lift the handset.
2. Dial “561”, the SLT 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**

1. Only the System Attendant or Administrator can assign the contents of Custom Messages 11-20.
2. The alphanumeric characters are displayed as they are entered; refer to Station Speed Dial for entry procedures.
3. Display Message is cancelled if you activate DND or Call Forward.
4. Custom Text Display Messages and Display Message status are stored in non-volatile memory to protect against loss during power failure.
5. The calling station must be an iPECS IP or LDP Phone with display to receive the message.
6. SLTs 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.
7. Activating Text Display Message does not affect normal operation of the station.
8. Pre-defined Messages 01 to 04 and 06 to 08 permit you to input auxiliary information such as time, date or number.
9. The Attendant station can activate Text Display Messages for other stations. However, the Text Display Message feature is not available to an Attendant.

**Programming**

**Related Features**

- Silent Text Message
- DND (Do Not Disturb)
- Call Forward
- Speed Dial

**Hardware**

- iPECS IP or LDP Phone required to receive Display Messages

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## **2.64 Prefix Dialing Table**

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### **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**

1. 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**

<b>ISDN LINE DATA</b>	Prefix Dialing Table (PGM 206)
<b>CO LINE DATA</b>	Prefix Table ID (PGM 142-Button 20)

### **Related Features**

### **Hardware**



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## 2.65 Remote Control from Mobile Phone

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### 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

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

1. 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.
2. Selecting ACD Agent Off-Duty will place you Off-duty with Reason code 1.
3. 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

<b>NUMBERING PLAN</b>	Remote MEX Control (PGM109-Button 17)
<b>STATION</b>	Call Forward (PGM 111-Button 2) VSF Access (PGM 113 Button 2)
<b>TABLES</b>	Mobile PGM Authority (PGM 236-Button 1) Announce (PGM 236-Button 13)

### Related Features

- Call Forward
- Mobile Extension

### Hardware

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## 2.66 Ringing Line Preference

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### Description

When Ringing Line Preference is enabled, you answer incoming calls by 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

1. 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.
2. Intercom calls are always given the lowest answering priority.
3. Ringing Line Preference overrides Prime Line assignment.
4. SLTs operate only in the RLP mode; when ringing, lifting the handset connects the SLT to the ringing call.

### Programming

<b>STATION</b>	Auto Speaker Select (PGM111-Button 1) Ringing Line Preference (PGM 112-Button 7)
<b>CO/IP</b>	CO Ring Assignments (PGM 144)
<b>SYSTEM</b>	Ring Line Preference Priority (PGM 173)

### Related Features

- Prime Line Immediately/Delayed
- Automatic Speaker Select

### Hardware

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## 2.67 *Speed Dial*

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### 2.67.1 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

1. The number is displayed when programming the Speed Dial number.
2. Display Security does not affect the SMDR output.
3. Display Security is provided on all outside calls including calls that are transferred or recall.
4. An “\*” in any digit position other than the first, will activate Pulse to Tone Switchover.

#### Programming

#### Related Features

- Dial Pulse to Tone Switchover
- Speed Dial

#### Hardware

## **2.67.2 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**

1. The timed pause is used only with analog CO Lines.

### **Programming**

**SYSTEM**                      Pause Timer (PGM 181-Button 10)

### **Related Features**

- Station Speed Dial
- System Speed Dial

### **Hardware**

## 2.67.3 Station Speed Dial

### Description

You can store commonly dialed numbers for easy access using Station Speed Dial. With the eMG80, your Station has access to 100 Speed Dial numbers each stored in a “bin”. Each Speed Dial number can be up to 23 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
<b>[DND]</b>	Insert a Flash command
<b>[DND]</b> as 1 <sup>st</sup> digit	Switch to <b>{Keypad Facility}</b>
Flash as 1st digit	Activate dial tone detect.
<b>[MESSAGE/CALLBACK]</b>	Insert pause dialing command..

Your iPECS IP and LDP Phone may 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. Dial the number to be stored.
6. Press the **[HOLD/SAVE]** button.
7. If desired, enter a name, see alphanumeric entry chart below.
8. Press the **[HOLD/SAVE]** button.

#### ***iPECS IP Phones with 3-Soft buttons***

To dial using a Station Speed Dial using an LIP 8000 series phones with soft buttons

1. Press the **[DIR]** soft button.
2. Press the **[SPEED]** soft button.
3. Dial the desired bin number (00-99).

To program a Station Speed Dial number using an LIP-8000 series phones with soft buttons

1. Press the **[DIR]** soft button.
2. Press the **[SPEED]** soft button.
3. Press the **[ADD]** soft button.
4. Dial the Speed Dial bin number (00-99).
5. Press Line button or dial the Line access code.
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.

**SLT**

To dial using a Station Speed Dial

1. Lift handset.
2. Dial '558', the SLT Speed Dial access code.
3. Dial the desired bin number (00-99).

To program a Station Speed Dial number

1. Dial 561, the SLT Programming code.
2. Dial 555, the SLT Speed Dial access code.
3. Dial the Speed Dial bin number (00-99).
4. Dial the Line access code.
5. Dial the number to be stored.
6. Momentarily press the hook-switch.
7. If desired, enter a name, see alphanumeric entry chart below.
8. Momentarily press the hook-switch.
9. Alphanumeric characters may be entered to name the Speed Dial number using the chart below.

**ALPHANUMERIC DIAL-PAD ENTRIES**

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

## **Conditions**

1. Accessing an empty Speed Dial bin will return error tone.
2. 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.
3. All Speed Dial numbers are stored in memory protected from power loss.
4. 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.
5. You may pre-select a Line for a Speed Dial number, overriding the Line assignment in the Speed Dial bin.
6. A name can be entered for a Speed Dial number to permit access from the Dial-by-Name directory.
7. 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**

**SYSTEM**

CO Dial Tone Detect (PGM 160-Button 3)

## **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

## **Hardware**

## 2.67.4 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, up to 3000 System Speed Dial numbers are available. Each Speed Dial number can be up to 23 characters in length and may include special instruction codes.

Special instruction codes available are:

'*' as 1st digit	Activate Display Security.
Flash as 1st digit	Activate dial tone detect.
<b>[DND]</b>	Insert a Flash command.
'*' not 1st digit	Switch from Pulse to DTMF dialing.
<b>[MESSAGE/CALLBACK]</b>	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 (2000-4999)

To dial a System Speed Dial number using an LIP-8000 series phone with soft buttons

1. Press the **[DIR]** soft button.
2. Press the **[SPEED]** soft button.
3. Dial the desired bin number (2000-4999)

#### **SLT**

To dial using a System Speed Dial

1. Lift handset.
2. Dial "558", the SLT Speed Dial access code.
3. Dial the desired bin number (2000-4999)



### ***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 (2000-4999)
4. Press the Line button or dial the Line access code.
5. Dial the number to be stored.
6. Press the **[HOLD/SAVE]** button.
7. If desired, enter a name, see alphanumeric entry chart under Station Speed Dial.
8. Press the **[HOLD/SAVE]** button.

#### To enter a Station Speed Dial number using an LIP-8000 series phone with soft buttons

1. Press the **[DIR]** soft button.
2. Press the **[SPEED]** soft button.
3. Press the **[ADD]** soft button
4. Dial the Speed Dial bin number(2000-4999)
5. Press the **{LINE}** button or dial the Line access code.
6. Dial the number to be stored.
7. Press the **[HOLD/SAVE]** button.
8. If desired, enter a name, see alphanumeric entry chart under Station Speed Dial.
9. Press the **[HOLD/SAVE]** button.

### **Conditions**

1. Accessing an empty Speed Dial bin will return error tone.
2. 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.
3. All Speed Dial numbers are stored in memory protected from power loss.
4. 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.
5. The user may pre-select an outside Line for a Speed Dial number, overriding the assignment in the Speed Dial bin.
6. A name can be entered for a Speed Dial number to permit access from the Dial-by-Name directory.
7. 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**

**STATION**

Speed Dial Access (PGM 112-Button 8)

<b>SYSTEM</b>	CO Dial Tone Detect (PGM 160-Button 3)
<b>TABLES</b>	System Speed Zone (PGM 232)

### **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)

### **Hardware**

## **2.67.5 System Speed Zone (Groups)**

### **Description**

System Speed Dial bins can be divided into zones or groups. iPECS systems have 10 zones available. To access a Speed Dial 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 Dial numbers in the zone. When assigned, the system will apply the Station and Line COS (Class of Service) to calls using Speed Dial numbers in the zone.

### **Operation**

Refer to System Speed Dial for operation.

### **Conditions**

1. The conditions of System Speed Dial apply to System Speed Dial numbers assigned to a zone.
2. A station not assigned access to a zone will receive error tone when trying to access a System Speed Dial number in the zone.
3. The COS (Class of Service) is applied to the Speed Dial number based on the active service mode, Day/Timed or Night.

### **Programming**

<b>STATION</b>	Speed Dial Access (PGM 112-Button 8)
<b>TABLES</b>	System Speed Zone (PGM 232)

### **Related Features**

- System Speed Dial
- Class of Service

### **Hardware**

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## 2.68 Station Call Coverage

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### Description

The Call Coverage feature permits your iPECS IP or LDP Phone to receive ring and answer calls directed to a covered station. This feature is generally employed to allow a Secretarial answering position to cover calls to other stations. When a covered station rings, the **{CALL COVERAGE}** button LED will flash and the covering station may receive ring (immediate or delayed) for the call. The covering station can answer the call using the **{CALL COVERAGE}** button, terminating ring at other stations. Once answered, the LED of **{CALL COVERAGE}** buttons for the station at other covering stations will extinguish.

Operation of this feature requires a **{CALL COVERAGE}** button at the covering iPECS IP or LDP Phone and the covered station must activate call coverage. A station can have multiple Call Coverage buttons each covering a different station and multiple stations can have a Call Coverage button for 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} + "67" + 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 ("0"- "9").
4. When a covered station receives a call, the covering station will receive the following display:

CALL FOR STA xxxx FROM yyyy time
-------------------------------------

## **Conditions**

1. An iPECS IP or LDP Phone user may cover for an SLT or other stations. However, since a Flex button is required, an SLT cannot provide coverage for other stations.
2. 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.
3. 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.
4. Call Coverage attributes can be assigned either by the covered station user or in the system database.
5. A station cannot provide call coverage for a station to which it is linked.

## **Programming**

<b>STATION</b>	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)

## **Related Features**

## **Hardware**

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## **2.69     *Station Groups***

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### **Description**

Stations can be grouped for incoming call routing and Call Pick-up purposes. Ten types of groups 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 Hunt, calls to a station in the group will go to the station, if unavailable or unanswered in the hunt no answer time; the call will be directed to the next station defined in the group. The call will continue to hunt until each station in the group has been tried. The call remains 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) so that calls to the pilot number will hunt. In this case, the call will be directed to the first station in the 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 voice mailbox.

#### Terminal Station Group

Calls to a station in a Terminal Station Group that encounter an unavailable or unanswered status will be routed through the hunt process. The call will proceed to the next listed station in the group until reaching the last listed station in the group. The call may remain at the last station or be routed to an Overflow destination.

A Terminal Hunt Group can be assigned with a pilot number (the Station Group number) so that calls 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 has been idle for the longest continuous time, Uniform Call Distribution. If all stations are busy or unavailable when the call is received, the call may be routed to an alternate location or may continue to wait (queue) for an available station in the group. After queuing to the group, the caller may be routed to an overflow destination, which can be a Station, Station Group or Voice Mailbox.

An ACD supervisor can be assigned to monitor the group and act to oversee operations of the group. The ACD Supervisor can print group statistics and activate alternate routing as well as assist agents.

### Ring Station Group

A call to any station in the Group will cause all stations in the group to ring and any station in the group may answer the call. If the call remains unanswered beyond the Overflow timer, the call 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 employs SLT ports to interface to the iPECS. An External AA/VM group is assigned for either circular or terminal hunt. The External AA/VM may employ either in-band signaling over the audio channel or SMDI protocol with a signaling connection to the system RS-232 channel.

### Pick-Up Station Group

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

### Integrated AA/VM Group

Incoming calls can be directed to one of 70 user-recorded System announcements, which may request further routing instructions from the user in the form of caller dialed digits. These digits are employed to 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 from a station. Such calls will receive the user's pre-recorded greeting and may leave voice messages. The user may call the integrated AA/VM Group to review and manage the integrated Voice Mail application.

### Feature Server UMS Group

The Feature Server is a PC based TAPI application with high-end Auto Attendant, Voice Mail and Unified Messaging Service (Voice/Fax and e-mail). The iPECS Feature Server receives calls, plays announcements, stores voice messages and forwards them as wave file attachments to the user's e-mail. The application also receives Faxes and forwards them as attachments to e-mail. The Text-to-Speech option permits listening to e-mails as well as voice mails.

### 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 central iPECS. 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 iPECS systems.

### Group Announcements

Station Group routing can be augmented with System announcements recorded in the integrated AA/VM memory. Callers are routed to one of several user-recorded System announcements. The system answers the call and plays the defined 1<sup>st</sup> announcement to the caller. The announcement may provide the caller with routing options for Caller Controlled Routing. The 1<sup>st</sup> announcement may be "Guaranteed" meaning it will play in full before routing the call. A 2<sup>nd</sup> announcement can be provided to the caller should queue timers expire.

## **Operation**

### **Conditions**

1. The eMG80 supports 40 Station Hunt Groups with up to 70 members for each group.
2. Station Group calls are not routed to member stations that are in DND.
3. 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.
4. A call transferred to a Station Group will follow the routing for the group and will not initiate the Transfer Recall process.
5. Calls to a Station Group receive either ring-back tone or MOH while queued to the group.
6. 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.
7. One of the 70 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**

<b>STATION GROUPS</b>	Station Group Assignment (PGM 190)
	Station Group Attributes (PGM 191)
	Pickup Group Assignment (PGM 192)

## **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

## **Hardware**

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## **2.70 SMDR (Station Message Detail Recording)**

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### **2.70.1 Call Cost Display**

#### **Description**

Each SMDR call record includes a “Cost” field, which is a calculated estimate for the cost of the call. The call cost updates in real-time and displays in the LCD of iPECS IP and LDP Phones in place of the call 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, or VoIP), 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 estimate the call cost.

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

#### ISDN

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

#### VoIP

For VoIP calls, the system employs the call duration, cost/pulse and decimal values to establish the 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 the SMDR RS-232 port

#### **Conditions**

1. The call cost display begins after the “SMDR Start Timer” expires, if enabled, or at receipt of the first Call Meter Pulse.

2. Once connected to the system, the call duration includes the total time the call is connected including periods when the call is on hold, in queue, etc.
3. To enable Call Cost Display, the “SMDR Cost per Unit Pulse” and “SMDR Decimal” must be assigned. When not assigned, the call duration is provided by the system.

### **Programming**

<b>CO/IP</b>	Metering Unit (PGM 142-Button 3)
<b>SYSTEM</b>	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)

### **Related Features**

- SMDR (Station Message Detail Recording)
- Lost Call Recording
- Traffic Analysis
- Outside Call Warning Tone Timer

### **Hardware**

- RS-323 device to capture SMDR

## 2.70.2 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 or hold duration 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**

1. When the Lost Call Count is reset, the SMDR port will provide a "count reset" message.
2. Individual Lost Call records are only available in real-time and not on-demand.
3. "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.
4. The fields of a Lost Call Record are the same as a normal SMDR Call Record.

## **Programming**

**SYSTEM**

SMDR Attributes (PGM 177)

## **Related Features**

- Call Cost Display
- SMDR Call Records
- Traffic Analysis

## **Hardware**

- RS-323 device to capture SMDR

### 2.70.3 SMDR Call Records

#### Description

SMDR (Station Message Detail Recording) provides detailed information on incoming and outgoing 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. Call records are output either upon completion of the call (real-time) or in response to a request from the System Attendant. SMDR may be sent periodically via e-mail to a defined e-mail address.

The SMDR record output for external calls is as shown in the figure below. There are two flexible 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/CPN NUM-1	COST	ACCOUNT CODE	DIAL/CLI/CPN NUM II	DC
XXXX	SSSS	BBB	DD:DD	FF/FF/FF EE:EE	HCCCCCCCCC CCCCCCC	sssss	Aaaaaaaa aaa	Hcccccccccc cccccc	

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-Button 24-5), 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	DIALED NUMBER	COST	ACCOUNT CODE	CPN NUMBER
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	1) Station 100 called station 101 and station 101 answered. 2) 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 figures above

### System Attendant

#### To print SMDR records

1. Press the **[TRANS/PGM]** button.
2. Dial "0111", the SMDR print code.
3. Enter the desired station range.
4. Press the **[HOLD/SAVE]** button.

#### To delete stored records

1. Press the **[TRANS/PGM]** button.
2. Dial "0112", the SMDR delete code.
3. Enter the desired station range.
4. Press the **[HOLD/SAVE]** button.

#### To abort SMDR printing

1. Press the **[TRANS/PGM]** button.
2. Dial "0114", the SMDR abort code.
3. Press the **[HOLD/SAVE]** button.

## Conditions

1. 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).
2. Except for DISA calls, the duration of ring for an incoming call is provided in the Dialed number field.
3. A header, including the assigned "Customer Site Id" is output after two blank lines and is repeated every 66th line.
4. The SMDR output is a simple ASCII stream of up to 80 characters per line.
5. When enabled, SMDR call record timing begins after the "SMDR Start Timer" expires and ends at call completion.



6. 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.
7. 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.

## **Programming**

### **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)

## **Related Features**

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

## **Hardware**

- RS-323 device to capture SMDR

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## **2.71 System Administration**

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### **2.71.1 Keypad Administration**

#### **Description**

The system database can be accessed and modified with the keypad and Flex buttons of an iPECS IP or LDP Phone. The display of the Phone is employed to view items in the iPECS database. You may be required to enter a password for access to Admin. Based on 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 ***iPECS eMG80 Administration & Programming Manual***.

#### **Operation**

Operation is detailed in the *iPECS eMG80 Administration and Programming Manual*.

#### **Conditions**

1. 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.
2. The Keypad has access to most of the system database. However, certain characteristics can be modified through Web Admin.

#### **Programming**

<b>STATION</b>	Admin Access (PGM 113-Button 1)
<b>SYSTEM</b>	Admin Access Authority (Web only)
	System Password (PGM 162)

#### **Related Features**

- Web Administration
- Multi-Level Admin Access

#### **Hardware**

## **2.71.2 Multi-Level Admin Access**

### **Description**

Access to the Web Admin database is password protected. Up to three (3) levels of access to the database can be established by assigning a different password to each level. The Maintenance level has access to the entire database, all maintenance routines and defines the Admin Access Authority of the two remaining passwords. The User password can only access the defined database items and cannot access the Station Program pages. The Admin password has 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 eMG80 Admin & Maintenance Manual.***

### **Conditions**

1. Admin Access Authority is defined only in the Web Admin; it cannot be defined when using Keyset Admin.
2. Admin Access Authority applies to all Admin access whether accessed via an iPECS IP or LDP Phone, ISDN, or IP channel.

### **Programming**

<b>STATION</b>	Admin Access (PGM 113-Button 1)
<b>SYSTEM</b>	Admin Access Authority (Web only)
	System Password (PGM 162)

### **Related Features**

- Keyset Administration
- Web Administration

### **Hardware**

## **2.71.3 Web Administration**

### **Description**

The system database is accessed and modified via an iPECS IP or LDP Phone, the LAN interface or an ISDN BRI/PRI line. Both the LAN and ISDN access the system's Web server, which delivers the 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 accessible IP address. The IP address should be fixed either as a public IP address or through a NAPT server with port forwarding. The browser should be pointed to the system's IP address and the TCP port assigned in the system database.

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

When accessed, the system will deliver the iPECS Admin and Maintenance Web page. From this page, selecting Admin & Maintenance will return the login page where the user must enter a password that matches an assigned password. Based on the password entered, the user is permitted access to specified system program codes.

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

### **Operation**

*Operation is detailed in the iPECS eMG80 Administration and Programming Manual.*

### **Conditions**

1. For Web Admin, the password can be encrypted using the Ericsson-LG Java Virtual Encryption plug-in. A Java Virtual Machine (MS or Sun) must be installed in the user's PC to support password encryption.
2. 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.

### **Programming**

<b>NUMBERING PLAN</b>	System IP Address Plan (PGM Code 102)
<b>SYSTEM</b>	Web Server TCP Port (PGM 161-Button 14)
	WEB Password Encryption (PGM 161-Button 15)
	Admin Access Authority (Web only)

ISDN/ICLID 1 BRI/PRI PPP Web Admin Attributes (PGM 205)  
Web Login ID Usage (PGM 160-Btn24-Btn12)  
System Password (PGM 162)

### **Related Features**

- Keypad Administration
- Multi-Level Admin Access

### **Hardware**

## **2.71.4 Web User Manual**

### **Description**

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

### **Operation**

*Operation is detailed in the iPECS eMG80 Admin and Maintenance Manual.*

### **Conditions**

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

### **Programming**

### **Related Features**

- Web Administration

### **Hardware**

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## 2.72 System Alarm Notification Control

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### Description

The system has several fault alarms that can be enabled or disabled. When enabled, a Station within each Zone can be assigned to receive a notification of the alarm event. Alarms that can be enabled are:

- Emergency call placed
- Digital (BRI or E1/PRI) Line fault
- SIP Trunk/Extension registration fault.

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 Stop code or, if assigned, pressing the **{ALARM STOP}** 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

### Operation

#### System

At detection of fault or Emergency Call, the Alarm signal is sent to assigned stations.

#### iPECS IP & LDP Phones

To assign a Flex button as an **{ALARM STOP}** 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 STOP}** button.

### Conditions

1. The station must terminate the Alarm signal at their phone. However, a Main Attendant can terminate the Alarm signal at the System Attendant.
2. When no station is assigned to receive an Alarm, the System Attendant will receive the notification.
3. 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.

4. 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**

<b>SYSTEM</b>	Emergency Call Notify (PGM 163-Button 5)
	DCOB Fault Notify (PGM163-Button 6)
	SIP Registration Fault Notify (PGM163-Button 7)
<b>ZONE DATA</b>	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)

## **Related Features**

- Emergency Call
- Emergency Call Attendant Alert
- SIP Service

## **Hardware**

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## 2.73 System Networking

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### 2.73.1 Distributed Control Network

#### Description

In the Distributed Control Network, each iPECS system maintains control over the devices registered 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 other systems in 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 DND operation 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

1. To use the networking features, the software lock-key installation is required. Two types of lock-keys relate to the networking protocols. One is for QSIG based networking protocol, and the other is for VoIP based networking. To obtain a lock-key, contact your local Ericsson-LG representative with the system serial number.
2. Each station must have a unique station number of up to eight (8) digits, which is based on their local system Station Numbering Plan.

#### Station to Station Network Call

1. 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 eMG80 database.
2. If no idle path is available over the Network, you will hear error tone.
3. Station to Station Network calls always signal in the Tone ring mode and are not subject to Intercom Caller Controlled ICM Signaling.
4. 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.
5. Station to Station Network calls are subject to the CO Call Restriction Timer.



### ***Call Transfer, Networked Station***

1. There are two kinds of standard transfer methods in QSIG and H.450; Transfer by join and Transfer by rerouting. The main difference is how control of the connecting path between transferring and transferred station is handled. In case of Transfer by join, an additional connecting path will be needed to transfer the call to another station. In case of Transfer by rerouting, a new connecting path is used to transfer the call and old connecting path of transferring station will be cleared.
2. The transfer will be canceled if you press the flashing **{LINE}** or the **[TRANS/PGM]** button.
3. A call transferred over the network is not subject to Transfer Hold recall back to the transferring station. However, if the remote system does not respond in the Network Fault Recall time, you will receive recall for the transferred call.
4. If no idle path is available over the Network, you will receive error tone.
5. The receiving station must be idle, if you attempt to complete a transfer to a network station which is busy, you will receive error tone.

### **Network Call Back (Call Completion)**

1. A station can leave or have only one callback message, and a new request cancels the previous Call Back activating a new Call Back request with the busy station.
2. 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. In this case you can leave a voice message only if the called station forwards calls to the Voice Mail.
3. 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 Fault Recall Timer, the call will be cancelled.

### **Network Call Offer (Call Waiting/Camp-on)**

1. Call Offer is only applied to a station that is busy on a call.
2. If the called station is in a conference or paging, call offer is not available.

### **Conference**

1. Multi-line conference time is not applied on 3 party conference of networking.
2. If any station in the network conference hangs up during the conference, the net conference will be cancelled and the network path will be cleared.
3. SIP phones cannot initiate a Network Conference.
4. When SIP phone sends an Invite to non-IP phone, the message will be rejected. When a non-IP phone invites a SIP phone, the normal call setup messages are sent to the ISP phone instead of Invite Setup message.

### **Call Forward**

1. System does not check the status of diverted-to station that is in DND, CFW or Empty.

### **DND**

1. When a station is in DND mode, the **[NET DSS]** of DND station will flash if BLF manager is activated.

### ***Paging***

1. The Network Page code must be programmed in the Network Numbering Plan Table (PGM 324) in the caller's system.
2. The Network Page code and page destination must be programmed in the Network Page Table (PGM 325) in the called system.
3. Network Page destination can be Internal, External, or All Call Page.
4. You will receive error tone if there is no idle networking path or if the page destination is invalid.
5. When employing Network Page, the speakerphone of an iPECS IP or LDP Phone can be used.

### ***Message Wait***

1. When you receive a call from a station in networked system, the caller can leave a Station Message Wait or Voice Mail message using the **[MESSAGE/CALLBACK]** button. In addition, because the network path is considered an outside Line, your iPECS IP or LDP Phone will receive a CLI Message Wait.

### **Programming**

#### **SYSTEM**

Networking Basic Attribute (PGM 320)  
Networking CO Line Attribute (PGM 322)  
Network Numbering Plan Table (PGM 324)  
Network Feature Code Table (PGM 325)  
Network CNIP/CONP Enable (PGM 320-Button 3 & 4)

### **Related Features**

### **Hardware**

### 2.73.1.1 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 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

1. 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.
2. The system can provide Net DSS BLF status for all stations.
3. The BLF Manager provides BLF for stations only, status for Lines is not provided over the network; CO BLF is not supported.
4. 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.
5. Each system updates the station status to the BLF Manager at intervals of the BLF status duration.
6. 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

##### 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)

#### Related Features

#### Hardware

- Windows PC running the BLF Manager application

### 2.73.1.2 (CAS) Attendant Call

#### Description

When you place an Attendant call (dial "0") from a station in any networked system, the call may be routed to a Centralized Attendant. The local System Attendant activates Attendant DND, which routes local Attendant calls to the Central Attendant. The Centralized Attendant must have a **{Net DSS}** button for the remote Attendant to receive the calls.

#### 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

1. Only DID/DISA calls which are assigned to route only to the Attendant destination are routed to the Centralized Attendant.
2. The Centralized Attendant can be assigned as a Main Attendant in a networked system. Also, CO lines may be assigned to ring the Centralized Attendant.
3. A System Attendant cannot be assigned a **{NET DSS}** button and thus should not be a Centralized Attendant.

#### Programming

<b>STATION</b>	Attendant DND Button (PGM 115 – STN PGM 83)
<b>SYSTEM</b>	Attendant Assign (PGM 164)
	DID/DISA Destination (PGM 167)
	CO Ring Assignment (PGM 144)

#### Related Features

#### Hardware

- iPECS IP or LDP Phone

### **2.73.1.3 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**

1. 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.
2. 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**

<b>SYSTEM</b>	Network Numbering Plan Table (PGM 324-Button 6) Station Groups (PGM 190) – Ext AA/VM Group
---------------	---

#### **Related Features**

#### **Hardware**

### **2.73.1.4 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 an extension 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**

1. CO Transit-In calls are not subject to recall or time restrictions at the originating system.
2. If no path is available over the network, the caller will receive busy tone.
3. 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**

<b>SYSTEM</b>	Networking Basic Attribute (PGM 320)
	Networking CO Line Attribute (PGM 322)
	Network Numbering Plan Table (PGM 324)
	Network CNIP/CONP Enable (PGM 320-Button 3 & 4)

#### **Related Features**

#### **Hardware**

### **2.73.1.5 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**

1. 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.
2. The Station COS of the networked system is applied for toll restriction.
3. For CO Transit-Out, any numbering sequence code can be assigned in the Network Routing Table – Network Numbering Plan (PGM 324). If a conflict exists between the System and Network Numbering Plans, the System Numbering Plan has priority.
4. The terminating PSTN lines must be assigned as Networked PSTN CO lines in the terminating system.
5. For proper operation of CO Transit-Out, the Network CO Line type and System Usage must be assigned as PSTN.
6. 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.
7. When enabled in the Network Routing Table – Numbering Plan, you must enter a valid Station or System Authorization code for access. The associated COS is implemented for the call.
8. The system, which terminates the call, will display two SMDR records for the call, the original incoming call from the station number and the transit-out call.
9. The station receives busy tone when a networking path is not available during transit-out.
10. In the main system, the attendant must have the CO access authority about the public connection of figure.

#### **Programming**

##### **SYSTEM**

Networking CO Line Attribute (PGM 322-Button 4)  
Network Numbering Plan Table (PGM 324)  
Networking Basic Attribute (PGM 320)

#### **Related Features**

#### **Hardware**

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## **2.74     *Traffic Analysis***

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### **Description**

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

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

Attendants enable Periodic Reporting. Once enabled, the system continues to monitor and output the requested report until Periodic Report is disabled. On-demand reports selected by the System Attendant are output only upon request. The Traffic Report is sent to the defined system 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”.
  - 0121 All Summary Traffic Reports
  - 0124 Attendant Traffic Report
  - 0125 Call Summary Report
  - 0126 Hourly Call Report
  - 0127 Hardware Usage Summary Report
  - 0128 CO Summary Report
  - 0129 Hourly CO Report
3. Press the **[HOLD/SAVE]** button.

**Conditions**

1. The Print All Summary Traffic Reports generates the Attendant, Call Summary and CO Summary Traffic Reports.

**Programming**

<b>SYSTEM</b>	RS-232 Port Settings (PGM 174)
	Print Port Selections (PGM 175)

**Related Features**

- SMDR Call Records

**Hardware**

- device to capture reports

## 2.74.1 Traffic Analysis, Attendant

### Description

The Attendant Traffic Report covers operational statistics for the Attendants. The report outputs periodically or the Attendant requests output of the report for a defined Analysis period. The following 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
    
```

<u>Field</u>	<u>Description</u>
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**

1. The Peak Hour is the hour when the system has the highest total call volume.

### **Programming**

<b>SYSTEM</b>	RS-232 Port Settings (PGM 174)
	Print Port Selections (PGM 175)

### **Related Features**

- SMDR (Station Message Detail Recording)
- ACD Statistics Report

### **Hardware**

- device to capture reports

## 2.74.2 Traffic Analysis, Call Reports

### Description

Call activity statistics are provided in the Hourly Call and Call Summary Reports.

#### Hourly Call Report

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 as shown 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.

### Conditions

#### **Programming**

##### **SYSTEM**

RS-232 Port Settings (PGM 174)

Print Port Selections (PGM 175)

#### **Related Features**

- SMDR (Station Message Detail Recording)
- ACD Statistics Report

#### **Hardware**

- device to capture reports

### 2.74.3 Traffic Analysis, H/W Usage

#### Description

The Hardware Usage report provides statistics for the system's special Hardware resources such 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          Num          Anal          Total          Total
Type          Unit          Hour          Req            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.

#### Conditions

#### Programming

- SYSTEM** RS-232 Port Settings (PGM 174)
- Print Port Selections (PGM 175)

#### Related Features

- SMDR (Station Message Detail Recording)
- ACD Statistics Report

#### Hardware

- device to capture reports

## 2.74.4 Traffic Analysis, Line Reports

### Description

The CO Traffic Summary and Hourly reports provide statistics on a summary or hourly basis for all Line Group activity. The following provides a sample report and description of the major fields 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.

## **Conditions**

## **Programming**

<b>SYSTEM</b>	RS-232 Port Settings (PGM 174)
	Print Port Selections (PGM 175)

## **Related Features**

- SMDR (Station Message Detail Recording)
- ACD Statistics Report

## **Hardware**

- device to capture reports

---

## **2.75 System Scenario Call Routing**

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### **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

### **Conditions**

### **Operation**

#### ***Attendant***

##### To activate a Scenario Group

1. Press DND in system attendant.
2. Dial or select item 5.
3. Dial the Scenario Group number.
4. Press **[HOLD/SAVE]** button.

### **Programming**

#### **SYSTEM**

System Call Routing (PGM251-Web only)  
Station ICM Group ( PGM 125 )  
CO tenancy group (PGM 141-B10)

### **Related Features**

- Day/Night/Timed/Scenario Ring Mode

### **Hardware**



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## 2.76 Universal Night Answer (UNA)

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### Description

UNA 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", the UNA feature 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*

To access an incoming UNA call

1. Lift the handset.
2. Dial "567", the UNA feature code, the oldest UNA call is connected.

### Conditions

#### Programming

SYSTEM

Universal Night Answer (PGM 141-Button 7)

#### Related Features

- LBC (Loud Bell Control)

#### Hardware

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## 2.77 *Wake-Up Alarm, Enhanced*

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### 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

### Operation

#### *Attendant*

##### To register a Wake-Up alarm

1. Press the **[TRANS/PGM]** button.
2. Dial "044", the Attendant Set Wake-Up code.
3. Dial the station range to be alerted. If a single station is to receive the alarm, enter "\*" as the second station number.
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 Wake-Up

1. Press the **[TRANS/PGM]** button.
2. Dial "045", the Attendant Erase Wake-up code.
3. Dial the desired station range, for a single station, enter an "\*" as the second station number.
4. 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**

To register Wake-Up

1. Lift the handset.
2. Dial "561", the SLT 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. 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. Hook-flash, conformation tone is provided.

To stop the alarm notification

1. Lift the handset.

To erase Wake-Up

1. Lift the handset.
2. Dial "561", the SLT Programming code, confirmation tone is heard.
3. Dial "42", the erase Wake-up code.
4. Enter Wake-Up alarm index number, "1" to "5".
5. Hook-flash, a conformation tone is provided.

### Conditions

1. 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.
2. 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.
3. 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
4. When the Attendant erases the Wake-up alarm, all five alarm indices are erased.
5. Changing the Wake-up alarms from standard to enhanced, will cancel and erase all active Wake-up alarms.
6. For the one time alarm, the year can be set from 2000 to 2031.

### Programming

<b>STATION</b>	Wake Up Announcement (PGM 114-Button 24-6)
<b>SYSTEM</b>	Enhanced Wake-Up Usage (PGM 161-Button 24-23)

### Related Features

- Wake-up Alarm, Standard

### Hardware

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## 2.78 *Wake-up Alarm, Standard*

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### 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 visual signal.

### Operation

#### *Attendant*

##### To register a Wake-Up Alarm

1. Press the **[TRANS/PGM]** button.
2. Dial "044", the Attendant Set Wake Up code.
3. Dial the desired station range, for a single station, enter an "\*" in place of the second station number.
4. Dial the 2-digit hour and 2-digit minute for alerting.
5. For a daily (repeating alarm), dial "#".
6. Press the **[HOLD/SAVE]** button.

##### To erase Wake-Up

1. Press the **[TRANS/PGM]** button.
2. Dial "045", the Attendant Erase Wake-up code.
3. Dial the desired station range, for a single station, enter an "\*" in place of the second station number.
4. Press the **[HOLD/SAVE]** button.

#### *iPECS IP & LDP Phones*

##### To register Wake-Up

1. Press the **[TRANS/PGM]** button.
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. 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.
2. Dial "42", the Erase Wake-up code.

3. Press the **[HOLD/SAVE]** button.

## **SLT**

### To register Wake-Up

1. Lift the handset.
2. Dial "561", the SLT 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. For a daily (repeating alarm), dial "#".
6. Hook flash, conformation tone is provided.

### To stop the alarm notification

1. Lift the handset.

### To erase Wake-Up

1. Lift the handset.
2. Dial "561", the SLT Programming code, confirmation tone is heard.
3. Dial "42", the erase Wake-up code.
4. Hook flash, a conformation tone is provided.

## **Conditions**

1. 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.
2. When receiving a Wake up signal, lifting the handset will return MOH.
3. 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.
4. Time must be entered in the Military 24-hour format (hh:mm).
5. The daily alarm will reset and repeat each day until erased (cancelled). The One-time alarm will reset and cancel automatically.
6. A System announcement can be recorded and assigned to play when the user responds to the wake up alarm.

## **Programming**

**STATION**                      Wake Up Announcement (PGM 114-Button 24-6)

## **Related Features**

- Enhanced Wake-up Alarm

## **Hardware**

## 2.79 Web Call Back

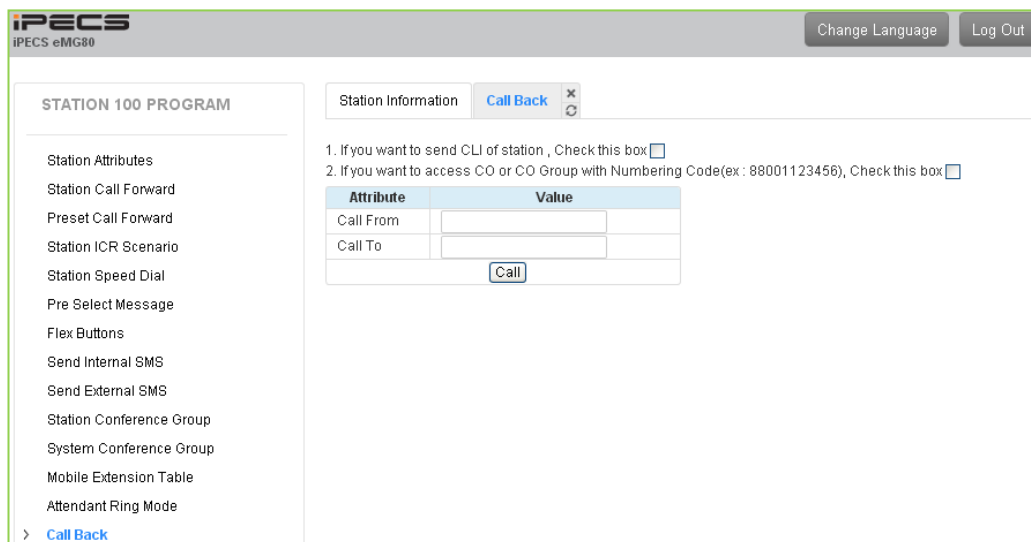
### Description

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 111 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.



The screenshot shows the iPECS eMG80 web portal interface. At the top, there are 'Change Language' and 'Log Out' buttons. The main content area is titled 'STATION 100 PROGRAM' and has two tabs: 'Station Information' and 'Call Back'. The 'Call Back' tab is active. Below the tabs, there are two instructions: '1. If you want to send CLI of station, Check this box ' and '2. If you want to access CO or CO Group with Numbering Code(ex : 88001123456), Check this box '. Below these instructions is a table with two columns: 'Attribute' and 'Value'. The table has two rows: 'Call From' and 'Call To', each with an empty input field. At the bottom of the table is a 'Call' button. On the left side of the page, there is a sidebar with a list of navigation options: 'Station Attributes', 'Station Call Forward', 'Preset Call Forward', 'Station ICR Scenario', 'Station Speed Dial', 'Pre Select Message', 'Flex Buttons', 'Send Internal SMS', 'Send External SMS', 'Station Conference Group', 'System Conference Group', 'Mobile Extension Table', and 'Attendant Ring Mode'. At the bottom of the sidebar is a link for '> Call Back'.

### Conditions

1. Web Call Back must be enabled in your Station’s attribute assignments.
2. 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

STATION DATA

Web Call Back Service (PGM 113-Button 24-11)

### Related Features

### Hardware

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## **2.80 Zone Management**

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### **2.80.1 Holiday Assignment**

#### **Description**

The Administrator can assign Holiday Zones (up to 40) and Vacation Zones (up to 5 for each Zone). Devices in a Zone follow the holiday and vacation assignments of the zone. When the date 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**

1. When the Tenancy group Ring mode for Flexible DID Conversion (PGM 231) is programmed, the priority is higher than the Zone Holiday Assignment.
2. Zone Holiday Assignment has a higher priority than the Attendant ring mode.
3. It is recommended to use the same zone for each outside Line and Station of a Tenancy group.

#### **Programming**

<b>SYSTEM</b>	Zone Holiday Assignment (PGM 444)
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#### **Related Features**

#### **Hardware**



## **2.80.2 Inter Zone Connection Limit**

### **Description**

To maintain voice quality, it may be necessary to limit the number of connected inter-Zone calls for a Zone so the available bandwidth is used effectively. Two parameters define the inter-Zone connection limits; the Zone maximum inter-zone connections and the maximum inter zone connections between Zones. As a default, no limit other than the system capacity is applied to the Inter Zone Connections.

### **Operation**

Operation of this feature is automatic when configured.

### **Conditions**

1. If maximum Inter-Zone call limit is set to 0, calls for the Zone are not limited.
1. When a Zone reaches the defined maximum Inter-Zone calls, you will receive error tone and the display "INTER CONN LIMIT OVER" when attempting an outside call.
2. An Inter-Zone call is determined when the call connection is made. If an incoming call is answered that places the Zone over the limit, the call is dropped and you receive error tone.
3. Emergency calls are counted as an outgoing outside call however, the Emergency call is always allowed.
4. If Two-way recording is active and the recording connection places the Zone over the limit, the call is not recorded.

### **Programming**

<b>ZONE DATA</b>	Zone Attributes (PGM 439-Web only)
	Inter Zone Attributes (PGM 441-Web only)

### **Related Features**

### **Hardware**

### **2.80.3 Remote Devices**

#### **Description**

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

- Nation code
- Language
- Page area
- Time-zone, GMT or system
- Holiday ring modes
- RTP relay
- RTP Relay group.

#### **Operation**

Zone operation is automatic once configured.

#### **Conditions**

1. 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.
2. Zone Attributes do not provide adjustment of time for DST.
3. It is recommended to assign outside Lines and stations of a Tenant group in the same Device Zone.
4. Wake-up time is based on the time displayed in the station's LCD.

#### **Programming**

<b>ZONE DATA</b>	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)

#### **Related Features**

#### **Hardware**

## **2.80.4 Zone Outgoing Call Connection Limit**

### **Description**

Every system has a limited number of outside Lines shared between all system users.. The number of outside Lines often is less than the number of other users. When all the Lines are in use, you may receive an All Trunks busy signal indicating no outside Lines are available.

To minimize this situation, outgoing outside calls from devices in a Zone can be limited from 1 to the maximum outside Line capacity of the system. As a default, the number of outgoing calls for each Zone is not limited.

### **Operation**

Operation of this feature is automatic based on zone maximum outgoing CO call number.

### **Conditions**

1. If maximum outgoing outside calls is set to "0", outgoing calls for the Zone are not limited.
2. When a Zone reaches the defined maximum outgoing calls, you will receive error tone and the display "OUTGOING CALL LIMIT OVER" when attempting an outside call.
3. Calls placed from a device in a Zone to a Networked system are considered outgoing outside calls for the Zone and will be limited by the outgoing call limit defined for the Zone.
4. Emergency calls are counted as an outgoing outside call however, the Emergency call is always allowed.

### **Programming**

**ZONE DATA**

Zone Attributes (PGM 439-Web only)

### **Related Features**

### **Hardware**

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## 3. INTERCOM

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### 3.1 Barge In

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#### 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 w/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. If Barge-in type 2 is assigned, press the **[JOIN]** soft button to join the conversation. The parties receive intrusion tone and you connect to the call.
2. Caller can join the conversation. The others hear “Intrusion Tone”.
3. To terminate the call and return the all parties to idle, press the **[DROP]** soft button.

###### To exit Barge-in

4. Hang-up the phone.

#### Conditions

1. This feature is only supported for iPECS IP and LDP Phone models with 3 soft buttons.
2. This feature is only available for active calls; you cannot barge in on a ringing or held call.
3. The Barge-In supervisor phone display will indicate the Id of the connected parties prior to the Barge-in operation.

## **Programming**

### **STATION**

Barge In Mode (PGM 113 Button 24-7)

## **Related Features**

- Intrusion

## **Hardware**

- iPECS IP or LDP Phones w/3-soft buttons

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## 3.2 Direct Station Selection/Busy Lamp Field (DSS/BLF)

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### Description

You can assign Flex buttons on iPECS IP and LDP Phones, or DSS Consoles to 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

To assign a Flex button for {DSS/BLF}

[TRANS/PGM] + {FLEX} + Station number + [HOLD/SAVE]

#### *iPECS IP & LDP Phones*

Operation of this feature is automatic for assigned Flex buttons.

### Conditions

1. A station receiving ICM ringing is considered busy and the associated DSS/BLF LED on all other stations will flash at 30 ipm.
2. A station receiving ICM ringing will receive visual indication by a flashing LED of the Flex button associated with the calling station.
3. When a station receives a Camp-On the LED of a DSS/BLF button associated with the calling station will flash.
4. The station is considered busy when in use, receiving ICM Ring at an iPECS IP or LDP Phone, or receiving any ring at an SLT.

### Programming

#### Related Features

- Intercom Call (ICM Call)
- Station User Programming & Codes

#### Hardware

- iPECS IP or LDP Phone

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## 3.3 Intercom (ICM) Call

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### Description

A non-blocking ICM is available to all stations in the system. You may place an intercom call to other stations in the system by dialing applicable digits as defined in the Station Numbering Plan.

### 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*

##### 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

1. Intercom dial tone will time-out if action is not taken within Dial-Tone Time or if the time between digits exceeds the Inter-digit Timer. Error tone is received should a time-out occur.
2. ICM Dial tone is removed after dialing the first digit.
3. If the called station is busy, Intercom Busy tone is provided for the Busy Tone time (7 seconds) then, 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.
4. 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.
5. 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

<b>STATION</b>	Prime Line (PGM 113-Button 7)
<b>SYSTEM</b>	ICM Dial Tone Timer (PGM 181-Button 6) Inter Digit Timer (PGM 181-Button 7)

### Related Features

- Intercom Signaling Mode
- Speakerphone

### Hardware

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## **3.4 Intercom Call Hold**

---

### **Description**

While on an active Intercom (ICM) call, using an iPECS IP or LDP Phone you can place the call on hold. The held station will receive the assigned Music-on-Hold. The call is placed on Exclusive Hold and recalls 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 [ICM] 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**

1. Only one Intercom call may be placed on hold at a station.

### **Programming**

#### **Related Features**

- MOH (Music-On-Hold)
- Intercom Call (ICM Call)
- Exclusive Hold
- Hold Recall

### **Hardware**

- iPECS IP or LDP Phone



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## ***3.5 Intercom Caller Controlled Signaling***

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### **Description**

You can change the signaling mode of an Intercom (ICM) call from Tone ring to Voice announce or 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***

##### 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**

1. 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.
2. If the signaling mode is changed, the call is not subject to Call Forward, No Answer.
3. The signaling mode for a specific Intercom call can only be changed once and cannot be changed back to the original signaling mode.
4. Changing the signaling mode does not affect privacy at the called station.

### **Programming**

**STATION**                                      Caller Controlled ICM Signaling (PGM 111-Button 15)

### **Related Features**

- Intercom Signaling Mode
- Linked Station Pairs

### **Hardware**

- iPECS IP or LDP Phone to receive Voice announced calls

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## 3.6 Intercom Lock-Out

---

### Description

If you take no action after going off-hook for the Dial Tone timer or fail to dial an additional digit within the Inter-digit timer, you will receive error tone for 30 seconds and be placed out-of-service (locked-out). The LED of associated {DSS/BLF} buttons as well as the 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 30 second 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

1. 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.

### Programming

<b>STATION</b>	Howler Tone (PGM 111-Button 5)
<b>SYSTEM</b>	ICM Dial Tone Timer (PGM 181-Button 6) Inter Digit Timer (PGM 181-Button 7)

### Related Features

- Intercom Call (ICM Call)

### Hardware

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## **3.7 Intercom Step Call**

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### **Description**

When busy tone is received on a dialed Intercom call, you may place a call to another station by dialing the last digit of the station number. The system replaces the last digit of the previously dialed busy station with the dialed digit and places an Intercom call to the new station number.

### **Operation**

#### ***iPECS IP & LDP Phones***

To activate step call, while receiving busy on a dialed Intercom call:

1. Dial a digit other than the last digit of the busy station's intercom number.

### **Conditions**

1. If the user dials the last digit of the busy station, Camp-On will be activated.
2. After receiving busy tone, if the user takes no action for the Busy Tone timer, 7 seconds, the system will start the Intercom Lockout procedure.

### **Programming**

### **Related Features**

- Intercom Lock-Out
- Intercom Call (ICM Call)

### **Hardware**

- iPECS IP or LDP Phone

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## **3.8 Intercom Tenancy Group**

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### **Description**

Stations in the iPECS are assigned to an Intercom Tenancy Group, group 01 ~ 15. Stations in a group are allowed or denied the ability to place intercom calls to stations in other groups on a group-by-group basis.

Each Intercom Tenancy Group is assigned an Attendant station. All dial "0" calls from a station in the group are routed to the assigned Attendant. In addition, the assigned Attendant can control 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) mode automatically during the day and night service mode. In addition, DID calls to the system can be routed to a specified Tenancy group. By defining the group as the destination in the 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**

1. Intercom calls from a station to a denied access Intercom Tenancy Group will return error tone.
2. 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.
3. The Attendant of a Tenancy Group can be any station in the system and it is not affected by Tenancy Group access.
4. When the Attendant of an Intercom Tenancy Group sets Day/Night/Timed mode, it will affect only the assigned Tenancy Group.
5. 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.
6. Intercom Tenancy Group 00 is the default or unassigned group.
7. 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**

<b>STATION</b>	ICM Group Number (PGM 111-Button 17) ICM Tenancy Group (PGM 125)
<b>CO LINE</b>	Tenancy Group (PGM141-Button 10)

**TABLES**

Flexible DID Conversion Table (PGM 231)

Weekly Time Table (PGM 233)

**Related Features**

- Intercom Call (ICM Call)
- Outside Line Ring Assignment
- Flexible Numbering Plan
- Call Transfer
- Auto Service Mode Control

**Hardware**

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## **3.9 Intercom Transfer**

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### **Description**

You can transfer an active Intercom call to other stations in the iPECS system. Intercom calls can be transferred after announcing the call (screened) or without announcing the call (unscreened).

The Intercom station is placed on Exclusive Hold. The Transfer Recall Timer is initiated and, if this timer expires before the Intercom call is answered, the call will recall your station until 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.
2. Dial the station to receive call.
3. At answer or Splash tone, announce call.
4. 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**

##### *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**

1. The **[ICM]** button, if configured, provides an appearance for the transferred station. The LED indicates status and pressing the button connects to the station.
2. If the receiving station is busy, the transferring station may camp the call on to the busy station, see Camp-On.
3. A station in DND or out-of-service cannot receive a transfer, and such attempts will result in error tone.

**Programming**

**Related Features**

- Exclusive Hold
- Hold Recall
- Call Waiting/Camp-On
- DND (Do Not Disturb)
- Call Transfer

**Hardware**

## 3.10 Message Wait/Call Back

### 3.10.1 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.
- 2.

ST	CL	VS	VM	FS	MS
001	001	005	006	001	004

3. Dial '2' to select CLI message shown as CL in the LCD. The LCD displays the first CLI message details including the Caller Id, time and number of missed calls from the caller.
4. Use the **[VOL UP]/[VOL DOWN]** buttons to scroll through the missed call list.

5551122323
21/11 09:10 CNT: 2

Or

5. Press the **{CLI MESSAGE WAIT}** button, the first CLI message is displayed.
6. Press the **[VOL UP]/[VOL DOWN]** button to scroll through the CLI Messages.

5551122323
21/11 09:10 CNT: 2



To delete the current CLI Message

1. Press the “\*” button, the next CLI message is displayed.

To delete all CLI Messages

1. Press the **[SPEED]** button.
2. Press the “\*” button twice.

To return a call to the current CLI Message

1. Press the **[HOLD/SAVE]** button.

**Conditions**

1. The total number of Message Wait indications in queue for the system is 1000. Should the message Wait buffer fill, the buffer full message, as below will be output over the RS-232 port.
2. “Warning: Message Wait Buffer Full.”
3. The CLI Message Wait display includes the date, time, message count and caller Id. The count is the number of attempts by a specific calling party with the time and date from the most recent attempt.
4. The CLI Message Wait status is stored in non-volatile memory to protect against power loss.
5. When an outside call is assigned to ring multiple stations, or a group, CLI Message Wait is not provided.
6. When an outside call with CLI is routed to an ACD/Terminal/Circular station group, the CLI Message Wait will be provided to the last station notified of the call before the call was abandoned.
7. Once a call is answered by a station in the system, CLI Message Wait is no longer available i.e.; transferred calls do not provide CLI Message Wait.
8. A CLI Message Wait indication is left at the originally called station even if the call is forwarded to another station.
9. While idle, the user may toggle between CLI number and CLI name, if available, by pressing the **[TRANS/PGM]** button.
10. The CLI and called station number will be shown in the SMDR record as an abandoned or lost call.
11. For ISDN calls, the CLI Message Wait will display and dial the number as shown in the Table below, modified based on assignments in PGM 200 for:

Incoming Prefix Code  
My Area Code  
My Prefix Area Code  
International Access Code

<b>Inc Prefix</b>	<b>Area Code</b>	<b>Country Code</b>	<b>Access Code</b>	<b>Received Caller ID</b>	<b>Display</b>
No	0343	82	00	National: 0343 50-7951	507951
No	0343	82	00	National: 0343 50-7951	507951
No	0345	82	00	National: 0343 50-7951	0343 507951
Yes	0343	82	00	National: 0343 50-7951	507951
Yes	0343	82	00	National: 0343 50-7951	0343 507951
Yes	0343	82	00	International: 492 888 7777	00492 8887777

### **Programming**

#### **STATION**

CLIP Display (PGM 114-Button 1)

CLI Message Wait (PGM 114-Button 10)

#### **SYSTEM**

Incoming Prefix Code Insertion (PGM 143-Button 12)

International Access Code (PGM 143-Button 14)

My Area Code (PGM 143-Button 15)

My Prefix Area Code (PGM 143-Button 16)

### **Related Features**

- Station Message Wait/Call Back
- ISDN (Integrated Service Digital Network)

### **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, the iPECS IP and LDP Phones can be sent a tone as a periodic reminder to you of message waits in queue. This tone is sent to the station only while idle and is heard over the speaker.

### **Operation**

#### **System**

Reminder tone is sent to stations automatically when assigned.

### **Conditions**

1. The interval between tones can be 00 to 60 minutes. A setting of 00 disables the reminder tone.
2. The reminder tone will continue until all messages have been retrieved.
3. A busy station or station in DND will not receive the Message Wait Reminder tone.
4. An SLT will receive stutter dial-tone when the handset is lifted to indicate messages are waiting.

### **Programming**

**SYSTEM**                      Message Reminder Tone Timer (PGM 181-Button 8)

### **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 the Phontage, 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 SEND} 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)

- Dial the desired station range to receive the message.

STA 100-124  
 SEND-(SAVE) CANCEL\_(#)

- Dial the message using two digits for each character as shown in the table below, use the **[REDIAL]** button to delete a character.

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

- Press the **[HOLD/SAVE]** button to send and store the message or “#” to cancel.

To retrieve SMS Messages:

- Press **[MSG]** or **[CALL BACK]** button. The message contents summary will be shown as below.

ST CL VS VM FS MS  
 001 001 005 006 001 004

- Dial “6” to select MS, SMS message wait.

“1” => ST: Station Message wait,

“2” => CL: CLI Message wait

“3” => VS: Integrated Voice message wait

“4” => VM: external Voice Mail

“5” => FS: Feature server

“6” => MS: SMS message wait.

The display shows the first line of received SMS messages.

MESSAGE: [0] EMERGENCY CONFER  
 [1] LETS GO TO LUNCH

3. Use the **[VOL UP]/[VOL DOWN]** to scroll through messages.

```
MESSAGE: [2] WE HAVE A MEETING A  
[3] CONFERENCE AT 9 AM
```

4. Dial the message number (“01”~“10”) to view the entire message, use the **[VOL UP]/[VOL DOWN]** to view each line of the message.

```
MESSAGE:> 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**

3. The system can store up to 10 received SMS messages and 1 sent SMS message for a station.
4. Once the received message storage capacity is reached, additional incoming SMS messages are ignored.
5. For operation with the Phontage, WLAN Phone or UCS Client, refer to the respective User Guide.

#### **Programming**

#### **Related Features**

#### **Hardware**

- iPECS IP or LDP Phone w/Display
- iPECS Phontage
- iPECS WLAN Phone
- iPECS UCS Client

### 3.10.4 Station Message Wait/Call Back

#### 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.

When a busy station is called, you may request to be placed in a queue to receive a Call Back. When the called station returns to idle, the system notifies you with Callback ring. When you answer, the previously busy station is called.

When a Message is waiting, the **[MESSAGE/CALL BACK]** button LED of iPECS IP or LDP Phones flashes and, for properly equipped SLTs, the MW Lamp will flash.

#### 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 leave a Call Back (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.

ST CL VS VM FS MS 001 001 005 006 001 004
--

2. Dial "1" to select ST (Station Message Wait)  
"1" => ST: Station Message Wait,  
"2" => CL: CLI Message Wait  
"3" => VS: Integrated VM Message Wait  
"4" => VM: external Voice Mail

“5” => FS: Feature server

“6” => MS: SMS message wait.

3. Press the **[VOL UP]/[VOL DOWN]** button to scroll through the Station Messages.



MSG : 100

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**

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”, the 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”, the Message Wait/Call Back code.
3. Hang up, Message Wait activated.

To retrieve a Station Message Wait:

1. Dial “557”, the 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”, the 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**

1. A Message Wait/Call Back return call will always ring at the receiving station overriding the Intercom signaling mode selected.
2. A station can leave only one callback request.
3. The **[MESSAGE/CALL BACK]** button LED will continue to flash until all Message Wait and Call Back requests, including Voice Mails, have been serviced.



4. If a station is attempting to leave a message and the system Message Wait queue is full, the station will receive ICM busy tone.
5. A Message Wait remainder tone can be enabled to remind the user of messages waiting.
6. A station in Call Forward can leave a message wait.
7. A Message Wait indication is left at the originally called station even if the call is forwarded.
8. 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.
9. Placing an Intercom call to a station will cancel any existing Message Wait from that station.

## **Programming**

**SYSTEM**                      Message Reminder Tone Timer (PGM 181-Button 8)

## **Related Features**

- Message Wait Reminder Tone
- CLI Message Wait

## **Hardware**

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## 3.11 *Paging*

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### 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
Internal All Call Page	543
External Page Zones	545
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/CALL BACK]** button.
2. Replace the handset returning to idle.

## **SLT**

### 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", the Call Back code.
2. Replace the handset returning to idle.

## **Conditions**

1. Stations, which are denied access to paging, will receive error tone when any Page Access code is dialed.
2. Stations dialing a Page Code will be queued when any of the other Internal or External Page zones are busy.
3. DTMF signals from stations are transmitted through the systems External Page port after access.
4. If an iPECS IP or LDP Phone user attempts to page using the speakerphone, pre-selection will be activated and the display shows "LIFT THE HANDSET TO PAGE".
5. Stations in DND or busy will not receive Page announcements.
6. Stations, which are not included in any Internal Page Zone, will not receive any page, including All Call.
7. For external paging, an external amplifier and speaker(s) are required.
8. The 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.
9. 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.
10. 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.
11. The queue recall is always in the tone ring mode regardless of the station's ICM signaling mode.
12. 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.
13. 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.
14. When the waiting station goes to idle, and both a "Page Queue" and "CO Call back Queue" exist, the Page Queue is given priority.

## **Programming**

<b>STATION</b>	Page Access (PGM 111-Button 7)
	Internal Page Zone Access (PGM 118)
<b>SYSTEM</b>	Privacy Warning Tone (PGM 161-Button 4)
	External Control Contacts (PGM 168)
	Paging Timeout Timer (PGM 181-Button 9)

## **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}** button.

### **Operation**

#### ***iPECS IP & LDP Phones***

To assign a Flex button as a **{MEET ME}** 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}** button.

#### ***SLT***

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**

1. A Meet Me Page must be answered within the Page Time-out timer.
2. A station may answer a Meet Me Page from any station regardless of pickup or paging group assignments and page access permission.
3. The paging party must remain off-hook until the paged party answers the Meet Me request. The initiator may press the Mute button to eliminate transmitting audio over the page circuit while waiting for the party to answer.

### **Programming**

**SYSTEM**                      Paging Timeout (PGM 181-Button 9)

### **Related Features**

- Internal/External & All Call Page

### **Hardware**

### 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

1. This feature can be only activated by pressing assigned flexible button in idle state.
2. The System announcement for SOS Paging can be recorded only from an Attendant station.
3. Paging zones includes Internal, External and All Call page zones.
4. 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

<b>STATION</b>	Page Access (PGM 111-Button 7)
<b>SYSTEM</b>	Paging Timeout Timer (PGM 181-Button 9)

#### Related Features

- Internal/External & All Call Page
- Integrated Voice Mail

#### Hardware

- iPECS IP or LDP Phone

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## 3.12 Push to Talk Paging

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### 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**

1. Conditions associated with Internal/External & All Call Page apply to Push-to-Talk paging.
2. To access PTT paging, the station must be permitted access to system paging and assigned to one or more PTT groups..
3. If allowed access to all PTT groups, a station may log-into all groups (PTT group 0) to place announcements to all groups simultaneously and receive announcements from any group.
4. A station can only login to PTT groups to which it is assigned access.
5. The station must have a **{PTT}** to place or receive PTT announcements. As a default, the iPECS WLAN phone is assigned a PTT button.
6. The station may be assigned and logged in to the default active PTT group in the system database.

### **Programming**

<b>NUMBERING</b>	PTT Log-in/out (PGM 109-Button 4)
<b>STATION</b>	Page Access (PGM 111-Button 7)
	Default Active PTT Group (PGM 111-Button 16)
	PTT Group Access (PGM 119)

### **Related Features**

- Internal/External & All Call Page

### **Hardware**

- iPECS IP or LDP Phone
- iPECS WLAN
- iPECS Phontage
- iPECS UCS Client



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## 4. *OUTSIDE (CO/IP) LINES*

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### 4.1 *Auto Fault Detection and Recovery*

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#### **Description**

If the system detects a Line fault on a PRI Line or SIP Trunk, the Line/Trunk is 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.

#### **Operation**

##### **System**

Operation of Fault Detection and Recovery is automatic.

#### **Conditions**

1. The “Unused” Line Group contains Lines that are not used or are temporarily blocked.
2. This feature is only available for PRI Lines and SIP trunks.

#### **Programming**

<b>CO/IP</b>	CO/IP Line Group (PGM 141-Button 1)
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#### **Related Features**

- Outside Line Groups

#### **Hardware**

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## **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**

### **Programming**

<b>CO/IP</b>	CO Additional Attribute - CID setting (PGM 148)
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### **Related Features**

### **Hardware**

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## **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***

While connected to an analog CO line

1. Momentarily depress the Hook-switch.
2. Dial “551”, the Flash feature code, the system generates a Flash on the analog CO Line.

### **Conditions**

1. 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.
2. During a Flash, the LED for the Line button will remain lit.
3. A Flash may be stored as a part of a Station or System Speed Dial number.
4. While connected to an internal call or dial tone, activating Flash will return internal dial tone.
5. While making a Page Announcement, activating Flash will terminate the Page Announcement and return internal dial tone.
6. The Flash function is not available on ISDN and IP Line calls.

### **Programming**

**CO/IP**

Flash Type (PGM 141-Button 6)

CO Flash Timer (PGM 142-Button 12)

### **Related Features**

- Station Speed Dial
- System Speed Dial

### **Hardware**

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## **4.4 CO Line Release Guard Time**

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### **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 the Line. 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.

### **Conditions**

### **Programming**

<b>SYSTEM</b>	CO Release Guard Timer (PGM 180-Button 16)
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### **Related Features**

### **Hardware**

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## **4.5 CO Line Ring Detect**

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### **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**

1. Ring On and Ring Off are assigned on a system basis.
2. The CO Ring Detect is applied to analog CO Lines only.

### **Programming**

<b>SYSTEM</b>	CO Ring OFF Timer (PGM 180-Button 17)
	CO Ring ON Timer (PGM 180-Button 18)

### **Related Features**

### **Hardware**

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## **4.6 Dial Pulse Signaling**

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### **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**

1. The break/make ratio is system programmable as 60/40 or 66/33.
2. You can change the signaling from pulse to DTMF by dialing '\*' while on a Line assigned for pulse signaling.

### **Programming**

<b>CO/IP</b>	CO Line Signal Type (PGM 141-Button 5)
<b>SYSTEM</b>	Pulse Dial (Break/Make) Ratio (PGM 176)
	Pause Timer (PGM 181-Button 10)

### **Related Features**

- Dial Pulse to Tone Switchover
- DTMF Signal Sending

### **Hardware**

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## **4.7 Direct Inward Dialing (DID)**

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### **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**

1. 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.
2. DID calls that encounter a busy signal, are not answered in the DID/DISA No Answer Timer, or are received at a vacant or invalid number 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.
3. For a station that is part of a non-pilot Station Hunt group, DID calls will follow the group hunt process if the Station is busy or does not answer the call. DID calls are subject to Group Call Pick-up and Directed Call Pick-up.
4. 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**

<b>SIP</b>	SIP User ID Attributes (PGM 126-Web only)
<b>CO/IP</b>	CO Service Type (PGM140)
	ISDN DID Remove the Number of Digit (PGM 143-Button 5)

	DID Conversion Type (PGM 145-Button 2)
<b>SYSTEM</b>	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)
<b>TABLES</b>	Customer Call Routing Tables (PGM 228)
	Flexible DID Conversion Table (PGM 231)

### **Related Features**

- Integrated Auto Attendant
- Directed Call Pick-Up
- Group Call Pick-Up
- IP Trunking
- IP Address Dialing

### **Hardware**

- IP, BRI or PRI Line



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## ***4.8 DID Line Digit Conversion***

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### **Description**

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.

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**

1. Each CO line can have a Digit Conversion Tables.
2. There are two (2) special entries for configuring digit conversion:
  - X: Mask Digit, any digit is accepted
  - F: Ignore Digit

### **Programming**

<b>SYSTEM</b>	Digit conversion Table (PGM 270)
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### **Related Features**

### **Hardware**

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## 4.9 Direct Inward System Access (DISA)

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### 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 to a 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

1. 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).
2. 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.
3. 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.
4. 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.
5. 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.

6. 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.
7. 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.
8. An iPECS IP or LDP Phone user can only receive a DISA call with an available DISA Line appearance button.

## **Programming**

<b>CO/IP</b>	DISA Account Code (PGM 142-Button 5) DISA CO Access (PGM 142-Button 11) DISA Attributes (PGM 146)
<b>SYSTEM</b>	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)
<b>TABLES</b>	CCR Audio Text Tables (PGM 228) Weekly Time Table (PGM 233)

## **Related Features**

- Day/Night/Timed/Scenario Ring Mode
- Dialing Restrictions
- Authorization Codes (Password)
- Unsupervised Conference
- Integrated Auto Attendant/Voice Mail
- Auto Service Mode Control

## **Hardware**

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## **4.10 DTMF Signaling**

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### **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**

1. The system mutes your voice transmission to reduce interference while sending DTMF tones.

### **Programming**

<b>CO/IP</b>	CO Line Signal Type (PGM 141-Button 5)
<b>SYSTEM</b>	Inter Digit Timer (PGM 181-Button 7)
	DTMF Duration Timer (PGM 182-Button 9)

### **Related Features**

- Dial Pulse Signaling

### **Hardware**

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## 4.11 IP Address Dialing

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### 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**

##### 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.

### Conditions

### Programming

<b>STATION</b>	CO/IP Group Access (PGM 117)
	Direct IP Call (PGM 122-Button 1)

### Related Features

### Hardware

- IP Line

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## **4.12 IP Trunking**

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### **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
- H.450.2
- H.450.3
- H.450.4
- H.450.5
- H.450.6
- H.450.7
- H.450.8
- H.450.9
- H.450.10
- H.450.11
- H.450.12

#### **Operation**

##### **System**

Operation of H.323 Service is automatic

#### **Conditions**

#### **Programming**

<b>STATION</b>	Station IP Attributes (PGM 122)
<b>BOARD</b>	H.323 VoIP Attributes (PGM 130)
<b>CO/IP</b>	CO/IP Line Group (PGM 141-Button 1)
	CO VoIP Mode (PGM 141-Button 11)

#### **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

### **Conditions**

### **Programming**

<b>STATION</b>	SIP User ID Table (PGM111-Button 19) Station IP Attributes (PGM 122)
<b>SIP</b>	SIP User ID Attributes (PGM 126-Web only)
<b>BOARD</b>	SIP Attributes (PGM 133, Web only)
<b>CO/IP</b>	CO/IP Line Group (PGM 141-Button 1) CO VoIP Mode (PGM 141-Button 11)

### **Related Features**

- System Networking
- H.323 v4 Service

### **Hardware**

- SIP Trunk

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## **4.13**    ***IP WAN Dialing after Answer***

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### **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**

1. 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**

### **Related Features**

### **Hardware**

- IP Line



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## 4.14 ISDN (Integrated Service Digital Network)

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### 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 switches on 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.

iPECS IP and LDP Phone display:

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:

- AA – ISDN CO line number
- BB...B –Calling/Connected Line Identification received
- CCC – Called station
- DD...D – Speed Dial name
- EEE – Speed Dial bin number
- 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

1. Press the **{CLIR}** or **{COLR}** Flex button.

### **Conditions**

1. This feature may not be available in the specific ISDN service area or may be a subscription service.

### **Programming**

<b>STATION</b>	CLIP LCD Display (PGM 114-Button 1) COLP LCD Display (PGM 114-Button 2)
<b>CO/IP</b>	COLP Table Index (PGM 143-Button 1) Type of Calling Number (PGM 143-Button 4)
<b>ISDN/ICLID</b>	CLIP Print to Serial Port (PGM 200-Button 2)

### **Related Features**

### **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**

1. 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).
2. This feature may not be available in the specific ISDN service area or may be a subscription service.

### **Programming**

<b>SYSTEM</b>	SMDR attributes (PGM 177)
<b>CO/IP</b>	Advice of Charge Type (PGM 143-Button 9)

### **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

1. This feature can be activated or deactivated only after an ISDN Line is seized.
2. 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.
3. 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.
4. This feature may not be available in the specific ISDN or may be a subscription service.
5. The system can handle only a single Call Reference; services that require handling of two simultaneous Call References, cannot be supported.

#### Programming

**STATION** Flex Button Assignment (PGM 115)

#### 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 served by a 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]

##### Conditions

1. 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.
2. 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.
3. 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.
4. An MSN loop button is overwritten by an individual CO line button, if it exists. On the other hand, an MSN loop button is not overwritten by CO Group button.
5. Multiple duplicate **{MSN LOOP}** buttons can be assigned on an iPECS IP or LDP Phone.
6. If enabled in SMDR, the MSN number is output as the Account code for outgoing calls employing an MSN button.
7. 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**

<b>STATION</b>	MSN Wait (PGM 114-Button 12) Flexible Button Assignment (PGM 115)
<b>CO/IP</b>	CO/IP Attributes III (PGM 143)
<b>ISDN/ICLID</b>	ISDN Attributes (PGM 200) MSN Table Attributes (PGM 202)
<b>System</b>	Print MSN on SMDR (PGM 177-Button 21)

## **Related Features**

- Direct Inward Dial (DID)

## **Hardware**

- ISDN or SIP Line



## 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

1. The ISDN must support the Malicious Caller Id Supplementary Service as defined by the EN300-130 standard protocol.
2. 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.
3. If the system is programmed to save SMDR records, the MCID is stored in the system memory as part of the call record.

### Programming

**CO/IP**                      MCID Request (PGM 109-Button 1)

**SYSTEM**                    SMDR Save (PGM 177-Button 1)

### Related Features

- ISDN Supplementary Services
- SMDR (Station Message Detail Recording)

### Hardware

- ISDN Line
- iPECS IP or LDP Phone



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## **4.16**    *Line Monitor*

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### **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**

### **Programming**

CO/IP

Line Monitoring (PGM148-Button 12)

### **Related Features**

### **Hardware**

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## 4.17 Multiple Calling Line Ids

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### 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

1. Your station can be assigned to restrict sending an Id for the call.
2. Your station can be configured to replace your Station CLI with the Station CLI of the Attendant.
3. 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.
4. The ISDN Line can be assigned to use different Table entries for the COLP and CLIP values.
5. For Transit calls additional options are available to employ the CLI of the original forwarding party.
6. The LCD of the station can display the CLI or COL Id if CLIP and COLP Display is enabled.
7. Multiple Calling Line Ids applies to SIP trunks when the “From Id is assigned to use the Extension CLI.

### Programming

<b>STATION</b>	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)
<b>CO/IP</b>	CLI Type (PGM 143-Button 24-1)
	COLP Table Index (PGM 143-Button 1)
	CLIP Table Index (PGM 143-Button 2)

### Related Features

### Hardware

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## **4.18    *Outside Line Groups***

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### **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 the eMG80.

Unused Lines are assigned to Group 21 in the eMG80. Private Lines are assigned to Group 00.

### **Operation**

#### **Conditions**

1. Unused outside Lines must be assigned to the appropriate group so that stations do not accidentally access these lines.
2. Outside Lines in Groups 01 through 20 can be accessed individually by dialing the code "88" and the Line number.
3. An outside Line in Group 00 can be accessed only with a Flex button assigned to the Line.
4. 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**

<b>STATION</b>	CO PGM (PGM 112-Button 6)
<b>CO/IP</b>	CO/IP Group (PGM 141-Button 1)
<b>SYSTEM</b>	CO Line Choice (PGM 160-Button 4)

#### **Related Features**

#### **Hardware**

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## **4.19 Outside Line Preset Forward**

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### **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**

1. Outside Line Preset Forward is disabled for calls initially routed to a station group.
2. Outside Line Preset Forward will override Call Forward No-Answer at a station.
3. Outside Line Preset Forward is disabled if the Preset Forward Timer is set to 0.
4. The outside Line Preset Forward destination cannot be the integrated Voice Mail group.

### **Programming**

<b>CO/IP</b>	Preset Forward timer (PGM 147-Button 1) Ring Table Index (PGM 147-Button 2) VMID Number (PGM 147-Button 3)
<b>ISDN &amp; ICLID</b>	ICLID Ring Assignment Table (PGM 204)

### **Related Features**

- Call Forward
- ICLID Call Routing

### **Hardware**

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## 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

1. Separate assignments are made for stations to ring in the Day, Night and/or Timed Ring mode.
2. A busy station receives muted ring or Call Waiting tones as appropriate for the station's Off-hook ring assignment.
3. 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.
4. The LCD of the Attendant station will display Night and Timed Ring Mode and the **[DND]** button LED will flash.
5. 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

<b>CO/IP</b>	CO Ring Assignment (PGM 144)
<b>TABLES</b>	Weekly Time Table (PGM 233)

### Related Features

- Day/Night/Timed/Scenario Ring Mode
- Auto Service Mode Control
- Off-Hook Signaling

### Hardware

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## 4.21 Private Line

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### Description

Your iPECS IP or LDP Phone can be assigned exclusive use of one or more outside Lines. 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

1. Private lines will not recall to the Attendant station.
2. Private lines are not subject to Call Pick-Up except by an assigned Secretary with an appearance of the Private line.
3. 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.
4. 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.
5. IP channels cannot be assigned as Private lines.

### Programming

<b>STATION</b>	Flex Button Assignment (PGM 115)
<b>CO/IP</b>	CO/IP Line Group (PGM 141-Button 1)

### Related Features

- Hold
- Call Forward
- Executive/Secretary Forward
- Call Forward, Preset

### Hardware

- iPECS IP and LDP Phones

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## 5. IPECS PHONE

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### 5.1 Answering Machine Emulation

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#### 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 your iPECS 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.

Your iPECS 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]

**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**

1. AME is supported only on iPECS IP and LDP Phones and an **{AME}** Flex button must be assigned on the phone.
2. 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**

**NUMBERING**                      AME Feature (PGM 109-Button 2)

### **Related Features**

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

### **Hardware**

- iPECS IP or LDP Phone



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## 5.2 Auto Called Number Redial (ACNR)

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### 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 while idle

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

##### To cancel ACNR during an ACNR attempt

1. Lift the handset or press the **[MUTE]** or flashing **[REDIAL]** button.

#### *SLT Phone*

##### 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

1. Four timers and a retry counter can be programmed.

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.

2. 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.
3. The ACNR Retry Counter decrements by one when the system completes the dialed number.
4. When the ACNR Pause Timer expires, if the station is in a busy state, the ACNR Delay Timer is invoked.
5. Upon completion of dialing, the system will monitor the call for progress signals.
6. To preserve ACNR feature in activated state, SLT user does not go On-Hook.
7. While ACNR dialing or waiting state, if SLT user goes On-Hook, then SLT ACNR is deactivated, automatically.
8. While ACNR is active for an SLT, the SLT user cannot receive or place a call.

## **Programming**

<b>SYSTEM</b>	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)
<b>NUMBERING</b>	SLT ACNR Code (PGM 109-Button 19)

## **Related Features**

- LNR (Last Number Redial)
- Speakerphone
- Mute

## **Hardware**

- iPECS IP or LDP Phone

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## 5.3 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 supported features, see conditions below.*

### Conditions

1. 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.
2. 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.

### Programming

### Related Features

### Hardware

- iPECS IP or LDP Phone

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## 5.4 Automatic Speaker Select

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### Description

The iPECS IP and LDP Phones programmed for Auto Speaker Select can access an outside Line or an internal call by pressing the appropriate button without the need to lift the handset or press the **[SPEAKER]** button. Audio from the 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

1. For iPECS IP and LDP Phones not equipped/assigned with speakerphone, the user must lift the handset to be heard.
2. 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

<b>STATION</b>	Auto Speaker Select (PGM 111-Button 1)
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### Related Features

### Hardware

- iPECS IP or LDP Phone

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## 5.5 Background Music (BGM)

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### 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 (00-10) BGM source.
  - 00 – no BGM
  - 01 – Music 1 (Internal)
  - 02 – Music 2 (External)
  - 03 – System Announcement MOH 1
  - 04 – SLT MOH 1
  - 05 – SLT MOH 2
  - 06 – SLT MOH 3
  - 07 – SLT MOH 4
  - 08 – SLT MOH 5
  - 09 – System Announcement MOH 2
  - 10 – System Announcement MOH 3
4. 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 and 2, respectively.
3. Dial the two (2)-digit (00-10) BGM source.
  - 00 – no BGM
  - 01 – Music 1 (Internal)
  - 02 – Music 2 (External)
  - 03 – System Announcement MOH 1
  - 04 – SLT MOH 1

- 05 – SLT MOH 2
  - 06 – SLT MOH 3
  - 07 – SLT MOH 4
  - 08 – SLT MOH 5
  - 09 – System Announcement MOH 2
  - 10 – System Announcement MOH 3
4. Press **[HOLD/SAVE]** to save your selection.

### Conditions

1. The speaker volume is adjustable at the station by using the **[VOL UP]/[VOL DOWN]** buttons.
2. BGM is delayed 1 second after a return to idle state.
3. Silence is provided if no BGM source is assigned.
4. For remote devices, BGM must be locally provided; BGM is not sent from the eMG80 system to remote devices.
5. System Announcement MOH 1 employs System announcement 001 and must be recorded properly. System announcement MOH2 and MOH3 may use any unused System announcement except for announcement 071 and 072. The announcement number must be defined in the Music Sources program (PGM 171).
6. For SLT MOH, the SLT port must be connected properly and the station number of the SLT port used for MOH must be assigned as the SLT MOH port.

### Programming

<b>SYSTEM</b>	BGM Type (PGM 171-Button 1)
	Internal MOH Type (PGM 171-Button 3)
	Assign SLT MOH Port (PGM 171-Button 4)
	VSF MOH2 (PGM 171-Button 5)
	VSF MOH3 (PGM 171-Button 6)

### Related Features

- MOH (Music-On-Hold)
- Internal/External & All Call Page

### Hardware

- An external music source properly connected to the KSU, refer to the ***iPECS eMG80 Hardware Description and Installation Manual***.

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## 5.6 Call Log Display

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### Description

Using an iPECS IP 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.

### Operation

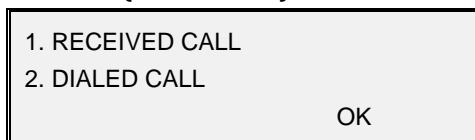
#### *iPECS IP & LDP Phones*

To assign a **{CALL LOG}** button:

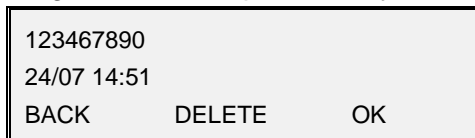
[TRANS/PGM] + {FLEX} + [TRANS/PGM] + "57" + [HOLD/SAVE]

To access the Call Log menu:

1. Press the **{CALL LOG}** button.



2. Using the volume Up/Down keys, select the desired log (received, dialed, or missed).



### Conditions

### Related Features

### Hardware

- iPECS IP and LDP Phones with LCD

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## **5.7 Call Profile Routing**

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### **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):
  - Type 0 – Station CID
  - Type 1 - All Station
  - Type 2 – CO CID
  - Type 3 - All CO call
  - Type 4 - All call
  - Type 5 - N/A
4. Enter the Scenario Start and End Date (YYMMDD).
5. Select day-of-week using Flex button:
  - Flex button 1 (MON) – Flex button 7 (SUN)
  - Flex 8 button (Holiday)
6. Enter scenario Start Time and End time (HHMM).
7. Select Type of destination (0 – 6):
  - Type 0 – Station
  - Type 1 - Hunt group
  - Type 2 - CO Line Number
  - Type 3 - CO Group number
  - Type 4 - CO Loop
  - Type 5 - CO Transit
  - Type 6 - N/A
8. Select Scenario Priority (0 to 9) (0: highest priority).
9. Select Forward From Net (0/1).
10. Select Call Profile Table Index (0-3).



11. Select Call Profile Timer (10–60 seconds).

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

1. 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.
2. The conditions associated with Station ICR apply to Call Profile Routing.
3. It is recommended hunt group is set last call profile destination.

## Programming

<b>STATION</b>	Call Forward (PGM 111-Button 2) Station Call Forward No Answer Timer (PGM 123-Button 1)
<b>SYSTEM</b>	System Call Forward No Answer Timer (PGM 181-Button 1)

## Related Features

- Call Forward
- Station ICR

## Hardware

- iPECS IP or LDP Phone

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## 5.8 DID Call Wait

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### Description

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. The DID call will queue for the DID/DISA No-answer time then will route based on the reroute assignments.

DID Call Wait is defined in the system database or can be controlled by you with a DID Call Wait Flex button.

### Operation

#### *iPECS IP & LDP Phones*

To assign a {DID/DISA CALL WAIT} button:

[TRANS/PGM] + {FLEX} + [TRANS/PGM] + "34" + [HOLD/SAVE]

To activate/deactivate DID call wait

1. Press the {DID/DISA CALL WAIT} button.
2. Dial activate/deactivate code, "1" or "0" respectively.

### Conditions

1. Other routing such as Preset Call Forward applies to DID calls even if DID Call Wait is active.
2. DID calls will follow the call routing defined in PGM 167 after the expiration of the DID/DISA no answer timer.
3. An iPECS IP or LDP Phone must have an appearance button for the DID line.
4. Assigning the ICLID Timer, which enables ICLID routing, for a DID line, disables DID Call Wait.

### Programming

<b>STATION</b>	DID/DISA Wait (PGM 113-Button 9)
<b>CO/IP LINE</b>	ICLID Timer (PGM 142-Button 14)
<b>SYSTEM</b>	DID/DISA Destination (PGM 167)
	DID/DISA No Answer Timer (PGM 181-Button 2)

### Related Features

- Direct Inward System Access (DISA)
- Direct Inward Dial (DID)
- ICLID Call Routing

### Hardware

- iPECS IP or LDP Phone

## **5.9 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 routed based 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**

1. Outside Line recalls will override One-Time DND.
2. Your station must be permitted to activate DND to activate One-time DND.
3. A station cannot reject a call from the Attendant using One-time DND.
4. One-time DND is not available to the Attendant.
5. One Time DND cancels existing Callback queues.

### **Programming**

<b>STATION</b>	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)
<b>SYSTEM</b>	System Call Forward No Answer Timer (PGM 181-Button 1)

### **Related Features**

- Call Waiting/Camp-On
- DND (Do Not Disturb)
- Call Forward
- Call Forward, Preset
- Station Call Coverage

### **Hardware**

- iPECS IP or LDP Phone

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## **5.10 Flex Button Direct Speed Dial Assignment**

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### **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 or, for iPECS 24D and 24DH phones, press the **[ICM]** 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**

1. 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.
2. The telephone number may include any of the special Speed Dial instructions (**[MESSAGE/CALLBACK]**, **[FLASH]**, display security, etc.).

### **Programming**

#### **Related Features**

- Station Speed Dial

#### **Hardware**

- iPECS IP and LDP Phone

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## **5.11 Flexible LED Flash Rates**

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### **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 35 different functions can be assigned any one of 15 different flash rates.

### **Operation**

#### **System**

System implements flash rates automatically based on database entries.

### **Conditions**

1. Available Flash rates and functions, which can be assigned, are given in the iPECS eMG80 Admin & Maintenance Manual.

### **Programming**

**SYSTEM** LED Flashing Rates (PGM 170-Button 1~35)

### **Related Features**

### **Hardware**

- iPECS IP or LDP Phone

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## 5.12 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

1. Press the **[SPEAKER]** button, speaker activates, the speakerphone microphone will be muted while the handset is off-hook.

### Conditions

1. While using the speakerphone, lifting the handset will turn off the speakerphone. To activate Group Listening, the **[SPEAKER]** button must be pressed while the handset is off-hook.
2. 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.
3. If full speakerphone operation is desired and permitted with Group Listen active, simply place the handset on-hook.

### Programming

<b>STATION</b>	Group Listening (PGM 113-Button 3)
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### Related Features

- Speakerphone
- Mute

### Hardware

- iPECS IP or LDP Phone

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## 5.13 Intercom Signaling Mode

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### Description

You can set the signaling you receive when your iPECS IP and LDP Phone receives 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, you hear 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, you receive a 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

1. Message Wait, Callback, Call Forward and Attendant Override will ring in the tone mode, regardless of ICM Signaling Mode you select.
2. The ICM signaling Mode Selection does not affect Page announcements.
3. The default ICM Signaling mode is Tone ring and the active mode is stored in battery-protected memory.
4. An SLT always functions in the Tone ring mode.

### Programming

#### Related Features

- Intercom Call (ICM Call)
- Paging
- Message Wait/Call Back
- Call Forward
- DND Override
- Intrusion

#### Hardware

- • iPECS IP or LDP Phone



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## 5.14 *Microphone Mute*

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### 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

1. Changing from speakerphone to handset or vice versa during a mute condition will return the microphone to an active status; your voice is transmitted to the connected party.
2. Returning to idle or placing another outside Line or intercom call will change the mute status to its normal (active microphone) condition.

### Programming

<b>STATION</b>	Headset Ring (PGM 111-Button 8)
	Speakerphone/Headset Selection (PGM 111-Button 9)

### Related Features

- Speakerphone
- Group Listening
- Headset Compatibility

### Hardware

- iPECS IP or LDP Phone

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## **5.15 Off-Hook Signaling**

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### **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. SLTs will only receive camp-on tones.

### **Operation**

#### **System**

Operation of Off-hook ring signals is automatically controlled.

### **Conditions**

1. While using the speakerphone, a Camp-On tone is provided over the speaker in place of the assigned Off-hook ring Signal.
2. Activating DND or One-Time DND places the station in DND, terminating any Off-hook signaling.
3. Off-hook ring signals terminate when the call forwards or is answered, or abandoned.
4. A station, which is receiving Off-hook ring signals, will receive normal ring upon return to idle status.

### **Programming**

<b>CO/IP</b>	CO Line Ring Assignment (PGM 144)
<b>SYSTEM</b>	Off-Hook Ring Type (PGM 161-Button 1)

### **Related Features**

- Call Waiting/Camp-On
- Outside Line Ring Assignment
- DND (Do Not Disturb)
- DND – One-Time DND

### **Hardware**

- iPECS IP or LDP Phone

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## 5.16 On-Hook Dialing

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### 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

1. 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.
2. When the speakerphone is used, the microphone is active unless the **[MUTE]** button is pressed and the **[MUTE]** button LED is On.

### Programming

<b>STATION</b>	Auto Speaker Selection (PGM 111-Button 1)
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### Related Features

- Mute
- Speakerphone
- Automatic Speaker Select
- Headset Compatibility
- Group Listening

### Hardware

- iPECS IP or LDP Phone

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## **5.17    *Outside Line Name Display***

---

### **Description**

When you receive or place an 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

1. Press the **{LINE}** button, or dial a Line/Group access code, LCD displays the Line name.

### **Conditions**

1. Each outside Line is assigned a name of up to 12 characters in the system database.

### **Programming**

<b>CO/IP</b>	CO Line Name Display (PGM 142-Button 1)
	CO Line Name Assign (PGM 142-Button 2)

### **Related Features**

### **Hardware**

- iPECS IP or LDP Phone w/Display

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## **5.18 Prime Line Immediate/Delayed**

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### **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

1. Lift the handset or press [**SPEAKER**] button and take no action, Prime Line as assigned will be accessed.

### **Conditions**

1. 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.
2. Selection of another Flex button or Feature button just prior to an off-hook event will override the Prime Line assignment.
3. An SLT can be assigned an Idle Line Selection with a Speed Bin Number.
4. 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.
5. 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**

<b>STATION</b>	Prime Line setting (PGM 113-Button 7)
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	Idle Line Selection (PGM 121)
<b>SYSTEM</b>	Prime Line Delay Timer (PGM 182-Button 6)

**Related Features**

- Speakerphone
- Intercom Call (ICM Call)
- Station Flexible Buttons

**Hardware**

- iPECS IP or LDP Phone

---

## 5.19 Ring Tone Selection and Download

---

### Description

You can select one of 14 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 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 ("1"~"15") to select the Ring Tone.
4. Press the **[HOLD/SAVE]** button.

### Conditions

1. The downloadable ring tone files are stored in system memory as \*.wav files with a maximum length of 4 seconds. These files can be replaced as desired using the Web Upload function. Refer to the eMG80 Administration and Programming Manual for details.

### Programming

#### Related Features

- Station User Programming & Codes
- Differential Ring

#### Hardware

- iPECS IP or LDP Phone

---

## 5.20 Save Number Redial (SNR)

---

### Description

The last dialed number on an outside 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 **[HOLD/SAVE]** button.

##### To save a dialed number using the LIP-8000/8000E series menu

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.
3. Dial “#”.

##### To dial a saved number using the LIP-8000/8000E series menu

1. Press the **[DIR]** soft button.
2. Press the **[SPEED]** soft button.
3. Dial “#”.

### Conditions

1. The saved number can be a maximum of 48 digits.
2. 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.
3. If you press the **[HOLD/SAVE]** button after seizing a Line without dialing, the Save Number Redial buffer will be erased.
4. If there is no **{LINE}** button, the call will be presented on a **{POOL}**, or **{LOOP}** button.
5. Save Number Redial is protected from power failure.
6. 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.

### Programming

#### Related Features

- Station Speed Dial
- System Speed Dial
- LNR (Last Number Redial)
- CO Line Flash

#### Hardware

- iPECS IP or LDP Phone



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## **5.21 Serial DSS/BLF Console**

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### **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. There are three models of the Serial DSS/BLF Console:

- 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
- LDP-7048DSS-48 button Console with paper button designation label
- LDP-9048DSS-48 button Console with paper button designation label

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 eMG80 Hardware Description and Installation Manual.

### **Operation**

#### ***iPECS IP Phone***

*Flex buttons on the Serial Console operate the same as the buttons on the phone*

### **Conditions**

1. The maximum number of Serial Consoles supported in an eMG80 system is 100 and no more than 60 can be the LSS model.
2. The iPECS IP Phone model LIP-8004 does not support a Serial DSS/BLF Console.
3. The LIP-8048DSS must always be powered by external adaptor available from Ericsson-LG.
4. An iPECS IP Phone can only provide power to two Serial Consoles, the remaining Consoles must be powered by the external adaptor
5. Linked pair secondary station cannot have a Serial Console.
6. 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, the 1st

Console starts from button 49, the 2<sup>nd</sup> starts from button 97, the 3rd Console starts from button 145, 4th starts from button 193.

7. 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.
8. If DSS configuration is changed and the phone restarts, programmed DSS data is initialized from the changed DSS.
9. 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.
10. Your station must be enabled to employ a Serial Console, if not enabled, any connected console will not function.

## **Programming**

<b>STATION</b>	Serial DSS Usage (PGM 111 –Button 21)
	Flex Button Assign (PGM 115)
	Serial LSS Label Edit (PGM 129)

## **Related Features**

### **Hardware**

- iPECS IP & LDP Phone with Serial DSS/BLF Console

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## 5.22 *Silent Text Message*

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### Description

Silent Text Messaging is used to respond to a Voice 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

1. Silent Text Messaging sends the DND display message, "STA IN DO NOT DISTURB" displays in the LCD of the calling Station.
2. When Silent Text Messaging is used to respond to an OHVO call, the existing call will not be disconnected.
3. When a SLT 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.

### Programming

**STATION**                      Voice Over (PGM 113-Button 6)

### Related Features

- Call Waiting/Camp-On
- Group Listening
- Voice Over
- Pre-defined & Custom Text Display Messages

### Hardware

- iPECS IP or & LDP Phone

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## 5.23 Speakerphone

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### 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

1. If Automatic Speaker Select is enabled for the station, pressing a DSS, outside Line or Speed Dial button will automatically activate the speakerphone.
2. The **[MUTE]** button LED indicates the status of the Microphone, when lit the Microphone is inactive.
3. When Group Listen is enabled, pressing the **[SPEAKER]** button while using the handset will send audio to both the Handset and Speaker. However, only the Handset microphone is active. In order to activate the Speakerphone Microphone, the Handset must be placed on-hook.
4. Each iPECS IP and LDP Phone equipped with Speakerphone is allowed/denied Speakerphone operation in the system database.
5. When Headset operation is assigned for the station, the Speakerphone is disabled and the **[SPEAKER]** button activates the Headset audio path instead of the speaker.

### Programming

<b>STATION</b>	Headset Ring (PGM 111-Button 8) Speakerphone/Headset (PGM 111-Button 9)
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### Related Features

- Mute
- Group Listening
- Automatic Speaker Select
- Headset Compatibility

### Hardware

- iPECS IP or LDP Phone

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## **5.24 Station Flexible Buttons**

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### **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. 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.

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.
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**

1. The **{LOOP}** and **{POOL}** buttons provide a status indication for the call as long as the station has supervisory control.
2. 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.
3. A station may have multiple **{LOOP}** and **{POOL}** buttons.
4. 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.
5. **{POOL}** buttons will access a Line from the designated group using the Round Robin or Last Used method as assigned in the database.
6. 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.
7. Station User Program Codes are defined in section 5.25.

### **Programming**

<b>NUMBERING</b>	Flexible Numbering Plan (PGM 105~109)
<b>STATION</b>	Flexible Button Assignment (PGM 115)

### **Related Features**

- Flexible Numbering Plan
- Station User Programming & Codes

### **Hardware**

- iPECS IP or LDP Phone

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## 5.25 Station Individual Call Routing (ICR)

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### 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 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):
  - Type 0 – Station CID
  - Type 1 - All Station
  - Type 2 – CO CID
  - Type 3 - All CO call
  - Type 4 - All call
  - Type 5 - N/A
4. Enter the Scenario Start and End Date (YYMMDD).
5. Select the Day-of-week using the Flex buttons:
  - Flex button 1 is Monday – Flex button 7 is Sunday
  - Flex 8 button is for holidays
6. Enter scenario Start and End time in 24-hour format (HHMM).
7. Select Type of destination (0 – 6):
  - Type 0 – Station
  - Type 1 - Hunt group
  - Type 2 - CO Line Number
  - Type 3 - CO Group number
  - Type 4 - CO Loop
  - Type 5 - CO Transit
  - Type 6 - N/A
8. Enter the specific Station, Hunt group, Line number, etc.

9. Select Scenario Priority (0 to 9).
10. 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**

1. Any station allowed access may define scenarios in the station's Web portal.
2. The conditions as listed under Call Forward apply.

**Programming**

<b>STATION</b>	Call Forward (PGM 111-Button 2) Station Call Forward No Answer Timer (PGM 123-Button 1)
<b>SYSTEM</b>	System Call Forward No Answer Timer (PGM 181-Button 1)

**Related Features**

- Call Forward

**Hardware**

- iPECS IP or LDP Phone



## **5.26 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.

<b>Code</b>	<b>Description</b>	<b>Entries</b>
10	Enblock Dial	
11x	Intercom Differential Ring	1~8
12x	Outside Line Differential Ring	1~8
13	Station Answer Mode	1: H, 2: T, 3: P
14x	Call Coverage Attribute	1~2
15x	Station Ring Download	5~ 8 + 0~9
19	Ear Mic. Headset	
21	Knock down Station COS (Class of Service)	
22	Restore Station COS (Class of Service)	Auth code
23	Walking COS (Class of Service)	Auth code
24	ICR Scenario	Scenario Data
25	LIP Phone Statistics	Real-time Packet Monitoring
26	Call profile Table Index	0-3
31	Station Message Wait Retrieve	
32	CLI/IP Message Wait Retrieve	
33	Register Authorization Code	Auth Code
34	DID/DISA Call Wait	
35	Choice Exec/Sec Message	1 On, 0 Off
36	SMS sending mode	
37	Register Mobile Extension	Tel. number
38	Activate Mobile Extension	1 On, 0 Off
39	Register Mobile Ext CLI	Tel. number
30	VM Mobile notify	1 On, 0 Off
41	Set Wake-Up Time	Once/Permanent & Hour/Min
42	Erase Wake-Up Time	
51	Custom/Pre-selected Message	00~20
52	Register Custom Message (00)	"Message"
53	Create Conference Room	Room number (1~9)

<b>Code</b>	<b>Description</b>	<b>Entries</b>
54	Delete Conference Room	Room number (1~0)
55	Monitor conference group	Hunt group number
57	<b>{Call Log Display}</b> button	
61	Headset/Speakerphone mode	0:H, 1:S
62	Select Headset Ring type	1:S, 2:H, 3:Both
67	<b>{Call Coverage}</b> button	
71	LCD Display Mode	Domestic/English
72	Sys version display	
73	Select BGM source	(00~10)
74	User Name registration	"name"
75	Display Phone IP Address	
76	Change IP Address (Dot:*)	
77	Display MAC IP Address	
78	Change Mode	
79	Display Phone Version	
7*	Display Key numbers	
80	<b>{RECORD}</b> button	
81	ISDN <b>{CLIR}</b> button	
82	ISDN <b>{COLR}</b> button	
83	<b>{ACD DND}</b> button	
84	<b>{ACCOUNT CODE}</b> button	
85	<b>{LOOP}</b> button	
86	<b>{INTRUSION}</b> button	
87	<b>{ICM}</b> button	
88	<b>{CAMP-ON}</b> button	
89	ISDN <b>{KEYPAD FACILITY}</b> button	
8#	<b>{OHVO}</b> button	
99	Push-To-Talk	
*6	Set forced forward destination (Hunt)	
*7	Enable/Disable forced forward (Hunt)	
*8	Register Bluetooth device	
*9	Bluetooth Usage	
*0	Agent Login	
**	Agent Logout	
91	<b>{Conference}</b> button	

Code	Description	Entries
92	{ <b>Callback</b> } button	
93	{ <b>DND</b> } button	
94	{ <b>Flash</b> } button	
95	{ <b>Mute</b> } button	
96	{ <b>Monitor</b> } button	
97	{ <b>Redial</b> } button	
98	{ <b>Call Forward</b> } button	

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 displays are 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 KEYSER STAT  
[6] CALL PROFILE

- Next top-level Menu selection

[3] MSG RET/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] MONITOR CONF-GROUP  
[1] SET PRE:CUST MSG

- 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] MPB 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.

## Programming

## Conditions

### **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

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## 5.27 Two-Way Record

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### Description

With your iPECS IP or LDP Phone you can record any active conversation to your Voice Mail mailbox or to the hard disk drive of an iPECS Phontage or 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.

### Operation

#### *iPECS IP & LDP Phones*

To assign a flexible button as a **{RECORD}** button

[TRANS/PGM] + **{FLEX}** + [TRANS/PGM] + 80 + recording destination (optional) +  
[HOLD/SAVE]

To activate Two-Way Record while on a call

1. Press the **{RECORD}** button, record warning tone heard, and recording starts.

To stop Two-Way Record while on a call

1. Press the **{RECORD}** again.  
Or,
2. Hang-up, return to idle.

To manage the recordings, use the procedures for the voice Mail or as outlined in the Phontage or UCS Client User Guide.

### Conditions

1. The **{RECORD}** button LED will flash at 240 ipm while recording.
2. 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.
3. With the Feature Server or integrated VM, internal calls can be recorded as well as external calls.
4. 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.
5. If a destination for the recording is not defined for the **{RECORD}** button, the Call Record Destination defined in PGM 112 is employed.

### Programming

<b>STATION</b>	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)

**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, Feature Server or External SMDI based AA/VM system
- Phontage or UCS Client Softphones



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## **5.28 Voice Over**

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### **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 the Voice 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***

##### *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**

1. When you respond to a Voice Over via Camp-On, all conditions and options available to Camp-On apply.
2. 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.
3. When a call is transferred with Voice Over, you will receive muted ringing after the transfer is complete.
4. 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.
5. Both the originating and receiving station must be programmed to allow OHVO calls.
6. When Silent Text Messaging is used to respond to a Voice over call, the existing call will not be disconnected.
7. 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**

**STATION**                      Voice Over (PGM 113-Button 6)

## **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

---

## 6. ATTENDANTS

---

### 6.1 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

1. Call routing order follows the order of entry in the Attendant Assignments program.
2. 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.
3. Calls to the Attendant's station intercom number are sent to the station dialed as with any intercom call.
4. When an Attendant calls another busy Attendant by dialing the station number, busy tone is received and camp-on is available.

#### Programming

<b>SYSTEM</b>	Attendant Assignment (PGM 164)
	MOH Type (PGM 171-Button 2)
	Attendant Call Queuing Ring back Tone (PGM 160-Button 1)

#### Related Features

- Attendant Positions
- Intercom Call (ICM Call)

#### Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

---

## **6.2 Attendant Positions**

---

### **Description**

The iPECS eMG80 system has the capacity for 4 Attendants (1 System Attendant / 3 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.

### **Operation**

#### **Condition**

1. As a default, the first Attendant (System Attendant) is assigned as Station 100 (logical number), and others are not assigned. In the eMG80, the first digital phone port is designated as the System Attendant and is Station number 100.
2. Attendant calls (dialing "0") and recalls are routed to first available attendant.

### **Programming**

<b>SYSTEM</b>	System Attendant Assignment (PGM 164-Button 1)
	Main Attendants Assignment (PGM 164-Button 2~4)

### **Related Features**

#### **Hardware**

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

---

## **6.3 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 outside Line to idle.

### **Operation**

#### **System**

Attendant recall operation is automatic.

### **Conditions**

### **Programming**

<b>SYSTEM</b>	Attendant Recall Timer (PGM 180-Button 1)
	ICM Hold Recall Timer (PGM 180-Button 5)

### **Related Features**

- Hold
- Call Transfer

### **Hardware**

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

---

## 6.4 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, reference Appendix C of the iPECS Admin & 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

[1] 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

[2] 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] LCD 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] Set Time from ISDN Message

enter 1= on, 0= off

[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 VSF slot number (001-999)
- [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.

## **Condition**

## **Programming**

<b>SYSTEM</b>	System Attendant (PGM 164-Button 1)
	Main Attendants Assignment (PGM 164-Button 2~4)

## **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

---

## 6.5 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

1. The Conditions of the Call Forward feature apply.
2. 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.
3. 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

<b>STATION</b>	Call Forward (PGM 111-Button 2) Station Call Forward No Answer Timer (PGM 123-Button 1)
<b>SYSTEM</b>	System Attendant (PGM 164-Button 1) Main Attendants Assignment (PGM 164-Button 2~4/5)

### Related Features

- Call Forward
- Alternate Attendant

### Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

---

## 6.6 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

1. 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.
2. The conditions of the Unsupervised conference feature apply.
3. The conditions of the Call Forward Off-Net feature apply.
4. When enabled, calls that forward Off-net will receive the Off net forward prompt.
5. When enabled, calls that forward Off-net will receive the DTMF repeat tone.
6. 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.

## **Programming**

<b>STATION</b>	Call Forward (PGM 111-Button 2)
	Off-Net Forward (PGM 111-Button 14)
<b>SYSTEM</b>	Off-Net Prompt Usage (PGM 160-Button 11)
	DTMF Repeat Tone Usage (PGM 160-Button 15)
	System Attendant Assignment (PGM 164-Button 1)
	Unsupervised Conference Timer (PGM 182-Button 5)
	Open Loop Detect Timer (PGM 142-Button 13)

## **Related Features**

- Call Forward
- Unsupervised Conference

## **Hardware**

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

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## **6.7 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 "73" to set Auto Mode control.

### **Conditions**

1. 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).
2. Stations receive incoming ring for outside Lines based on the database assignments and the system mode (Day/Night/Timed/Scenario) when the call arrives.
3. 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.
4. The System Attendant always has manual control of the system mode by enabling/disabling the Auto Service Mode Control.
5. To employ the Scenario mode, scenarios must be assigned in the System Call Routing program.

### **Programming**

<b>CO/IP</b>	CO Line Ring Assignment (PGM 144)
<b>SYSTEM</b>	External Control Contact (PGM 168)
<b>TABLES</b>	Weekly Time Table (PGM 233)

System Call Routing (PGM 251) Web only

**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

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## **6.8 DSS/BLF Consoles**

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### **Description**

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 an eMG80. 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.

### **Operation**

### **Conditions**

1. Each DSS Console is assigned to operate in connection with a specific station.
2. 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.
3. Assignments for the DSS button maps are detailed in the iPECS eMG80 Admin & Programming manual. Note that the individual button assignments can be changed as with the iPECS IP and LDP Phone Flex buttons.

**Table 6.8-1 DSS/BLF button maps — iPECS eMG80**

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	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**

### **STATION**

Station Type (PGM 110-Button 1)

DSS Map (PGM 110-Button 2)

Flexible Button Assignment (PGM 115)

## **Related Features**

- Attendant Positions
- Station Flexible Buttons

## **Hardware**

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended



---

## 6.9 Disable Outgoing Line Access

---

### Description

The System Attendant can place outside Lines 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 the Line. 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

1. The System Attendant may disable a busy Line. The feature will take effect after the desired Line goes to idle.
2. 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.
3. The outside Line status is stored in battery-protected memory in case of a power failure.
4. Multiple Lines may be enabled/disabled without redialing the Attendant Station Program code. Confirmation tone is heard after each Line is enabled/disabled.
5. When the system detects a fault on a Line, the Line is disabled for outgoing access automatically.
6. Incoming calls on a disabled Line will continue to operate normally.

### Programming

**SYSTEM**                      Attendant Assignment (PGM 164-Button 1)

### Related Features

- Attendant Positions

### Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

---

## 6.10 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

1. An Attendant may use Override to transfer an outside call to a station in DND.

### Programming

**SYSTEM**                      Attendant Assignment (PGM 164)

### Related Features

- Intrusion
- DND (Do Not Disturb)
- Executive/Secretary Forward

### Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

---

## 6.11 *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 STOP} button to terminate the alert:

[TRANS/PGM] + {FLEX} + "565" + [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 STOP} 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..

## **Conditions**

## **Programming**

<b>STATION</b>	Emergency CO (PGM 112, BUTTON 18)
<b>TABLES</b>	Emergency Service Call Table (PGM 226)
<b>SYSTEM</b>	EMR CALL ATD NOTIFY (PGM 161, BUTTON 24-5)

## **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

---

## **6.12    *ez-Attendant***

---

### **Description**

The ez-Attendant is a Windows based PC application that provides a visualization of the Attendant functionality to simplify Attendant control of features and functions including displays of call, user and system status. ez-Attendant operates in conjunction with the Attendant's iPECS IP or LDP Phone to simplify operation and expand features and functions available to the Attendant.

For further information on ez-Attendant, refer to the ez-Attendant Installation and User Guide.

### **Operation**

#### ***System Attendant***

*Operation of the ez-Attendant is provided in the ez-Attendant Installation and User Guide.*

### **Conditions**

1. iPECS ez-Attendant requires installation of a system Lock-key.

### **Programming**

<b>SYSTEM</b>	Attendant Assignment (PGM 164)
---------------	--------------------------------

### **Related Features**

### **Hardware**

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

---

## 6.13 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.

### Conditions

### Programming

**SYSTEM** Attendant Assignment (PGM 164-Button 1)

### 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

---

## **6.14 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**

1. IP-Attendant requires installation of a system Lock-key.

### **Programming**

<b>SYSTEM</b>	Attendant Assignment (PGM 164)
	Station User Login (PGM 443) Web only

### **Related Features**

### **Hardware**

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

---

## 6.15 *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.

### Conditions

1. 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.
2. The conditions of the Conference feature apply.

### Programming

<b>STATION</b>	Override Privilege (PGM 113-Button 4)
<b>SYSTEM</b>	Automatic Privacy (PGM 161-Button 3) Privacy Warning Tone (PGM 161-Button 4)

### Related Features

#### Hardware

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended



---

## **6.16 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.

### **Conditions**

### **Programming**

<b>SYSTEM</b>	Attendant Assignment (PGM 164-Button 1)
	LCD Date Display Mode (PGM 169-Button 1)
	LCD Time Display Mode (PGM 169-Button 2)

### **Related Features**

- Attendant Positions

### **Hardware**

- iPECS IP or LDP Multi-button Phone, a 24-button phone is recommended

---

## 6.17 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

1. The time is entered as a 24-hour format.
2. 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

<b>SYSTEM</b>	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)

### **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

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## 7. SLT (SINGLE LINE TELEPHONE)

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### 7.1 Broker Call

---

#### Description

Broker Call allows an SLT 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

Camped On Broker Call – the second call camped on the SLT.

#### Operation

##### SLT

##### 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 feature code "600", camped on call is connected.

##### To alternate between the calls

1. Momentarily press the hook-switch.
2. Dial the Camp-On Answer feature code "600".

#### Conditions

1. After a hook-switch flash, if the call results in an error, busy, no answer or an abnormal state, the SLT user may momentarily press hook-switch to retrieve the held call.
2. During a Transfer Broker Call, if the SLT user goes on-hook, the Broker Call parties are connected completing a Call Transfer.
3. During a Transfer Broker Call, if the active caller disconnects from the SLT user, the held party, if another station, is connected to the SLT. If the held party is an outside call,

the SLT user receives error tone and may go on-hook to receive recall and retrieve the held call.

4. During a Camp-On Broker Call, if the SLT user goes on-hook, the active call is disconnected and the held call recalls to the SLT.
5. During a Camp-On Broker Call, if the active party disconnects from the SLT, the SLT user receives error tone. The SLT user may momentarily press the hook-switch to retrieve the held party or go on-hook and receive recall.
6. If the SLT user presses the hook-switch twice in less than 2 seconds, a 3-way conference is established.
7. If after a hook-switch flash, the user takes no action for the dial tone timer, the SLT will receive error tone.
8. If the SLT goes to an on-hook state with a call on hold, the SLT will receive recall ring automatically.
9. The SLT Flash mode must be set to Transfer.

## **Programming**

**STATION**                      SLT Flash Mode (PGM 113-Button 24-8)

## **Related Features**

- Message Wait/Call Back
- Call Waiting/Camp-On
- Exclusive Hold
- Call Transfer
- Conference

## **Hardware**

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## **7.2 Howler Tone**

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### **Description**

When an SLT 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**

1. Howler Tone is sent after a period, of about 30 seconds of error tone.
2. Lock-out occurs when howler tone starts.

### **Programming**

<b>STATION</b>	Howling Tone To Station (PGM 111-Button 5)
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### **Related Features**

- Intercom Lock-Out

### **Hardware**

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## **7.3 SLT Flash Mode**

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### **Description**

When the SLT 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 user goes on-hook, the held Line is disconnected.

### **Operation**

### **Conditions**

### **Programming**

SLT Flash Mode (PGM 113-Button 24-8)

### **Related Features**

### **Hardware**

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## **7.4 SLT Message Wait Indication**

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### **Description**

All SLT 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 SLTs indicating a Message Wait.

### **Conditions**

1. The system switches a 90 VDC supply On and Off to flash the SLT's neon Message lamp.
2. Although the SLT Battery Feed is removed during the 90 VDC On cycle, the system will recognize an SLT Off-hook event.
3. The SLT must incorporate a 90 VDC neon lamp that is connected directly across the tip and ring of the voice network.

### **Programming**

<b>STATION</b>	Station Type (PGM 110-Button 1)
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### **Related Features**

- Message Wait/Call Back

### **Hardware**

- SLT w/90 VDC Neon lamp



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## **7.5 SLT Name Entry**

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### **Description**

A SLT 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**

##### To register the name at the SLT

1. Lift the handset.
2. Dial "561", the SLT Programming code, confirmation tone is heard.
3. Dial "74", the SLT 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:

1. Lift the handset.
2. Dial "561", the SLT Programming code, confirmation tone is heard.
3. Dial "74", the SLT Name Program Code.
4. Momentarily depress the hook-switch, receive confirmation tone.

### **Conditions**

### **Programming**

### **Related Features**

- Dial-by-Name
- Station Speed Dial

### **Hardware**

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## **7.6 Transfer CLI to SLT**

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### **Description**

A SLT phone may 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 when configured.*

### **Conditions**

1. For proper operation, use of the Original CLI should be assigned to the Transfer CLI.

### **Programming**

<b>STATION</b>	Transfer CLI to SLT (PGM 114-Button 18)
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### **Related Features**

### **Hardware**

- SLT w/display

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## 8. SIP EXTENSION

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### 8.1 SIP Phone Support

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#### Description

The iPECS eMG80 incorporates 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 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. Some of the features supported are listed below.

In addition to the Numbering Plan features, for the Ericsson-LG 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 3<sup>rd</sup> party SIP phone is employed, the phone 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 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 SIP phones
- Emergency Page
- Emergency Calling
- Integrated Voice Mail
- Last Number Redial
- Paging
  - Internal/External Page
  - Meet Me Page
  - 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 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 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**

1. A lock-key is required to use a 3<sup>rd</sup> party SIP phone, contact your local Ericsson-LG representative.
2. SIP phones must be configured in accordance with the manufacturer's instruction and have an appearance of the station assigned in the eMG80 system.
3. When using Feature Sync, the Station must be allowed Call Forward and DND for proper operation.
4. In addition to the feature set supported for 3<sup>rd</sup> party SIP phones, the Ericsson-LG SIP phones support Voice Over, Intrusion, Forced Hands-free ICM.
5. 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.
6. 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.
7. The ELG and most 3<sup>rd</sup> 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.
8. Ericsson-LG takes no responsibility of the compatibility of any 23<sup>rd</sup> party SIP phone with the eMG80 system. It is the user's responsibility to assure compatible operation.
9. Configuring authentication will require the SIP phone provide authentication at

registration and during the initiation of a call.

## **Programming**

### **SIP DATA**

SIP Attributes (PGM 210) Web only

SIP Phone Attributes (PGM 211) Web only

SIP Phone Provisioning (PGM 212) Web only

## **Related Features**

## **Hardware**

- eMG80 supports 32 SIP Extensions with 16 simultaneous calls

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## **8.2 SIP Phone Provisioning, Ericsson-LG Phones**

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### **Description**

The Provisioning feature automatically configures the Ericsson-LG SIP phones from the eMG80 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 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 eMG80 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**

1. Provisioning is only available for Ericsson-LG SIP desk-top phone.
2. The download employs TFTP as the file transfer protocol.

### **Programming**

#### **SIP DATA**

SIP Phone Attributes (PGM 211) Web only

SIP Phone Provisioning (PGM 212) Web only

### **Related Features**

### **Hardware**

- IP-8800 & IP8800E series phone

## **Thanks for purchasing iPECS eMG80**

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