iPECS - MG

Functional Specification
For
Version 2.0

Revision History

ISSUE	DATE	DESCRIPTION OF CHANGES
0.1A	2011 June	Initial Draft

Table of Contents

1. ENHANCING TERMINAL PORTFOLIO

1. LIP-8000G SERIES

Description

iPECS-MG supports LIP-8000G series through IPKTS protocol interface and provides the related features as follows:

- LLDP-MED
- OpenVPN
- 802.1X/EAP-MD5

Operation

Conditions

Programming

PGM 101 Slot Assignment
PGM 103 Logical Slot Assignment
PGM 104 DECT/IP/SIP Max Port
PGM 106 IP-Phone Registration

Related Features

Hardware

LIP-8002, LIP-8002A, LIP-8008G, LIP-8012G, LIP-8024G, LIP-8040G

2. WIT-400H

Description

iPECS-MG is capable of connecting to WIT-400H as the standard SIP. The WIT-400H has some specific functions or restriction so that iPECS-MG provides the related features as follows:

- Preselected-Message Indication when WIT calls to a station in preselected message mode.
- SIP Call recording
- WIT Push-To-Talk
- Presents One Digit Busy Service
- Handling Message button
- ADD-HOC Conference with SIP which has not mix capability

Operation

Conditions

WIT-400H no need lock key for SIP phone.

Programming

PGM 380 SIP STA Basic Registration Table
PGM 381 SIP STA Additional Registration Attributes

Related Features

Hardware

WIT-400H

3. Mobile SIP Client

Description

The System supports the iPECS Communicator(Mobile SIP Client). Compatible SIP phones support the Internet Engineering Technical Committee standard RFC3261 for real-time communications over the Internet. Once registered, the System will deliver services to the SIP Phone. Operation of the SIP Phone generally follows the steps outlined for an SLT.

Operation

Conditions

Programming

PGM 380 SIP STA Basic Registration Table
PGM 381 SIP STA Additional Registration Attributes

Related Features

Hardware

4. Wireless Keyset

Description

Wireless Keyset (WK) based on DECT Phone is supported in MG System.
WK is developed with DECT Phone, so phone image seems like LIP-8K Series, but actual operation is similar with DECT phone.
WTIB board is needed for using WK phone in MG system. Attendant can register WK phone, and also through WEB admin menu WK can be registered.

Trans/PGM, DND, MSG, HOLD, MUTE, SPEAKER, and Volume Up/Down: fixed buttons is served in default. And also 12 Flex Button is served in WK phone.

Operation

WK Registration from Attendant:

1. Dial [TRANS/PGM] and choose attendant sub-menu.
2. Dial # for WTU Subscribe menu
3. Press Flex 1 and enter DECT phone number
4. Enter Phone Type 4 for WK
5. Hold.

Device Limitation

1. Max 8 WK can be installed in MG System.
2. Only LED1~3 can support dual color, so LED 4~12 supports only red color.
3. LED Flashing is fixed always 640msec on and 640msec off.
4. WK has 3 lines LCD for message, but only 1 line is used for System message.

Conditions

1. When WK registers in MG system, Wk does not need to enter AC code. WK can be installed automatically.
2. WK conditions...

Programming

DECT Registration 1. DECT (Un)Subscribe (DECT Data Page in WEB Admin)

Related Features

Hardware

2. BUILDING UP APPLICATION PORTFOLIO

2.1 AIM

Description

AIM is an abbreviation of Application Interface Message of the LG-Ericsson System. LG-Ericsson supports the AIM as DLL type. The 3rd party software vendors can use this AIM to implement call center or office CTI solution.

There are two kinds of AIM modes. One is the 3rd party mode and the other is the 1st party mode.

In the 3rd party mode, the application can connect to system in only one PC (namely, Server). This type of AIM does not support any functions between Server and Client. Windows 2000 Pro or higher OS is allowed. (Windows 2000 Pro/ Server, Windows XP Home/Pro, Windows 2003 Server, Windows Vista)

In the 1st party mode, the application can connect to system in many PCs. Windows 2000 Pro or higher OS is allowed too.

Operation

Operation of this feature is automatic.

Conditions

1. Either 3rd party mode or 1st part mode should be selected. Both modes can't work at the same time. If CTI Server IP Address (PGM133-Index12) is assigned then only 3rd party mode is available. When CTI Server IP address (PGM124-Index10) is not assigned and when 1st part CTI IP Address is assigned, 1st party mode is available.
2. 3rd party mode needs CTI (TAPI 3rd) lock-key but 1st part mode doesn't.

Programming

System Data	1. CTI Server IP Address (PGM 133-Index12)
Station Data	1. CTI IP Addres (PGM 124–Index10)

Related Features

Hardware

2.2 SMB CC

Description

SMB CC is one of TAPI applications. The 3rd party software vendors can use this SMB CC to implement call center or office CTI solution.

Operation

Operation of this feature is automatic.

Conditions

Programming

System Data 1. CTI Server IP Address (PGM 133-Index12)

Related Features

Hardware

2.3 IP ATTENDANT

Description

IP Attendant is a windows-based PC application that provides a visualization of the Attendant functionality to simplify Attendant control of Features and Functions including Call Display, and User and System status. IP Attendant works independently from digital phone (i.e. hardware terminal) while ez-Attendant operates in conjunction with the Attendant Digital Phone. IP Attendant has its own station number and is registered to system like phontage or UCS client.

Operation

Attendant

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

Conditions

Programming

Tenant Data 1. Attendant Group Assign (PGM 270)

Related Features

Hardware

2.4 IPCR

Description

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

Operation

System

IPCR Equipment Registration:

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

IPCR Server

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

Set IPCR before registration to system.

PBX registration(system IP, SIP ID, SIP Password)
IPCR Server registration
User registration
Channel registration (Agent ID Registration)

Agent ID Registration

Match the Agent ID number to the station number (with Tenant Prefix Number) which has to be programmed to record voice of its call.
You can use ACR (Auto-call recording) by configuring IPCR Auto Record(PGM147-6).

Conditions

1. This feature requires a license key based on the number of agent to be recorded.
2. A VOIB channel will be needed for a TDM terminal such as DKT or SLT to use this feature.
3. You can search the recorded voices using Web Admin of IPCR.
4. If recording type on a station is ODR (On Demand Recording, i.e. IPCR Auto Record is OFF), the station user has to press the [Two Way Record] button to start recording.
5. If a user press [HOLD]/[TRANS] button while recording, the recording will be stopped.

Programming

System Attribute PGM 223-21- IPCR Server IP address
System Attribute PGM 223-22- SIP Extension Number For IPCR

Related Features

Hardware

IPCR Server

2.5 WEB PHONE

Description

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

Operation

Web

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

Conditions

1. A lock key is required to use the Web Phone.
2. Currently, the Web phone is designed for the voice call.
3. Only MS Internet explorer is available.
4. If the TLS option is enable in system, user should install the certification file before web access.
5. The web phone does not support “zoom level” in the IE 7.0. (Use 100%)

Programming

Station

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

Related Features

Hardware

3. ENHANCING SYSTEM CAPABILITY

4. FEATURE SYNC. WITH IPLDK

4.1 EMERGENCY SUPERVISOR

Description

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

- Voice Over
- Override
- Override & Disconnect

Operation

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

Conditions

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

Programming

Station Port Data	1. Emergency supervisor (PGM 123 – Index9)
Station Number Data	1. Voice Over Access (PGM 133-Index9)
	2. Rejection of Voice Over(PGM 133-Index10)

3. Auto Privacy(PGM 134-Index11)

System Data 1. Intercom Busy One-digit Attributes (PGM 237)
Numbering Plan 1. Feature Numbering, Override & Disconnect (PGM 113)

Related Features

Hardware

4.2 Override and Disconnect

Description

When a user calls to a station and receives a busy signal, the user can request Override & Disconnect.

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

Operation

With One Digit Service

To activate a Override & Disconnect while receiving Intercom busy tone:

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

With Feature Code

To activate a Override & Disconnect while receiving Intercom busy tone:

1. Dial the feature code for Override & Disconnect, the called station will receive the Call Wait Alarm tone
2. The existing conversation is terminated and a conversation with overriding station will be established automatically.

Conditions

Programming

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

Related Features

Ошибка! Источник ссылки не найден.

Hardware

4.3 CALL DURATION RESTRICTION

The System can be programmed to limit the length of calls at specified stations.
Administrator can make restriction rule in each ICM Call / Incoming Call / Normal Outgoing Call / Outgoing Call in Prefix Table (Local, Long, International Call) / Dedicated CO Line / Mobile Call.
In each case, only single alarm tone can be set as restriction rule after restrict timer. And repeated alarm tone can be set as restriction rule. And also after restriction time, forced release rule can be set, automatically.

If only single alarm tone is assigned as restriction rule, specific station can hear single alarm tone after restriction time.
If repeated alarm tone is assigned as restriction rule, specific station can hear alarm tone periodically in programmed cycle after restriction time.
If forced disconnection rule is assigned as restriction, specific station can hear warning tone and then after timer, call will be released forcibly.

Max 30 rules can be assigned in iPECS-MG100/300 system. Each station has to refer to one of these rules. And each station will follow one of assigned restriction rule.

4.3.1 ICM Call Duration Restriction

Description

ICM Call Restriction rule can be defined from Call Restriction admin PGM284, 285.
If call restriction is set, after restriction time, assigned restriction rule will be operated automatically. Single or repeated alarm tone can be served, or call can be released forcibly.

If only single or repeated alarm tone is set, after restriction time, alarm tone will be served automatically.
If disconnection rule is set, before defined warning tone time, released warning tone will be served to user.

Operation

System

Operation of this feature is automatic when assigned:

Conditions

6. ICM Call Duration Time Display to station's LCD option is added for ICM call conversation in PGM123.
7. Call Duration Restriction Admin is moved from DN base admin PGM134 to Station Base admin PGM121.

Programming

Station
1. Call Duration Restriction Table (PGM 121-Index14)
2. ICM Call Duration Time Display (PGM 123-Index13)

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

Related Features

Hardware

4.3.2 Incoming Call Duration Restriction

Description

Incoming Call Restriction rule can be defined from Call Restriction admin PGM284, 285.
If call restriction is set, after restriction time, assigned restriction rule will be operated automatically. Single or repeated alarm tone can be served, or call can be released forcibly.

If only single or repeated alarm tone is set, after restriction time, alarm tone will be served automatically.
If disconnection rule is set, before defined warning tone time, released warning tone will be served to user.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. Call Duration Restriction…

Programming

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

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

Related Features

Hardware

4.3.3 Normal Outgoing Call Duration Restriction

Description

Normal Outgoing Call Restriction rule can be defined from Call Restriction admin PGM284, 285.
Normal Outgoing Call means unmatched all of outgoing call from Local / Long / International Prefix table.
If call restriction is set, after restriction time, assigned restriction rule will be operated automatically. Single or repeated alarm tone can be served, or call can be released forcibly.

If only single or repeated alarm tone is set, after restriction time, alarm tone will be served automatically.
If disconnection rule is set, before defined warning tone time, released warning tone will be served to user.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. Call Duration Restriction…

Programming

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

Related Features

Hardware

4.3.4 Local / Long / International Outgoing Call Duration Restriction

Description

Outgoing Call can be divided Local / Log / International Call with comparison between dialed digit from user and Local / Long / International Prefix table.
Each Local / Long / International Call can be set restriction rule from Call Restriction admin PGM284, 285. If call restriction is set, after restriction time, assigned restriction rule will be operated automatically. Single or repeated alarm tone can be served, or call can be released forcibly.

If only single or repeated alarm tone is set, after restriction time, alarm tone will be served automatically.
If disconnection rule is set, before defined warning tone time, released warning tone will be served to user.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. Call Duration Restriction…

Programming

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

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

Tenant Data 1. Local Call Prefix Table (PGM 286)
2. Long Call Prefix Table (PGM 287)
3. International Call Prefix Table (PGM 288)

Related Features

Hardware

4.3.5 Dedicated Line Call Duration Restrict

Description

CO line can be defined as Normal CO line or Dedicated Line in CO Access Mode admin in PGM162-Index1.
If CO line is dedicated line, all of outgoing call will be handled as dedicated line. And Outgoing Call through dedicated line can be set restriction rule from Call Restriction admin PGM284, 285. If call restriction is set, after restriction time, assigned restriction rule will be operated automatically. Single or repeated alarm tone can be served, or call can be released forcibly.

If only single or repeated alarm tone is set, after restriction time, alarm tone will be served automatically.
If disconnection rule is set, before defined warning tone time, released warning tone will be served to user.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. Call Duration Restriction…

Programming

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

CO 1. CO Access Mode (PGM 162-Index1)

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

Related Features

Hardware

4.3.6 Mobile Call Duration Restrict

Description

In case of Mobile Call, restriction rule can be defined from Call Restriction admin PGM284, 285 in Tenant base admin. Administrator can define mobile call prefix digits in PGM289 Mobile Prefix Table.

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

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

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

Operation

Operation of this feature is automatic when assigned:

Conditions

1. Call Duration Restriction…

Programming

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

Related Features

Hardware

4.4 Fail Over PSTN

Description

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

Operation

To Register Fail over PSTN:

1) Select DN number at PGM 311
2) Set destination number on Failure.

Conditions

Fail over PSTN only used for extension.
In case of CO lines fail over PSTN, can be supported by ARS services.

Programming

PGM 180 CO Group Access Code Attribues
PGM 310 Fail over PSTN System Attributes
PGM 311 Fail over PSTN DN Attributes
PGM 362 Incoming Attributes
PGM 364 H.323 Check Message Information

Related Features

Hardware

4.5 DISA Destination When No VM Channel Is Available

Description

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

Operation

Conditions

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

Programming

CO Line Data 1. Wait If VM busy (PGM 161 – Index 12)
2. Incoming CO Alternate Destination (PGM 169)

Related Features

Hardware

AAFU, AAIB, or VMIB

4.6 Virtual Subscriber Service

Description

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

Virtual Subscriber

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

Virtual Subscriber Table

This table contains incoming CLI, called number, incoming CO group, day/night/timed class, tenant, maximum virtual calls, Virtual CLI table index and destination. The incoming CLI can be assigned up to 24 digits. In addition, the incoming CLI can be assigned if the incoming CLI is identical but the tenant is different. The table can be assigned up to **300/100** tables.

Virtual CLI Table

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

Virtual Subscriber Service Option

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

Operation

System

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

4. System receives a call from CO.
5. System processes a virtual subscriber with temporary day/night/timed class and tenant if CLI of incoming call fulfills the condition of virtual subscriber, and system sends the call to the specific destination of virtual subscriber.

6. If the destination of virtual subscriber is CO access code, system sends the CO access code and called party number incoming through CO.
7. If there is no destination of virtual subscriber, system just sends called party number incoming through CO.

1) Virtual Subscriber Service Option

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

2) Virtual subscriber's CLI

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

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

- **Length Matching**: Length of CLI should be same with one of virtual subscriber and any digits are available. Ex) The case that pre-assigned digits of virtual subscriber are ‘XXXXXXX’ and received CLI is ‘4504875’. Length of incoming CLI is 7.
- **Beginning Masked Numbers**: The last part of digits of CLI should be same with one of virtual subscriber and length of CLI should be same with or more than same digits of virtual subscriber. Ex) The case that pre-assigned digits of virtual subscriber are ‘C4875’ and received CLI is ‘4504875’. ‘C’ means any one and more digits. Length of incoming CLI is more than 4. The last 4 digits are same with one of virtual subscriber.
- **End Masked Numbers**: The first part of CLI should be same with one of virtual subscriber and length of CLI should be same with or longer than virtual subscriber. Ex) The case that pre-assigned digits of virtual subscriber are ‘450C’ and received CLI is ‘4504875’. Length of incoming CLI is more than 3. The first 3 digits are same with one of virtual subscriber.
- **No Number**: There is no CLI in incoming CO call. The case that pre-assigned digit of virtual subscriber is N. The priority is same with ‘Whole Numbers’.
- Incoming CLI only has numbers from 0 to 9.
- In Admin programming, ‘N’, ‘X’ and ‘C’ can’t be used in one CLI type at the same time. And ‘N’ and ‘C’ can’t be assigned more than one in one CLI type.
- If incoming CLI fulfills one more conditions above, selection of condition follows above priority. The order of priority is as next: Whole Numbers and No Numbers > Prefix Masked Numbers > Postfix Masked Numbers > Beginning Masked Numbers > End Masked Numbers > Length Matching.

3) Virtual subscriber’s Called Party Number

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

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

virtual subscriber. Ex) The case that pre-assigned digits of virtual subscriber are ‘C4875’ and received CPN is ‘4504875’. ‘C’ means any one and more digits. Length of incoming CPN is more than 4. The last 4 digits are same with one of virtual subscriber.

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

4) Incoming CO Group

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

5) Virtual Subscriber’s Day/Night/Timed Class

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

6) Virtual Subscriber’s Tenant

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

7) Virtual Subscriber’s Maximum virtual calls

Maximum number of CO incoming calls that virtual subscriber service is applied to simultaneously. If the number isn’t assigned, there is no limit of incoming calls. Otherwise, the number can be assigned from 0 to 254

8) Digit Conversion Table

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

9) Virtual CLI

- Type : There are two types to make Virtual CLI. ALL - The assigned Virtual CLI is applied for all CO groups and extension. IND(Individual) - The system can make CLI for extensions, CO groups(max. 6) and the others.
- O/G CO Group Number : If the Type ALL is selected, the assigned Virtual CLI is used for all CO groups. In case of IND, user is able to assign the Virtual CLI for each O/G CO groups up to 6. And “ext” is used only for extension display and “else” is used for all other CO groups which are not assigned in O/G CO groups.

- Table Index : Index of virtual CLI table for the outgoing CO groups. When virtual subscriber makes a call, this Virtual CLI is used. But real incoming CLI is be used if virtual CLI table index is not assigned.

10) Virtual Subscriber's Destination

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

11) Virtual CLI Table

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

- **Whole CLI**: Assigned virtual CLI is sent whatever CLI comes in. Ex) The case that assigned virtual CLI is '2793914' and received CLI is '4504875'. CLI for O/G call is virtual CLI '2793914'.
- **Begin Copied CLI**: Virtual CLI includes some beginning digits of received CLI. Ex) The case that assigned virtual CLI is '12BXXXX0' and receive CLI is '4504875'. CLI for O/G call is '1245040'. B means to copy digits from the beginning of CLI.
- **End Copied CLI**: Virtual CLI includes some end digits of received CLI. Ex) The case that assigned virtual CLI is 'EXXX1234' and receive CLI is '4504875'. CLI for O/G call is '8751234'. E means to copy digits from the end of CLI.
- **Combined Copied CLI**: Virtual CLI includes some beginning digits and some end digits of received CLI. Ex) The case that assigned virtual CLI is 'BXXXEX123' and receive CLI is '4504875'. CLI for O/G call is '4505123'.
- In Admin programming, Virtual CLI is assigned up to 24 digits in case of not including 'B' or 'E', 25 digits in case of including 'B' or 'E' and 26 digits in case of including both 'B' and 'E'.

Conditions

1. Virtual Subscriber Service should be processed before Digit Conversion process and ICLID process.
2. Duplicated CLI can be entered in Virtual CLI Table.
3. It is not allowed to assign duplicated Virtual CLI table index in Virtual Subscriber's Table. If user want to make same CLI in different O/G CO groups, a) duplicate the CLI in Virtual CLI Table and assign different index which has same CLI to Virtual Subscriber's Table.
4. The Virtual Subscriber Table and Virtual CLI Table can be assigned up to **300(MG300)/100(MG100)** tables.

Programming

System Data 1. Virtual Subscriber Service (PGM 24?)

Related Features

Hardware

4.7 USB ALWAYS CALL RECORD

Description

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

Operation

To enable "USB ~~Always~~ Call Record":

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

To disable "USB ~~Always~~ Call Record":

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

Conditions

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

Programming

Numbering Plan 1. USB Call Record Feature Code (PGM 113)
Station Data 1. USB Auto Record Service (PGM 147 – Index 3)

Related Features

Hardware

LDP 7000 keyset

4.8 PRIVATE CO GROUP

Description

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

Each station has own private CO access code and CO lines and he can access CO line with only private CO access code.

There is service priority for each station to seize normal CO line when all private CO lines are busy.

Operation

Conditions

1.Private CO Group code should be one of CO group access codes.

Programming

CO LINE Data 1. CO Access Mode (PGM 162-Index1)

5. IR & REQUESTS

5.1 VM PASSWORD INPUT

Description

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

1. DN number + password
2. Password
3. No password

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

Operation

DN number + password (authorization code)

Users can access to voice mailbox by dialing an DN number and its password.
With this mode, it is possible to access to other DN’s voice mailbox if you know the password of the DN.

Password

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

No Password

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

Conditions

1. If a password is not registered, you cannot access to voice mailbox in the configuration requesting authorization code or password.
2. Voice mailbox of which VM password input is No Password may be accessed through external calls.

Programming

PGM 147-5 : VM PASSWORD INPUT

Related Features

Hardware

5.1 VM SLOT NO ASSIGNMENT

Description
If VM Slot No is assigned for a DN, all voicemail messages for the DN will be stored at that board.

Operation

Conditions

Programming
PGM 147-4: VM SLOT NO

Related Features

Hardware

5.1 SMTP PORT FOR VM E-MAIL NOTIFICATION

Description

Each DN can have its own SMTP port number for VM E-mail notification.

Operation

Conditions
1. If SMTP port number is not assigned, the default value 25 will be used for the port.

Programming
PGM 147-3 : SMTP PORT NUMBER

Related Features
VM E-mail Notification

Hardware

5.2 Tenant Group Access

Description

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

There are four tables for Tenant group access.

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

Operation

Allow/Deny access to other groups:

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

Conditions

1. The Tenant Group Access is one way access. To allow access from group 1 to group 2 it should be allowed 1→2 and 2→1.

Programming

Tenant Data 1. Tenant Group Access (PGM 283)

Related Features

Hardware

5.3 WAIT USER RELEASE FOR IN-BAND INFORMATION

Description

When ISDN CO receives DISCONNECT message with IN-BAND information in PROGRESS INDICATOR, system waits for releasing from external user.

Operation

Conditions

1. This is only applied to ISDN CO lines.

Programming

CO Line Data 1. Outgoing CO Line Attributes (PGM 170-Index 19)

Related Features

Hardware

5.4 TRANSIT DTMF BYPASS

Description

iPECS-MG can set the SIP/H323 lines to bypass the DTMF.

Operation

Conditions

Programming

Tenant Attribute 1. Tenant Attributes (PGM 281-Index 12)

Related Features

Hardware

5.5 SEND DTMF AFTER DIAL TONE

Description

Send digit to CO lines after detecting dial-tone.

Operation

Conditions

1. This is applicable only for analog CO lines.

Programming

Outgoing CO Attribute 1. Outgoing CO Line Attributes (PGM 171-Index 12)

Related Features

Hardware

5.6 SIP Phone Voice Mail Notification

Description

When a user has left a voice message on an SIP terminal, some SIP terminals including IP-8800/6800 can notify the user of voice messages. This function requires the VM notification function to be supported in SIP terminals.

The [MSG] LED will flicker if there is a voicemail saved for the user.

Operation

SIP Phone

To retrieve voice mail:

1) When **[MSG]** notification is present, press the **[MSG]** button; the number of new messages and saved messages are displayed.
2) Dial the {Voice Mail Access} feature code.

Conditions

To use this function, the SIP terminals must support VM notification.

Programming

Related Features

Hardware

SIP phone supporting extended “Alter-Info” function (IP-6800, IP-8800)

5.7 SIP Phone BLF

Description

The BLF function is available in the terminals with BLF device (IP-8800 series). This function enables users to check the status of other extension, and to perform functions like pick-up and call transfer.

Operation

SIP Phone

To register BLF:

1) On the terminal program, assign the BLF function to the FLEX button.
2) Check the status of other users with the registered button.

To use BLF:

1) Access the SIP terminal web setting screen.
2) Register an extension in the BLF button of the SIP terminal.
3) The status of the registered extension is displayed on the [BLF] button.
   - Busy: LED on
   - Incoming ring: LED flickers
   - Not used: LED off

To perform Call Pick-up:

1) The [BLF] button flickers due to an incoming ring to the registered BLF user.
2) By pressing the flashing button, you can pick up the incoming call.

Conditions

This function is only supported in the SIP terminals that comply with the LG-Ericsson SIP-extended I/F spec (IP88xx).

Programming

Related Features

Hardware

LG-Ericsson SIP phone (IP88xx)

5.8 SIP PHONE DISTINCTIVE RING

Description

This function enables the system to provide different rings depending on whether the call is from an extension user or a trunk user.

Operation

SIP Phone

To program distinct ring based on the call type:

1) Access the SIP phone web setting window.
2) Set different rings for internal calls and trunk calls.
3) When a call is incoming, the ring is provided depending on whether the call is from internal or trunk user.

Conditions

This function is only supported in the SIP terminals that comply with the LG-Ericsson SIP-extended I/F spec (IP88xx).

Programming

Related Features

Hardware

LG-Ericsson SIP phone (IP88xx)

5.9 SIP PHONE INTERCOM

Description
This function automatically opens the speaker for incoming calls, so that the users can speak without lifting the handset.

Operation
SIP Phone

If the terminal receiving mode is Handsfree:

1) On Web-ADMIN, change the internal call receiving mode to "H".
2) For incoming call, the terminal automatically opens the speaker and answers the call.

If "forced change of extension call answer mode" is activated for the extension with the terminal receiving mode "T":

1) The calling party presses the {Forced Hands Free Call} feature code to place a call to an extension.
2) If a call is incoming, the terminal automatically opens the speaker to answer the call.

Conditions
This function is only supported in the SIP terminals that comply with the "Alert-Info" function (IP-68xx, IP-88xx)

Programming

Related Features

Hardware
SIP phone supporting extended "Alter-Info" function (IP-6800, IP-8800)

5.10 SIP PHONE CALL-BACK

Description
If the dialed number is on busy, the caller may register "call-back" to the called party. If call-back is assigned, the system sends the ring to the calling party if the call is terminated, and the called party returns to idle.

Operation
SIP Phone

To register call-back:

1) Place a call to a busy extension user.
2) Receive a busy tone.
3) On-hook, and dial the "Call Back/Queuing Register" feature code + Extension Number.
4) You will hear the confirmation tone.

To answer call-back:
1) The busy extension returns to idle.
2) The call from the extension is incoming automatically.
3) Answer the call.

Conditions
This function is only supported in the SIP terminals that comply with the “Alert-Info” function (IP-68xx, IP-88xx)

Programming

Related Features

Hardware
SIP Phone

5.11 SIP PHONE CALL INTRUSION

Description
Users can attempt intrusion into the bus user. If intrusion is made successfully, a 3-party conference is made including the extension that has attempted an intrusion.

Operation
To perform call intrusion:
1) Place a call to a busy extension.
2) Receive the busy tone.
3) On-hook, and dial the “Intrude Request” feature code + Extension number.
4) A 3-party conference is made between the parties that have been in the call, and the new user who intruded into the call.

Conditions
Allowing intrusion into busy SIP phone is only supported in the terminals that comply with LG-Ericsson SIP extended I/F spec. (LIP-88xx)
Call intrusion is supported in any SIP terminal.

Programming

Related Features

Hardware
LG-Ericsson SIP phone which is supporting SIP extension I/F (LIP-88xx)

5.12 SIP PHONE CALL OVERRIDE

Description

If a user attempts call override, the user in the current call is put on hold, and a connection is made between the called party and the user that attempted call override.

Operation

SIP Phone

To connect a call override

1) Place a call to a busy extension.
2) Receive the busy tone.
3) On-hook, and dial the "Call Override" feature code + Extension number.
4) The current call of the called party is put on hold, and the connection is made with the extension that attempted call override.

Conditions

Allowing override into busy SIP phone is only supported in the terminals that comply with LG-Ericsson SIP extended I/F spec. (LIP-88xx)

Call override can be attempted in any SIP terminal.

Programming

Related Features

Hardware

LG-Ericsson SIP phone which is supporting SIP extension I/F (LIP-88xx)

5.13 V2.0 Enhanced Database Management

Description

DB management controller is changed from under V1.7 version for V2.0 enhanced feature.
DB controller is redesigned for bellow 2 key points.

1. Extension feature
2. Compression for spacing

For next enhanced feature, database structure has to be extended and redesigned. In current space of SRAM memory for database is small to append new functions for meeting customer hope, because iPECS-MG100 can support only 4Mb, iPECS-MG300 can support 8Mb.
So DB controller is changed in V2.0 version.

Compatibility

1. MPB S/W Upgrade
   When MPB S/W is upgraded from under V1.7 to V2.0, all of previous data will be kept. User can use previous functions. All of features work well.

2. MPB S/W Downgrade
   In case of Downgrade from V2.0 to under V1.7 S/W, data of DB has to be initialized.

3. V1.7 DB file Upload to V2.0 MPB
   Saved V1.7 DB file can upload to V2.0 system.

4. V2.0 DB file Upload to V1.7 MPB
   V2.0 DB file cannot upload to V1.7 System. V1.7 system cannot recognize V2.0 BD file.

5. Individual Database
   Any individual V1.7 DB file cannot upload to V2.0 System,
   Any individual V2.0 DB file cannot upload to V1.7 System,

Operation

Conditions

1. Condition Issue…

Programming

Related Features

Hardware

5.14 ACD GROUP SUPERVISOR FEATURE

Description

Supervisor and Sub-Supervisor can check and monitor agent's status. And Supervisor and Sub-Supervisor can overhear agent's conversation.

Supervisor and Sub-Supervisor also can record agent's conversation during monitoring with two-way recording feature code or Record menu on 3 Soft-Button. When supervisor try to record agent's conversation, S-Monitor alert tone will be served to agent and called-party. And then agent's conversation will be recorded.

Supervisor and Sub-Supervisor can make conference with agent's conversation. And also during conference supervisor can release one of member forcibly.

Operation

Agent's call Recording:

1. Dial {ACD Supervisor Silent Monitor} feature code.
   Or
2. Press flex button registered as {ACD Supervisor Silent Monitor} feature code.
3. Dial desired Agent number
4. Press Two-way recording feature flex button during conversation monitoring.
   Or
5. Press Record menu on 3 Soft-Button
6. If supervisor want to stop recording,
7. Press again Two-way recording feature flex button
   Or
8. Press Record-Stop menu on 3 Soft-Button

Agent's call Conference:

1. Dial {ACD Supervisor Silent Monitor} feature code.
   Or
2. Press flex button registered as {ACD Supervisor Silent Monitor} feature code.
3. Dial desired Agent number
4. Press Conf menu on 3-soft button.

Conditions

2. Supervisor has to have Two-way recording authority.
3. If supervisor make conference during recording, recording feature will be stopped.
4. If agent's conversation is already recorded, supervisor cannot record conversation.
5. While agent's conversation is recorded, even if agent's conversation is over, recording feature will not be stopped. Supervisor has to stop recording feature.

Programming

Station Group
1. Main Supervisor (PGM200)
2. Sub Supervisor (PGM200)

Related Features

Hardware

5.15 SMDR NEW OPTION

Description

Each CO line can be set to print SMDR data or not. There are 5 type options for charge mode. (Free, Report according to SMDR report type, only for Incoming Call, Outgoing or Transit Call) Incoming or Outgoing Call option has high priority than SMDR report type rule. So regardless of SMDR report type, specific CO line can be reported as SMDR data.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. In case of Interface SMDR data, there are transferring information and also original calling information about Transit Call.
2. SMDR condition…

Programming

CO	1. Charge Mode (PGM160)

Related Features

Hardware

5.16 VOIB/VMIB CHANGEABLE WEB PORT NUMBER

Description

Administrator can change web page port number of VOIB/VMIB board.
Default WEB port number of all of board is 80, and this number can be changed and then after changing VOIB or VMIB will be restarted for adjusting port number.

After board reset, changed port number will be used for WEB page of VOIB or VMIB board

Operation

Conditions

2. If port number is changed, target board will be restarted automatically.
3. Conditions...

Programming

Board Data 1. WEB Port Number (PGM305)

Related Features

Hardware

5.17 TONE SERVICE FOR DECT SWITCHED-OFF CASE

Description

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

Operation

When DECT terminal is switched on:

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

When DECT terminal is switched off:

1. The caller will hear "Wireless Station Searching Tone" first and the system will search for the called DECT station.
2. In about ten seconds, the search will fail and the call will be routed to "Error Destination" in case it is a CO call. If the caller is an internal station, "Internal No Answer Tone" will be heard and the call will be disconnected.

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

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

Conditions

8. Internal call will not be rerouted even if DECT station is not found.
9. If the CO ring is assigned only to one DECT station, the CO call will be rerouted in case that DECT terminal is switched off. But if there're multiple ring-assigned stations ~~other than DECT phone~~, the call will not be rerouted. ~~Finally, if multiple DECT stations are assigned ring and all of them are switched off, the call will be rerouted.~~

Programming

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

Related Features

Hardware

5.18 DIRECT/INDIRECT HELD CO RETRIEVE

Description

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

Operation

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

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

To retrieve a held CO indirectly:

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

Conditions

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

Programming

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

Related Features

Hardware

5.19 VM Private Message

Description

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

Operation

To mark a voice message as private:

1. Record the desired message after hearing the user greeting and beep tone.
2. Dial '#' after message recording is finished.
   Or
   Dial '*' for further options and then dial '#'.
3. The following prompt will be heard.
   "For regular delivery, press one. To mark urgent, press two.
   To mark private, press three. To mark urgent and private, press four"
4. Dial '3' for a normal delivery in private option
   Or
   Dial '4' for an urgent delivery in private option

Conditions

1. This feature needs recording of new VMIB prompts. In addition, "Enhanced VM Features" must be enabled in System Attributes (PGM 223 – Index 20).
2. In case the caller is an internal station, "Private Message Mark" attribute must be set ON in VM COS (PGM 243 – Index 8). Otherwise, private message cannot be left.
3. If the caller is an external party, it is always possible to leave a private message.

Programming

Station Data	1. VM COS (PGM 145 – Index 20)
System	1. Enhanced VM Features (PGM 223 – Index 20)
	2. Private Message Mark (PGM 243 – Index 8)

Related Features

Hardware

VMIB

5.20 VM MESSAGE DELIVERY CONFIRMATION

Description

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

Operation

To mark a voice message as delivery confirmation:

1. Record the desired message after hearing the user greeting and beep tone.
2. Dial '#' after message recording is finished.
   Or
   Dial '*' for further options and then dial '#'.
3. The following prompt will be heard.
   "For regular delivery, press one. To mark urgent, press two.
   To mark private, press three. To mark urgent and private, press four
   To request delivery receipt of the message for future, press 5"
4. Dial '5' to set delivery confirmation on the message

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

1. A confirmation message is sent back to the sender's mailbox.
2. The sender will see the notification of message through LCD display or LED button.
3. If the sender accesses the mailbox, the following message will be played.
   "Message for XXX was listened to on HH:MM MM/DD".
   XXX stands for mailbox number or recorded name while HH:MM and MM/DD are the time and date information, respectively.

Conditions

1. This feature is available only if the caller is an internal party.
2. This feature needs recording of new VMIB prompts. In addition, "Enhanced VM Features" must be enabled in System Attributes (PGM 223 – Index 20).
3. "Confirm Message Receipt" attribute must be set ON in VM COS (PGM 243 – Index 7) for the sender's VM COS. Otherwise, it is not available and the corresponding prompt will not be heard.
4. The confirmation message is treated as a normal VM message and all options apply except for rewind and fast forward.
5. The name of mailbox can be recorded by using the feature code {Record VM Subscriber Name}.

Programming

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

Related Features

Hardware

VMIB

5.21 VM MESSAGE FUTURE DELIVERY

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

Operation
To mark a voice message as future delivery:

1. Record the desired message after hearing the user greeting and beep tone.
2. Dial '#' after message recording is finished.
   Or
   Dial '*' for further options and then dial '#'.
3. The following prompt will be heard.
   "For regular delivery, press one. To mark urgent, press two.
   To mark private, press three. To mark urgent and private, press four
   To request delivery receipt of the message for future, press 5
   For future delivery, press 6"
4. Dial '6' to set future delivery on the message.
5. System will play the prompt like
   "Enter date and time and press one of the following options, 1 for AM, 2 for PM"
6. User dials 4 digits MM/DD and 4 digits HH:MM.
7. And finally, 1 digit for AM or PM should be dialed.
   For example, if user dials 0903 0830 1 for MM/DD, HH:MM, and AM or PM, it means Sep 3rd 8:30 AM.
8. If the user finishes dialing, message will be sent to the destination in future.

Conditions
1. This feature needs recording of new VMIB prompts. In addition, "Enhanced VM Features" must be enabled in System Attributes (PGM 223 – Index 20).
2. "Future Delivery Message" attribute must be set ON in VM COS (PGM 243 – Index 6) for the receiver's VM COS. Otherwise, it is not available and the corresponding prompt will not be heard.

Programming

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

Related Features

Hardware

5.22 VM MESSAGE FAST FORWARD / REWIND

Description

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

Operation

To forward the voice message while hearing:

1. Dial '#' while the message is played.
2. System will fast forward the message as the programmed amount of time.

To rewind the voice message while hearing:

1. Dial '*' while the message is played.
2. System will rewind the message as the programmed amount of time.

Conditions

3. This feature is available only for the voice messages left in the mailbox. And only VMIB supports the feature.
4. Delivery confirmation message doesn't support fast forward and rewind features.

Programming

Related Features

Hardware

VMIB

5.23 VM MESSAGE PAUSE / START

Description

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

Operation

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

1. Dial '8' while the message is played.

To resume the paused playback of a voice message:

1. Dial '8' in a paused status.

Conditions

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

Programming

Related Features

Hardware

VMIB

5.24 DIGIT SENDING MODE

Description

iPECS-MG provides the option for digit sending method.(Information Message / Inband DTMF) Basically, it sends Inband DTMF signal to destination after it receives a Call Proceeding message. Some systems such as LIK, ipLDK needs Information Message after it sends Call Proceeding message.

Operation

This can be applied automatically according to PGM.

Conditions

1. This option can be set in case CO line is QSIG/H.450 Type.

Programming

CO Data 1. Digit Sending Mode (PGM 170-Index18)

Related Features

Transit Out

Hardware

5.25 OFF NET CALL FORWARD TONE

Description

The Off Net CFW Tone can be provided when a call is forwarded to Off Net.
There are 3 kinds of option to provide Off Net CFW Tone.

1. **No Tone:** a call can be routed to Off Net FWD Destination without any tone.
2. **Tone:** a call can be routed to Off Net FWD Destination with tone.
3. **After Tone:** a call can be routed to Off Net FWD Destination after a tone is finished

This can be set according to type of CO line (Normal CO, Digital CO-R2)

0. Normal CO(No Tone), R2 CO(No Tone)
1. Normal CO(No Tone), R2 CO(Tone)
2. Normal CO(No Tone), R2 CO(After Tone)
3. Normal CO(Tone), R2 CO(No Tone)
4. Normal CO(Tone), R2 CO(Tone)
5. Normal CO(Tone), R2 CO(After Tone)
6. Normal CO(After Tone), R2 CO(No Tone)
7. Normal CO(After Tone), R2 CO(Tone)
8. Normal CO(After Tone), R2 CO(After Tone)

Conditions

Programming

Tenant Data	1. Off Net CFW Tone Usage (PGM 281-Index13)
	2. Off Net Call Forward Tone (PGM 290-Index64)

5.26 TONE TABLE BASED ON CO LINE

Description

System can provide 9 Tone tables and each tenant has 1 tone table.

If tone table index is programmed in each CO line, system provides tone according to tone table. Or, system provides the tone with tone table for tenant.

Operation

When an user does not assign a tone table index in CO line

1. System check tone table index for tenant.
2. With tone table index, system provides TONE in PGM 290.

When an user assigns a tone table index in CO line

1. System check tone table index with CO line
2. With tone table index, system provides TONE in PGM 290.

Conditions

Programming

CO LINE Data	1. Tone Table Index (PGM 161-Index 12)
Tenant Data	1. Tenant Tone Table Index (PGM 280-Index9)
	2. Tone Table (PGM 290)

5.27 Mobile Extension Button Prgoram

Description

System allows programming the activation button of mobile extension.

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

Operation

To program a Mobile Extension Activation Button

5. Press [Trans/PGM] button and select the flexible button to program

3. Dial '1' to program number

4. Dial 'Mobile Extension Activation' feature code

5. Dial index of Mobile Extension to program.(1-2)

6. Press [Hold/Save] button.

To activate/deactivate Mobile Extension feature.

1. Press [Mobile Extension Activation] button

6.

Conditions

Programming