iPECS-CM

Feature Description & Operation Manual

iPECS is an Ericsson-LG Brand

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

Revision History

ISSUE	DATE	DESCRIPTION OF CHANGES
1.0	15-Apr-10	Preliminary release
2.0	15-Oct-10	Initial Release
3.0	30-Jul-11	LIP-8000E, LDP-9000, Related WMS menu update Wake-Up, Multi Language Prompt, Speed Dial , Name Registration are revised
3.1	26-Sep-11	SIP BLA, Door Open, IP Call Recording
3.2	02-Dec-11	New Edition including Version 3.0
3.3	23-Dec-11	Update for changed WMS Menu
3.4	07-Apr-12	Major format and editing throughout
	08-May-12	Review issue 3.4
3.5	27-OCT-12	New Edition including Version 4.0
3.6	26-MAY-13	New Edition including Version 5.0
3.7	27-DEC-13	New Edition including Version 5.5 and Changed Ericsson-LG to Ericsson-LG Enterprise

Table of Contents

1. Introduction

1.1 Overview

This document provides detailed information covering description and operation of the numerous features available in the iPECS-CM Release 3 system software.

1.1.1 Manual Structure

The features are grouped based on common characteristics into the following two (2) groups:

- System
- SIP Phone

Within the two groups, features are organized alphabetically.

1.1.2 Feature Information

Each feature is explained with the following six (6) sections:

- *Description*: explains the nature of the feature.
- *Operation*: gives detailed step-by-step operation of the feature for iPECS IP Phones and SLTs.
- *Conditions*: explains known feature interactions and constraints related to the feature.
- *Related WMS Menu*: lists Menus of WMS (Web Management System) for database entries that may be required for proper feature operation.
- *Related Features*: lists related topical information to aid in understanding the feature.
- *Hardware*: lists hardware required for proper feature operation

1.2 System Capacities

IPECS-CM CAPACITY TABLE

<table>
<tr><th>ITEM</th><th colspan="2">CAPACITY</th><th colspan="2">REMARKS</th></tr>
<tr><td rowspan="2">Total Slots</td><td colspan="2">8000 Slots / System</td><td colspan="2"></td></tr>
<tr><td colspan="2">6 Slots/CM-MGC</td><td colspan="2">CM-MGC(5U)</td></tr>
<tr><td rowspan="8">Port Capacity</td><td></td><td>Total</td><td>Trunk</td><td>Extension</td></tr>
<tr><td>S30K</td><td>30,000</td><td>10,000</td><td>30,000</td></tr>
<tr><td>S10K</td><td>10,000</td><td>5,000</td><td>10,000</td></tr>
<tr><td>S4K</td><td>4,000</td><td>2,000</td><td>4,000</td></tr>
<tr><td>S2K</td><td>2,000</td><td>1,000</td><td>2,000</td></tr>
<tr><td>S1K</td><td>1,000</td><td>500</td><td>1,000</td></tr>
<tr><td colspan="4">* Total Ports = Trunk Ports + Extension Ports</td></tr>
<tr><td colspan="4">* S1K is used only for an LCM</td></tr>
<tr><td>Tenant Group</td><td colspan="2">254</td><td colspan="2"></td></tr>
<tr><td>Numbering Plan</td><td colspan="2">8 digits</td><td colspan="2">Extension Number/Feature Code
/Trunk Access Code</td></tr>
<tr><td>Attendant</td><td colspan="2">30/Tenant
7620/System</td><td colspan="2"></td></tr>
</table>

ITEM	CAPACITY	REMARKS
Attendant Night Subscriber	10/Tenant	
Attendant Overflow Subscriber	5/Attendant	
DSS/BLF Console	10 per Extension	
Conference Group	100/System	128/Goup, 256/CM-VPCM
Internal Paging Zone	100/Tenant	
System Speed Dial	3,000(24 digits)	
Station Speed Dial	100 (32 digits)	
SMDR	1,000,000 Records	
Authorization Code	30,000 (Max. 12 digits)	
Trunk Group	500/System	500/Group
Trunk Access Code	2,000	
Individual Trunk Group	500/System	20/Group
Trunk Call Routing Table	500/System	30/Table
Key Number Group	1000/System	1000/Group
Call Pick-up Group	2,000/System	100/Group
Command Call Group	500/System	128/Group
Interphone Group	100/System	100/Group
Paging Group	100/Tenant	128/Group
PTT Group	10/Tenant	128/Group
Phone Number	60,000/System	30 Phones/Number 48 Number/Phone
Hot Desk IDs	30,000	
System Numbering Plan	127,000/System 500/Tenant	
Digit Restriction Table	Class: 128	
	1,600/Class	
	24 digits	
Digit Conversion Table	Class: 128	
	1,600/Class	
	24 digits	
IP DECT	200 Subscribers	
	Up to 40 Base	

1.3 System Numbering Plan

Description

A System Numbering Plan defines which and how many digits are used for Extension numbers, Trunk access codes and feature codes in the system.

Users can access system resources and functions of the system with the feature codes. Numbering plans, including feature codes, are not defined in the default system database. The Numbering Plans must be defined by administrator to match the site environment. The Numbering Plans must be established through the WMS (Web Management System). All programmable feature codes available are listed in the chart below.

Feature Code List

Internal Page	Internal Page Answer	Forward Register (Normal)
Forward Cancel	Call Wait Request	ACD Agent Headset Ring Mode Change
Forward Register at Remote Extension	Pre-Defined Text Register	ACD Agent Release Call
Pilot Hunt Group Forward (Register)	Force Handsfree	ACD Supervisor Display Q Wait Count
Pilot Hunt Group Forward (Cancel)	Anonymous Call Reject (Register)	ACD Supervisor Group Night Mode
DND Register/Cancel (Toggle)	Anonymous Call Reject (Cancel)	ACD Supervisor Group Holiday Mode
Force DND Cancel	Caller ID Display Restrict (Call Base)	ACD Supervisor Group Call Forward
Account Code	Caller ID Display Restrict (On/Off)	ACD Supervisor Silent Monitor
Trunk Flash	Connected ID Display Restrict (On/Off)	ACD Supervisor ACD Q Overflow Count Change
Last Number Redial	Pilot Hunting Request	ACD Call Indicator
Station Speed Dial (Register)	Command Group Call (Page)	Non ACD Call Indicator
Station Speed Dial	Command Group Call (Conference)	ACD Announcement Play
Message Wait Register	Intrude Request	Override Register (SIP)
Message Wait Answer	Camp On Register	Two Way Record
Message Wait Cancel	OHVO(Off Hook Voice Over) Request	Delete All Voice Mail
Extension Call Back/Trunk Queuing	Mobile Extension Register	Voice Mail Record for Internal Page
Extension Call Back/Trunk Queuing Cancel	Mobile Extension CID Register	Direct VM transfer
Call Pick-Up (Group)	Mobile Extension Activate/Deactivate	Hotel Room Maid Status
Call Pick-Up (Direct)	System Tone Listen	Hotel Guest Info.
Walking COS	VPCM Prompt/Announcement Listen	Hotel Minibar
Call Park (Register/Answer)	Announcement Table Access(CCR)	Hotel Room Monitor
Station Program Mode	Announcement Table Access & Release	Internal Page (Recorded Voice Mail)
AME(Answering Machine Emulation)	Call Hold (Register)	Internal Page (Annc. Serial No)
Voice Mail Access	Information Voice Announce	Terminal BLF Registration
Trunk Line Access	DISA Tone Access	Key Number Group Forward (Register)
MCID Report	All Feature Cancel	Ken Number Group Forward (Cancel)

PTT Group Login/Logout	Conference Member Add	User Answer Greeting Play (On/Off)
Hot Desk Login/Logout	System Service Time Mode Change	Call Log List Display
Conference Room Activate	ACD Agent Not Ready Mode	PSTN Failover Forward Register
Conference Room Deactivate	ACD Agent Log On/Off	PSTN Failover Forward Cancel
Extension Class Down	ACD Agent Work Mode	Transfer Monitor
Extension Class Restore	ACD Agent Auto Work Mode After Call	ISDN MCID Request
Password Change	ACD Agent Auto Answer(On/Off)	Prepaid Money Register
Interphone Call	ACD Agent Head/Hand Set	Virtual Desk Login/Logout
Forced Call Record Stop	ACD Supervisor Traffic Info	Company Directory Service
VM Name Recording (Company Directory)	VMIB Greeting Recording	ACD Agent Help Request
Forward Group Register	Forward Group Cancel	Change Billing Targer

Operations

Conditions

・Up to eight (8) digits can be assigned for the Extension, Trunk and feature codes.
・Duplications are not allowed in and across the Numbering Plans.

Related WMS Menu

System Management > System Environment > “Number of Tenant in System”
> System Environment > “System Numbering Plan”
Data Management > Numbering Plan Information > Numbering Plan
> Numbering Plan Information > Feature Code

Related Features

Hardware

2. System Features

2.1 Account Code

Description

Extension users 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.

Operation

iPECS Multi-button Phone

To assign a Flex button for {Account Code} operation using the Extension User program:

- [PGM] + {FLEX} + Button Feature Type (2) + Feature Code (Account Code) + [SAVE]

To enter an Account Code using the {Account Code} button prior to placing a call:

1. Lift the handset or press the [SPEAKER] button.
2. Press the {Account Code} button.
3. Enter the account code (up to 12 digits).
4. Dial *, the intercom dial tone will be heard.
5. Place a trunk call.

To enter an account code using the {Account Code} button during a Trunk call:

6. Press the {Account Code} button.
7. Enter the account code (up to 12 digits).
8. Dial *.

Single Line Telephone (SLT)

To enter an account code prior to placing a call:

1. Lift the handset.
2. Enter {Account Code} feature code
3. Enter the Account Code (up to 12 digits).
4. Dial *, the intercom dial tone will be heard.
5. Place a Trunk call

To enter an account code using the {Account Code} button during a trunk call:

1. Make a hook-flash, the intercom dial tone will be heard.
2. Enter {Account Code} feature code.
3. Enter the Account Code (1 to 12 digits).
4. Dial *, you will be reconnected to the trunk.

Conditions

- When entering an account code, DTMF tones will not be heard by the connected party.

Related WMS Menu

Data Management > Numbering Plan Information > Feature Code > “Account Code”

Related Features

- Authorization Code
- Flexible Button Assignment

Hardware

2.2 Alternative Route Selection (ARS)

Description

This function is useful if there are numbers of routes to the destination. If the route selected by the user is not available for any reason (all busy, line failure), the system automatically routes the call to Alternative Route assigned through WMS.

Operation

Conditions

- The system can route calls even if there is no directly connected route. In this case, there must be a preset alternative route.
- Up to 10 alternative routes can be assigned to a single route.
- This function is supported for the functions like Last Number Redial, Station Speed Dial, and System Speed Dial.
- The administrator may select whether the alternative routes will be used.
- The Auto Network Dialing function can be used when an alternative route is used.
- If the “Digits to dial” option is set to “Before change”, and the Trunk access code is contained in the dialed digits, the system will send the digits to the Trunk excluding the access code.
- The ARS table can be set as the access code or route.
- Administrator can “Digits to dial” 4 option as belows:
- After Digit Conversion: send digits of after digit conversion to the trunk
- Before Digit Conversion: send digits of before digit conversion to the trunk.
- Use Digit Conversion Class(After Digit Conversion) : use another digit conversion class to decide digits to turnk. At this time, use digits of after digit conversion to compare with digits which is added in “Digit Conversion Table”
- User Digit Conversion Class(Before Digit Conversion): use another digit conversion class to decide digits to trunk. At this time, use digits of after digit conversion to compare with digits which is added in “Digit Conversion Table”

Related WMS Menu

[To add ARS]

Data Management	> Trunk Information > Trunk Basic Information > Trunk Access Code > “ARS Trunk Access Code”
	> Trunk Information > Trunk Basic Information > Trunk Access Code > "ARS Access Code"
	> Trunk Information > Trunk Basic Information > Trunk Access Code > "ARS AND Digit"

> Trunk Information > Trunk Basic Information > Trunk Access Code > "Digits to dial"

[To enable ARS]
Data Management > Trunk Information > Trunk Access Code > “ARS Service”-> “ON”

Related Features

Hardware

2.3 Answering Machine Emulation (AME)

Description

When an Extension/Trunk call is sent to a Voice Mailbox, the associated Extension can be assigned to notify the user and allow the user to screen the call. Two methods of notification and call screening are provided, Silent or Speaker mode.

In the Silent mode, the user is notified by flashing of the AME Flex button LED. The user 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 the user’s multi-button phone.

The user may terminate the screening leaving the caller in the Voice Mailbox to record a message, talk with the caller and record the conversation in the mailbox, or answer the call and disconnect the Voice Mail.

The user’s multi-button phone must be assigned with an AME Flex button for proper operation.

Operation

To assign an {AME} button to Flex button using the Extension User program:

- [PGM] + {FLEX} + Button Feature Type (2) + AME (Answering Machine Emulation) code + [SAVE]

To set the Silent mode:

1. Lift the handset or press the [SPEAKER] button.
2. Press the {AME} button.
3. Dial ‘1’.
4. Press the [SAVE] button.

To set the Speaker mode:

1. Lift the handset or press the [SPEAKER] button.
2. Press the {AME} button.
3. Dial ‘2’.
4. Press the [SAVE] button.

To deactivate the AME function:

1. Lift the handset or press the [SPEAKER] button.
2. Press the {AME} button.
3. Press ‘0’.
4. Press the [SAVE] button.

To screen a call in the Silent mode:

1. Press the flashing {AME} button, the caller’s voice is broadcast over the Extension 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, the caller’s voice is recorded in Voice Mailbox.

To talk with the caller and record the conversation in Voice Mail:

1. Press the illuminated [MUTE] button to record the voice while continuing the call.

To answer the call and cancel the voice message:

1. Press the illuminated {AME} button to continue the call without recording it in Voice Mailbox.

Conditions

- The AME function operates when the Extension is forwarded to internal Voice Mail.
- The {AME} button must be assigned to Flex button of the user’s iPECS Multi-button phone for AME.

Related WMS Menu

[To assign the AME (Answering Machine Emulation) feature code]

Data Management > Numbering Plan > Feature Code > “AME(Answering Machine Emulation)”

[To set Answering Machine Emulation]

Data Management > Extension Information > Number(DN) Information > DN Feature Registration > “Answering Machine Emulation”

Related Features

Hardware

- iPECS Multi-button phone
- VPCM

2.4 Auto Called Number Redial (ANCR)

Description

This feature allows an Extension user to request the system to retry a busy external call until the call is connected or the feature is cancelled.

Operation

iPECS Multi-button Phone

To activate ACNR while hearing busy tone:

1. Press the [MSG/CALLBK] button or [ACNR] soft button.
2. Hang up the call or press the [SPEAKER] button.

To cancel ACNR while idle:

1. Press the flashing [MSG/CALLBK] button or [ACNR] soft button.

System

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

Conditions

- ACNR Pause Timer and Retry Count can be assigned through WMS.
- ACNR Pause Time defines the time allowed between ACNR retries.
- ACNR Retry Count determines the number of times the system will retry before ACNR is automatically cancelled.

Related WMS Menu

[To enable ACNR]

Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > "ACNR"

[To set ACNR Pause Time]

Data Management > Tenant Information > Tenant Time Information > Tenant Timer > "ACNR Pause Time"

[To set ACNR Retry Count]

Data Management > Tenant Information > Tenant Basic Information> Tenant Attribute > "ACNR Retry Count"

Related Features

- Last Number Redial (LNR)
- Speakerphone
- Mute

Hardware

- iPECS Multi-button Phone

2.5 Auto Trunk Access Service

Description

A user can place an external call by dialing the external telephone number without a trunk access code. The system can be set to dial a specific Trunk Access code for the Extension and then sends the user dialed external telephone number.

Operation

Conditions

- Only one Auto Trunk Access code is allowed for an Extension.
- This feature can be configured with Digit Conversion.

Related WMS Menu

[To set Auto Dialing Digit and Auto Dial Pause Time]

Data Management > Extension Information > Number(DN) Information > DN Attribute > "Auto Dialing Digit"
> Extension Information > Number(DN) Information > DN Attribute > "Auto Dial Pause Time (sec)"

[To set Digit Conversation Map and Group]

Data Management > System Feature Information > Digit Conversion Information > "Digit Conversion Map"
> System Feature Information > Digit Conversion Information > "Digit Conversion Table"

[To set System Time Zone]

System Management > System Time Zone

Related Features

Hardware

2.6 Automatic Network Dialing (AND)

Description

Automatic Network Dialing (AND) dials assigned digits automatically when Trunk is seized using the Trunk Access code.

Operation

Conditions

- AND digits cannot include a Pause.
- Up to ten (10) AND digits can be assigned to a Trunk Access code.

Related WMS Menu

[To set AND digits]
Data Management > Trunk Information > Trunk Access Code > "AND Digit"

Related Features

Hardware

2.7 Automatic Pause Insertion

Description

A pause may be included in a Speed Dial number to stop dialing for a period. A pause may also be automatically inserted when configured in the Outgoing Route Information.

Operation

Conditions

- The pause entered by the user is indicated as "P" in the Trunk dial string on the LCD of the iPECS Multi-button phone.
- A pause inserted automatically by the system is not indicated in the LCD.
- The programmed pause is applied to only DTMF signaling, not dial pulse or R2 signaling.

Related WMS Menu

[To set Outgoing Route DTMF Sending Time]
Data Management > Trunk Information > Outgoing Route Information > Outgoing Route DTMF Sending Time

Related Features

Hardware

2.8 Automatic Privacy

Description

This feature allows a user in conversation not to be disturbed by Intrusion/Call Wait/Camp-On/OHVO

Operation

iPECS Multi-button Phone

To intrude on a call when Privacy is disabled:

1. When a call is placed to a busy Extension, busy tone will be received.
2. Intrusion/Call Wait/Camp-On/OHVO features can be utilized.

To enable/disable Privacy during a call (temporary Privacy enable):

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

Conditions

- Automatic Privacy/Branch Line is disabled as a default and the service must be enabled in the WMS menu.
- Only an iPECS Multi-button phone can intrude when Privacy is disabled. A tone is provided to the parties when Privacy is overridden.
- If temporary privacy is enabled during a call (One-Time DND), Privacy will be disabled automatically after the call is terminated

Related WMS Menu

[To use Automatic Privacy]	
Data Management	> Extension Information > Number (DN) Information > DN Feature Registration > “Automatic Privacy “
[To use Call Wait, Voice Over, Camp-On]	
Data Management	> Extension Information > Number (DN) Information > DN Feature Allow/Deny > “Call Wait” Extension Information > Number (DN) Information > DN Feature Allow/Deny > “Voice Over” Extension Information > Number (DN) Information > DN Feature Allow/Deny > “Camp-On”
[To set One Digit Service on Busy]	
Data management	> Tenant Information > Tenant Basic Information > One Digit Service on Busy

Related Features

- Busy One Digit Service
- Call Wait
- Voice Over
- Camp-on
- Call Intrusion

Hardware

2.9 Broker Call

Description

Broker call allows an SLT user to be engaged 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 and Call-Wait.

- Transfer Broker Call – 2nd Call is originated by SLT user.
- Call-Wait Broker Call – 2nd Call is delivered to the SLT through a Call-Wait.

Operation

Single Line Telephone (SLT)

To activate a transfer broker call:

1. Make or receive an intercom or external call.
2. Press and release the hook-switch (hook-flash), intercom dial tone is received and the active call is placed in Exclusive hold state.
3. Place the second call.
4. To alternate between calls, make a hook-flash.

To activate Call-Wait, Broker Call:

1. Receive Call Wait tone.
2. Press and release the hook-switch (hook flash) to alternate between calls.

Conditions

- To activate a Transfer Broker call, Hook Action During Transfer must be set to “Broker Call”.
- During a Transfer Broker Call, if the user goes on-hook, the Broker Call parties are connected.
- During a Call-Wait Broker Call, if the user goes on-hook, the active call is disconnected and the held call is recalled.

Related WMS Menu

[To use Broker Call]

Data Management	> Terminal Information > Terminal Option > “Hook Action During Transfer”->Broker Call

Related Features

- Call Waiting
- OHVO

Hardware

2.10 Built-In Auto Attendant/Voice Mail

2.10.1 VPCM/VMIM

Description

VPCM includes processing and memory for the iPECS-CM integrated Auto Attendant, Voice Mail and system announcement applications. The memory is employed to store Auto Attendant announcements, Voice Mail, greetings and messages, and various system prompts. The system prompts (time, date, etc.) are employed by the Auto Attendant and Voice Mail applications as well as other system features.

2.10.2 Built-In Auto Attendant

Description

Incoming internal and Trunk calls through DID or DISA lines can be routed to one of 1000 announcements established in the Announcement Table. An announcement can be assigned as a Key Number Group announcement or as an Auto Attendant announcement with Audio/text tables that permit CCR (Caller Controlled Routing). Key Number Group announcements are played when a call is routed to the group based on definitions in the Key Number Group Attributes.

For an Auto Attendant announcement, the system will play the announcement and monitor for digits from the connected external party. A CCR Audio/Text table defines a dialed digit (0-9, *, #) to route. Each digit can be set to one of the following paths:

- Extension
- Key Number Group
- Speed Dial number
- Paging zone
- Voice mail
- VPCM announcement

In addition, the system will monitor digits for an Extension number. If the user dials an Extension number, the system will route the call to the appropriate Extension.

When error case(Busy, No Answer, etc) happen While system wait dial digit of user, system can route the call to another destination by below options which can be set based on CCR table.

- Release
- Attendant
- Trunk Call Routing Table
- System Tone

Operation

iPECS Multi-button Authorized Extension

To record an Auto Attendant announcement:

1. Press the [PGM] button.
2. Dial 9 + 7 (the Message Record code).
3. Enter a VPCM slot number (0001~8000).
4. Enter an announcement number (01~70).

5. With Multi Language support, enter the language number (1~9), the current announcement is played followed by the “Press # to record” prompt.
6. Dial ‘#’.
7. After the beep-tone, record message.
8. Press the [SAVE] button to stop recording and save the message.

To delete a recording:

1. Press the [PGM] button.
2. Dial 9 + 7 (Message Record code).
3. Enter a VPCM slot number (0001~8000).
4. Enter an announcement number (01~70).
5. With Multi Language support, enter the language number (1~9), the current announcement is played followed by the “Press # to record” prompt.
6. Press the [SPEED] button, or press the [Delete] soft button during playback to erase the message.

Conditions

- There are no individual time limits on an Auto Attendant announcement.
- External callers will hear Ring back tone until the system answers the call or a VM greeting begins.
- Authorized Extensions must save a recording before hanging-up. Otherwise, the existing message will be used.
- The external caller may dial at any time during an Auto Attendant announcement and must dial prior to the expiration of the CCR Analysis timer.
- If the external caller dials an invalid selection or Extension, the system will return the ‘Invalid Entry’ message and allow re-entry using DISA retry count.
- If a caller dials more than a single digit, the call is routed based on the numbering plan.
- The calls answered by an Auto Attendant announcement (CCR) are interactive DISA calls and are subject to the conditions of DISA call.
- A CCR Announcement may be programmed to disconnect after playing.
- The Auto Attendant announcement (CCR) feature is supported for DISA and DID calls.
- Each VPCM provides up to 256 channels. Total recording time depends on the capacity of the storage. The V MIM (Voice Mail Interface Module) also can be used to support eight (8) channels and nine (9) hours of voice storage.
- CCR No answer error destination is applied after routing the call by CCR feature. That is, when CCR feature activate and user dial completely to route, but the call is not answered by anyone. In this case, CCR No answer option is applied.

Related WMS Menu

[To assign VPCM/VMIM to Zone]
Data Management > Zone Information > Zone Attribute > “Using Slot No”
[To assign Prompt Language of Extension]
Data Management > Zone Information > Channel Attribute > “Zone”
[To assign Announcement Table]
Data Management > System Feature Information > Announcement Table

[To assign CCR Table]
Data Management > System Feature Information > CCR Table
[To assign Extension Password]
Data Management > Extension Information > Number (DN) Information > DN Attribute > "Extension Password"
[To set Authorized User]
Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > "Privileged User"
[To assign feature codes for Voice Mail]
Data Management > Numbering Plan Information > Feature Code > "Voice Mail Access", "Announcement Table Access(CCR)", "Announcement Table Access & Release"

Related Features

- Key Number Group
- DISA (Direct Inward System Access)
- DID (Direct In Dial)

Hardware

- VPCM
- VMIM

2.10.3 Delay of DISA CCR Prompt Play

Description

When an incoming trunk through DISA line is activated, the announcement (CCR) can be played after sending answer.
To prevent caller party doesn't hear the first part of announcement. user can specify the delay time of starting to play CCR Prompt.

Operation

if the value of time delay is set, this feature is operated automatically.

Conditions

Related WMS Menu

[To set time dealy]
Data Management > Trunk Information > Incoming Route Information > Incoming Route Options > VM Play Delay Time

Related Features

DISA

Hardware

2.10.4 Built-In Voice Mail

2.10.4.1 Message Storage

Description

When an Extension activates Call Forward to the {Voice Mail Access} feature code, calls are routed to a VPCM Mailbox, where the call is handled by the iPECS Voice Mail application. The caller connects to the called Extension's Mailbox and receives the User greeting followed by a beep tone.

The remote caller can record a message, hang up the call, or dial the "*" key to hear other options. At disconnect, the VM application stores the message in the called user's Voice Mailbox and activates the Message Waiting indication at the user's Extension.

Operation

Remote Caller

To leave a voice message after Greeting and beep leave the message:

1. Hang up to terminate the call or dial "*' for further options.
2. If the caller dials "*', the option prompt ("dial 1 to Replay, 2 to Record, or # to End") is received.

Conditions

- Two timers are provided to control voice message length. The VM Msg Min Recording Time establishes the minimum voice message length. Voice messages shorter than this timer are not stored. The VM Msg Max Recording Time establishes the maximum voice message length. When the VM Msg Max Recording Time expires while recording a voice message, confirmation tone is heard and the message is saved for the destination Extension.
- If all VPCM channels are busy, Ring Back tone is provided until a VPCM channel becomes available.
- All Extensions including SLTs may leave or receive voice messages.
- Individual User Greetings and Voice Mails are protected from loss of AC power.
- In order to store messages for a DN that activates Call Forward to the VM, the system searches an idle gateway in a specific zone. The first member's zone number of the DN is used. If the DN has no member, the zone of the caller is used as an exception. Voice messages are stored in an idle gateway in the selected zone. If there is no free channel in all gateways of the selected zone, the caller will hear a Ring Back tone until a channel becomes idle.

Related WMS Menu

[To assign Using Slot No]
Data Management > Zone Information > Zone Attribute > "Using Slot No"
[To assign Zone]
Data Management > Zone Information > Channel Attribute > "Zone"
[To set Zone VM Msg Max Recording Time, VM Msg Min Recording Time]
Data Management > Tenant Information > Tenant Time Information > Tenant Timer > "VM Msg Max Recording Time (sec)"

> Tenant Information > Tenant Time Information > Tenant Timer > "VM Msg Min Recording Time (sec)"

[To assign Extension Password]

Data Management > Extension Information > Number (DN) Information > DN Attribute > "Extension Password"

[To assign Voice Mail Access Feature code]

Data Management > Numbering Plan Information > Feature Code > "Voice Mail Access"

Related Features

- Call Forward
- Message Wait
- VPCM Voice Mail
- Transfer, Voice Mail

Hardware

- VPCM
- VMIM

2.10.4.2 Message Retrieval

Description

Users can access a Voice Mailbox by dialing the {Voice Mail Access} feature code, pressing the [MSG/CALLBK] button on the Multi-button phone, or pressing a {Voice Mail Access } Flex button.

Prompts are then provided to guide the user in the Voice Mailbox operation. Users must enter the Authorization Code, the Extension number and the password, in response to the Request for Password ("Please enter your password code.") prompt.

If the user enters a valid and matching Mailbox and password number, the Number of Messages prompt ("You have xx new messages. You have yy saved messages.") is received. At this point, the user 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 the user responds 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 attach a memo to message, press seven, 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 the user dials 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 new message is played and the No Message prompt ("No Messages") is played.

In addition to the options indicated in the prompt, a user can 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 the user dials 9 in response to the Number of Messages prompt or during or at the end of a message the VM long Options prompt is played.

Operation

Multi-button Phone

To assign {Voice Mail Access} to a Flex button using the Extension User program:

- [PGM]+ {FLEX} + 2 + {Voice Mail Access} Feature Code+ [SAVE]

To retrieve voice mail:

1. Lift the handset or press the [Speaker] button.
2. Press the [MSG/CALLBK] button; a simple summary of the messages is shown as below.

MWI (03) VMS (05) SMS (08)
Select (MWI:1, VMS:2, SMS:3)

3. Press '2' to select VMS; the Enter the Authorization Code prompt is played.
4. Enter the Authorization code (DN and Password) and *; the Number of Messages prompt is played:
5. Enter 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 the {Voice Mail Access} button.
3. Enter the User Authorization Code and *; the Number of Messages prompt is played.
4. Enter 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/Old Message Option prompt, dial '7'.
2. At the beep, record the memo.
3. Dial "*" to stop recording and save the memo.
4. During or after the New/Old option prompt, dial four (4) to forward the message and memo.
5. Dial the destination number.

Single Line Telephone (SLT)

To retrieve voice mail:

1. Lift the handset.
2. Enter the {Voice Mail Access} feature code; the Enter the Authorization Code prompt is presented.

3. Enter the authorization (Extension number and Password) code and *; the Number of Messages prompt is presented:
4. Enter 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 save the memo.
4. During or after the New/Old option prompt, dial ‘4’ to forward the message and memo.
5. Dial the destination number.

Conditions

- If no new/old messages are available, pressing ‘1’ or ‘2’, is an invalid operation and the user receives the Invalid Entry prompt or No Message prompt.
- If the dialed number is not recognized, the Invalid Entry prompt is played. After the second invalid entry, the user is disconnected.
- The user may dial digits at any time during a message playback, system prompt or silence. The user must dial a digit in response to a system prompt within the CCR Analysis timer or the system will disconnect and return error tone.
- Messages can be retrieved in either a FIFO (First in First out) or LIFO (Last in First out) order to meet the desire of each user.

Related WMS Menu

[To assign Using Slot No]
Data Management > Zone Information > Zone Attribute > “Using Slot No”
[To assign Zone]
Data Management > Zone Information > Channel Attribute > “Zone”
[To assign Phone Flexible Button]
Data Management > Extension Information > Terminal Information > Phone Flexible Button
[To assign Extension Password]
Data Management > Extension Information > Number (DN) Information > DN Attribute > “Extension Password”
[To assign Voice Mail Access Feature code]
Data Management > Numbering Plan Information > Feature Code > “Voice Mail Access”

Related Features

- Message Retrieval Options
- Remote Message Retrieval
- Multiple Voice Mail

Hardware

- VPCM / VMIM

2.10.4.3 Remote Message Retrieval

Description

The system permits remote users access to their mailbox. After accessing the VPCM Voice Mail, operation follows the local procedures.

Operation

Remote user

To access a Voice Mailbox from a remote location:

1. Dial DISA line assigned for a VPCM Auto-Attendant.
2. Upon answer, dial {Voice Mail Access} feature code, the Request for Mail Box Number prompt will be presented.
3. Follow Message Retrieval procedures.

Conditions

Related WMS Menu

[To assign Using Slot No]
Data Management > Zone Information > Zone Attribute > “Using Slot No”
[To assign Zone]
Data Management > Zone Information > Channel Attribute > “Zone”
[To assign Extension Password]
Data Management > Extension Information > Number (DN) Information > DN Attribute > “Extension Password”
[To assign Voice Mail Access Feature code]
Data Management > Numbering Plan Information > Feature Code > “Voice Mail Access”

Related Features

- Message Retrieval Options
- Auto Attendant
- Message Retrieval
- Direct Inward System Access (DISA)

Hardware

- VPCM / VMIM

2.10.4.4 Message Retrieval Options

Description

The user may dial the digit ‘9’ to receive the VM long Options prompt while in the Voice Mailbox, including during or after a voice message or system prompt except when an option has been selected that requires user dialing. The later may occur when the user 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 Extension 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 VPCM Voice Mail will respond to incoming digits as shown in the following table

DIGIT	FUNCTION
1	New message play
11	Replay message
12	Listen to the next message
13	Delete message
14	Forward message
15	Call the sender
16	Skip message
17	Attach a memo to message
19	Return to main menu
2	Stored message play
21	Replay message
22	Listen to the next message
23	Delete message
24	Forward message
25	Call the sender
27	Attach a memo to message
29	Return to main menu
7	Call forward settings
8	Voice mailbox settings
81	Greeting set
82	Password set
89	Return to main menu
9	Message retrieval options
#	End
0	Attendant consol call

Operation

To access the message retrieval option:

1. At any time after the Number of Messages prompt, dial a Message Retrieval Option digit. The system initiates the selection providing any needed prompts.

Conditions

- The user 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 the user will receive error tone.
- If the call option is an external call, dialing restriction is applied based on the external COS.

- If the user remains off-hook after a call placed through the Voice Mail is complete, the user will be returned to the previous place in the Voice Mailbox. If the user hangs up, the VMIM/VSF will recall the user and, upon answer, the system will play “Request Mailbox Number” prompt.

Related WMS Menu

[To assign Using Slot No]
Data Management > Zone Information > Zone Attribute > “Using Slot No”
[To assign Zone]
Data Management > Zone Information > Channel Attribute > “Zone”
[To assign Extension Password]
Data Management > Extension Information > Number (DN) Information > DN Attribute > “Extension Password”
[To assign Voice Mail Access Feature code]
Data Management > Numbering Plan Information > Feature Code > “Voice Mail Access”

Related Features

- Mailbox Access-Message Retrieval
- Mailbox Access-Remote Message Retrieval
- Voice Mail Setup
- Call Back
- Dialing Restriction/COS

Hardware

- VPCM / VMIM

2.10.4.5 E-mail Notification

Description

With the VPCM, the system stores the voice message and can send a notification to the e-mail address associated with the Extension. The voice message can be attached to the e-mail as a wav file.

Operation

System

If configured, System automatically sends e-mail to notify User of new Voice Messages

Conditions

- Voice messages are stored in the VPCM, and may be transmitted with e-mails; settings determine if voice messages are attached to e-mails and if they are deleted automatically or stored after being sent.
- The e-mail is sent to the address assigned for the Extension with the “sender” address defined for the VPCM. Note the latter is required, as many e-mail servers will reject anonymous e-mails.
- E-mail addresses of the Extensions and VPCM are defined in WMS.

・If e-mail server require secure authentication; TLS, SSL supported.

Related WMS Menu

[To assign options for E-mail Notification]

Data Management > Extension Information > Number (DN) Information > Voice Mail Information > “SMTP Mail Server IP”
> Extension Information > Number (DN) Information > Voice Mail Information > “SMTP Mail Server Port Number”
> Extension Information > Number (DN) Information > Voice Mail Information > “SMTP User ID”
> Extension Information > Number (DN) Information > Voice Mail Information > “SMTP User Password”
> Extension Information > Number (DN) Information > Voice Mail Information > “Receiver Email Address”
> Extension Information > Number (DN) Information > Voice Mail Information > “VM Attach to E-mail Indication”
> Extension Information > Number (DN) Information > Voice Mail Information > “Delete VM after E-mail Attach”
> Extension Information > Number (DN) Information > Voice Mail Information > “SMTP Security”

Related Features

- VPCM Auto Attendant/Voice Mail

Hardware

- VPCM
- VMIM

2.10.4.6 Voice Mailbox Setting

Description

The user can program the Mailbox settings for their Mailbox including a security password and a greeting. When a user presses ’8’ while retrieving messages, the Mailbox Setting prompt, (“To edit your greeting, press one, to edit you password, press two. To return to main menu, press nine”) is played.

Operation

To program Mailbox settings while in the Voice Mailbox

1. Dial ‘8’, for Mailbox settings, the Mailbox Setting prompt is received.

To set password:

1. Dial ‘2’ and receive the Password Entry prompt (Please enter your new password and press pound when finished).
2. Enter new password.

3. Dial ‘*’ or ‘#’ and receive the Reenter Password prompt ("Please re-enter your password to confirm and press pound when finished.").
4. Enter new password again.
5. Dial ‘*’ or ‘#’ and receive Password Confirmation prompt ("Your password is saved.").

To set a greeting:

1. Dial ‘1’ and receive 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 greeting.

Or,

1. Dial ‘7’ and receive Record Greeting prompt, ("At the tone, record your new greeting, press # when done.").
2. After beep, record greeting speaking in a normal voice.
3. Dial ‘#’ and receive Greeting Confirmation prompt ("Your greeting is saved.").

To return to main menu:

1. Dial ‘9’; the Mailbox setting prompt (("To play New Messages, press 1. To play Saved Messages, press 2. To set Station Forwarding, press 7 (available only for remote access). To set Greeting or Password, press 8. To Disconnect, press #. Press 0 for the Operator. Press 9 to hear this message again.").

Conditions

- If the user is external, the user must begin dialing within the CCR Analysis time, if not the call is released.
- If the entered number is not recognized, the Invalid Entry prompt will be presented.
- The user must assign a password (up to 12 digits) before access to the Mailbox will be allowed.

Related WMS Menu

[To assign VPCM/VMIM to Zone]
Data Management > Zone Information > Zone Attribute > “Using Slot No”
[To assign Prompt Language for Extension]
Data Management > Zone Information > Channel Attribute > “Zone”
[To assign Extension Password]
Data Management > Extension Information > Number (DN) Information > DN Attribute > “Extension Password”
[To assign Voice Mail Access feature code]
Data Management > Numbering Plan Information > Feature Code > “Voice Mail Access”

Related Features

- Message Storage
- Message Retrieval
- Remote Message Retrieval
- Message Retrieval Option

Hardware

- VPCM

2.10.4.7 Call Forward from VPCM

Description

External users can activate or deactivate Call Forward for their Extension. Pressing ‘7’ while retrieving messages will return the Mailbox Set Forward prompt (“To forward a call to another Extension, press ‘1’. To cancel forwarding, press ‘2’. To return to the main menu, press ‘9’”).

Operation

To activate Call Forward while in the Voice Mail;

1. Dial ‘7’, for Mailbox set forward, the Mailbox Set Forward prompt is received.

To set call forward:

1. Dial ‘1 and receive the Password Entry prompt (“Please enter the number to forward to ...”).
2. Dial the desired number.
   - To forward to another Extension, dial the Extension number.
   - To forward Off-net, dial the Trunk access code and telephone number.
3. If the forward number is valid, confirmation tone is presented.

To deactivate call forward:

1. Dial ‘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 receives the Mail Box Settings prompt.

Conditions

- External users must begin dialing within the CCR analysis time, and finish dialing within the VPCM inter-digit timer. If not, the call is disconnected.
- This Mailbox Set Forward is only available for external users.

Related WMS Menu

[To assign VPCM/VMIM to Zone]
Data Management > Zone Information > Zone Attribute > “Using Slot No”
[To assign Prompt Language of Extension]
Data Management > Zone Information > Channel Attribute > “Zone”
[To assign Extension Password]
Data Management > Extension Information > Number (DN) Information > DN Attribute > “Extension Password”
[To assign Voice Mail Access feature code]
Data Management > Numbering Plan Information > Feature Code > “Voice Mail Access”

Related Features

- Message Storage
- Message Retrieval
- Remote Message Retrieval
- Message Retrieval Option

Hardware

- VPCM

2.10.4.8 User Answer Greeting

Description

With the VPCM, the system can play individual User Greeting when a call is answered.

Operation

To activate User Answer Greeting Play:

1. Lift handset or press the [Speaker] button.
2. Enter the {User Answer Greeting Play (On/Off)} feature code.
3. Dial “1”.

To deactivate User Answer Greeting Play:

1. Lift handset or press the [Speaker] button.
2. Enter the {User Answer Greeting Play (On/Off)} feature code.
3. Dial “0”

Conditions

· The individual User Greeting must be recorded.

Related WMS Menu

[To assign User Answer Greeting Play feature code]

Data Management > Numbering Plan Information > Feature Code > ”User Answer Greeting Play(On/Off)”

[To set User Answer Greeting Play]

Data Management > Extension Information > Number (DN) Information > DN Feature Registration > “User Answer Greeting”

Related Features

- VPCM Auto Attendant/Voice Mail

Hardware

- VPCM
- VMIM

2.10.4.9 DND Forward to Voice Mail

Description

This feature modifies the operation of calls presented to a station in the DND condition. The calls would be directed to the voice mail of the called party if this feature is enabled.

Operation

DND forward to voice mail is automatic if the feature is enabled:

1)

Conditions

Related WMS Menu

Data Management > Extension Information > number(DN) Information > Voice Mail Information> « VM SVC for DND »

Related Features

Hardware

2.10.4.10 Announce only mailbox

Description

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

Operation

Previous Menu option (CCR Flow)

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

Previous Menu option (CCR Flow)

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

Hang-up option

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

Conditions

Related WMS Menu

Data Management > Extension Information > number(DN) Information > Voice Mail Information> » Announce Only(Greeing Voice Mail) »
Data Management > Extension Information > number(DN) Information > Voice Mail Information> » Announce Only Option(Previoue Menu/Hang Up)»

Related Features

Hardware

2.10.4.11 Outcall Notification of Voicemail Message

Description

This feature provides a way to notify the arrival of voicemail messages to the specified telephone number. If the user enables the notification of mailbox and program a phone number including CO access code, system gives a call by dialing the programmed number and allows the access to voice mail after checking the user's password. If the user doesn't answer the call, system quits the call. But by programming, the notification can be retried after some time interval and retry count can be set also.

Operation

Called Party

To retrieve voice messages after receiving notification call:

1) When the called party answers the notification call, system will announce the prompt like "This is the voice mail system. There is a message for [recorded name] or [mailbox number (xxxxxxxx)]."
2) And then "Enter your password followed by pound" prompt will be heard.
3) If the called party enters the station number, its password and '#', system checks the validity of entered password.
4) In case the verification is successful, the called party will hear the main menu of voice mail and can access its own voice mail. After that, all the mailbox features will be available.
5) System will retry for 3 times before disconnecting the call.

Conditions

- Analog CO lines cannot be used for this feature since there is no explicit signal when the call is answered.
- Only CO party can receive the notification call. So, the notification phone number should start with a CO access code

Related WMS Menu

Data Management > Extension Information > number(DN) Information > Voice Mail Information > »OutCall Notification (Notify Arrival)»
Data Management > Extension Information > number(DN) Information > Voice Mail Information

> »OutCall Notification Retry Count»
Data Management > Extension Information > number(DN) Information > Voice Mail Information > »OutCall Notification Phone Number»

Related Feature

Hardware

2.10.4.12 Voice Mail Class Service

Description

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

Operation

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

1) Greeting Length (00 - 99 seconds): Maximum user greeting time can be set.
2) Message Length (001 - 600 seconds): Maximum recording time for each voice mail message can be set
3) Number of messages (001 - 250): Maximum number of voice mail messages
4) Retention time (01 - 99 days): Voice mail messages will be automatically deleted after this amount of days.

Conditions

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

Related WMS Menu

Data Management > Extension Information > number(DN) Information > Voice Mail Information> « VM COS »
Data Management > System Feature Information > Voice Mail Class Information

Related Feature

Hardware

2.10.5 Enhanced Voice Mail Feature

Description

iPECS-CM provides 2 kind of voice mail service(Basic VM feature and Enhanced VM feature). Enhanced VM feature is supported from system version 3.5.
When user access Voice Mail service using Enhanced VM, Main menu is strengthened as following.

[Main Menu]

1	Play new message
2	Play saved message
3	Play urgent message
4	Send message
5	Set Personal option
6	Administrator menu
7	Set Station Forwarding
8	Set Greeting or Password

If user want to use Enhanced VM feature, user should set VM service type in WMS

Operation

Conditions

Related WMS Menu

Data Management > Tenant Information > Tenant Basic Information > Tenant Attribute

Related Features

Hardware

2.10.5.1 Feature Code

The following features are added.

Feature Code	Feature
[146]	Company Directory Service
[147]	VM Name Recording(Company Directory Service)

2.10.5.2 Urgent Message

Description

When user want to leave a urgent message, user can assign message type as urgent. After recording the message, user select message type as urgent or normal.

If user want to send a message as normal, press “1”.
If user want to send a message as urgent , press “2”.
If user hang up the phone without assign message type, message type is normal.

When user access to Voice Mailbox , user can listen how many urgent messages are left and user can listen urgent message.

Operation

To send message

1) Set Extension A direct transfer to VM.
2) Call A and Listen Confirm tone to leave a message.
3) Record message to A.
4) Press “#”
5) Press “‘2” (Urgent delivery)

Conditions

Related WMS Menu

Data Management > Extension Information > number(DN) Information > Voice Mail Information> « Number of Message (Urgent) »

Related Feature

Hardware

2.10.5.3 Distribution List

Description

This allows a mailbox owner to setup a group of mailboxes and send/forward a message using one number instead of having to enter each mailbox individually
One user can have up to 5 distribution List , and A distribution List can have up to 25 members.
User can add , delete and listen distribution list personally using personal option menu.

Operation

To access distribution list menu:

1) Lift handset or press the **[SPEAKER]**.
2) Dial the {Voice Mail Access} Code
3) Dial mailbox (station) number.
4) Dial the mailbox(station) password
5) Press the ‘*’ key (password end mark)
6) In the main menu prompt, the user will hear a prompt like “For personal options, press five”.
   The new main menu for voicemail is like the following.
   “To play new messages, press one

To play saved messages, press two
To play urgent messages, press three
To send a message, press four
For personal options, press five
To access administrative options, press six
To set station forwarding, press seven
To set greeting or password, press eight
To disconnect, press pound
Press 0 for the operator
Press nine to hear this message again.”

7) User dials 5 to use distribution list feature.
8) The prompt will state “to edit a list press 1”
9) User dials ‘1’ at the distribution list menu.
10) After hearing "Enter list number [1-5]", user dials distribution list number.
11) System will prompt “to add a mailbox press 1, to delete a mailbox press 2, to listen to mailboxes in list, press 3”

To add a mailbox to a distribution list [1]:
1) At the prompt for distribution list editing menu, user dials ‘1’.
2) The prompt will state "Please enter the mailbox number"
3) User dials a mailbox number.
4) Then, “mailbox XXX added” will be heard.

To delete a mailbox to a distribution list [2]:
1) At the prompt for distribution list editing menu, user dials ‘2’.
2) The prompt will state "Please enter the mailbox number"
3) User dials a mailbox number.
4) Then, “mailbox XXX deleted” will be heard.

To list to mailboxes in a distribution list [3]:
1) At the prompt for distribution list editing menu, user dials ‘3’.
2) The prompt will state "mailbox XXX" for all the mailboxes in the list.

To send a message to the mailboxes in a distribution list
1) At the main menu prompt, user dials ‘4’.
2) “At the tone, please leave a message and to stop recording press pound key” prompt is heard.
3) User records a message to send and then dials ‘#’ to finish recording.
4) System announces “Enter the mailbox number or distribution list number followed by pound. To spell the name, press *” prompt.
5) If the user dials mailbox number or list number followed by pound, “For regular delivery, press one. To mark urgent, press two.” prompt will be heard.
   If the user dials ‘*’, user can select the destination with the same flow mentioned in company directory feature.

6) User dials '1' or '2' to select the delivery option and "Your message has been sent" prompt is heard.
7)

Then, user hears the main menu again.

Conditions

- A maximum of 5 distribution lists can be set up per mailbox.
- The maximum number of mailboxes in a distribution list is 25.

Related WMS Menu

Data Management > Extension Information > number(DN) Information > Voice Mail Information > VM Distribution List

Related Feature

Hardware

2.10.5.4 Company Directory

Description

This feature allows a caller to utilize the DTMF keys to "spell" the name of a subscriber and be directed to the selected extension. Company directory feature is available by using its feature code in feature numbering plan.
Internal stations can use this feature by dialing {Company Directory Service} feature code. And also external caller can dial this feature code in DISA or DID line.
To use this feature, first name and last name for company directory have to be preprogrammed and station's subscriber name has to be recorded in each station. Users can record the subscriber name by dialing {Record VM Name Recording(Company Directory Service) } feature code in feature numbering plan.

Operation

To record VM Subscriber Name

1) Dial the {Record VM Subscriber Name} feature code
2) Enter desired Station's number and password followed by '#'.
3) After hearing prompt to record the name, dial '#'.
4) Record subscriber name.
5) Dial '#' to finish recording.

To use Company Directory Feature

1) When internal stations or outside caller presses {Company Directory} feature code, the system will prompt "Press one to search by first name" and "Press two to search by last name".

2) If the user dials ‘1’, system will prompt “Enter the first 3 characters of the person’s first name”. Or if the user dials ‘2’, system will prompt “Enter the first 3 characters of the person’s last name”.

   For example, let’s say that the person’s first name is “JAMES”. If the caller wants to talk with Mr. James, the first three characters must be dialed. For ‘J’, 5 should be dialed, for ‘A’, 2 and for ‘M’ 6. So, the caller should dial 5 2 6. Please, refer to the following table.

Dial pad for Company Directory

1	2 A B C	3 D E F
4 G H I	5 J K L	6 M N O
7 P Q R S	8 T U V	9 W X Y Z
*	0	#

3) Once 3 characters have been entered, the system will check the entered characters for a match to the programmed name in VMIB Attribute: Company Directory First Name and Last Name. System will search an entry only among those who have recorded the subscriber name.
4) If only one match is found, the system will play “transferring to [programmed name]” and transfer the call to the station. Call will be transferred once the prompt is finished.
5) If more than one matches are found, the system will play:
   ” for [subscriber name], press 1”
   ” for [subscriber name], press 2”
   …
   ” for [subscriber name], press 9”
6) When the caller enters the desired digit (1-9), the system will route the caller to that station after announcing the prompt “transferring to [programmed name]”

Conditions

- The current field for station user name will remain as a user-settable option and will not affect the searchable name directory for this feature.
- The first name and last name fields will contain a maximum of 12 characters each.
- To be searched in company directory, subscriber name must be recorded. And the first and last name also should be programmed properly.

Related WMS Menu

Data Management > Extension Information > number(DN) Information > Voice Mail Information> »Company Directory(First Name) »
Data Management > Extension Information > number(DN) Information > Voice Mail Information> »Company Directory(Last Name) »
Data Management > Extension Information > number(DN) Information > Voice Mail Information> »Name (Company Directory) Recording»

•

Related Feature

Hardware

2.10.5.5 Administrator Mailbox

Description

This feature provides a mailbox that has administrative interface via telephone commands to perform common tasks associated with the VM. Administrator can add mailbox, delete mailbox, reset VM password.
Also, administrator can record user greeting or VM Name instead of user.

Operation

To access administrative mailbox menu:

1) Lift handset or press the **[SPEAKER]**.
2) Dial the {VMIB Access} Code
3) Dial mailbox (station) number.
4) Dial the mailbox(station) password
5) Press the '*****' key (password end mark)
6) In the main menu prompt, the user will hear a prompt like "To access administrative options, press six".
   The new main menu for voicemail is like the following.
   "To play new messages, press one
   To play saved messages, press two
   To play urgent messages, press three
   To send a message, press four
   For personal options, press five
   To access administrative options, press six
   To set station forwarding, press seven
   To set greeting or password, press eight
   To disconnect, press pound
   Press 0 for the operator
   Press nine to hear this message again."
7) User dials 6 to use administrative mailbox features.
8) "Administrative main menu" prompt will be heard.
   The following is the administrative main menu.

"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"

To add a mailbox from mailbox administrator [1]:
1) User dials '1' at the administrative main menu.
2) After hearing "Please enter the mailbox number", user dials mailbox number.
3) When "Enter COS 1-5" is heard, user dials one-digit class of service for voice mail.
4) Then, "Press 1 to confirm or # to cancel and go back to administration main" is heard.
5) If '1' is dialed, the VM access is set ON.

To delete a mailbox from mailbox administrator [2]:
1) User dials '2' at the administrative main menu.
2) After hearing "Please enter the mailbox number", user dials mailbox number.
3) Then, "Press 1 to confirm or # to cancel and go back to administration main" is heard.
4) If '1' is dialed, the VM access is set OFF.

To reset a mailbox from mailbox administrator [3]:
1) User dials '3' at the administrative main menu.
2) After hearing "Please enter the mailbox number", user dials mailbox number.
3) Then, "Press 1 to confirm or # to cancel and go back to administration main" is heard.
4) If '1' is dialed, the password of the mailbox (DN's password) gets empty.

To record a mailbox greeting from mailbox administrator [4]:
1) User dials '4' at the administrative main menu.
2) After hearing "Please enter the mailbox number", user dials mailbox number.
3) Then, "To listen press 1, to record press 2, to delete press 3" and "Press # to cancel and go back to administration main" prompts are heard.
4) According to the user input, the relevant feature will be performed.

To record a broadcast message from mailbox administrator [5]:
1) User dials '5' at the administrative main menu.
2) After hearing "Enter broadcast message number" (01-10), user dials message number.
3) Then, "To listen press 1, to record press 2, to delete press 3, to send press 4" and "Press # to cancel and go back to administration main" prompts are heard.
4) According to the user input, the relevant feature will be performed.

To record a mailbox name from mailbox administrator [6]:
1) User dials '6' at the administrative main menu.
2) After hearing "Please enter the mailbox number", user dials mailbox number.
3) Then, "To listen press 1, to record press 2, to delete press 3" and "Press # to cancel and go back to administration main" prompts are heard.

4) According to the user input, the relevant feature will be performed.

Conditions

・There are no limits on the number of mailboxes that can be marked as administrators.

Related WMS Menu

Data Management > Extension Information > number(DN) Information > Voice Mail Information > »VM Administrator(Administrator) »

Related Feature

Hardware

2.10.6 System Information Announcement

Description

The system provides several general Voice Memos such as system time and data, and Extension number, as well as, other settings over the iPECS Multi-button phone speaker or SLT handset.

Operation

Multi-button Phone

To hear date/time prompt:

1. Enter the {Information voice announce} feature code + ‘1’. The date and time is heard.

To hear the Extension number prompt:

1. Enter the {Information Voice Announce} feature code + ‘2’. The Extension number announcement for Extension is heard, “This is station 150”.

To hear the Extension settings:

1. Enter the {Information voice announce} feature code + ‘3’; Status for Extension is reported as follows:
   - Station ICM Mode (Hands free/Tone/Privacy)
   - Station IP Address
   - Station Mac Address
   - Listed message x (x: number of all message waiting)
   - Wake-Up Time (hh:mm)
   - Do not disturb
   - Forwarded to xxxxxxxxx
   - Forwarded to VM
   - Queued Trunk xx
   - Locked (temporary COS change)
   - COS x

Single Line Telephone (SLT)

To hear date/time prompt:

1. Lift the handset.
2. Enter the {Information Voice Announce} feature code + ‘1’, the announcement for date and time is heard, “Date is May 2nd. Time is xx:xx pm”.

To hear the Extension number prompt:
1. Lift the handset.
2. Enter the {Information Voice Announce} feature code + ‘2’; The Extension number announcement for Extension is heard, “This is station 150”.

To hear the Extension settings:
1. Lift the handset.
2. Enter the {Information Voice Announce} feature code + ‘3’; Status for Extension is reported as follows:
    - Station number
    - Station ICM Mode (Hands free/Tone/Privacy)
    - Station IP Address
    - Station Mac Address
    - Listed message x (the number of all message waiting)
    - Wake-Up Time (hh:mm)
    - Do not disturb
    - Forwarded to xxxxxxxxx
    - Forwarded to VM
    - Queued Trunk xx
    - Locked (temporary COS change)
    - COS x

Conditions

- In Station Settings, the entries (from Message X to COS X) will only be announced if they are enabled to do so.

Related WMS Menu

[To assign VPCM/VMIM to Zone]
Data Management > Zone Information > Zone Attribute > “Using Slot No”
[To assign Prompt Language for Extension]
Data Management > Zone Information > Channel Attribute > “Zone”
[To assign Extension Password]
Data Management > Extension Information > Number (DN) Information > DN Attribute > “Extension Password”
[To assign Information Voice Announce feature code]
Data Management > Numbering Plan Information > Feature Code > “Information Voice Announce”

Related Features

Hardware

- VPCM

2.10.7 System Tone/Prompt/Announce Message Retrieval

Description

Authorized users can retrieve prompts or announcements stored in VPCM.

Operation

Authorized User, Multi-button Phone

To listen to prompts:

1. Press the [PGM] button and dial ‘9’ + ‘6’, or enter the {VPCM Prompt/Announcement Listen} feature code.
2. Dial ‘1’ to select a prompt.
3. Enter a VPCM slot number (0001~8000).
4. Enter a prompt number (001~255).
5. Enter a language (1~9).
6. The system will play from the selected prompt to the last prompt in order, and the prompt number is displayed on the LCD (ex., 50 entered would present 50-255 in order).
7. Dial ‘2’ to pause prompt playback, and dial the 3-digit prompt number to resume the prompt.
8. Dial ‘1’ to play the previous prompt while a prompt is being played, or dial ‘3’ to play the next prompt.
9. To finish prompt playing, press the [SAVE] button.

To listen to announcements:

1. Press the [PGM] button and dial ‘9’ + ‘6’, or enter the {VPCM Prompt/Announcement Listen} feature code.
2. Dial ‘2’ to select an announcement.
3. Enter a VPCM slot number (0001~8000).
4. Enter a message number (01~70).
5. Enter a language (1~9).
6. The system will play from the selected message to the last message in order, and the message number is displayed on the LCD (ex., 50=50-70).
7. Dial ‘2’ to pause during playback, and dial the 2-digit message number to resume the message.
8. Dial ‘1’ to play the previous message while a message is being played, or dial ‘3’ to play the next message.
9. To finish playing an announcement, press the [SAVE] button.

To listen to System Tones:

1. Lift the handset or press the [SPEAKER] button.
2. Enter a {System Tone Listen} feature code.
3. Enter a System Tone ID, which is the index of Tenant System Tone on WMS.

Authorized User, Single Line Telephone (SLT)

To listen to prompts:

1. Lift handset.
2. Enter the {VPCM Prompt/Announcement Listen} feature code.
3. Dial ‘1’ to select a prompt.

4. Enter a VPCM slot number (0001~8000).
5. Enter a prompt number (001~255).
6. Enter a language (1~9).
7. The system will play from the selected prompt to the last prompt in order, and the prompt number is displayed on the LCD (ex., 50 entered would present 50-255 in order).
8. Dial ‘2’ to pause prompt playback, and dial the 3-digit prompt number to resume the prompt.
9. Dial ‘1’ to play the previous prompt while a prompt is being played, or dial ‘3’ to play the next prompt.

To listen to announcements:

1. Lift handset.
2. Enter the {VPCM Prompt/Announcement Listen} feature code.
3. Dial ‘2’ to select an announcement.
4. Enter a VPCM slot number (0001~8000).
5. Enter a message number (01~70).
6. Enter a language (1~9);
7. The system will play from the selected message to the last message in order, and the message number is displayed on the LCD (ex., 50=50-70).
8. Dial ‘2’ to pause during playback, and dial the 2-digit message number to resume the message.
9. Dial ‘1’ to play the previous message while a message is being played, or dial ‘3’ to play the next message.

Conditions

- If no prompt or message is stored, the system will pause playback, and then play the next prompt or message.
- When the user stops playback and enters a new prompt or message number, the system will stop the function if no input is made during the Extension Button Input Timer.

Related WMS Menu

[To set Extension Button Input Time]
Data Management > Tenant Information > Tenant Time Information > Tenant Timer > “Extension Button Input Time”
[To set Prompt Language Configuration]
System Management > Prompt Language Configuration
[To assign VPCM Prompt/Announcement Listen feature code]
Data Management > Numbering Plan Information > Feature Code > ”VPCM Prompt/Announcement Listen”
[To set Authorized User]
Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > ”Privileged User”

Related Features

Hardware

- VPCM

2.10.8 Delete All VM Message

Description

This feature allows users to delete all VM messages of the selected Extension.

Operation

To delete all VM messages of the selected DN:

1. Lift the handset.
2. Enter the {Delete All Voice Mail} feature code.
3. The prompt for User Authorization Code input will be provided.
4. Enter the Extension number.
5. Enter the password (up to 12 digits).
6. Dial '*'.
7. Dial '1' to delete all or dial '2' to cancel.
8. When all messages are deleted, a confirmation tone will be presented.

Conditions

・If you stop deleting messages during the deletion process, the remaining messages can be retrieved and played.

Related WMS Menu

[To assign Delete All Voice Mail feature code]
Data Management > Numbering Plan Information > Feature Code > "Delete All Voice Mail"

Related Features

- Message Storage
- Message Retrieval

Hardware

- VPCM

2.10.9 Voice Mail Back-up Extension

Description

With the VPCM, an iPECS soft phone (Phontage and UCS Client) user receives notification of new messages for assigned Extensions. The soft phone will indicate the total messages for the assigned Extensions and the soft phone. The soft phone user can download the messages for other Extensions 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 VPCM memory.

Operation

Phontage or UCS Client

Refer to the Phontage or UCS Client Guide

Conditions

- Voice messages are stored in and deleted from VPCM; deleting messages from the soft phone hard drive does not delete the message from the VPCM memory.

Related WMS Menu

[To set options for VM backup]

Data Management	> Extension Information > Number(DN) Information > Voice Mail Information > “Phone for Internal VM backup”
	> Extension Information > Number(DN) Information > Voice Mail Information > “Delete Backuped VM to Phontage”

Related Features

Hardware

- VPCM

2.11 Busy Lamp Field (BLF)

Description

When a Flex button on an iPECS Multi-button phone or DSS Console is assigned as a {DN} or {Extension DSS} button, it also serves as a status LED. The LED indicates the status of the Extension based on the configuration of the system.

The chart below indicates the states that can be indicated via a DN or DSS button.

DN Button	Extension DSS Button
DN in use at user phone	
DN in use at other phone	Busy
DND	DND
Incoming call	Incoming call
On hold	On hold
Call Forward	Call Forward
Conference by user	Preset message set
Conference by others	Phone in failure (Station lock out)
Conference as a supervisor	

Operation

iPECS Multi-button Phone

To assign a BLF button

1. Press the [PGM] button.
2. Press the desired [FLEX] button.
3. Dial 2.
4. Dial DN.
5. Press the [HOLD/SAVE] button.

Conditions

- If the Attendant Number is assigned to DSS button, the LED does not indicate the status of the Attendant.

Related WMS Menu

[To assign Phone Flexible Button]
Data Management > Extension Information > Terminal Information > Phone Flexible Button
[To assign DSS Console Connection]
Data Management > Extension Information > Terminal Information > DSS Console Connection
[To assign DSS Console Flexible Button]
Data Management > Extension Information > Terminal Information > DSS Console Button
[To set LED color & flash rate for buttons]
Data Management > Tenant Information > Tenant Basic Information > Phone LED Control

Related Features

- Internal Call
- Extension User Program Code

Hardware

- iPECS Multi-button Phone

2.12 Busy One Digit Service

Description

When a caller receives Busy tone, the caller can select a busy feature service, such as Camp-On, Call-Wait, Off-Hook Voice Over, Intrusion, Pilot Hunt, Call Back and Call Snatch. The system can be configured to provide access to these or other services with a single digit.

Operation

To use One Digit Service, while receiving a busy tone:

1. Enter a digit defined for Busy One Digit Service

To use a busy service feature with feature code while receiving a busy tone:

1. Press the [TRANS] button, or, for an SLT, press and release the hook-switch (hook-flash).
2. Enter the feature code for the desired busy feature.

Conditions

- If Step Call is used, Busy One Digit Service does not operate, because the input digit is used to direct the step call.

Related WMS Menu

[To set One Digit Service on Busy]

Data Management > Tenant Information > Tenant Basic Information > One Digit Service on Busy

Related Features

- Camp On
- Call Wait
- Call Back
- Call Intrusion
- Pilot Hunt Group
- Off Hook Voice Over(OHVO)
- Call Snatch

Hardware

2.13 Call Admission Control (CAC)

Description

This function restricts the number of Trunk channels for outgoing purposes.

Operation

Conditions

・The number of channels available for a Trunk route can be restricted.

Related WMS Menu

[To set options for CAC]
Data Management > Trunk Information > Outgoing Route Information > CAC (Call Admission Control)

Related Features

Hardware

2.14 Call Back

2.14.1 Station Call Back

Description

When the called Extension is busy, the caller may request to be placed in queue to receive a call back when the called Extension returns to idle. When the called Extension returns to idle, the system sends the Call Back Alert Ring to the caller. When the caller answers the Call Back Alert Ring, the previously busy Extension is called.

Operation

Multi-button Phone

To leave a Call Back while receiving a busy tone:

1. Press the [MSG/CALLBK] button, the user receives confirmation tone.
2. Hang up the call.

To respond to a Call Back recall when the busy Extension is available

1. Lift the handset or press the [Speaker] button.
2. Previously busy Extension is called.

Single Line Telephone (SLT)

To leave a Call Back while receiving busy tone:

1. Press and release the hook-switch (hook-flash).
2. Enter the {Extension Call Back/Trunk Queuing} feature code.
3. Hang up.

To respond to a Call Back recall when the busy Extension is available:

1. Lift the handset.
2. Previously busy Extension is called.

Conditions

・Extensions can leave only one call back request.

Related WMS Menu

[To assign feature codes for Call Back, Queuing]

Data Management	> Numbering Plan Information > Feature Code > " Extension Call Back/Trunk Queuing"
	> Numbering Plan Information > Feature Code > “Extension Call Back/Trunk Queuing Cancel”

Related Features

- Busy One Digit Service

Hardware

2.14.2 Trunk Queuing

Description

When Trunks are busy, users may request to be placed in queue awaiting the Trunk to become available. When the Trunk becomes available, the system calls the waiting Extension.

Operation

iPECS Multi-button phone

To request queuing for a busy trunk:

1. Enter the Trunk Access code or press the programmed Trunk Access Code [FLEX] button.
2. Press the [MSG] or [CALLBK] button, and then hear the confirmation tone.
3. Hang up; the [MSG] or [CALLBK] LED will flash.

To cancel queuing on a busy trunk:
1. Press the [MSG] or [CALLBK] button, the [MSG] or [CALLBK] LED will extinguish.

<u>Single Line Telephone (SLT)</u>
To request queuing on a busy trunk:
1. Press and release the hook-switch (hook-flash).
2. Enter the {Extension Call Back/Trunk Queuing} feature code.

To cancel queuing on a busy trunk:
1. Lift the handset.
2. Enter the {Extension Call Back/Trunk Queuing Cancel} feature code.

<u>System</u>
When the trunk line becomes available:
1. The system provides a Queue recall to the Extension with oldest queue, the Trunk and Extension are indicated as busy for all other users.

Conditions
- A Trunk Access code can have any number of simultaneous queue requests.
- An Extension may only have a single active Trunk queue request; if an Extension requests a new Trunk queue, the existing one is cancelled.
- If the Extension is idle and Trunk becomes available, the Extension will receive Queue recall. If the Extension fails to respond to the Queue recall, the system will cancel the queue.

Related WMS Menu
[To use Trunk Queuing]
Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > “Trunk Queuing”
[To assign Queuing feature code]
Data Management > Numbering Plan Information > Feature Code > “Extension Call Back/Trunk Queuing”
> Numbering Plan Information > Feature Code > “Extension Call Back/Trunk Queuing Cancel”

Related Features
- Trunk Route Groups

Hardware

2.15 Call Duration Restriction (CDR)

Description

The system can be programmed to restrict the call duration for each Extension. If the CDR timer expires while a user is on an internal or Trunk call, the system will terminate the call, and the user will hear error tone. An audible alert tone to notify of call termination will be presented to the both party by a preprogrammed time before the CDR timer expires. When the CDR timer expires, the system releases the call, and returns the Trunk to idle.

Operation

Conditions

・ Trunk CDR operates for a forwarded call.

Related WMS Menu

Data Management	> Tenant Information > Tenant Time Information > Tenant CDR Time Information > “Restriction” > Tenant Information > Tenant Time Information > Tenant CDR Time Information > “Service after Call Restriction Time” > Tenant Information > Tenant Time Information > Tenant CDR Time Information > “CDR Tone Play Interval” > Tenant Information > Tenant Time Information > Tenant CDR Time Information > “Warning Tone Send Time before Release” > Tenant Information > Tenant Time Information > Tenant CDR Time Information > ”Call Restriction Time” > Tenant Information > Tenant Basic Information > Tenant Local/DDD/ISD Prefix > Extension Information > Terminal Information > Terminal Option > “CDR Usage”

Related Features

Hardware

2.16 Call Forward

Description

Users can select incoming calls to be rerouted automatically to another Extension, a Key Number Group, or an outside telephone number according to Extension status.

Call Forward Options;

- Code 0 (Remote Call Forward, Follow-me Forward) – Users can register unconditional call forwarding for their Extension from another phone after entering the authorization code.

- Code 1 (Unconditional Call Forward) – All incoming calls except recalls forward regardless of the Extension status.
- Code 2 (Busy Call Forward) – All Incoming calls forward if the Extension is busy.
- Code 3 (No Answer Call Forward) – All incoming calls not answered by the Extension forward after the no answer timer is expired.
- Code 4 (Busy/No Answer Call Forward) - Calls forward if the Extension is busy or does not answer the calls.
- Code # (Call Forward Cancel) – the registered Call Forward is canceled.

Operation

iPECS Multi-button Phone

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

1. Lift the handset or press the [Speaker] button.
2. Press the [DND] button.
3. Enter a Call Forward Option Code (range: 1~4)
4. Enter the destination number or outside telephone number including trunk access code.
5. Dial * or press [HOLD/SAVE] button to save. If it is allowed, DND LED will be flashed and confirmation tone will be provided

To activate Call Forward from remote Extension:

1. Lift the handset or press the [Speaker] button.
2. Press the [DND] button.
3. Dial 0 (Call Forward Option code).
4. Enter the User Authorization Code (Extension number + password) of the remote Extension.
5. Dial *.
6. Enter the destination number. The [DND] button will be flashing and a confirmation tone will be provided if allowed.)

To deactivate Call Forward;

1. Lift the handset or press the [Speaker] button.
2. Press the [DND] button.
3. Dial # (Call Forward Cancel option). If it is allowed, the [DND] button LED will be off and conformation tone will be provided.

Single Line Telephone (SLT)

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

1. Lift the handset.
2. Enter the {Forward Register (Normal)} feature code.
3. Enter a Call Forward Option Code (range: 1~4).
4. Enter the destination number.
5. Dial *. If it is allowed, confirmation tone will be provided.

To activate Call Forward from remote Extension:

1. Lift the handset.

2. Enter the {Forward Register (Normal)} feature code, and dial 0 (Remote Call Forward Option Code).
   Or
   Enter the {Remote Forward Register} feature code.
3. Enter the User Authorization code of the remote Extension (Extension number + password); confirmation tone will be heard.
4. Enter the destination number.
5. Dial *.
6. If successful, confirmation tone is provided

To cancel Call Forward:
1. Lift the handset
2. Enter the {Forward Register (Normal)} feature code, and dial # (Call Forward Cancel Option Code).
   Or
   Enter {Forward Cancel} feature code.
   Or
   Enter {All Feature Cancel} feature code. If it is allowed, DND LED will be off and confirmation tone will be provided.

Conditions

- Call Forwarding can be registered through WMS as well as by the user at the phone.
- A forwarded call can be transferred to the forwarding Extension.
- The Call Forward feature may be enabled or disabled by system programming.
- Call forward, DND and predefined text display operate with the same button, they cannot be active simultaneously.
- Call Forward activated by a user has higher priority than Preset Call Forward set by system programming.
- If a Trunk call forwards Off-net, and the Speed Dial used for Off-net Call Forward contains a flash, only the digits before the flash are dialed.
- Chained Call Forward registration (1000 to 1001, and 1001 to 1002, and 1002 to 1003, and 1003 to XXXX, and XXXX to 1000) is allowed.
  - A call to 1000, will route to XXXX.
  - Even though telephone numbers in the Call Forward registration chain are not limited, the maximum Call Forward nodes may be limited through WMS. The default Call Forward node value is 3.
- If Preset Call Forward is set up in a Call Forward chain, and the chain includes an Extension in DND, calls forward to the next Extension, skipping this DND Extension. If the DND Extension is the last element of this chain, the previous node becomes the final destination of Call Forward.
- Station Ring Lock time is used as the incoming call No Answer timer.
- The Call Forward destination number cannot start with * or #, because they are used during the normal activation of Call Forward.
- The Call Forward destination number cannot start with #, because they are used during the normal activation of Call Forward.

- The Call Forward destination number can start with *. But After the first digit, it cannot be used because they are used as the normal activation of Call Forward.

Related WMS Menu

[To register Call Forward]
Data Management > Extension Information > Number (DN) Information > DN Feature Registration > "Call Forward Type" and "Call Forward Destination"
[To assign Call Forward (Normal) feature code]
Data Management > Numbering Plan Information > Feature Code > "Forward Register (Normal)"
[To set Max. count of chained Call Forward]
Data Management > Tenant Information > Tenant Basic Information > Tenant Attributer > "Call Forward Chain Allow Step"
[To set time for No Answer Call Forward]
Data Management > Tenant Information > Tenant Time Information > Tenant Timer > "No Answer Forward Time"

Related Features

- User Authorization Code (Password)
- Preset Call Forward

Hardware

2.17 Call Hold

Description

Users can put internal or Trunk calls on hold, the user is released from the connection and audio to the caller is provided from a defined hold source.

2.17.1 Call Hold

Description

During conversation, the iPECS Multi-button phone user can place the call on hold. The held user will hear the music on hold. And the station which pushed the hold button will hear Hold Service Set Tone. If left on hold, the call recalls after expiration of the Normal Hold Tone (Extension or Trunk) timer.

Operation

To put an intercom call on hold

1. Press the [HOLD] button. The [DN] button LED flickers.

To retrieve the held the intercom call

2. Press the flashing [DN] button, the intercom call is connected.

Related WMS Menu

[To set Normal Hold Tone (Station or Trunk)]

Data management > Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > “Normal Hold Tone(Station or Trunk)”

[To set Hold Service Set Tone]

Data management > Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > “Hold Service Set Tone”

[To enable Automatic Call Hold by other DN Button]

Data Management > Extension Information > Terminal Information > Terminal Option > "Automatic Call Hold by other DN Button"

Related Features

- Music-On-Hold
- Directory Number (DN)
- Hold Recall

Hardware

- iPECS Multi-button Phone

2.17.2 Hold Recall

Description

When a user places an internal or Trunk call on hold, the Hold Tone Timer is activated. When the timer expires, the held call recalls the Extension. If the Extension does not answer the call, the held call is disconnected, and the Trunk returns to idle.

Operation

Conditions

- Different timers for the various types of hold such as Normal Hold, Transfer Hold, Call Park Hold, Call Wait Hold, and Camp On Hold are assigned.

Related WMS Menu

[To set Hold Indication LED]

Data Management > Tenant Information > Tenant Basic Information > Phone LED Control

[To set Hold Tone and Recall Time]

Data Management > Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > “Normal Hold Tone”

[To set Recall Ring]

Data Management > Tenant Information > Tenant Tone/Ring Information > Tenant System Ring

Related Features

- Call Hold
- Call Transfer
- Call Park
- Call Wait
- Camp On

Hardware

2.17.3 System Hold

Description

A user can place a DN (Directory Number) connected in an Extension or Trunk call on hold. The button LED of held DN blinks. Other users can answer the call on hold by pressing the blinking DN button on their phone.

Operation

iPECS Multi-button phone

To place a call on hold:

1. Press the [HOLD] button.

To answer the call on hold:

1. Press the held DN button.

Single Line Telephone (SLT)

To place a call on hold:

1. Press and release the hook-switch, hook-flash.
2. Enter the {Call Hold(Register)} feature code.

To answer the call on hold:

1. Lift the handset.

Conditions

- When a DN is on hold, the LED of DN button blinks as defined in Phone LED Control.
- Extension users can answer the held call by pressing the [DN] button.
- For SLTs, system hold may be activated for the call on the Prime DN only.
- System hold cannot be activated for a DN participating in a conference.

Related WMS Menu

[To set System Hold LED]
Data Management > Tenant Information > Tenant Basic Information> Phone LED Control
[To assign Hold register Feature code]
Data Management > Numbering Plan Information > Feature Code > “Call Hold (Register)”

Related Features

- Directory Number(DN)

Hardware

2.17.4 Automatic Hold

Description

While busy on a call, the user may select another DN or S-DN (Sub-Directory Number) to place or receive a call. In this case, the original call is placed on hold automatically.

Operation

To use the automatic hold function

1. While on a call, press the desired S-DN button, to answer or place a call; the current call is automatically placed on hold.

Conditions

・The number of calls that can be put on hold is equivalent to the number of the S-DN buttons.
・Automatic Call Hold must be enabled.

Related WMS Menu

[To set Automatic Hold by other DN Button]
Data Management > Extension Information > Terminal Information > Terminal Option > "Automatic Call Hold by other DN Button"

Related Features

- Directory Number(DN)
- Call Hold

Hardware

- iPECS Multi-button Phone

2.18 Call Intrusion

Description

A user can intrude on the call of another busy Extension, establishing a three-party conference. The intruded Extension user will hear Intrusion Alarm tone.

Operation

To intrude with Busy One Digit Service while receiving busy tone:

1. Enter a the digit defined for Busy One Digit Service {Intrude Request}

To intrude with feature code while receiving a busy tone from an iPECS Multi-button phone:

1. Press the [TRANS] button, or press and release the hook-switch (hook-flash).
2. Enter the {Intrude Request} feature code.

Conditions

- Call Intrusion uses the conference feature, so the VPCM resource is needed.
- Call Intrusion is not available when the called Extension is using Privacy.
- If Call Intrusion is requested when the called Extension is dialing, the called Extension can be connected to the caller.

Related WMS Menu

[To assign Intrude Request Feature code]
Data Management > Numbering Plan Information > Feature Code > “Intrude Request”
[To set One Digit Service On Busy]
Data Management > Tenant Information > One Digit Service On Busy > “Intrusion”
[To enable Intrusion]
Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > “Intrusion”

Related Features

- Busy One Digit Service
- Call Snatch
- Conference

Hardware

- VPCM

2.19 Call Log

Description

An iPECS Multi-button phone with soft button allows users to view a log of incoming, outgoing, busy and missed calls in the display.

Operation

1. Press the {LOG} soft button, a LOG(M) soft button displays if there are missed calls.
2. Using Volume Up/Down button or [Navigation] button, select desired call logs (Incoming, Outgoing, Missed or Busy). Each call log will be displayed as below.
   ▶: Outgoing Call
   ◀: Incoming Call.
   M: Missed Call
   B: Incoming Call on Busy
   X: Call from unknown telephone number
   F: Forwarded Call
   D: Call in DND
   P: Call picked up by another phone.

3. Call Logs can be handled by user with following sub-soft button menus.
   {SEND}: Call to the selected number.
   {SELECT}: Show the date and time of the call log.
   {SAVE}: Save the call log as Station Speed dial.
   {DEL_SEL}: Delete the selected call log.
   {DEL_ALL}: Delete all call logs.

To assign {Call Log List Display} to a Flex button using the Extension User program:
- [PGM] + {FLEX} + 2 (program type for number) + {Call Log List Display} feature code + [SAVE]

Conditions
- Up to 100 entries can be displayed in a call log for each Extension.
- Call Logs are not recorded in permanent memory, if power is lost, the Call log is lost.
- Call Logs are maintained when Active and Standby Call Server operation is switched, but the call logs will be deleted if both the Active and Standby call servers restart simultaneously.
- Phontage, UC client, WIT-300 have separate Call logs, so this feature is not supported.
- If the Catch Call Log Save Option is Not Use, only Missed, Incoming, Outgoing call information is saved.

Related WMS Menu
[To set Call Log access protect]
Data Management > Extension Information > Number(DN) Information > DN Feature Registration > "Call Log Access Protect"
[To use Catch Call Log Save Option]
Data Management > Extension Information > Terminal Information > Terminal Option > "Catch Call Log Save Option"

Related Features

Hardware
- iPECS Multi-button Phone with soft button

2.20 Call Park

Description
A call can be placed on hold so that other users can access the call. The call is placed in a designated Call Park 'Orbit' using a three (3) digit identifier (000 - 999) in order to be answered by another Extension including an SLT. Call Park is often used with paging, the call is parked and then the desired party is paged to answer the parked call.

Operation
iPECS Multi-button Phone
To park a call:
1. In conversation, press the [TRANS] button.

2. Enter the {Call Park (Register/Answer)} feature code.
3. Enter the 3-digit Call Park Identifier (000~999); if it is accepted, confirmation tone is provided and the Extension returns to the idle.

To retrieve a parked call:
1. Lift the handset or press the [Speaker] button,
2. Enter the {Call Park (Register/Answer)} feature code.
3. Enter a 3-digit Call Park Identifier (000~999).

Single Line Telephone (SLT)
To park a call:
1. In conversation, press and release the hook-switch (hook-flash).
2. Enter the {Call Park (Register/Answer)} feature code.
3. Enter the 3-digit Call Park Identifier (000~999); confirmation tone is provided.

To retrieve a parked call:
1. Lift the handset.
2. Enter the {Call Park (Register/Answer)} feature code.
3. Enter the 3-digit Call Park Identifier (000~999).

Conditions
- If the Call Park timer expires, the call recalls following Call Hold Recall process.
- If a button which is assigned as {Call Park (Register/Answer)} + Call Park Identifier is exist, the LED will indicate status of Call Park Identifier.

Related WMS Menu
[To use Call Park]
Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > “Call Park”
[To assign Call Park Feature code]
Data Management > Numbering Plan Information > Feature Code > “Call Park (Register/Answer)”

Related Features
- Hold/Hold Recall

Hardware

2.21 Call Pick-up

2.21.1 Direct Call Pick-up

Description
Except for Call Backs, calls ringing at an Extension can be picked-up by other Extensions.

Operation

To pick up a call ringing at another Extension:

1. Lift the handset.
2. Enter the {Call Pick-Up (Direct)} feature code.
3. Enter the ringing Extension number.

Conditions

- Call Backs are not subject to call picked up.

Related WMS Menu

[To assign Pick-up (Direct) feature code]
Data Management > Numbering Plan Information > Feature Code > “Call Pick-Up (Direct)”

Related Features

- Group Call Pick-up

Hardware

2.21.2 Group Call Pick-up

Description

All calls ringing at an Extension can be picked up by another Extension belonging to the same Pick-up Group

Operation

iPECS Multi-button Phone

To answer a call to another Extension in the same group:

1. Lift the handset or press the [Speaker] button.
2. Enter the {Call Pick-Up (Group)} feature code.

Single Line Telephone (SLT)

To pick up a call ringing at another Extension in the same group:

1. Lift the handset.
2. Enter the {Call Pick-Up (Group)} feature code.

Conditions

- When there are several calls ringing in the same Pick-up Group, calls are answered in order of arrival, first in first answered.
- Call Backs are not subject to Call Pick-up.

Related WMS Menu

[To assign Call Pick-Up (Group) Feature code]
Data Management > Numbering Plan Information > Feature Code > “Call Pick-Up (Group)”
[To assign Pick-up Group]

Data Management > Extension Information > Group Information > Pick-up Group

Related Features

- Direct Call Pick-up

Hardware

2.22 Call Recording

2.22.1 Two-Way Recording

Description

An Extension can record a conversation saving the recording to the internal VM, Phontage or external VM. The Two-way record device is configured in WMS. Users can start two-way recording in two ways; automatic recording when a call is connected, or the user presses the preconfigured {Two Way Record} Flex button. If the {Two Way Record} Flex button is not available, the {Two Way Record} feature code may be used.

Operation

iPECS Multi-button phone

To assign {Two Way Record} to a Flex button using the Extension User program:

- [PGM] + {FLEX} + 2 (program type for number) + {Two Way Record} feature code + [SAVE]

To reserve recording before starting a call:

1. Lift the handset and receive a dial tone.
2. Press the {Two Way Record} Flex button or dial the feature code.
3. Receive second dial tone.
4. Place a call to the desired number.
5. When the called party answers, the user will hear the Record Alarm(ODR) tone.
6. After the tone, the call is recording.

To cancel recording reservation:

1. Press the {Two Way Record} Flex button or dial the feature code.

To start Two-Way recording manually with a {Two Way Record} Flex button:

1. Press the {Two Way Record} Flex button during a call.
2. After the Record Alarm(ODR) tone, the call is connected, and the system starts recording the call.

To start Two-Way recording manually without a {Two Way Record} Flex button:

1. Press the [TRANS] button, or, for an SLT, press and release the hook-switch (hook flash).
2. Enter the {Two Way Record} feature code.
3. After the Record Alarm(ODR) tone, the call is connected, and the system starts recording the call.

To stop two-way recording:

1. Press the flashing {Two Way Record} Flex button.
   Or
1. Hang up on the call.

Conditions

- The call is recorded in the internal VM (VPCM), Phontage or external VMS depending on the WMS settings. An external VMS employing in-band (DTMF) signaling will not support the Two-Way Record function.
- There are two kind of Recording Alarm Tone, one is Record Alarm Tone(ACR) which is used for automatic record, and the other is Record Alarm Tone(ODR) which is used for maual record.
- Use of this feature when the Two-Way Record Alarm(ODR/ACR) 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.

Related WMS Menu

[To set Two-Way Record Warning Tone]
Data Management > Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > “Record Alarm Tone(ODR)”, “Record Alarm Tone(ACR)”
[To assign Two-Way Record Feature code]
Data Management > Numbering Plan Information > Feature Conditions > Two-Way Record”
[To set Two-Way Record Device, Two-Way Record Start Mode]
Data Management > Extension Information > Number(DN) Information > DN Voice Mail Information > “Two-Way Record Device”
> Extension Information > Number(DN) Information > DN Voice Mail Information > “Two-Way Record Start Mode”

Related Features

Hardware

- iPECS Multi-button phone
- WLAN Phone (WIT-400H).

2.22.2 Call Recording - Phontage

Description

DNs can be configured to record all calls to the hard disk drive of the iPECS Phontage client. iPECS Phontage enables users to retrieve, 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 User Guide.

Conditions

- Phontage may record one call at a time and cannot place or receive a call during recording.
- Phontage must be idle when set to record a call.
- When recording a call, the Extension user receives the Record Alarm(ODR) tone.
- The call recording feature is not accessible during a conference.

Related WMS Menu

[To set options for Call recording]

Data Management > Extension Information > Number(DN) Information > DN Voice Mail Information > “Two-Way Record Device”
> Extension Information > Number(DN) Information > DN Voice Mail Information > “Two-Way Record Start Mode”

Related Features

Hardware

- PC with Phontage

2.22.3 Call Recording - IPCR

Description

The system can record calls automatically or manually using the optional IPCR server application. The subscriber is assigned with an agent ID in IPCR to allow call recording.

. Extension call recording :

- ✓ An extension call recording is configured as two ways, automatic call recording and manual call recording. ACR(Automatic Call Recording) is started after extension call is connected to counter party, while ODR(On Demand Recording) is controlled by selecting press the pre-configured button or dialing the feature code.
- Agent id of IPCR Device means the extension number to be recorded in iPECS-CM.
- and, Once if agent id is created in IPCR Device, then agent id and recording start mode will sent to iPECS-CM ,and set in DN voice mail information automatically.

. Trunk call recording :

- ✓ Trunk call recording supports only ACR Mode, doesn’t support manual recording.
  And each trunk is required to have an Agent ID of IPCR.
- ✓ In this trunk call recording, agent id doesn’t mean extension number of iPECS-CM,
  So agent id for recording trunk call should not match with extension number of iPECS-CM.
- ✓ In IPCR Server, agent id for trunk call should be assigned as like
  “trunk line access code + trunk serial number”.
- ✓ The ACR tone is automatically played when call recording is started default.
  But, recording alarm announcement is played if IPCR announcement using option is on.

Operation

Automatic recording
ACR recording of extension and trunk call are operated automatically.

On-demand recording (only extension call recording)

Press the {Two Way Record} Flex button during a call, the entire call from the beginning is recorded to the IPCR application server.

Conditions

- During recording, iPECS-CM use channels of a VPCM or an MCIM.
- A maximum of five (5) IPCRs per tenant can register to iPECS-CM.
- If the recording process use standard encryption, then iPECS-CM needs a VPCM or a VOIM to support standard encryption.
- If trunk call is connected to internal extension, and this trunk and extension are set to be recorded at the same time, an extension call recording will operate and trunk call recording will be canceled. That is, extension call recording is prior to trunk call recording.

Related WMS Menu

[To add an IPCR server]
Data management > Extension Information > Terminal Information> Terminal Attribute > "Proprietary SIP"
[To assign Two-Way Record Device]
Data Management > Extension Information > Number(DN) Information > DN Voice Mail Information > "Two-Way Record Device"
[To assign IPCR Device]
Application Interface > IPCR Device Information

Related Features

Hardware

2.22.4 IPCR Record Alarm Announcement

Description

CM can request to play announcement of IPCR when the call recording is started and during recording.

Operation

To play a recording start announcement:

1. ACR (Automatic Call Recording)
   a. The announcement is automatically played when call recording is started
2. ODR (On Demand Recording)
   a. The announcement is played when user press 'call recording' button.

To play the announcement during recording:
- Add a ‘IPCR recording alarm announcement play’ feature code.
  1. The announcement is played when user press ‘IPCR recording alarm announcement play’ flexible button on IP phone.

Conditions

- SIP/SLT is not able to play the announcement during call recording. (SIP phone cannot be handle flexible button feature on the call/ There is not flexible button on SLT phone.)
- System tone is played when IPCR announcement using option is off.
- Announcement is mixed with calling voice when it is played during call recording.

Related WMS Menu

[To add feature code]
Data management > Numbering Plan Information > Feature Code > “IPCR recording alarm announcement play”
[To set IPCR alarm using option]
Data management > Extension Information > Number(DN) Information > DN Voice Mail Information > “Use IPCR record alarm announcement”

Related Features

Hardware

IPCR Server

2.22.5 Call Recording using IP Phone conference without VPCM

Description

CM required VPCM conference group resource to record call, so VPCM should be installed depending on the number of users using call recording.
To decrease required number of VPCM, SIP phone which is activated call recording performs voice mixing internally and send mixed voice packet directly to the call recording device.

Operation

1. SIP phone press “call-record” button to start call recording in call status.
2. When it activate call recording successfully, the LED of call-record button start to blink and send alert tone to both party.
3. The user can stop call recording to press “call-record” button again or disconnect call.

Conditions

- IP-Phone should support internal voice mixing for call recording. It can be checked in “SIP Terminal Configuration” menu. If it’s not enabled, please check s/w version and configuration of provisioning.
- Call recording is not supported in transfer call.
- Call recording device should support SIP protocol.

Related WMS Menu

[To set use of Call Recording in System]
Data Management > Number (DN) Information > Voice Mail Information > Two-way Record Device
Data Management > Number (DN) Information > Voice Mail Information > Two-way Record Start Mode

Version Management > SIP Phone Provisioning > SIP Phone Type Configuration > Call Recording Use : Use

Related Features

Hardware

・Call recording device supporting SIP protocol.
・IP8815/20/30/40E supports this feature.

2.23 Call Intercept

Description

When a called Extension is busy, the calling Extension can connect to the called Extension forcing the release of the connected party. The caller must have an Intercept level that is higher than the called party to activate Call Intercept.

Operation

To intercept a call with Busy One Digit Service while receiving a busy tone:

1. Enter the Busy One Digit Service digit for {Intercept}

To intercept a call with the feature code while receiving a busy tone:

1. Press the [TRANS] button, or, for an SLT, press and release the hook-switch (hook-flash).
2. Enter the {Intercept Request} feature code.

Conditions

・The Intercept authority is determined by the Intercept Level from level 1 to level 20; level 1 is the highest and the level 20 is the lowest level. Subscribers with a lower level cannot intercept a call to a higher level subscriber.
・Intercepting is not possible for Networking or Trunk calls.

Related WMS Menu

[To assign Intercept Request Feature code]
Data management > Numbering Plan Information > Feature Code > “Intercept Request”
[To use Intercept]
Data management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > “Intercept”
[To set Snatch Level]

Data management > Extension Information > Number(DN) Information > DN Feature Registration > “Intercept Level”

[To set One Digit Service On Busy]

Data management > Tenant Information > Tenant Basic Information > One Digit Service On Busy > “Intercept”

Related Features

Hardware

2.24 Call Transfer

2.24.1 Call Transfer to Extension

Description

A Trunk or Extension call can be transferred to an Extension. Calls can be transferred announcing the call (screened) or without an announcement (unscreened).

If a call is transferred with the unscreened method, the Transfer Call Ring timer starts. If the timer expires before the call is answered, the recall process will be activated.

Operation

iPECS Multi-button Phone

To use screened transfer during a Trunk or Extension call:

1. Press the [TRANS] button.
2. Enter an Extension number to receive the transfer.
3. After the Extension answers, announce the call and hang up.
   Or
1. Press the [TRANS] button.
2. Press the [DN DSS/BLF] button for the desired Extension.
3. After the Extension answers, announce the call and hang up.

To use unscreened transfer during a Trunk or Extension call:

1. Press the [TRANS] button.
2. Enter an Extension number to receive transfer.
3. Hang up the call before the call connected.
   Or
1. Press the [TRANS] button.
2. Press [DN DSS/BLF] of the desired Extension.
3. Hang up the call before the call connected.

To use transfer with monitoring during a Trunk or Extension call;

1. Press the [TRANS] button.
2. Enter {Transfer Monitor} feature code.
3. Enter the desired Extension number.

4. The transferred Extension goes off-hook, the call is automatically transferred and the transferring Extension is held.
5. The transferred Extension goes on-hook. The call is reconnect to the waiting Extension.

Single Line Telephone (SLT)

To use screened transfer during a Trunk or Extension call:
1. Press and release the hook-switch (hook-flash).
2. Enter an Extension number to receive the transfer
3. After Extension answers, announce the call and hang up.

To use unscreened transfer during a Trunk or Extension call
1. Press and release the hook-switch (hook-flash).
2. Enter an Extension number to receive the transfer.
3. Hang up.

Conditions

- If the called party Extension is busy, the user can use the Camp-On feature.

Related WMS Menu

[To set Recall Time]

Data management	> Tenant Information > Tenant Tone/Ring Information > Tenant System Ring > "Transfer Call Ring (Trunk)"
	> Tenant Information > Tenant Tone/Ring Information > Tenant System Ring > "Transfer Call Ring (Station)"

Related Features

- Hold / Hold Recall
- Call Transfer to Trunk
- Call Waiting / Camp-On
- Flexible Button
- Broker Call

Hardware

2.24.2 Call Transfer to Trunk

Description

A Trunk or Extension call can be transferred to another Trunk. If the transferred call is not answered before the Transfer Hold Recall timer expires, the Hold Recall process is activated.

Operation

To use screened transfer during a Trunk call:
1. Press the [TRANS] button, or, for an SLT, press and release the hook-switch (hook-flash).
2. Enter an outside telephone number.

3. When the called party answers, announce and hang up.

To use unscreened transfer during a Trunk call:
1. Press the [TRANS] button, or, for an SLT, press and release the hook-switch (hook-flash).
2. Enter an outside telephone number.
3. When Ring Back tone is heard, hang up.

Conditions

- When a Trunk call is transferred to another Trunk, the call is allowed or denied based on “Transit Service” setting in WMS.
- When a Trunk call is transferred to a Trunk, and the call termination is not detected from both Trunks in the designated timer, the system may terminate the call automatically.
- In case of a Trunk call with ‘answer’ signals such as with an ISDN or a VoIP call, the Transfer Recall function is performed if the call is not answered.

Related WMS Menu

[To set Transit Call Time, Release Method]

Data Management	> Trunk Information > Transit Service > ”Transit Call Time”, “Release Method”

[To set Unsupervised Conference Time, Unsupervised Conference Extension]

Data Management	> Trunk Information > Outgoing Route Information > Outgoing Route Options > “Unsupervised Conference Time”
	> Trunk Information > Outgoing Route Information > Outgoing Route Options > “Unsupervised Conference Extension”

Related Features

- Hold / Hold Recall
- Call Transfer to Extension
- Transit Service

Hardware

2.24.3 Call Transfer to VM

Description

Extension users may transfer a call directly to the Voice Mailbox of another Extension (VPCM or external voice mail system), where the caller may leave a message.

Operation

Multi-button Phone

To transfer a call directly to another Extension’s VM:
1. Press the [TRANS] button.
2. Press the [MSG/CALLBK] button or enter the {Direct VM Transfer} feature code; phone LCD will display “Direct Transfer to VM.”

3. Enter the desired Extension number.
4. Hang up to complete the transfer.

Single Line Telephone (SLT)
To transfer a call directly to a VM of other Extension:
1. Make a hook-flash.
2. Enter the {Direct VM Transfer} feature code.
3. Enter the desired Extension number.
4. Hang up to complete the transfer.

Conditions

・To transfer a call to VM, the "Voice Mail Device (Phone No)" of the Extension number must be set as the internal or external VM. The {Voice Mail Access} feature code (internal VM, VPCM) or the external VM key number may be set as the "Voice Mail Device (Phone No)".

Related WMS Menu

[To assign Voice Mail Device (Phone No)]
Data Management > Extension Information > Number(DN) Information > Voice Mail Information > "Voice Mail Device"
[To assign Direct VM Transfer feature code]
Data Management > Numbering Plan Information > Feature Code > "Direct VM Transfer"

Related Features

- Hold / Hold Recall
- Call Transfer to Station/Call Transfer to Trunk
- Flexible Button

Hardware

2.24.4 Call Transfer Restriction

Description

Extensions may be configured so that they may not transfer a call to other Extensions. When pressing the [TRANS] button, the system may restrict the transfer based on the dialed digits compared to the Digit Restriction Map in the Transfer COS of the Extension.

Operation

Conditions

・When transferring a call, the dialed digits of an Extension that is enabled to use transfer COS are compared with digits of the Digit Restriction Table and, if they match, then the transferor hears error tone and the call is not transferred.
・When an Extension is set to use temporal COS and Extension user make a transfer call, the transfer COS is mapped by New Digit Restriction Class by Temporary COS Activation. For

example, if an Extension's Original Digit Restriction Class is 1 and if the New Digit Restriction Class by Temporary COS Activation is 10, then system will search the transfer COS of the Original Digit Restriction Class 10 in Transfer COS menu.

- If a SIP phone uses Blind Transfer and the dialed digits for the Blind Transfer are blocked, then the call is not transferred, and the user receives Error tone.

Related WMS Menu

[To assign Digit Restriction Class for Transfer]
Data Management > Extension Information > Number (DN) Information > Transfer COS
[To assign Digit Transfer COS of DN]
Data Management > Extension Information > Number (DN) Information > DN Feature Allow/Deny > Transfer COS

Related Features

- Temporary COS - Station Lock

Hardware

2.24.5 Transferring a trunk call in alerting status

Description

Extension can transfer the trunk call in call proceeding status to other extension.

Operation

To use unscreened transfer
1. Extension (A) makes external call through trunk.
2. Extension (A) press "Transfer" button or "Transfer" soft-menu and dial extension number (B) to transfer.
3. Extension (A) hang up.
4. Extension (B) hangs off, and then Extension (B) hears alert tone.

To use screened transfer
1. Extension (A) makes external call through trunk.
2. Extension (A) press "Transfer" button or "Transfer" soft-menu and dial extension number (B) to transfer.
3. Extension (B) hangs off and then A and B is in talk state.
4. Extension (A) hangs on, and then Extension (B) hears alert tone.

To use blind transfer
1. Extension (A) makes external call through trunk.
2. Extension (A) press "Blind Transfer" soft-menu and dial extension number (B) to transfer.
3. Extension (B) hangs off and then Extension (B) hears alert tone.

Conditions

It's available only for trunk call and extension call can't be transferred in alerting status.

Related WMS Menu

[To set use of transferring a trunk call in alerting status]
Data Management > Extension Information > Terminal Information > Transfer the Trunk Call in Alerting Status

Related Features

Hardware

2.25 Call Wait

Description

Call Wait is used to notify a busy Extension user that a call is waiting to be answered. The system provides a Call Wait Alert Tone to the busy Extension user. With an iPECS Multi-button phone, the HOLD button LED will flash.

The called Extension user with waiting calls, may place the current call on hold, and answer the waiting call.

Operation

iPECS Multi-button phone

To activate Call Waiting after receiving busy tone for an internal call:

1. When a busy tone is heard while attempting a call, enter the one digit for Call-Wait register.

To answer the waiting call:

1. Press the [HOLD] button; the current call is placed on hold, and the new call is connected with the waiting Extension user.

Single Line Telephone (SLT)

To activate Call Waiting after receiving a busy tone for an Extension call:

1. When a busy tone is heard while attempting a call, enter the one digit for Call-Wait register.

To answer the waiting call:

1. Press and release the hook-switch (hook-flash); the current call is placed on hold, and the new call is connected with the waiting Extension user.

Conditions

- Call Wait may be used when encountering a busy Extension; it cannot be used with an Extension in DND, conference or receiving a page.
- If the waiting Extension user terminates the internal call, Call Wait is cancelled.

- An Extension may have only one Call Wait.

Related WMS Menu

[To use Call Wait]
Data Management > Extension Information > Number (DN) Information > DN Feature Allow/Deny> "Call Wait"
[To set One Digit Service on Busy]
Data Management > Tenant Information > Tenant Basic Information > One Digit Service on Busy
[To set Call Wait Request feature code]
Data Management > Numbering Plan Information > Feature Code > "Call Wait Request"

Related Features

- Do-Not-Disturb
- Internal Call
- Broker Call

Hardware

2.26 Called Number Service

Description

iPECS-CM provides two types of service for incoming Trunk calls for a special Called Number, range or all calls. The services provide call progress information as 180 series messages for SIP or progress tones for other call types.

Operation

Conditions

- In the Called Number Service Group menu, "ELSE' means that the system provides the service for all calls except the assigned called number. 'ELSE' can be entered in Called Number field.
- To assign a range of called numbers in the Called Number Service Group menu, enter an, "X" as the wild card entry for any digit.
- Up to 10 tones can be assigned to one Trunk Tone Group.
- This feature is available for incoming trunk calls but not available for QSIG trunk.
- When a called number is SIP phone, information messages (180 ~ 189) from the SIP phone can be transferred to associate system.

Related WMS Menu

[To assign Called number Service Group for Incoming Route Group]
Data Management > Trunk Information > Incoming Route Information > Incoming Route Options > "Called Number Service Group"
[To assign Called number Service Group]

Data Management > Trunk Information > Incoming Route Information > Called Number Service Group

[To assign Trunk Tone Group]

Data Management > Trunk Information > Trunk Tone Group

Related Features

- Direct In Dial(DID)
- Direct Inward System Access(DISA)
- Direct In Line(DIL)

Hardware

2.27 Camp On

Description

A caller may Camp-On to a busy Extension until the called Extension user answers. The Camp- On user will be on hold until the call is answered.

Operation

To activate Camp-On with Busy One Digit Service:

1. When busy tone is heard while attempting a call, enter the Camp-On digit.
2. The called party will hear the Camp-On Alarm tone.

To activate Camp-On with the feature code:

1. When busy tone is heard while attempting a call, press [TRANS] button or, for an SLT, press and release the hook-switch (hook-flash).
2. Enter the {Camp-On Register} feature code.
3. The called party will hear the Camp-On Alarm tone.

Conditions

- The user may not camp-on to an Extension in DND.
- The user may Camp-On to an Extension that already has a camped call, up to five (5) calls may Camp On.
- If an Extension has multiple camped calls, the service is provided in the order activated.
- If a Camped on Extension terminates the internal call, Camp-On is cancelled.

Related WMS Menu

[To enable Camp-On]

Data Management > Extension Information > Number (DN) Information > DN Feature Allow/Deny> “Camp-On”

[To set One Digit Service on Busy]

Data Management > Tenant Information > Tenant Basic Information > One Digit Service on Busy

[To set Camp On Register feature code]

Data Management > Numbering Plan Information > Feature Code > “Camp On Register”

Related Features

- Do-Not-Disturb
- Internal Call

Hardware

2.28 Caller ID Conversion

Description

The Caller ID associated with a call can be modified by the system. In certain cases such as dialing from the Call Log or other automated dialing, the system cannot determine the call route. For this case, CID conversion can be used to attach a Trunk Access code.

Operation

Conditions

- CID Conversion Group should be assigned in “Incoming Route Options”, or “Outgoing Route Options” of trunk route, and the maximum value is 128.
- One CID Conversion Group can use maximum 20 CID Conversion Table, and each CID Conversion Table can assign 80 CID numbers.
- Maximum length of “CID before conversion”and “CID after conversion” in CID Conversion Table is 24 digits.
- Maximum length of “Called Number” in CID Conversion Table is 32 digits.
- If a ‘Before CID Conversion Table’ is duplicated, for example, if ‘9’ is registered as ‘A’ and 9019 as ‘B’, the digits are converted into B if 4 digits match or, into A if 1 digit matches.
- For a transit call, digits are not converted for incoming calls, but conversion is applied using the CID Conversion Group assigned in ”Outgoing Route Options” for the outgoing side of the transit call.

1 CID Conversion Service Option in Incoming Route Option
   - **ALLOW**: If received CLI and called number meet the searching condition of CID Conversion, CID Conversion service would be applied to the incoming call. But if received CLI and called number doesn’t meet searching condition, the incoming call would be released.
   - **DENY**: If received CLI and called number meet searching condition of CID Conversion, the incoming call would be released. But if received CLI and called number doesn’t meet searching condition, the incoming call would be processed as a normal trunk incoming call.
   - **MATCH**: If received CLI and called number meet searching condition of CID Conversion, CID Conversion service would be applied to the incoming call. But if received CLI and called number doesn’t meet searching condition, the incoming call would be processed as a normal trunk incoming call.

2 CID before Conversion in CID Conversion Table

   Incoming CLI and called number through trunk should meet one of following condition.
   - **Whole Numbers**: Whole received CLI should be same with pre-assigned “CID before

conversion" in WMS. It has the highest priority.
Ex) The case that pre-assigned "CID before conversion" digits are '4504875' and received CLI is '4504875'.

- **Prefix Masked Numbers**: Length of CLI should be same as the length of pre-assigned "CID before conversion" digits, and post part of CLI should be same as numeric part of "CID before conversion" digits.
  Ex) The case that pre-assigned "CID before conversion" is '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 as numeric part of 'XXX4847', and any beginning 3 digits are acceptable.
- **Postfix Masked Numbers**: Length of CLI should be same as the length of pre-assigned "CID before conversion" digits, and pre part of CLI should be same as numeric part of "CID before conversion".
  Ex) The case that pre-assigned "CID before conversion" is '450XXXX' and received CLI is '4504875'. Length of incoming CLI is 7. The first 3 digits of CLI are same as numeric part of '450XXXX', and any last 4 digits are acceptable.
- **Length Matching**: Length of CLI should be same as the length of pre-assigned "CID before conversion" digits
  Ex) The case that pre-assigned "CID before conversion" is 'XXXXXXX' and received CLI is '4504875'. Because the length of incoming CLI is 7, this CLI meet the searching condition.
- **Beginning Masked Numbers**: The last part of CLI should be same as numeric part of "CID before conversion" digit, and length of CLI should be same or longer than "CID before conversion" digit. "CID before conversion" digit should be start with 'C'.
- Ex) The case that pre-assigned "CID before conversion" is '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 as '4875'. So this CLI meet the searching condition.
- **End Masked Numbers**: The first part of CLI should be same as numeric part of "CID before conversion" digit, and length of CLI should be same or longer than "CID before conversion" digit. "CID before conversion" digit should be end with 'C'.
  Ex) The case that pre-assigned "CID before conversion" is '450C' and received CLI is '4504875'. Length of incoming CLI is more than 3. The first 3 digits are same as '450'. So this CLI meet the searching condition.
- **No Number**: There is no CLI in incoming trunk call. The case that pre-assigned "CID before conversion" is 'N'. The priority is same with 'Whole Numbers'.
- Numeric part of CLI include from 0 to 9 and *, #.
- In WMS, '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 "CID before conversion".
- In WMS, 'C' can be used only in the first or last digit.
- If incoming CLI meet two or more conditions, selection of condition follows the priority of each condition. 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 Called Number in CID Conversion Table

The Called Party Number from incoming trunk call should meet one of following condition.

- **Whole Numbers**: Whole received Called Party Number(CPN) should be same with pre-assigned "Called Number" in WMS. It has the highest priority.
  Ex) The case that pre-assigned "Called Number" are '4504875' and received CPN is '4504875'.
- **Prefix Masked Numbers**: Length of CPN should be same with pre-assigned "Called Number" digits, and post part of CPN should be same as numeric part of "Called Number".
  Ex) The case that pre-assigned "Called Number" is '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 as numeric part of ‘XXXX4875’, and any beginning 3 digits are acceptable.

- **Postfix Masked Numbers**: Length of CPN should be same as the length of pre-assigned “Called Number” digits, and pre part of CPN should be same as numeric part of “Called Number”.
  Ex) The case that pre-assigned “Called Number” is ‘450XXXX’ and received CPN is ‘4504875’. Length of incoming CPN is 7. The first 3 digits of CPN are same as numeric part of ‘450XXXX’, and any last 4 digits are acceptable.
- **Length Matching**: Length of CPN should be same as the length of pre-assigned “Called Numbre” digits.
  Ex) The case that pre-assigned “Called Number” is ‘XXXXXXX’ and received CPN is ‘4504875’. Because the length of incoming CPN is 7, this CPN meet the searching condition.
- **Beginning Masked Numbers**: The last part of CPN should be same as numeric part of “Called Number” digit, and length of CPN should be same or longer than “Called Number”. “Called Number” digit should be start with ‘C’.
  Ex) The case that pre-assigned “Called Number” is ‘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 as ‘4875’. So this CPN meet the searching condition.
- **End Masked Numbers**: The first part of CPN should be same as numeric part of “Called Number” digit, and length of CPN should be same or longer than “Called Number”.
  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 as ‘450’. So this CPN meet the searching condition.
- **No Number**: There is no CPN in incoming trunk call. The case that pre-assigned “Called Number” is N. The priority is same with ‘Whole Numbers’.
- Numeric part of CPN include from 0 to 9 and *, #.
- In WMS, ‘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 “Called Number”.
- In WMS, ‘C’ can be used only in the first or last digit.
- If incoming CPN meet two or more conditions, selection of condition follows the priority of each condition. 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 CID after Conversion

- It is used for making converted CID.
- **Whole Numbers**: Pre-assigned “CID after Conversion” number would be used as a new converted CID.
  Ex) The case that pre-assigned “CID after Conversion” are ‘2793914’ and received CLI is ‘4504875’. CID would be converted as ‘2793914’.
- **Masked Numbers**: Masked part of received CLI could be used in converted CID.
  Ex) The case that pre-assigned “CID before Conversion” is ‘450XXXX’, “CID after Conversion” is ‘8064XXXX’, and received CLI is ‘4504845’. Converted CID would be ‘80644845’.
- **Eliminated Numbers**: Some part of received CID can be deleted at converted CID.
- Ex) The case that pre-assigned “CID before Conversion” is ‘XXX450XXXX’, “CID after Conversion” is ‘---8054XXXX’, and received CLI is ‘0314504845’. ‘-’ means elimination of the ‘X’ in “CID before Conversion”. So converted CID would be ‘80544845.

Related WMS Menu

[To assign Incoming CID Conversion Group]

Data Management > Trunk Information > Incoming Route Information > Incoming Route Options > “CID Conversion Group”

[To assign Outgoing CID Conversion Group]

Data Management > Trunk Information > Outgoing Route Information > Outgoing Route Options > “CID Conversion Group”

[To assign CID Conversion Map]

Data Management > System Feature Information > CID Conversion Information > CID Conversion Map

[To assign CID Conversion Table]

Data Management > System Feature Information > CID Conversion Information > CID Conversion Table

Related Features

Hardware

2.29 Virtual Subscriber

Description

This feature allows considering trunk incoming call with CLI (Calling Line Identification) as a virtual subscriber. The virtual subscriber is processed with designated digit restriction class, digit conversion class, and tenant number. The virtual subscriber call can be routed specific destination by digit conversion class. This feature can be used 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-CM extension subscriber, but which can be identified by received CLI number and/or Called Number

CID Conversion/Virtual Subscriber Map

CID Conversion/Virtual Subscriber Group is assigned in Trunk Incoming Route Options, and one CID Conversion/Virtual Subscriber Group can assign maximum 20 CID Conversion/Virtual Subscriber Table.

CID Conversion/Virtual Subscriber Table

This table allow a incoming call get tenant number, digit restriction class, digit conversion class, allowed line number, and CLI conversion according to CLI and called number. This table can be assigned up to 2,000.

Virtual Subscriber Service Option

This option contains how to apply virtual subscriber service in Trunk Incoming Route Options.

Operation

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

-. System receives a call from trunk.

-. System processes a virtual subscriber with digit restriction class, digit conversion class, and tenant number if CLI and called number of incoming call fulfills the condition of virtual subscriber.

1 Virtual Subscriber Service Option in Incoming Route Option

- **ALLOW**: If received CLI and called number meet the searching condition of virtual subscriber, virtual subscriber service would be applied to the incoming call. But if received CLI and called number doesn't meet searching condition of virtual subscriber, the incoming call would be released.
- **DENY**: If received CLI and called number meet searching condition of virtual subscriber, the incoming call would be released. But if received CLI and called number doesn't meet searching condition of virtual subscriber, the incoming call would be processed as a normal trunk incoming call.
- **MATCH**: If received CLI and called number meet searching condition of virtual subscriber, virtual subscriber service would be applied to the incoming call. But if received CLI and called number doesn't meet searching condition of virtual subscriber, the incoming call would be processed as a normal trunk incoming call.

2 CID before Conversion in CID Conversion/Virtual Subscriber Table

Incoming CLI and called number through trunk should meet one of following condition.

- **Whole Numbers**: Whole received CLI should be same with pre-assigned "CID before conversion" in WMS. It has the highest priority.
  Ex) The case that pre-assigned "CID before conversion" digits are '4504875' and received CLI is '4504875'.
- **Prefix Masked Numbers**: Length of CLI should be same as the length of pre-assigned "CID before conversion" digits, and post part of CLI should be same as numeric part of "CID before conversion" digits.
  Ex) The case that pre-assigned "CID before conversion" is '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 as numeric part of 'XXX4847', and any beginning 3 digits are acceptable.
- **Postfix Masked Numbers**: Length of CLI should be same as the length of pre-assigned "CID before conversion" digits, and pre part of CLI should be same as numeric part of "CID before conversion".
  Ex) The case that pre-assigned "CID before conversion" is '450XXXX' and received CLI is '4504875'. Length of incoming CLI is 7. The first 3 digits of CLI are same as numeric part of '450XXXX', and any last 4 digits are acceptable.
- **Length Matching**: Length of CLI should be same as the length of pre-assigned "CID before conversion" digits
  Ex) The case that pre-assigned "CID before conversion" is 'XXXXXXX' and received CLI is '4504875'. Because the length of incoming CLI is 7, this CLI meet the searching condition.
- **Beginning Masked Numbers**: The last part of CLI should be same as numeric part of "CID before conversion" digit, and length of CLI should be same or longer than "CID before conversion" digit. "CID before conversion" digit should be start with 'C'.
  Ex) The case that pre-assigned "CID before conversion" is '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 as '4875'. So this CLI meet the searching condition.

End Masked Numbers: The first part of CLI should be same as numeric part of "CID before conversion" digit, and length of CLI should be same or longer than "CID before conversion" digit. "CID before conversion" digit should be end with 'C'.
Ex) The case that pre-assigned "CID before conversion" is '450C' and received CLI is '4504875'. Length of incoming CLI is more than 3. The first 3 digits are same as '450'. So this CLI meet the searching condition.

- **No Number**: There is no CLI in incoming trunk call. The case that pre-assigned "CID before conversion" is 'N'. The priority is same with 'Whole Numbers'.
- Numeric part of CLI include from 0 to 9 and *, #.
- In WMS, '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 "CID before conversion".
- In WMS, 'C' can be used only in the first or last digit.
- If incoming CLI meet two or more conditions, selection of condition follows the priority of each condition. 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 Called Number in CID Conversion/Virtual Subscriber Table

The Called Party Number from incoming trunk call should meet one of following condition.

- **Whole Numbers**: Whole received Called Party Number(CPN) should be same with pre-assigned "Called Number" in WMS. It has the highest priority.
  Ex) The case that pre-assigned "Called Number" are '4504875' and received CPN is '4504875'.
- **Prefix Masked Numbers**: Length of CPN should be same with pre-assigned "Called Number" digits, and post part of CPN should be same as numeric part of "Called Number".
  Ex) The case that pre-assigned "Called Number" is '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 as numeric part of 'XXXX4875', and any beginning 3 digits are acceptable.
- **Postfix Masked Numbers**: Length of CPN should be same as the length of pre-assigned "Called Number" digits, and pre part of CPN should be same as numeric part of "Called Number".
  Ex) The case that pre-assigned "Called Number" is '450XXXX' and received CPN is '4504875'. Length of incoming CPN is 7. The first 3 digits of CPN are same as numeric part of '450XXXX', and any last 4 digits are acceptable.
- **Length Matching**: Length of CPN should be same as the length of pre-assigned "Called Numbre" digits.
  Ex) The case that pre-assigned "Called Number" is 'XXXXXXX' and received CPN is '4504875'. Because the length of incoming CPN is 7, this CPN meet the searching condition.
- **Beginning Masked Numbers**: The last part of CPN should be same as numeric part of "Called Number" digit, and length of CPN should be same or longer than "Called Number". "Called Number" digit should be start with 'C'.
  Ex) The case that pre-assigned "Called Number" is '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 as '4875'. So this CPN meet the searching condition.

- **End Masked Numbers**: The first part of CPN should be same as numeric part of "Called Number" digit, and length of CPN should be same or longer than "Called Number".
  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 as '450'. So this CPN meet the searching condition.
- **No Number**: There is no CPN in incoming trunk call. The case that pre-assigned "Called

Number" is N. The priority is same with 'Whole Numbers'.

- Numeric part of CPN include from 0 to 9 and *, #.
- In WMS, '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 "Called Number".
- In WMS, 'C' can be used only in the first or last digit.
- If incoming CPN meet two or more conditions, selection of condition follows the priority of each condition. 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 CID after Conversion

It is used for conversion of CID for virtual subscriber. If there is no need to convert CID, this field should be remain empty.

- **Whole Numbers**: Pre-assigned "CID after Conversion" number would be used as CID of virtual subscriber.
  Ex) The case that pre-assigned "CID after Conversion" are '2793914' and received CLI is '4504875'. CID of the virtual subscriber would be converted as '2793914'.
- **Masked Numbers**: Masked part of received CLI could be used in converted CID for virtual subscriber.
  Ex) The case that pre-assigned "CID before Conversion" is '450XXXX', "CID after Conversion" is '8064XXXX', and received CLI is '4504845'. Converted CID of virtual subscriber would be '80644845'.
- **Eliminated Numbers**: Some part of received CID can be deleted at converted CID of virtual subscriber.
  Ex) The case that pre-assigned "CID before Conversion" is 'XXX450XXXX', "CID after Conversion" is '---8054XXXX', and received CLI is '0314504845'. '-' means elimination of the 'X' in "CID before Conversion". So converted CID of virtual subscriber would be '80544845.

5 Virtual Subscriber's Tenant Number

It is used for regular tenant services for virtual subscriber. A trunk call which meet the Searching Condition(CID before Conversion, and Called Number) is processed with designated tenant number. Tenant number should be set to act as a virtual subscriber.

6 Virtual Subscriber's Digit Restriction Class

It is used for regular digit restriction class services for virtual subscriber. A trunk call which meet the Searching Condition(CID before Conversion, and Called Number) is processed with designated digit restriction class. Digit Restriction Class should be set to act as a virtual subscriber.

7 Virtual Subscriber's Digit Conversion Class

It is used for regular digit conversion class services for virtual subscriber. A trunk call which meet the Searching Condition(CID before Conversion, and Called Number) is processed with designated digit conversion class. Digit Conversion Class should be set to act as a virtual subscriber.

8 Virtual Subscriber's Maximum virtual calls

It is used for restriction of number of simultaneous call for each virtual subscriber. If the number isn't assigned, there is no limitation of incoming calls. Otherwise, the number can be assigned from 1 to 254. If a virtual subscriber call comes into over pre-assigned Max. Virtual Call, the call is processed as the Virtual Subscriber Service Option. For example, if Virtual Subscriber Service Option is set to "Allow", the overflowed call should be terminated, and when Virtual Subscriber Service Option is set to "Match", the overflowed call should be processed as a normal trunk call.

Conditions

- Virtual Subscriber Group should be assigned in "Incoming Route Option" of incoming trunk route, and the maximum value is 128.
- One Virtual Subscriber Group can use maximum 20 Virtual Subscriber Table, and each Virtual Subscriber Table can assign 80 virtual subscriber.
- Maximum length of "CID before conversion"and "CID after conversion" in Virtual Subscriber Table is 24 digits.
- Maximum length of "Called Number" in Virtual Subscriber Table is 32 digits.
- Virtual Subscriber feature has higher priority than Digit Conversion or ICLID Routing feature.
- Maximum number of Virtual Subscriber Table is 2,000.
- "Tenant Number", "Digit Restriction Class", and "Digit Conversion Class" should be set in Virtual Subscriber Table.
- If "Max. Virtual Calls" field is empty, there is no llimitation of the virtual subscriber.
- If "Use Option" field is set to Not Use, virtual subscriber feature isn't supported.

Related WMS Menu

[To assign Incoming CID Conversion Group]

Data Management	> Trunk Information > Incoming Route Information > Incoming Route Options > "CID Conversion Group"

[To assign Outgoing CID Conversion Group]

Data Management	> Trunk Information > Outgoing Route Information > Outgoing Route Options > "CID Conversion Group"

[To assign CID Conversion Map]

Data Management	> System Feature Information > CID Conversion Information > CID Conversion Map

[To assign CID Conversion Table]

Data Management	> System Feature Information > CID Conversion Information > CID Conversion Table

Related Features

- Digit Conversion
- Digit Restriction

Hardware

2.30 Command Call

Description

A user can place a call to all members in a Command Call Group simultaneously. A Command Group can be established for paging or conference.

The following options are available for a Command Call:

- Allow Member On-Hook during Command Call – If a busy Command Call Group member returns to idle during a Command call, the caller may recall the previously busy member. This feature is activated automatically if the "Internal Page Group Access Table" is set to 'Recall' in the Command Call Group option. The recall is not applicable to the Trunk members of a group.

- Member Busy Option – If a member in a Command Call Group is dialing or busy, the Command caller may terminate the call and connect the member to the Command Call. If the member is off-hook, the system releases all resources and connects the call to the member. If the member is receiving a ring, the system releases the caller, sends the Command Call ring to the member. When the member answers the incoming call, a Command Call is established. To activate this feature, the Member Busy Option must be set to "Call after Forced Release" in the Command Call Group option.

The Command caller can call or release a member by pressing the Flex button assigned for the member. If the Command caller calls the member Extension by pressing the Flex button and the Extension is not engaged in a Command Call, the member will receive the Command Call. If the user is engaged in or receiving a Command Call, the member is released from the Command Call. 2^{nd} member can be assigned for the busy state of 1^{st} member.

Operation

To place a Command Call:

1. Enter the {Command Group Call, Page} or {Command Group Call, Conference} feature code and Command Call Group (00~99).
2. A call is placed to the internal and external members of the group.
3. At answer, a conference is established between members.

To transfer a Command Call:

1. The Command caller presses [TRANS] and dials the Extension number.
2. After the transferring call is answered, the Command caller returns to idle.

Conditions

- The system supports 100 Command Call groups. Command Call group 00 calls all Command Groups allowed to be accessed. Other groups call only members of the individual group.
- A Command Call Group can have 128 members, Extension or Trunk users.
- A conference board (VPCM) is required to support the Command Call function.
- An Extension with a held conference is not allowed to place a Command Call. In this case, the user receives error tone.
- If the "Call after Forced Release" option is set, members can be recalled if they are receiving normal ring, busy, dialing, receiving Ring Back tone, busy tone or other normal tones. Members cannot be recalled if they are receiving paging, PTT, Command Call or Wake-up alarm, blocked, on hold or in conference.
- When Trunks are called by Command Call, Trunks are seized in a round-robin order regardless of system settings.
- "Recall on hang up" is not applicable to a SIP phone.
- When an Ericsson-LG Enterprise SIP phone with LCD is used as a member of Command Call, "Command Call" is displayed on the LCD.
- Command Call codes dialed on PC phone (UCS Client or Phontage) are not saved in Call Log List.

- A Command Call must be answered by more than one called member, for the Command caller to transfer the call.
- A Command Call cannot be transferred to an Extension that is busy or with an active forward.
- A SIP Phone cannot establish a transfer of a Command Call.
- 2^{nd} member works only Member Busy Option is “Busy”.

Related WMS Menu

[To set LED for Command Call]
Data Management > Tenant Information > Tenant Basic Information> Phone LED Control > “[External No. Button]: Command Call Member Ringing”
> Tenant Information > Tenant Basic Information> Phone LED Control > “[External No. Button]: Command Call Member Answer”
[To allow access to Command Call]
Data Management > Extension Information > Number(DN) Information > DN Feature Registration > “Command Call Group Access Table”
[To set options & members of Command Call Group]
Data Management > Extension Information > Group Information > Command Call Group
[To set Command Call Group Access]
Data Management > Extension Information > Group Information > Command Call Group Access Table
[To assign Command Call feature code]
Data Management > Numbering Plan Information > Feature Code > “Command Group Call (Page)”
> Numbering Plan Information > Feature Code > ”Command Group Call (Conference)”

Related Features

Hardware

- VPCM

2.31 Conference

2.31.1 Conference Room

Description

Internal and external parties can be joined in a conference by entering a Conference Room number. Conference rooms can be password protected so that only parties that enter the password are allowed to join the Room.

Operation

Authorized Extension

To deactivate a Conference Room:

1. Press the [PGM] button.
2. Dial ‘9’ + ‘5’.

3. Enter the tenant number (001~254).
4. Enter the conference room number.
5. Dial * or press the [SAVE] button.

To view the number of participants in a Conference Room:
1. Press the [PGM] button.
2. Dial '9' + '4'.
3. Enter the tenant number (001~254).
4. Enter the Conference Room number.
5. Dial * or press the [SAVE] button.

<u>iPECS Multi-button Phone</u>
To activate a Conference Room:
1. Press the [PGM] button.
2. Dial '5' + '4' or the {Conference Room Activate} feature code.
3. Enter the desired Conference Room number and '*'.
4. Enter a password for the Conference Room (up to six (6) digits).
5. Press the [SAVE] button.

To join a Conference Room:
1. Enter the Conference Room number. If password is set, password dial tone is provided.
2. Enter a password for the Conference Room (up to six (6) digits) and '*'.

To deactivate a Conference Room:
1. Press the [PGM] button.
2. Dial '5' + '5' or the {Conference Room Deactivate} feature code.
3. Enter the desired Conference Room number and '*'.
4. Enter the password.
5. Dial * or press the [SAVE] button.

To transfer a call to a Conference Room:
1. Press the [TRANS] button during a call.
2. Enter the Conference Room number.
3. Enter the password
4. Dial *.
5. Hang up the call.

<u>Single Line Telephone (SLT) / SIP</u>
To activate a Conference Room:
1. Lift the handset.
2. Enter the {Conference Room Activate} feature code.
3. Enter the desired Conference Room number and '*'.
4. Enter a password for the Conference Room.
5. Dial *.

To join a Conference Room:
1. Lift the handset.
2. Enter the Conference Room number. If password is set, Password Dial Tone is provided.
3. Enter the password
4. Dial *.

To deactivate a Conference Room:
1. Lift the handset.
2. Enter the {Conference Room Deactivate} feature code. If password is set, password dial tone is provided.
3. Enter the Conference Room number and ‘*’.
4. Enter the password and “*”.

To transfer a call to a Conference Room:
1. Make a hook-flash while on a call.
2. Enter the Conference Room number. If a password is set, Password Dial tone is provided.
3. Enter the password and “*”.
4. Hang up the handset.

Conditions

- A VPCM is required. The number of available channels of VPCM depends on the codec (128 parties with g.711 codec, 80 parties with g.723 codec, and 100 parties with g.729 codec).
- Up to 100 Conference Rooms per tenant can be assigned and each can support a maximum of 128 parties.
- The Conference Room Number is configured through WMS.
- Once a Conference Room is created, users may enter the Conference Room until it is deactivated or deleted.

Related WMS Menu

[To assign Conference Room]
Data management > Tenant Information > Tenant Basic Information > Conference Room
[To use Conference Call]
Data management > Extension Information > Number (DN) Information > DN Feature Allow/Deny > “Conference Call”,
[To assign features code for Conference Room]
Data Management > Numbering Plan Information > Feature Code > “Conference Room Active”
> Numbering Plan Information > Feature Code > “Conference Room Deactivate”

Related Features

- Conference/Multi-Party Voice Conference
- Conference/Unsupervised Conference

Hardware

- VPCM

2.31.2 Add On Conference

Description

The system allows multiple internal and external parties to be connected in a call, in an ad hoc conference.

Operation

iPECS Multi-button Phone

To establish a conference (Conference Member Add Method: “Immediately On Answer”):

1. Establish a call.
2. Press the [CONF] button. The LED will light, the connected party is placed on hold and the user receives dial tone.
3. Place a call to the Extension/Trunk to be included in the conference.
4. When connected, all parties are joined in the conference.

To establish a conference (Conference Member Add Method: “[CONF] Button during a Call”):

1. Establish a call.
2. Press the [CONF] button. The LED will light, the connected party is placed on hold and the user receives a dial tone.
3. Place a call to the Extension/Trunk to be included to the conference.
4. When connected, press the [CONF] button; the new party is added to the conference.
5. To add another Extension/Trunk to the conference, repeat steps 3 and 4.

To hold a conference for a while:

1. Press the [CONF] button; the [CONF] button LED will flash.

To retrieve a conference:

1. Press the flashing [CONF] button or the [DN] button used during the conference.

Single Line Telephone (SLT)

To establish a conference (Conference Member Add Method: “Immediately On Answer”):

1. Establish a call.
2. Press and release the hook-switch (hook-flash); the connected party is placed on hold and the user receives dial tone.
3. Enter the {Conference Member Add} feature code; dial tone is provided.
4. Place a call to the Extension/Trunk to be included in the conference.
5. When connected, all parties are joined in the conference.

To establish a conference (Conference Member Add Method: “[CONF] Button during a Call”):

1. Establish a call.
2. Press and release the hook-switch (hook-flash); the connected party is placed on hold and the user receives dial tone.
3. Enter the {Conference Member Add} feature code, dial tone is provided.
4. Place a call to the Extension/Trunk to be included in the conference.
5. When connected, press and release the hook-switch (hook-flash), the new party is added to the conference.

6. To add another Extension/Trunk to the conference, repeat Steps 3, to 5.
7. Press and release the hook-switch (hook-flash) to return to the conference.

Conditions

- A VPCM is required. The number of available channels depend on codec (128 parties with g.711 codec, 80 parties with g.723 codec, and 100 parties with g.729 codec).
- A member can leave the conference momentarily, the [CONF] button will light.
- An unlimited number of 3-party conferences may be established using iPECS Phones. A VPCM is required if no iPECS Phone is in the 3-party conference or for conferences of more than three (3) parties.
- If all VPCM channels are busy, the user will receive error tone and the display will indicate that no Conference channels are available.
- If an error occurs while attempting to add members, the conference initiator may return to the conference by pressing the [CONF] button or, for an SLT, press and release the hook-switch (hook-flash).
- An Extension that is busy, in DND or other non-idle state cannot be added to a conference
- An Extension with a held conference is not allowed to place a Command Call, conference, conference room, page and PTT page . In this case, the user receives error tone.

Related WMS Menu

[To set Conference Member Add Method]
Data management > Tenant Information > Tenant Basic Information > Tenant Attribute > "Conference Member Add Method"
[To enable Conference Call]
Data management > Extension Information > Number (DN) Information > DN Feature Allow/Deny > "Conference Call"
[To assign VPCM to Zone]
Data management > Zone Information > Zone Attribute > "Using Slot No."
[To assign Zone]
Data management > Zone Information > Channel Attribute > "Zone"

Related Features

- Unsupervised Conference

Hardware

- VPCM.

2.31.3 Consultation Conference

Description

Extension users can establish a 3-way conference by placing the current user on hold, and initiating a call to another user.

Operation

Multi-button Phone

To establish a Consultation Conference:

1. Press the [TRANS] button while on a call.
2. Enter the Extension or Trunk user number.
3. When connected, press the [CONF] button.

Single Line Telephone (SLT)

To establish a Consultation Conference:

1. Press and release the hook-switch (hook-flash) while on a call.
2. Enter the Extension or Trunk user number.
3. When connected, momentarily press and release the hook-switch.

Conditions

- To activate Consultation Conference with an SLT, “Hook Action During Transfer “option must be assigned as “Conference Call”.

Related WMS Menu

[To set Hook Flash option for conference]

Data Management > Extension Information > Terminal Information > Terminal Option > “Hook Action During Transfer” -> Conference call

Related Features

Hardware

2.31.4 Unsupervised Conference

Description

Extension users may establish a conference call with Trunk users, and may exit the conference, allowing the Trunk users to converse privately without supervision (Unsupervised Conference).

The system will disconnect the Unsupervised Conference if Disconnect Supervision is detected with only two parties connected or at expiration of the Unsupervised Conference timer. If enabled, either Trunk user in an Unsupervised Conference can request the Unsupervised Conference timer be extended. The trunk user dials a digit 1 to 9 indicating the Timer extension multiplier, then the system will extend the timer based on the dialed digit multiple of the Timer.

Operation

iPECS Multi-button phone

To make an Unsupervised Conference:

1. During a conference call, press the [CONF] button to exit from the conference call, the [CONF] button LED will flash to indicate the Unsupervised Conference.

To re-enter an Unsupervised Conference:

1. In either on-hook or off-hook status, press the flashing [CONF] button
   Or
   Press the [CONF(J)] soft button.

To terminate an Unsupervised Conference:

1. Press the [CONF (E)] soft button.

Members of Conference

To extend an Unsupervised Conference timer:

1. Dial 1~9 (ex. if dialed digit is 4 and Unsupervised Conference time is 2 min, extended conference time will be eight (8) minutes).

Conditions

- The Unsupervised Conference feature applies to DISA users also.
- An Unsupervised Conference will not recall.

Related WMS Menu

[To set options for Unsupervised Conference of Incoming Route]

Data management	> Trunk Information > Incoming Route Information > Incoming Route Options > ”Unsupervised Conference Time”
	> Trunk Information > Incoming Route Information > Incoming Route Options > ”Unsupervised Conference Extension”

[To set options for Unsupervised Conference of Outgoing Route]

Data management	> Trunk Information > Outgoing Route Information > Outgoing Route Options > ”Unsupervised Conference Time”
	> Trunk Information > Outgoing Route Information > Outgoing Route Options > ”Unsupervised Conference Extension”

Related Features

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

Hardware

- VPCM

2.32 Data Line Security

Description

System tones such as Camp-On or Intrusion during data transmission (MODEM or FAX) between Extensions or over a Trunk may cause distortion or errors in the data. To eliminate such errors, Extensions that use analog data can be assigned to block system tones.

Operation

Conditions

- The Extension user will hear error tone when attempting to Camp-On or Intrude on an Extension with Data Security enabled.
- If Data Security is enabled, the system does not control gain.

Related WMS Menu

[To set Data Line Type]
Data Management > Extension Information > Terminal Information > Terminal Option > “Data Line Type”

Related Features

- Call Waiting/Camp-On

Hardware

2.33 Delayed Auto-Attendant

Description

Incoming Trunk calls can be routed to the VPCM Auto-Attendant either immediately upon detection or after expiration of the VPCM Auto-Attendant delay timer. This feature allows another Extension to answer the call.

Operation

Conditions

- When Delayed Auto-Attendant Ring is assigned, after the delay, the call will no longer ring assigned Extensions and will only ring to the VPCM Auto-Attendant.

Related WMS Menu

[To set options for Delayed Auto Attendant]
Data Management > System Feature Information > Trunk Call Routing Table > “Service Type” -> “Multi Ring”
> System Feature Information > Trunk Call Routing Table > “Digit Information” -> “Announcement Table Access(CCR)”

> System Feature Information > Trunk Call Routing Table >" Ring Count before following Digit Information"

[To assign Announcement Table]

Data Management > System Feature Information > Announcement Table

Related Features

Hardware

2.34 Dial Pulse Signaling

Description

An analog CO line will send dial pulse signals to the central office. If programmed as a pulse CO line, the system will send open loop pulses at 10 pps with the assigned break/make ratio.

Operation

Conditions

・ The break/make ratio (60/40 or 66/33) can be configured.

Related WMS Menu

[To set Dialing Type]

Data Management > Trunk Information > Trunk Basic Information > Trunk Attribute > "Dialing Type"

Related Features

- DTMF Signal Sending
- Station Speed Dial
- System Speed Dial

Hardware

2.35 Differential Ring

Description

Differential Ring allows any one of 14 different audible Ring signals to be assigned to an iPECS Phone, allowing users to determine which phone is ringing and the type of call (internal or trunk). When the phone receives an incoming call, the selected ring signal is provided over the speaker. Different selections are assigned for Intercom and internal or Trunk calls.

Eight different tones are stored in the iPECS Phone. Four of these tones are permanent while the other four are assigned from the 10 ring-tones stored in the system. Note the system ring tones may be replaced with any 8 second **.wav* file through iPECS-CM Web Management System (WMS).

Operation

To select a ring tone in the phone:

1. Press the [PGM] button.
2. Dial 2 for Ring selection.
3. Dial 1 (Extension ring) or 2 (Trunk ring).
4. Enter a ring tone (1~8).

To download a ring tone from the system:

1. Press the [PGM] button.
2. Dial 2 for Ring selection.
3. Dial 3, (Ring Download)
4. Enter the ring tone storage number (5~8).
5. The ring sources (0~9) stored in the system are listed. You can hear the ring tones by selecting the numbers.
6. Dial 1 to register the ring.

Conditions

- To use a system ring tone, the tone must first be downloaded to the phone memory then select the tone from the phone menu.
- iPECS-CM Phontage does not support system ring tones; iPECS-CM Phontage supports download of *wav files.
- The 14 ring tones stored in the system can be changed using the iPECS-CM Web Maintenance System (WMS).

Related WMS Menu

[To set Differential Ring]

Data Management	> Extension Information > Terminal Information > Terminal Option > "Differential Ring ID for Internal Call"
	> Extension Information > Terminal Information > Terminal Option > "Differential Ring ID for External Call"

[To set Phone/GW Ring]

Data Management	> Phone/GW Information > Phone/GW Tone/Ring Information > Phone/GW Ring

[To set SLTM GW Ring]

Data Management	> Phone/GW Information > Phone/GW Tone/Ring Information > SLTM GW Ring

[To upload Ring]

Version Management	> Tone, Ring Upload

Related Features

Hardware

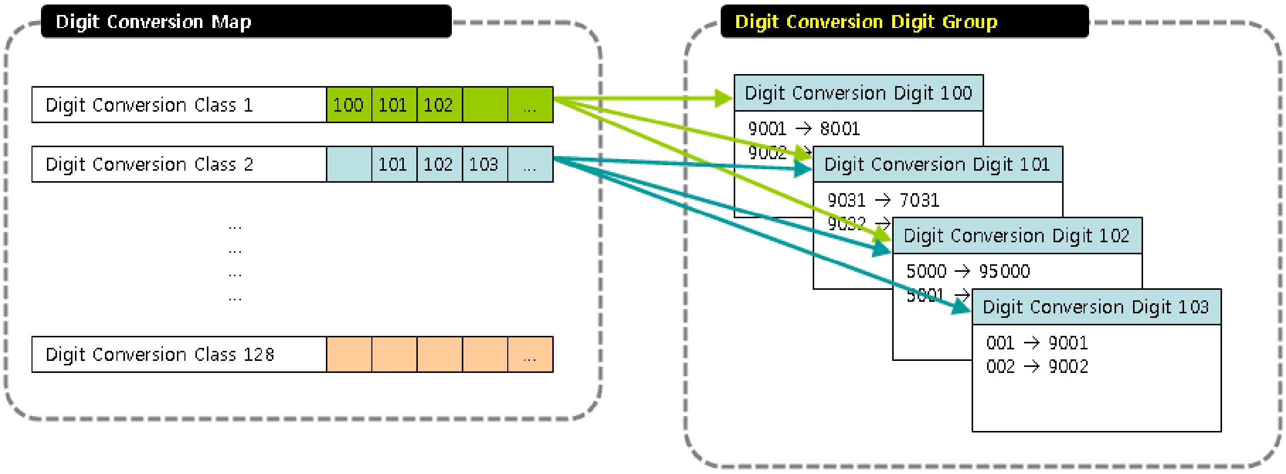
- Multi-button Phone

2.36 Digit Conversion

Description

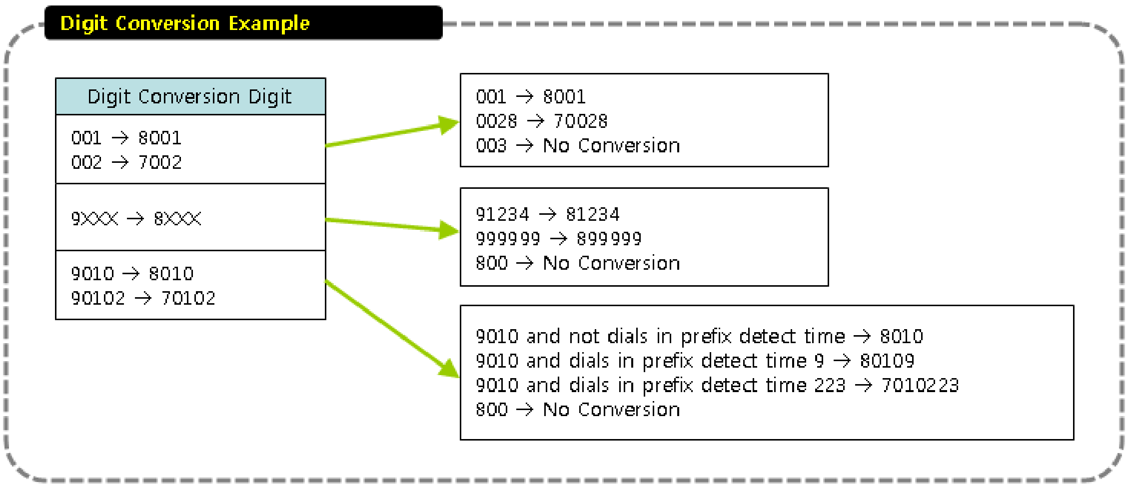
The system converts the digits dialed by the user before servicing the number according the Numbering Plan. A Digit Conversion class is assigned to the DN or Trunk, so digit conversion is applied when the Extension user dials a digit or the Extension receives an incoming Trunk call.

A Digit Conversion class has a maximum of 20 Digit Conversion Tables, and a Digit Conversion Table has maximum of 80 Digit Conversion entries. A Digit Conversion Table can be assigned to multiple Digit Conversion classes.



Digit Conversion Rules are shown below:

- The system ignores the dialed digit stream until a match occurs.
- An 'X' is employed as a wild card digit (any digit)
- A long prefix or short prefix can be supported. For example, if the digit stream "9010" and "90101" are listed in Digit Conversion Table, and the user dials 9010, the system waits for the prefix detect time to decide the prefix. If the user does not dial in the prefix detect time, the short prefix will be applied, otherwise the long prefix will be applied.



Options for digit conversion are listed below:

- Digit Conversion Applied Time Type – The Administrator can program the time that a conversion is to be applied for each digit, using the LCR (Least cost routing) time data or Day/Night time data.
    Unconditional: Digit conversion is always performed
    Depend On Day/Night Time: Digit conversion is performed based on each Day/Night/Timed time
    Depend On LCR Time: Digit conversion is performed by each of the nine (9) LCR time zones.
- Digit Conversion Virtual Tone – Because the Trunk is not seized until digit conversion is finished, the system cannot provide dial tone directly from the Trunk. This function enables the system to provide a virtual tone after a specific digit.
- Digits before/after Conversion – This option enables users to decide whether the system should display the digit dialed by the Multi-button phone user or the digit after conversion, and whether the system should send the digit before conversion or after conversion depending on the program.

Operation

Conditions

- There are 128 Digit Conversion classes are available for each tenant.
- Up to 1600 Digit Conversion tables are supported for a class.
- Up to 24 digits can be entered for conversion.
- Digits dialed by the user before conversion are stored in the Call Log.

Related WMS Menu

[To set System Time Zone]
System management > System Time Zone
[To set time to detect prefix]
Data Management > Tenant Information > Tenant Time Information > Tenant Timer > "Prefix Detection Time on Duplicate Num Plan"
[To set Digit Conversion Class]
Data Management > Extension Information > Number (DN) Information > DN Attribute >"Digit Conversion Class"
Data Management > Trunk Information > Trunk Basic Information > Trunk Attribute >"Digit Conversion Class"
[To assign Digit Conversion Map/Group/Option]
Data Management > System Feature Information > Digit Conversion Information > Digit Conversion Map
Data Management > System Feature Information > Digit Conversion Information > Digit Conversion Table
Data Management > System Feature Information > Digit Conversion Information > Digit Conversion Option
[To assign Digit Conversion Virtual Tone]
Data Management > System Feature Information > Digit Conversion Virtual Tone

Related Features

Hardware

2.37 Digit Map

Description

Digit Map feature is used when user want to send outgoing digits at once.

2.37.1 Digit Map Option

Description

User can specify whether to use the option by tenant.
If Digit Map option index specified, function is operated by Enbloc-outgoing digit of the corresponding table.

Operation

System collects digits until defined digits are gathered.
During delay timer, system collect digit. If delay timer expired, send outgoing call automatically

Conditions

- 20 digit map table can be assigned per tenant
- Test - Simulate the digit map digit information. It shows the simulation results.

Related WMS Menu

[To set Digit Map Option by tenant]
Data Management > System Feature Information > Digit Map Information > Digit Map Option
[To set Digit Map Digit]
Data Management > System Feature Information > Digit Map Information > Digit Map Digit

Related Features

Hardware

2.37.2 Digit Map Group

Description

Every outgoing trunk and voice network can have one digit map group optionally.
If digit Map Group index specified, feature is operated by Enbloc-outgoing digit of the corresponding table.

Operation

System collects digits until defined digits are gathered.

During delay timer is running, system collects digit until they are valid and matched with the Specified Digit Map table.
If delay timer is expired or gathered digits are not matched with the Digit Map tables, then stop gathering digit and will send outgoing call automatically.

Conditions

- 20 digit map tables can be assigned per Digit Map Group.
- Digit Map option and Digit Map Group are able to set at the same time, but Digit Map group is prior to
- Digit Map option.
- Only when numbering plan type of voice networking numbering plan is transit out, Digit Map Group is applied.
- Test - Simulate the digit map digit information. It shows the simulation results.

Related WMS Menu

[To set Digit Map Group by Outgoing Route]

Data Management	> Trunk Information > Outgoing Route Information > Outgoing Route Options > Digit Map Group
Data Management	> System Feature Information > Digit Map Information > Digit Map Group

[To set Digit Map Group by voice networking numbering plan]

Data Management	> Voice Network > Voice Network Numbering Plan > Digit Map Group

[To set Digit Map Digit]

Data Management	> System Feature Information > Digit Map Information > Digit Map Digit

Related Features

Hardware

2.37.3 Digit Map Digit

Description

Digit Map Digit is used to collect digit when user want to send outgoing digits at once.
System collects digits until defined digits are gathered.
In case of ISDN or H323 trunk, User can specify called party number information.(Number of Type / Numbering Plan / Sending Complete)

Operation

System collects digits until defined digits are gathered. After collecting the digits, system send digits
automatically.
Symbol helps user to define Digit map digit more flexible and easy –to-use.

Select Symbol (9|8)	:	(9|8)01 --> 901, 801
Range Symbol [7-9]	:	[7-9]019 --> 7019, 8019, 9019
Number Symbol x	:	9019xxxx --> 8 Digit beginning with 9019
Option Symbol ?	:	90xxxx?? --> 6, 7, or 8 Digit beginning with 90. ?? symbol is determined by Delay Timer of Digit Map Option
Replace Symbol T	:	T019xxx --> If Replace Symbol is (9|8), it is same with (9|8)019xxx

Conditions

- Table index is up to 200.
- User can collect digit up to 32 digits.
- ?? symbol is determined by delay timer of Digit Map Option.

Related WMS Menu

[To set Digit Map Option by tenant]
Data Management > System Feature Information > Digit Map Information > Digit Map Option
[To set Digit Map Digit]
Data Management > System Feature Information > Digit Map Information > Digit Map Digit

Related Features

Hardware

2.38 Digit Restriction

Each Extension and Trunk has a Digit Restriction class that determines the dialing capability of the user.

2.38.1 Class of Service

Description

Dialing restrictions from Class of Service (COS) are applied for:

- Internal calls
- Outgoing trunk calls
- Incoming trunk calls to an Extension
- Incoming trunk calls to a trunk

A COS from 1 to 128 is assigned to each Extension and Trunk.

Operation

Conditions

- Digit restriction is based on the Trunk and Extension COS.

- When placing an outgoing call using authorization code, COS assigned to the Authorization Code is applied to the call.
- Up to 1600 Deny/800 Allow digits can be programmed for each COS.
- Deny/Allow digits can be set differently depending on Day/Night/Timed mode.

Related WMS Menu

[To set Digit Restriction Class]

Data Management	> Extension Information > Number (DN) Information > DN Attribute >“Digit Restriction Class”
Data Management	> Trunk Information > Trunk Basic Information > Trunk Attribute >“Digit Restriction Class”

[To assign Digit Restriction Map/Group]

Data Management	> System Feature Information > Digit Restriction Information > Digit Restriction Map
Data Management	> System Feature Information > Digit Restriction Information > Digit Restriction Table

Related Features

- Temporary COS - Station Lock
- Walking COS
- Authorization Code

Hardware

2.38.2 Day/Night/Timed COS

Description

Digit restriction can be serviced differently depending on Day, Night and Timed system operation mode.

Operation

Conditions

Related WMS Menu

Related Features

- Dialing Restriction / COS
- System Service Time Mode
- System Time Zone

Hardware

2.38.3 Temporary COS - Station Lock

Description

Users can temporarily change the Extension COS, or lock the Extension, in order to prevent unauthorized dialing from the Extension. Before using the Station Lock feature, a Password, Temporary COS Information and Digit Restriction Tables must be programmed. The COS is changed according to the Temporary COS Information. Even when the COS is changed, the Extension users is allowed to make emergency calls and internal calls. To restore the COS of the Extension, enter {Extension Class Restore} feature code.

Operation

iPECS Multi-button phone

To active temporarily COS:

1. Enter the {Extension Class Down} feature code, or press the [PGM] button and dial the Extension Class Down Program code “31”.
2. Enter the Extension Password.
3. Press the [SAVE] button.

To restore the assigned COS of Extension:

1. Enter the {Extension Class Restore} feature code, or press the [PGM] button and dial the Extension Class Restore Program code “32”.
2. Enter the Extension Password.
3. Press the [SAVE] button.

Single Line Telephone (SLT)

To activate temporary COS of Extension:

1. Lift the Handset.
2. Enter the {Extension Class Down} feature code.
3. Or,
4. Enter the {Station Program Mode} feature code and dial the Extension Class Down Program code “31”.
5. Enter the Password and dial *.

To restore assigned COS of Extension:

1. Lift the Handset.
2. Enter the {Extension Class Recover} feature code.
3. Or,
4. Enter the {Station Program Mode} feature code and dial the Extension Class Restore Program code “32”.
5. Enter the Password and dial *.

Conditions

- The Extension is restored to the Extension COS as defined in the active System Service Time Mode (Day, Night or Timed).
- A Password, Temporary COS and Digit Restirction Map/Group must be set to use this service.

Related WMS Menu

[To set Extension Password, Digit Restriction Class]

Data Management	> Extension Information > Number (DN) Information > DN Attribute > "Extension Password", > Extension Information > Number (DN) Information > DN Attribute > "Digit Restriction Class"

[To use Temporary COS]

Data Management	> Extension Information > Number (DN) Information > DN Feature Registration > "Temporary COS"

[To assign Digit Restriction Map]

Data Management	> System Feature Information > Digit Restriction Information > Digit Restriction Map

[To assign Digit Restriction Group]

Data Management	> System Feature Information > Digit Restriction Information > Digit Restriction Table

[To assign feature code for Temporary COS]

Data Management	> Numbering Plan Information > Feature Code > "Extension Class Down", " Extension Class Recover"

Related Features

- Dialing Restriction/COS
- Dialing Restriction/Walking COS
- System Service Time Mode
- Password

Hardware

2.38.4 Walking COS

Description

A user may temporarily override the Digit Restriction of an Extension to place calls from a normally restricted Extension. The user must input a User Authorization code in order to activate Walking COS and is subject to the COS of the Authorization Code. The COS associated with the User Authorization Code is applied after the code is entered for the next call.

Operation

Multi-button Phone

To temporarily change the Extension COS:

1. Enter the {Walking COS} feature code.
   Or
   Press the [PGM] button and dial the Walking COS program code "'33".
2. Enter the User Authorization Code (Extension number + password) and dial *.

3. Place the Trunk call.

Single Line Telephone (SLT)
To temporarily change Extension COS:
1. Dial {Walking COS} feature code.
2. Dial the User Authorization Code (Extension number + password) and dial *.
3. Place the Trunk call.

Conditions

- The Walking COS applies the COS for only one call. Terminating the call will automatically return the Extension to the assigned COS. The user may reactivate Walking COS to place another call, or use FLASH to maintain the Walking COS.

Related WMS Menu

[To set Extension Password, Digit Restriction Class]

Data Management	> Extension Information > Number (DN) Information > DN Attribute > “Extension Password”, > Extension Information > Number (DN) Information > DN Attribute > “Digit Restriction Class”

[To assign Digit Restriction Map/Group]

Data Management	> System Feature Information > Digit Restriction Information > Digit Restriction Map
Data Management	> System Feature Information > Digit Restriction Information > Digit Restriction Table

[To assign Walking COS Feature code]

Data Management	> Numbering Plan Information > Feature Code > “Walking COS”

Related Features

- Dialing Restriction/COS
- System Service Time Mode Change
- Password

Hardware

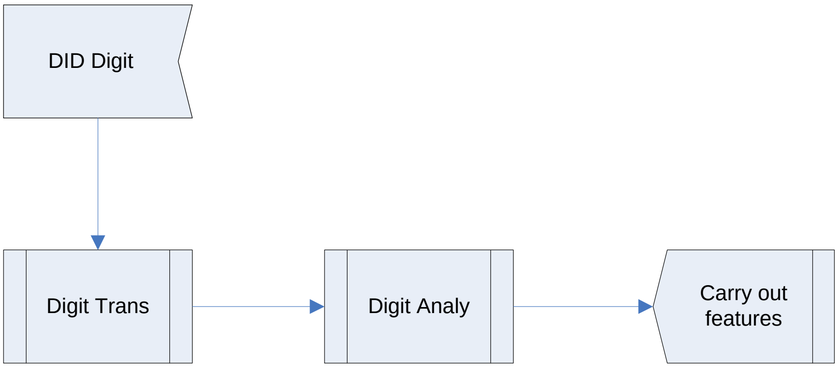
2.39 Direct In Dial (DID)

Description

The PSTN service, known as Direct In Dial (DID), sends the last several digits to the system so that calls may be routed directly to a specific Extension or system facility. DID service is available over analog, digital and packet network.

With analog DID service a special one-way analog Trunk (incoming calls only) is provided. DID gateway modules provide the special interface to receive digits from the analog Trunk.

DID digits are processed as in the following flow diagram.



- When DID digits are received, the system checks the Digit Conversion table, and converts matching digit strings. The digit conversion process varies according to Unconditional, Day/Night/Timed or LCR rule.
- Through digit analysis, the system recognizes the feature code of converted digit.
- The system performs the function of the feature code which is the destination of the DID digits. The destination can be an Extension, a Key Number Group, including an external VM, an external number, a voice announcement, a network Extension, a page or a Conference Room.

Before digit analysis, CID based routing service is supported according to the Incoming Route Options as follows:

- Not used – The CID routing service is not enabled.
- Used (Normal if there is no CID)
  - If there is CID, the system checks if the CID is assigned in the "Tenant ICLID Routing". The call routes according to the "Trunk Call Routing Table" only when the CID matches.
  - If there is no CID, the call is routed according to normal call processing flow.
- Used (Trunk Call Routing Table if there is no CID)
  - If there is CID, the system checks if the CID is assigned in the "Tenant ICLID Routing". The call routes according to the "Trunk Call Routing Table" only when the CID matches.
  - If there is no CID: The call is routed according to "Incoming Route Options > Trunk Call Routing Table".

If the converted DID digits are an Extension, the name or Key Number Group name is displayed.

Operation

Conditions

- If the Tenant ICLID Routing is assigned to the Trunk, the system compares the received Caller ID to the Tenant ICLID Routing table. If the Caller ID does not match an entry in the Tenant ICLID Routing table, the general DID call routing is applied.
- If there is no Caller ID provided, the call can be routed to a specific destination (Trunk Call Routing Table).
- Calls can be routed to designated destinations (Incoming Route Error Destination) if the DID number is not valid, if the Extension is busy, or if the Extension does not answer the call.

- For an Extension in a Pilot Hunt Group, DID calls are processed according to the group hunt process if the Extension is busy or does not answer the call.
- Group call picked-up and directed call pick-up are applied to DID calls.

Related WMS Menu

[To set System Time Zone]
System Management > System Time Zone
[To assign Tenant ICLID Routing]
Data Management > Tenant Information > Tenant basic Information > Tenant ICLID Routing
[To set CID Display and Generation]
Data Management > Extension Information > Number(DN) Information > CID Display and Generation
[To assign Trunk]
Data Management > Trunk Information > Trunk Basic Information > Trunk Attribute
[To set Incoming Route Group/Options/Error Destination]
Data Management > Trunk Information > Incoming Route Information > Incoming Route Group
> Trunk Information > Incoming Route Information > Incoming Route Options
> Trunk Information > Incoming Route Information > Incoming Route Error Destination
[To assign Digit Conversion Map/Group]
Data Management > System Feature Information > Digit Conversion Information > Digit Conversion Map
> System Feature Information > Digit Conversion Information > Digit Conversion Table
[To assign CID Conversion Map/Table]
Data Management > System Feature Information > CID Conversion Information > CID Conversion Map
> System Feature Information > CID Conversion Information > CID Conversion Table

Related Features

- Call Pick-up, Direct Call Pick-up
- Call Pick-up, Group Call Pick-up

Hardware

- MDTM(PRI/R2)
- VPCM

2.40 Direct In Line (DIL)

When the system detects an incoming ring from a Trunk that does not deliver DID digits, the system can route the call to a pre-defined destination.

2.40.1 Delayed Trunk Ring

Description

When an incoming Trunk call is detected, ring signals are sent to the members of the Trunk Call Routing Table after a time delay.

Operation

Conditions

- If no delay is assigned, the Extension will receive immediate ring.
- If no Extension or Key Number Group is assigned for immediate ring, the call will ring at Attendant.

Related WMS Menu

[To set Service Type, Ring Delay Count]

Data Management > System Feature Information > Trunk Call Routing Table > "Service Type"
> System Feature Information > Trunk Call Routing Table > "Ring Count before following Digit Information"

Related Features

- Key Number Group

Hardware

2.40.2 Trunk Ring Assignment

Description

When Trunks are set to DIL/DISA and an incoming call is detected on the Trunk, the call can be routed to designated Extensions. Separate call routing can be made for Day/Night/Timed mode.

Operation

Conditions

- DIL Routing Number of Trunk Attribute is highest priority to route the incoming call.
- Trunk ring assignment is determined by Trunk Call Routing Table defined in Incoming Route DIL/DISA Service.
- Separate ring can be assigned for Day/Night/Timed mode.

- When a Trunk is assigned to ring one Extension only and the Extension is in Call Foward, the incoming call will forward.

Related WMS Menu

[To route incoming trunk call]
Data Management > Trunk Information >Trunk Basic Information > Trunk Attribute > "DIL Routing Number"
> Trunk Information > Incoming Route Information > Incoming Route DIL/DISA Service > "Type (Day, Night or Timed)"
> Trunk Information > Incoming Route Information > Incoming Route DIL/DISA Service > "Trunk Call Routing Table"
[To set Time Zone No.]
Data Management > Trunk Information > Incoming Route Information > Incoming Route Options > "Time Zone No."
[To use DIL/DISA]
Data Management > Trunk Information > Trunk Basic Information > Trunk Attribute > "Service Type" ->: DIL/DISA
[To set Trunk Call Routing Table]
Data Management > System Feature Information > Trunk Call Routing Table

Related Features

- System Service Time Mode
- Built -in Auto Attendant

Hardware

2.40.3 Trunk Ring Detect

Description

For an analog Trunk, the Trunk Ring On and Trunk Ring Off time are used to determine if there is an incoming call on the Trunk. Ring for less than the Ring On time is not detected as an incoming call. If ring is not detected on the Trunk for longer than the Ring off time, the call is considered abandoned.

Operation

Conditions

- Ring On and Ring Off are assigned by system.
- Trunk ring detection is applied to analog trunk lines only.

Related WMS Menu

[To set Trunk Ring On Time, Trunk Ring Off Time]
System Management > GW Initial Information > MATM/LGCM > "Trunk Ring On Time"
> GW Initial Information > MATM/LGCM >, "Trunk Ring Off Time"

Related Features

Hardware

2.41 Direct Inward System Access (DISA)

Description

Each trunk line may be assigned for DISA service, which allows an incoming caller to gain access to the system resources and/or features. The system will answer the outside call and provide intercom dial tone or route the call based on Caller Controlled Routing (CCR). The DISA caller may then access the desired resource using dial codes.

The system can be set to perform authentication for the callers who wish to use the system resources to place outgoing calls.

Operation

DISA Caller

To access the system resource from the Trunk:

1. Place call to the DISA facility of the system.
2. At receipt of intercom dial tone, dial the desired number. Dial the password after receiving the password input prompt.

Conditions

- DISA callers can be routed to the Built-in Auto Attendant announcement in place of intercom dial tone. The announcement can be associate with the CCR table, or assigned to disconnect after playback announcement.
- DISA callers can be required to enter a User 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 User Authorization Code based on the DISA Retry count. Continued failure results in disconnect.
- DISA callers are subject to COS dialing restrictions. The COS of incoming Trunk line applies for this purpose.
- The transit service options are applied when DISA callers place outgoing calls.

Related WMS Menu

[To assign DISA Tone Access Feature code]

Data Management > Numbering Plan Information > Feature Code > "DISA Tone Access"

[To set Service Type, Digit Restriction Class]

Data Management > Trunk Information > Trunk Basic Information > Trunk Attribute > "Service Type: DIL/DISA"
> Trunk Information > Trunk Basic Information > Trunk Attribute > "Digit Restriction Class"

[To set Options for DISA]

Data Management > Trunk Information > Incoming Route Information > Incoming Route DIL/DISA Service > “Trunk Call Routing Table”
> Trunk Information > Incoming Route Information > Incoming Route DIL/DISA Service >, “DISA Transit”
> Trunk Information > Incoming Route Information > Incoming Route DIL/DISA Service >, “DISA Retry Count”
> Trunk Information > Incoming Route Information > Incoming Route DIL/DISA Service >, “Authorization code Usage for Trunk Seizure”

[To set Trunk Call Routing Table]
Data Management > System Feature Information > Trunk Call Routing Table

Related Features

- System Service Time Mode
- Digit Restriction / COS
- User Authorization Code
- Built-In Auto Attendant/Voice Mail

Hardware

2.42 Direct Station Select Console (DSS)

Description

The system allows an unlimited number of DSS/DLS boxes to be installed in the system. However, only up to 10 DSS/DLS Consoles may be associated with an Extension. Each button on the console can be assigned as a Trunk, Extension, or feature code.

Users or the administrator may change individual Flex button. The operation of the Flex button on a DSS/BLF Console is the same as the Flex button on an iPECS Multi-button phone.

Operation

Conditions

- DSS/DLS Consoles are programmed to work with a specific Extension.
- There is no restriction on the number of DSS/DLS consoles in the iPECS-CM system.
- The individual button assignments can be changed as with the iPECS Multi-button phone Flex buttons.

Related WMS Menu

[To assign Phone Flexible/DSS Button]
Data Management > Extension Information > Terminal Information > Phone Flexible Button
> Extension Information > Terminal Information > DSS Console Button

[To connect DSS Console]
Data Management > Extension Information > Terminal Information > DSS Console Connection

Related Features

- Flexible Button

Hardware

2.43 Directory Number (DN)

Description

In the iPECS-CM, Directory Numbers are not unique numbers matched with physical phones, but virtual numbers that can be shared by multiple phones. A DN can be assigned to multiple iPECS Multi-button phones or analog phones and used like one phone. Multiple DNs can be assigned to a single phone and user can have multiple Extension numbers on the same phone.

DN Types:

- SADN (Single-Assign Directory Number) can be assigned to only one phone
- MADN (Multi-Assign Directory Number) can be assigned to multiple phones
- M-DN (My Directory Number) is the number associated with the physical phone
- S-DN (Sub Directory Numbers), are DNs other than the M-DN assigned to a phone
- P-DN (Prime Directory Number) is the DN the user will access when going off-hook and subject to the DND/Call Forward features activated by the user.

2.43.1 Basic Functions

Description

Outgoing Call

A user may place a call after selecting a DN button, by picking up the handset without selecting a DN button, or through on-hook dialing. When a user selects to place a call without selecting a DN button, the system automatically selects the P-DN.

If the user selects the DN for an outgoing call, the tenant, phone number, digits restriction class and digits conversion class of the selected DN in the system are used. The selected DN is also used for charging and for CLI for the destination.

Incoming Call

Regardless of whether the phone is busy or not, calls can be assigned to the idle DNs at the same time. Ring is provided to an idle phone or an off-hook signal is sent to a busy phone to indicate that there are incoming calls.

The user can answer a call by selecting a DN button or by picking up the handset without selecting a DN button. If the user answers a call without selecting a DN button, the system automatically selects the P-DN.

When an Extension has an incoming call, the user may elect to place a call instead of answering the ringing call by simply pressing an idle DN button.

Hold
A busy DN can be placed on hold. Users of the phones with an appearance (Flex button) for the DN can answer the call by pressing the held DN button.

For SADN, since it can be assigned to a single phone, calls are put on hold individually, and for MADN, system hold is applied.

P-DN
When multiple incoming calls are ringing at an Extension, the P-DN will have priority in the answer queue if a DN button is not pressed; calls on the P-DN are answered first. A P-DN can be either M-DN or S-DN. If no P-DN is designated for phone, it is assigned to the first registered DN button.

A MADN can be assigned as the P-DN at multiple phones. In this case, if the state of the DN is changed at one phone, it is applied to the other phones with an appearance.

DN Button LED
Depending on the state of the DN, the LED of the DN button indicates the following information.

- The DN in use locally GREEN ON
- The DN in use remotely RED ON
- The DN on hold AMBER FLASH
- The DN incoming call GREEN FLASH
- No Ring(LCD Display): No ring signals is used, but the caller information is displayed.

Tenant/COS by number
Each DN is assigned to a Tenant. A phone can have DNs with different tenants. Also, each DN can be assigned with a different COS, which is often used to direct outgoing calls over specific DNs

Auto Hold
Answering an incoming call while engaged in a call, places the active call on hold and connects the incoming caller. The DN button displays the hold status and allows access to the call.

Branch Line
When a DN is assigned for Branch Line operation and appears on multiple phones, a user can access a busy DN establishing a conference.

Auto Dialing Digit (Immediate/Delayed)
A DN can be assigned Auto Dialing. When assigned, if a user selects an Auto Dialing DN, the system dials the number. If assigned for Auto Dial Pause, the system will delay for the assigned pause time. The user may dial before the expiration of the pause time and will deactivate the Auto Dialing feature.

Ring Options
A DN that appears on multiple phones can be assigned different ring options at each phone. Ring options include Immediate Ring, Delayed Ring and No Ring:

- Immediate Ring: The system sends the ring signal upon receiving an incoming call.
- Delayed Ring: Ring signal is sent after the delay timer.
- No Ring: No ring signal is used.

When ringing, the LED will flash to indicate the incoming call on the DN.

Operation

iPECS Multi-button Phone or SLT
To place a call:

1. Lift the handset, or select a DN.
2. Dial the desired number.

Note that the system will select the P-DN if the user does not select a DN prior to going off-hook.. If the user selects the DN, the tenant, phone number, digit restriction class and digits conversion class of the selected DN are used. The selected DN is also used for charging and for CLI for the destination.

To answer an incoming call to a specific DN:

1. Lift the handset, or select the ringing DN.

Note that the user may select an idle DN to place a call.

To place a call on hold:

1. While on a call, press the Hold button; the call can be retrieved by pressing the appropriate DN button.

Note that for SADN, calls are put on individual hold and, for MADN, system hold is applied.

To activate Auto Hold:

1. While on a call, press the DN with an incoming call, the active call is placed on hold automatically. The held call can be accessed by pressing the associated DN button.

Conditions

- Directory Numbers must comply with the system Extension numbering plan (i.e., up to eight (8) digits).
- Up to 65,536 Directory Numbers can be created regardless of the system capacity.
- A single Directory Number can be assigned to up to 30 phones.
- A single phone can be assigned with multiple Directory Numbers to the maximum number of Flex buttons available, including any DSS Console.

Related WMS Menu

[To set Numbering Plan]

Data Management > Number Plan Information > Numbering Plan
[To create Number]
Data Management > Extension Information > Terminal Information> Terminal Attribute
[To assign DN to Phone]
Data Management > Extension Information > Terminal Information> Phone Flexible Button
[To set DN Seize Method]
Data Management > Extension Information > Terminal Information> Terminal Option > "Automatic Call Hold by other DN Button"
[To set DN options]
Data Management > Extension Information > Number(DN) Information > DN Attribute
[To query Extension which use DN & set Ring Option]
Data Management > Extension Information > Number(DN) Information > DN Member
[To set Feature of DN]
Data Management > Extension Information > Number(DN) Information > DN Feature Registration
[To set LED Color & Flash]
Data Management > Tenant Information > Tenant Basic Information> Phone LED Control

Related Features

Hardware

2.43.2 Additional Functions

The features described below can be implemented by the application of the DN function.

System Password
If a DN and the password are shared by multiple users, the system password function can be used.

Executive/Secretary
The DN of an Executive can be assigned for delayed ring at the Executive phone and immediate ring at the Secretary. A second DN can be assigned to the Secretary with the Force Handsfree feature and Auto Dialing to a third DN at the Executive phone. This allows the Secretary to answer the Executive calls and then notify the Executive of a call on the Executive DN

If the Secretary does not answer, after the ring delay timer expires, both the Executive and Secretary phone will ring.

Individual Hold
A SADN user can place the call on hold such that others will not see the call on hold. A call on Individual hold cannot be changed to system hold.

System Hold
A MADN can be placed on hold. When placed on hold other users will see and be able to access the held call.

Linked Station Pairs
A single DN can be assigned to up to 30 phones. If this number is assigned as the P-DN of multiple phones, the phones act as a single Extension. If the number is in use by a phone, the DN cannot be used by other phones. Users can place or answer calls with other DNs assigned to the phones.

Mobile Extension
Up to five (5) external numbers can be assigned to a DN. If one of the external numbers is used as mobile station, the LED of the DN indicates busy. Other features are the same as those of Linked Station Pairs.

Call Coverage
A DN can be assigned at a phone to "Allow only Incoming call". In this case, if the DN is assigned for Call Coverage the user can answer calls on the DN.

Auto Trunk Seizure
The Auto Dialing feature associated with DNs can be configured to auto dial the Trunk access code and other digits, such as the Call Forward code to automatically forward calls.

One Time DND
If a phone receives ring, the user can activate One-time DND by pressing the DND button. The phone stops ringing.

Off-Hook Signaling
When on a call, the user receives off-hook signaling, muted ring over the speaker and flashing LED, for incoming calls.

Hot/Warm Line
An Extension or external number can be defined as the DN Auto Dial digits for a DN. When the user goes off-hook, the call is placed to the Extension or external number as a Hot Line. If the Delay timer is set, the call is placed after the delay as a Warm Line. This permits the user to access a different DN or dial digits to override the Auto Dial feature.

UCD DND
UCD DND activates DND for the calls incoming to a specific key number only. To use this function, DNs A and B are assigned to a single phone, assign A to the key number (A) and B to the key number (B). To apply DND for the calls to the key number (B), set DND for the line B only.

Presence Lamp

This function is linked with the Executive/Secretary function. If a secretary presses the Presence lamp button, the lamps for this function on other phones turn on to indicate that the executive is in room . To use this function the same DN is assigned to other phones, and DND feature code must be set as the Auto Dialing digit. When the secretary presses this button, the LED turns on to indicate DND. If the secretary presses this button again, DND is deactivated and LED is off. In addition, “Deny All Outgoing Call”.

Operation

Conditions

Related WMS Menu

Related Features

- DN/ Basic Functions

Hardware

2.44 Disable Hook-Flash

Description

This function is used to restrict use of hook-flash at Single Line Telephones. Hook-flash is not allowed for users with this feature activated.

Operation

Conditions

Related WMS Menu

[To disable Hook Flash of SLT]

Data Management	> Extension Information > Number (DN) Information > DN Feature Allow/Deny> “Hook Flash Restriction”

Related Features

Hardware

2.45 Do Not Disturb (DND)

Description

An Extension allowed Do Not Disturb, can be placed in DND to block incoming ring for Trunk, and Intercom calls, transfers and paging announcements.

Operation

Multi-button Phone

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

1. Press the [DND] button;
2. Dial the P-DN number only when DND Register Confirm is set to Use
3. The [DND] LED turns on.

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

1. Press the [DND] button when in an idle state; the [DND] LED turns off.

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

1. Press the S-DN button.
2. Enter the {DND Register/Cancel} feature code or press the DND soft button;
3. Dial the P-DN number only when DND Register Confirm is set to Use.
4. The S-DN button LED blinks.

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

1. Press the DN button.
2. Enter the {DND Register/Cancel} feature code; the S-DN button LED turns off.

Single Line Telephone (SLT)

To activate DND:

1. Lift the handset.
2. Enter the {DND Register/Cancel} feature code;
3. Dial the P-DN number only when DND Register Confirm is set to Use.
4. Confirmation tone is received.

To deactivate DND:

1. Lift the handset.
2. Enter the {DND Register/Cancel} feature code; confirm tone is received.

To deactivate DND forcibly:

1. Lift the handset.
2. Enter the {Force DND Cancel} feature code; confirmation tone is received.
   Or
   Enter {All Feature Cancel} feature code.

Conditions

- An Extension will receive error tone if not allowed access to DND.

- When an Extension user presses the [DND] button while receiving ring signal, the caller receives Ring Back tone. For the Extension user only the ring is blocked and LED will blink showing that an incoming call is waiting. This waiting call can be answered by pressing the blinking LED button. DND is applied for the incoming call only.
- When calling an Extension in the DND state, an iPECS Multi-button phone display indicates the DND state.
- When DND Register Confirm is set to Use, P-DN number is used for confirmation regardless of destination DN for DND registering.

Related WMS Menu

[To enable DND]

Data Management	> Extension Information > Number(DN) Information > DN Feature Allow/Deny >"DND"

[To change LED for DND indication]

Data Management	> Tenant Information > Tenant Basic Information> Phone LED Control > "[DN Button]: DND"

[To set DND feature code]

Data Management	> Numbering Plan Information > Feature Code > "DND Register/Cancel (Toggle)"
Data Management	> Tenant Information > Tenant Basic Information > Tenant Attribute > "DND Register Confirm"

Related Features

Hardware

2.45.1 Timed Do Not Disturb (DND)

Description

This feature allows users to set DND for range of designated time. When the selected range is not reached, DND function is not worked.

Operation

iPECS Multi-button Phone

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

1. Enter the {Timed DND Register/Cancel (Toggle)} feature code.
2. Dial the P-DN number only when DND Register Confirm is set to Use.
3. Enter the type of DND (1: Once, 2: Daily, 3: Monday~Friday, 4: Monday~Saturday, 5: Date).
4. Enter the start time and end time. (HH:MM~HH:MM, 8 digits in 24 Hour Format)
5. If DND type is date, enter six (6) digits (YY/MM/DD) for year, month and day.
6. Press the [SAVE] button.

To deactivate DND for P-DN (Prime Directory Number):
1. Enter the {Timed DND Register/Cancel (Toggle)} feature code; the [DND] LED turns off.

To activate DND for S-DN (Sub Directory Number):
1. Press the S-DN button.
2. Enter the {Timed DND Register/Cancel (Toggle)} feature code.
3. Dial the P-DN number only when DND Register Confirm is set to Use.
4. Enter the type of DND (1: Once, 2: Daily, 3: Monday~Friday, 4: Monday~Saturday, 5: Date).
5. Enter the start time and end time. (HH:MM~HH:MM, 8 digits in 24 Hour Format)
6. If DND type is date, enter six (6) digits (YY/MM/DD) for year, month and day.
7. Press the [SAVE] button; the S-DN button LED blinks.

To deactivate DND for S-DN (Sub Directory Number):
1. Press the S-DN button.
2. Enter the {Timed DND Register/Cancel (Toggle)} feature code; the S-DN button LED turns off.

<u>Single Line Telephone (SLT)</u>
To activate DND:
1. Lift the handset.
2. Enter the {Timed DND Register/Cancel (Toggle)} feature code.
3. Enter the type of DND (1: Once, 2: Daily, 3: Monday~Friday, 4: Monday~Saturday, 5: Date).
4. Enter the start time and end time. (HH:MM~HH:MM, 8 digits in 24 Hour Format)
5. If DND type is date, enter six (6) digits (YY/MM/DD) for year, month and day.
6. Dial '*'; a confirmation tone will be heard.

To deactivate DND:
1. Lift the handset.
2. Enter the {Timed DND Register/Cancel (Toggle)} feature code; confirm tone is received.

To deactivate DND forcibly:
1. Lift the handset.
2. Enter the {Force DND Cancel} feature code; confirmation tone is received.
   Or
   Enter {All Feature Cancel} feature code.

<u>Conditions</u>
- An Extension will receive error tone if not allowed access to DND.
- Five (5) DND options are supported: Once, Daily, Monday~Friday, Monday~Saturday, or a Date.
- DND registration for Daily, Monday~Friday or Monday~Saturday is maintained until it canceled by the user.
- DND registration for Once or Date mode is canceled automatically after end time.

Related WMS Menu

[To set Timed DND feature code]

Data Management	> Numbering Plan Information > Feature Code > “Timed DND Register/Cancel (Toggle)”

[To set Timed DND]

Data Management	> Extension Information > Number(DN) Information > DN Feature Registration > “Timed DND Service Type”
Data Management	> Extension Information > Number(DN) Information > DN Feature Registration > “Timed DND Start Time”
Data Management	> Extension Information > Number(DN) Information > DN Feature Registration > “Timed DND End Time”

Related Feature

Hardware

2.46 Door Open

Description

The CM-S1K hardware is equipped with four (4) relays that activate External Control Contacts. The contact can be assigned to activate as a Door Open Contact, activating a Door-lock release mechanism. The assigned user may dial the Door Open Code to activate the contact.

Operation

To assign a {Door Open} feature code

1. In WMS, add the {Door Open} code to Feature Codes.

To assign a {Door Open} authority

1. In WMS, change the {Door Open} feature allowance in DN Feature Allow/Deny menu.

To assign a {Door Open} button

1. [PGM] + {Flex} + {Door Open} feature code + LCM Site Index(3Digit)+Relay no.(digit:1~4) + [Save]
2. LCM Site Index means the index of LCM using S1K and it is displayed in Overall CM Server Configuration menu.
3. Controlling CCM’s Door Open Contact, LCM site index should be 000.

To activate the Door Open Contact

1. Lift handset or press [Speaker] button by the authorized DN.
2. Enter {Door Open} feature code, the LCM Site Index and Relay Number).
3. Hang-up to return to idle.
   Or,
1. Lift handset or press [Speaker] by the authorized DN.
2. Enter the {Door Open} Flexible button
3. Hang-up to return to idle.

Conditions

- The Door Open time can be assigned from 500 ms to 10 seconds.
- If the Door Open Flexible button is used, the user may activate this feature while idle, busy or ringing. , An SLT or SIP phone may activate this feature only in the Idle state.

Related WMS Menu

[To assign Door Open feature code]
Data Management > Numbering Plan Information > Feature Code>"Door Open"
[To allow Door Open]
Data Management > Extension Information > Number (DN) Information > DN Feature Allow/Deny > "Door Open"
[To assign Door Open Time]
System Management > MISC Detail Information > "Door Open Time"

Related Features

Hardware

- CM-S1K

2.47 DTMF Tone Control

Description

DTMF (Dual Tone Multi-Frequency) signals are used with the Trunk assigned for DTMF signaling. The duration of the DTMF signal can be adjusted.

Operation

Conditions

- The system mutes user's voice transmission to reduce interference while sending DTMF tones.
- The allowable time between digits (Inter-digit time) can be set.

Related WMS Menu

[To set DTMF sending time for Outgoing Route]
Data Management > Trunk Information > Outgoing Routing Information > "Outgoing Route DTMF Sending Time"
[To set Inter Digit & Wait Seize Ack Time]
Data Management > Trunk Information > Outgoing Route Information > Outgoing Route Inter Digit Time > "Outgoing Inter Digit Time", "Wait Seize Ack Time"

Related Features

- Dial Pulse Signaling

Hardware

2.48 Emergency Call

Description

Users can place Emergency calls using the Emergency Code, regardless of the Extension Digit Restriction (COS).

Operation

Conditions

・If user is Hotel Guest Station, Emergency Call is notified to IPECS Hotel Attendant.

Related WMS Menu

[To assign Emergency Code]
Data Management > System Feature Information > Emergency Code Table

Related Features

- Digit Restriction/COS

Hardware

2.49 Executive/Secretary Service

Description

Extension users can be assigned as Executive/Secretary pairs. Up to 500 Extensions can be assigned to Executives and Secretaries, respectively. Executives can forward Trunk, internal, and forward calls to the Secretary unconditionally based on the system configuration. When an Executive calls another Executive, the call is forwarded directly to the called Executive according to "Call Option between Executives" of the Executive Group or if the Secretary is in DND. With "Secretary Call option" assigned in the Executive Group, users can select the way Secretaries calls are placed. If "Direct Call to Executive in other Group" is enabled in the Secretary Group, Secretaries can call Executives in other groups.

When a call between Executives is allowed to any Executive, the user can call other Executives directly regardless of Executive/Secretary Forward. In addition, Executive/Secretary Service can be configured using Sub Directory Number (S-DN).

Operation

Conditions

・An Executive can have multiple Secretaries, and a secretary can be assigned to multiple Executives. If the first Secretary is busy when a call for the Executive is received, the call is transferred to the next Secretary.
・If a Secretary Extension is busy, the caller receives a busy tone.
・A Secretary may override DND of an Executive.

- If the Executive sets call forward to an Extension other than the Secretary, calls are forwarded according to the setting.
- Internal or Trunk calls to Executives can be forwarded to the Secretaries.
- Callers leave messages to the Executive or the Secretary according to the “Extension Message Wait Destination” option of the Executive group.
- The Attendant calls can be direct to the Executives according to the “Direct Call from Attendant to Executive” option.
- The maximum number of Secretary members for an Executive group is ten (10), and the maximum Executive members for a Secretary group is ten (10).

Related WMS Menu

[To set Executive/Secretary Group & access option]
Data Management > Extension Information > Group Information > Executive Group, Group Information Secretary Group
Data Management > Extension Information
[To allow or deny access between Executives]
Data Management > Extension Information > Group Information > Access Control between Executives
[To allow Attendant to call Executive]
Data Management > Tenant Information > Tenant Basic Information> Tenant Attribute > “Direct Call from Attendant to Executive”

Related Features

Hardware

2.50 Extension User Program Code

Description

iPECS Multi-button phone users can program an array of features, access status information and assign special feature codes to Flex buttons. The following table shows the range of features available through the Extension User program. The Extension User program can also be configured in the Extension User Web portal. Refer to the Extension User Web Manual.

CODE	DESCRIPTION	ENTRIES
1	User Organizer	
11	Answer mode (H/T/P)	1:H, 2:T, 3:P
12	Station Name Program	
13	Set wake-up	Index (1~5)
14	Reset wake-up	Index (1~5)
15	Language program	00~17
16	LCD date mode	DDMM/MMDD
17	LCD time mode	12H/24H
18	Set back light	0~3
2	Ring	

CODE	DESCRIPTION	ENTRIES
21	Intercom Ring Type	1~8
22	Trunk Ring Type	1~8
23	Ring download	5~8 + 0~9
3	COS/Password	
31	Temporary COS	Auth code
32	Retrieve COS	Auth code
33	Walking COS	Auth code
34	Register password	
35	Call Log Protect	
36	SMS Message Protect	
4	Multi message	
41	Preselected Message Program	
42	Set User Message	
43	Send SMS Message	
44	Received SMS Message	
5	Mobile Ext./conference room	
51	Mobile-Ext. Enable	1~5
52	Mobile-Ext. Number	1~5
53	Mobile-Ext. CLI	1~5
54	Create Conference Room	
55	Delete conference room	
6	Speaker/headset	
61	Speaker/Headset	0:S, 1:H, 2:B
62	Headset Ring	0:S, 1:H, 2:Both
63	Bluetooth	
7	Station ICLID	
71	Register Station ICLID	
72	View Station ICLID	
8	Keyset information	
81	View IP address	
82	View MAC address	
83	View keyset version	
80	Network Settings	
9	System Feature	
91	Set system date	
92	Set system time	
93	Day/Night	001~254
94	Monitor Conference Room	
95	Forced Delete Conference Room	
96	Listen VM Announcement	
97	Record VM Announcement	
98	Room Prepaid Money	

The Extension User program menu is shown in the LCD of iPECS Multi-button phones to guide users in programming features and functions. The [VOL UP]/[VOL DOWN] button is used to scroll menu items, and the dial pad is used to enter a selection. The codes are also used to assign functions or features to a Flex button.

Operation

iPECS Multi-button phone

To assign an Extension user program code to a FLEX button:

1. Press the [PGM] button. The Extension User program menu is displayed.
2. Press the desired Flex button.
3. Dial 2 for a User Program code, and dial the {Station Program Mode} feature code and the desired Extension User program code. You may need to enter further data depending on the function. Note the {StationProgram Mode}feature code is defined in Feature Code menu of WMS.
4. Press the [SAVE] button.

To dial the Extension User program code directly:

1. Press the [PGM] button. The Extension User program menu is displayed.
2. Use the [VOL UP]/[VOL DOWN] button to find a menu, or dial the desired Extension User program code. Additional entries may be required depending on the function.

Conditions

- If Bluetooth is available for the phone, 'Headset/Bluetooth' is indicated under Code "6".

Related WMS Menu

Related Features

- Flexible Button
- Pre-defined Text Display/Pre-defined & Custom Text display Messages
- Headset Compatibility

Hardware

- iPECS Multi-button Phone

2.51 External Voice Mail System (VMS) Interface

Description

iPECS-CM provides interfaces for several types of external Voice Mail System (VMS). A Voice Mail service can be provided based on the user's status (DND, No Answer, Call Forward) by sending information about the user's status to the VMS.

The following interfaces are supported:

- Digital voice (E1) with LAN, MSI protocol
- Analog voice with LAN, MSI protocol
- Analog voice with in-band DTMF protocol

Operation

Conditions

・iPECS CM can be connected to eight (8) external Voice Mail systems and up to 250 voice lines per VMS.

Related WMS Menu

[To set VMS connection & DTMF information]
Application Interface > VMS Information > VMS Connection, In-band DTMF Command
[To set VMS number]
Data Management > Extension Information > Number(DN) Information > Voice Mail Information > "Voice Mail Device"

Related Features

- Built-In Auto Attendant/Voice Mail

Hardware

- External VMS

2.51.1 In-band DTMF Signaling

Description

The system employs in-band signaling for communication with external Auto Attendant (AA)/Voice Mail systems (VMS). When a call is routed to an AA/VMS connected to SLT ports, the system sends the AA/VMS DTMF signals to indicate the reason for the call. DTMF digit strings are assigned to various functions allowing the AA/VMS to respond appropriately to each call.

Operation

Conditions

・The In-band DTMF table can be configured independently for each AA/VMS.

Related WMS Menu

[To set DTMF information for VMS]
Application Interface > VMS Information > In-band DTMF Command

Related Features

Hardware

- External AA/VMS System

2.51.2 MSI Interface

Description

The system employs the MSI (Manufacture Specific Interface) protocol to communicate with the external AA/VMS system. When a call is routed to an AA/VMS SLT port, the system will send the AA/VMS MSI messages for the call.

Operation

Conditions

Related WMS Menu

[To set VMS type]

Data Management > Extension Information > Terminal Information> Terminal Attribute > "Terminal Sub Type"

[To connect VMS]

Third Party App Interface > VMS Information > "VMS Connection"

Related Features

Hardware

- External AA/ VMS

2.52 Fax Bridge

Description

Fax Bridge service supports fax transmission without fax representative number. Customer can receive fax easily without another DID fax number (One number service)

CM System detects Fax tone automatically, when user assigned fax bridge device number. If fax toned detected, the call routed to fax device automatically. If fax tone is not detected during fax tone detection timer, incoming call to original called number

Operation

CM System detects Fax tone automatically, when user assigned fax bridge device number. If fax toned detected, the call routed to fax device automatically. If fax tone is not detected during fax tone detection timer, incoming call to original called number

Conditions

- CM System detects Fax tone automatically, when user assigned fax bridge device number. If fax toned detected, the call routed to fax device automatically in WMS (Voice Mail Information).
- Fax bridge service Trunk type : E1 DTMF, PRI, SS7, H323, SIP Trunk.

Related WMS Menu

[To set use of Fax Bridge in System]

Data Management > Number (DN) Information > Voice Mail Information > Fax Bridge service

[To set Fax Bridge detection timer]

Data Management > Number (DN) Information > Voice Mail Information > Fax Bridge Tone Detect Timer

[To define Fax Bridge Number]

Data Management > Number (DN) Information > Voice Mail Information > Fax Bridge Device

Related Features

Hardware

2.53 Flexible Button Assignment

Description

Numbers (internal or external telephone numbers), Feature Codes and Fixed button codes (REDIAL, PTT, FLASH, etc) can be assigned to the Flexible buttons of the iPECS Multi-button phones for one-touch feature access.

Operation

To assign the fixed button to a Flex button:

1. Press the [PGM] button.
2. Press the desired FLEX button.
3. Dial 1.
4. Select the desired function with [VOL UP]/[[VOL DOWN]
5. Press the [HOLD/SAVE] button.

To assign a number to a Flex button:

1. Press the [PGM] button.
2. Press the desired Flex button.
3. Dial 2.
4. Dial the external number with Trunk Access code, Extension number or feature code.
5. Press the [HOLD/SAVE] button.

To delete the function from a Flex button:

1. Press the [PGM] button.
2. Press the desired Flex button.
3. Dial 0.

To place a call with the Speed Dial number assigned to the Flex button:

1. Lift the handset or press the [Speaker] button.
2. Press the desired Flex button.

Conditions

・This feature is available for the iPECS Multi-button phone users only.
・Only the Fixed button that is not assigned on the phone can be assigned to the Flexible button.

Related WMS Menu

[To assign flexible button]
Data Management > Extension Information > Terminal Information > Phone Flexible Button
[To assign & check feature code]
Data Management > Numbering Plan Information > Feature Code

Related Feature

- Direct Station Select Control (DSS)

Hardware

- iPECS Multi-button Phone

2.54 Flexible LED Flashing Rate

Description

The flashing rate of the various buttons of the iPECS Multi-button phone may be adjusted to meet the needs of the customer. The system supports 15 different flash rates for each feature.

Operation

Conditions

Related WMS Menu

[To change LED color & flash rate]
Data Management > Tenant Information > Tenant Basic Information > Phone LED Control

Related Features

Hardware

- iPECS Multi-button phone

2.55 Forced Handsfree

Description

Users can change the Intercom Call Answer Mode from the Tone ring mode (T mode) to Voice announcement with handsfree reply mode (H mode)

Operation

To change the Intercom Call Answer Mode:

1. Place an internal call.
2. Dial '#' while hearing Ring back tone, the Intercom Call Answer mode is changed from Tone ring to Voice Announce mode.

Conditions

・The called Extension must be iPECS Multi-button phone.
・If the answer mode is changed, "No Answer" is not applied.

Related WMS Menu

[To answer Handsfree]
Data Management > Extension Information > Number(DN) Information > DN Member.
[To use Forced Handsfree]
Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny >" Forced Handsfree"

Related Features

- Intercom Call Answer Mode
- Linked Station Pairs

Hardware

2.56 Forward Group

Description

If an Extension or a Key Number is a member of Forward Group, a call can be routed to Forward Group according to the defined conditions. Call condition is always Unconditional.

Operation

Conditions

・Calls cannot be routed to an Extension in DND or Call Forward.
・A Forward Group can have up to 500 members.
・Up to 200 Forward Groups can be supported.

Related WMS Menu

[To assign Pick-up Group]
Data Management > Extension Information > Group Information > Forward Group

Related Features

- Do Not Disturb
- Call Forward

Hardware

2.56.1 Forward Group Call Forward

Description

A Forward Group can register Call Forward to reroute incoming calls to any member of the group to another Extension, Key Number Group, or outside telephone number.

Operation

To activate Call Forward:

1. Lift the handset or press the [Speaker] button.
2. Enter the {Forward Group Register} feature code.
3. Enter the destination number. For an external destination, dial the outside telephone number including Trunk Access code.
4. Dial * or Press [HOLD/SAVE] button to save.

To deactivate Call Forward:

1. Lift the handset.
2. Enter the {Forward Group Cancel} feature code. If allowed, confirmation tone is provided.

Conditions

- To activate the Forward Group feature, the DN of the Extension must be included in a specific Forward Group.
- Call forward set by a Forward Group has priority over any other Call Forward.

Related WMS Menu

[To set Pilot Hunt Group Call Forward]
Data management > Extension Information > Group Information > Forward Group
[To assign Pilot Hunt Group Forward feature code]
Data Management > Numbering Plan Information > Feature Code > “Forward Group Register”
> Numbering Plan Information > Feature Code > “Forward Group Cancel”

Related Features

- Call Forward
- Do Not Disturb (DND)
- Preset Call Forward

Hardware

2.57 Group Listening

Description

Every iPECS Multi-button phone has a built-in Speakerphone. The user can activate the speaker after placing a call in the handset mode without activating the microphone. This feature allows local users to hear the conversation through the speaker without being heard over the microphone.

Operation

While on a call using the handset:

1. Press the [Speaker] button. The speaker is activated. The microphone is muted while the handset is off-hook.

Conditions

- While using the Speakerphone, lifting the handset will turn off the Speakerphone. To activate Group Listening, the [Speaker] button must be pressed while the handset is off-hook.
- While in Group Listening Mode, pressing the [Mute] button will cause the TX path from the handset to be muted. However, the voice is still heard over both the handset receiver and the speaker.
- If full Speakerphone operation is desired, place the handset on-hook to cancel Group Listening Mode.
- This feature is not available for LIP-8004D phones.

Related WMS Menu

[To use Group Listening]
Extension Information > Terminal Information> Terminal Attribute > “Group Listening Use”

Related Features

- Speakerphone
- Mute

Hardware

- iPECS Multi-button phone

2.58 Headset Compatibility

Description

An industry standard headset can be connected to an iPECS Phone in place of or, in addition to, the handset. The Extension is then 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.

Operation

To change the operation mode from Speakerphone to Headset:

1. Press the [PGM] button.
2. Dial 61, the Speaker/Headset Program code; current mode is indicated in the LCD.
3. Dial 0 for Speakerphone or 1 for Headset
4. Press the [SAVE] button.

To change the device to receive ring signals:

1. Press the [PGM] button.
2. Dial 62, the Headset Ring Program code; current setting is displayed.
3. Dial 1 for Speakerphone Only, 2 for Headset Only, or 3 for Both.
4. Press the [SAVE] button.

To receive/place calls in the Headset mode:

1. Press the [SPEAKER] button.

Conditions

- The intercom signaling mode can be set in the Headset mode as in the Speakerphone mode.
- If an Extension is in Headset mode, the system does not monitor hook-switch status.

Related WMS Menu

[To set Headset ring Mode]

Data Management > Extension Information > Terminal Information > Terminal Attribute > "Headset Ring Mode"

Related Features

- Speakerphone

Hardware

2.59 Hot/Virtual Desk

2.59.1 Hot Desk

Description

Hot Desk enables users to select Extensions in a flexible manner through a process of login/logout rather then having fixed Extensions. This may be particularly useful in a call center environment, where a desk can be shared by multiple users.

When no user is logged in, the Hot Desk Extension has no specific Extension number, and calls to any logged out user will be forwarded to the destination designated at the time of logout. After login, users may use the system features and resources based on the configuration of the user's Extension number.

Operation

To login to a Hot Desk from an iPECS Multi-button phone:

1. Enter the {Hot Desk Login/Logout} feature code.
2. Enter the Extension number and password.
3. Dial * or press the [SAVE] button.

To logout from a Hot Desk:

1. Enter the {Hot Desk Login/Logout} feature code.
2. Enter the phone number of the destination for call forward.
   Or,
   Dial # if call forward is not required. A number is not required if you wish to have calls forwarded to the destination used in the previous logout.
3. Dial '*' or press the [SAVE] button.

Conditions

- Only an iPECS Multi-button phone may be used as a Hot Desk phone.
- Hot desk Extensions can be programmed to logout an active user automatically if no action has been taken by the agent in the Hot Desk Agent Auto Log-off Time.
- An active (logged in) agent can login to another inactive Hot Desk. However, this will log the agent off the previously active Hot Desk.
- A user may only logout from the Extension where they login; the user will hear an error tone if attempting to logout from any other Hot Desk Extension.
- The Flex button map of a Hot Desk Extension is fixed depending on the type of the phone.
- When a user logs in to a Hot Desk, the Flex buttons may be changed to match to the user settings. Flex button will return to the previous value when the user logs out; the extended button map (DSS console), however, does not change.
- Extension messages, call logs and SMS will remain unchanged when users log in to or out from a Hot Desk.
- The number of Hot Desk Extensions is limited by the system's Extension capacity, and the number of Hot Desk users is limited by the DN capacity; each Hot Desk Extension and Hot Desk user requires an Extension and a DN in the system respectively.

Related WMS Menu

[To use terminal for Hot desk]

Data Management	> Extension Information > Terminal Information> Terminal Attribute > "Hot Desk Terminal Use"

[To set options for Hot desk]

Data Management	> Extension Information > Number(DN) Information > DN Attribute > "Hot Desk Terminal Use"
	> Extension Information > Number(DN) Information > DN Attribute > "Extension Password"
	> Extension Information > Number(DN) Information > DN Attribute > "Hot Desk Log-in Number"
	> Extension Information > Number(DN) Information > DN Attribute > "Forced-Log off"

[To set Hot Desk Agent Auto Log-Off Time]
Data Management >Tenant Information > Tenant Time Information > Tenant Timer >" Hot desk Agent Auto Log-Off Time (Hour)"
[To set feature code for Hot Desk]
Data Management > Numbering Plan Information > Feature Code > "Hot Desk Login/Logout"

Related Features

Hardware

- Multi-button Phone

2.59.2 Virtual Desk

Description

Virtual Desk enables users to select the LIP Extension M-DN in a flexible manner through a process of login/logout on any LIP Phones. After Virtual Desk login, the Phone will act as the original phone, so all features of original phone are supported.

Operation

To login to a Virtual Desk

1. Enter the {Virtual Desk Login/Logout} feature code.
2. Enter the Extension number (M-DN) and password.
3. Dial "*" or press [SAVE] button. The original phone goes to an unused state.

To logout a Virtual Desk

1. Enter the {Virtual Desk Login/Logout} feature code.

To login to a Virtual Desk

1. Enter the Extension number and password.
2. Dial "*" or press [SAVE] button.

To verify login status of phone

1. Press [PGM] and [PGM]. The user's M-DN of login and M-DN of phone are displayed.

Conditions

- The Virtual Desk feature is possible only between iPECS Multi-button Phones located in the same CCM or LCM.
- When a user logs out, the M-DN of the phone is returned unless this M-DN is not used on another phone.
- If the desired login M-DN is in use, the user cannot log in to the Virtual Desk.
- A Hot Desk terminal cannot use Virtual Desk feature.

Related WMS Menu

[To assign a Virtual Desk Login/Logout feature code]

Data Management > Numbering Plan Information > Feature Code > "Virtual Desk Login/Logout"

Related Features

- Hot Desk

Hardware

- iPECS Multi-button

2.60 Hot/Warm Line

Description

An Extension can be assigned to access a pre-assigned number (Hot Line/Warm Line) automatically upon an off-hook state. Warm Line may be configured, in which case, the Extension user receives normal intercom dial tone for the Hot Line delay time and may dial based on the assigned dialing restrictions. When the delay timer expires, if the user has taken no action, the Hot Line call is placed.

Operation

To activate the Hot Line function:

1. Lift the handset or press the [Speaker] button. Immediately, the pre-assigned number is dialed automatically.

To activate the Warm Line function:

1. Lift the handset or press the [Speaker] button and take no action for the Auto-Dial Pause Time.
2. The pre-assigned number is dialed automatically after the Auto-Dial Pause Time.

Conditions

- The destination for the Hot/Warm Line is configured as the Auto Dialing Digit.
- To assign a Hot Line, the Auto-Dial Pause Time must be set to 0. To assign a Warm Line, the Auto-Dial Pause Time must be set to over 1 sec.
- When the user lifts the handset or presses the [Speaker] button, the system will act as if the user had dialed the pre-defined digits after going off-hook.
- When Warm Line is set, the user must wait, taking no action until the the Auto-Dial Pause Time expires before the Warm Line is accessed. The user receives intercom dial tone during this period and may dial any valid numbering plan digit(s) or select a Flex button or feature button.
- If the Warm Line waiting time (Auto-Dial Pause Time) is longer than the Dial Tone Timer, the Warm Line feature will not operate properly.
- Hot/Warm Line is not available to SIP Phones.

Related WMS Menu

[To set option for Auto Dial]

Data Management > Extension Information > Number (DN) Information > DN Attribute > "Auto Dialing Digit",
> Extension Information > Number (DN) Information > DN Attribute > "Auto-dial Pause Time"

Related Features

- Directory Number

Hardware

- iPECS Multi-button phone

2.61 ICLID Routing

Description

The iPECS-CM system can route Trunk calls based on ICLID (Incoming Calling Line ID). The system compares the received ICLID with the Tenant ICLID Routing Table and, if a match is found, routes the call to the destination assigned in the table. The destination can be a VPCM, an external VM announcement, an Extension or a Key Number Group.

Operation

Conditions

- For analog Trunks, which send the CID after the first ring cycle, CID Detect Time can be adjusted so that the system receives ICLID. After the CID Detect Time expires or upon receiving CID, the system immediately routes external calls.
- If the received ICLID has no matching entry in the Tenant ICLID Routing Table, the system routes calls to the destination by normal Trunk service.
- Up to 10 ICLID Routing entries can be assigned to each DN (Directory Number).
- Up to 300 ICLID Routing entries can be assigned to each tenant.

Related WMS Menu

[To adjust CID Detect Time for analog trunk]

Data Management > Trunk Information > Incoming Route Information > Incoming Route Options > "CID Detect Time"

[To use CID based Routing Service]

Data Management > Trunk Information > Incoming Route Information >Incoming Route Options > "CID based Routing Service"

[To set CID based routing destination]

Data Management > Tenant Information > Tenant Basic Information > Tenant ICLID Routing

[To set normal routing destination]

Data Management > Trunk Information > Incoming Route Information>Incoming Route Options > “Trunk Call Routing Table”
[To set Trunk Call Routing Table]
Data Management > System Feature Information > Trunk Call Routing Table

Related Features

Hardware

2.62 Individual Trunk Access

Description

The system may provide Individual Trunk Group access for Extensions allowed to access the group. A Trunk group may operate as follows according to the options configured.

- Busy Service: When all Trunks in a Group are in use, the caller receives busy and may be provided busy services.
- Normal Trunk Service: When all Trunks in a Trunk Group are in use, a normal Trunk group is accessed.
- Individual Trunk Service: When all Trunks of normal Trunk Group are in use, an Individual Trunk Group is accessed.

Operation

Conditions

- Up to 10 Individual Trunk access codes may be assigned per user.

Related WMS Menu

[To allow a DN to access Individual Trunks]
Data Management > Extension Information > Number (DN) Information > DN Attribute > “Individual Trunk Use”
[To assign an Individual Trunk Access code for a DN]
Data Management > Extension Information > Number (DN) Information > DN Individual Trunk Access Code
[To assign Individual Trunk Group]
Data Management > Trunk Information > Trunk Basic Information > Individual Trunk Group
[To check the Trunk serial number for an individual Trunk Group]
Data Management > Trunk Information > Trunk Basic Information > Individual Trunk Group Verify
[To allow the Attendant to use individual Trunk access]
Data Management > Attendant Information > Attendant Attribute >” Individual Trunk Use”

Related Features

Hardware

2.63 Intercom Call

Description

Extension users can place a call to other Extensions in the system by dialing the Extension number as defined in the Numbering Plan.

Operation

To place an internal call:

1. Lift the handset or press the [Speaker] button. You will hear intercom dial tone.
2. Dial the Extension number or, for iPECS Multi-button phones, press the desired {DSS} button.

Conditions

- If no action is taken before expiration of dial tone time (1st Dial Tone time), or if the interval between digits exceeds the inter-digit timer (Extension Button Input Time), error tone is provided.
- If the called Extension is busy, busy tone is provided. The caller may hang up the phone or use the Message Wait/Call-Back feature.
- For iPECS Multi-button phones users, consecutive intercom calls can be placed without the need to regain intercom dial tone (no need to hang-up) between calls; the user simply presses another {DSS} button.

Related WMS Menu

[Numbering Plan]
Data Management > Numbering Plan Information > Numbering Plan
[To assign DSS on flexible button]
Data Management > Extension Information > Terminal Information > Phone Flexible Button
[To set Dial/Busy/ Error Tone & tone time-out]
Data Management > Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > “1st Dial Tone”
> Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > “Internal Busy Tone”
> Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > Error Tone”
[To set Extension Button Input Time]
Data Management > Tenant Information > Tenant Time Information > Tenant Timer > “Extension Button Input Time”

Related Features

- Intercom Call Answer Mode
- Speakerphone
- Busy Lamp Field (BLF)
- Flexible Button Assignment
- Forced Handsfree

Hardware

2.64 Intercom Call Answer Mode

Description

An iPECS Multi-button phone can select the Intercom Call Answer Mode used for answering incoming intercom calls. Three answer modes are available.

- H - Call announce with Handsfree answerback
  When an intercom call is received, the phone is automatically answered after splash tone. Therefore, the user may respond to the caller without lifting the handset or pressing [Speaker] button.

- P - Call Announce with Privacy:
  When an intercom call is received, the phone is automatically answered with the microphone muted after Splash tone. To respond to the caller, the user must lift the handset or press the [Mute] button.

- T - Tone ring:
  An intercom call will cause the phone to ring. The user must lift the handset or press [Speaker] to answer. An SLT always functions in the Tone ring mode.

Operation

To change the Intercom Call Answer Mode:

1. Press the [PGM] button. The [Speaker] button does not flash, but remains on.
2. Dial the Extension User program code ‘11’.
3. Enter the Intercom Call Answer mode (1: H mode, 2: T mode or 3: P mode).
4. Press the [SAVE] button.

Conditions

- The Intercom Call Answer Mode does not apply to paging; pages are sent to the speaker of the iPECS Multi-button phones.
- The default Intercom Call Answer Mode is Tone ring
- The Intercom Call Answer mode is maintained even if power to the system is lost.

Related WMS Menu

[To set ICM Answer Mode]

Data Management > Extension Information > Terminal Information > Terminal Option > “ICM Answer Mode”

Related Features

- Intercom Call
- Paging
- Message Wait
- Call Back
- Call Forward
- DND Override
- Busy One Digit Service

Hardware

- Multi-button Phone

2.65 Interphone Group

Description

The Interphone Group allows a user to make an Intercom Call between members of an Interphone group using one or two digit dialing.

Operation

1. Lift the handset and enter the {Interphone call} feature code.
2. Enter Interphone number (0~9 or 00-99) according to the programmed digit count.
3. The Extension assigned to the interphone number is called.

Conditions

- An Extension can be assigned to one Interphone Group only.
- Users that are not assigned to a group cannot use the Interphone Group calling.
- Only an Extension can be a member of Interphone Group.
- If a group has less than 10 users, the Interphone Group numbers can be 1 digit. The interphone numbers must be two (2) digits if a group has more than 10 users.

Related WMS Menu

[To assign Interphone Group]
Data Management > Extension Information > Group Information > Interphone Group
[To assign Interphone Call feature code]
Data Management > Numbering Plan Information > Feature Code > “Interphone Call”

Related Features

Hardware

2.66 IP Transcoding and RTP Relay

Description

The system employs either IEEE G.711, G.729 or G.723 codec to digitize and compress voice signals sent as RTP packets between iPECS devices. iPECS terminals and gateway modules incorporate DSP functions to support codec conversion. Available VPCM/VOIMs include DSP circuitry used to support transcoding (converting) codecs for incoming VoIP calls to devices that have no built-in codec. The VPCM/VOIMs will transcode the incoming voice codec (G.711, G.723 or G.729) to the system codec and reverse the process for the outgoing packets. When the external VoIP connection can only support G.729 and the system codec is G.723, the DSP must implement a complex trans-coding operation, which requires two (2) DSP channels. In all other cases, transcoding only requires a single DSP channel.

When a call is connected between a local phone and a remote phone, the system relays the RTP packets employing the VOIM/VPCM channel,

- Automatic RTP relay by the system
  Calls between a local phone or gateway and a remote phone or gateway

Calls between a remote phone or gateway and remote phone or Gateway. RTP relay is not used when two remote phones are in the same zone, or the remote phones or gateways use the public IP address.

- RTP relay by WMS setting
  RTP relay between RTP relay groups in a zone
  RTP relay between zones

In this case, one or two VOIM/VPCMs can be used for RTP relay.

Operation

Conditions

- The System codec for VOIM/VPCM can be changed at any time during an IP call.
- VOIM/VPCM DSP can generate and delete in-band DTMF and call processing tone to support the DISA function.
- Transcoding between complex codecs (G.723 and G.729), requires use of two (2) VOIM/VPCM channels.
- If there is no channel available for IP transcoding, the status is displayed on the Status Message window of WMS.
  For a call between phones and gateways, the codecs are used with the following priority:
  - Connection between Zones > Channel Attribute > Zone Attribute > Tenant Attribute
  - The codec assigned on Connection between Zones is applied to RTP relay between zones only. The codec is not used for a connection within same zone. If the codec is available in the phone/GW, the codec is used for the call between VOIM/VPCM and phone/GW. If a phone/GW does not support the codec, transcoding occurs between VOIM/VPCM and phone/GW.
  - If the Codec option is “Follow Zone Codec” in the Channel Attribute menu, the system analyzes the Codec on Zone Attribute to determine the Codec.
  - If the Codec on Zone Attribute menu is set to “Follow Tenant Codec”, the system analyzes the default codec in Tenant Attributes to determine the codec.

Related WMS Menu

[To set Codec rule between Zones]
Data Management > Zone Information > Connection between Zones
[To set Zone & Codec of channel]
Data Management > Zone Information > Channel Attribute
[To set RTP Relay Group]
Data Management > Codec Zone Information > Zone RTP Relay Group
[To set Zone Attribute]
Data Management > Codec Zone Information > Zone Attribute
[To set codec of tenant]
Data Management > Tenant Information > Tenant Basic Information> Tenant Attribute > “Default Codec”

Related Features

- System Networking
- IP Trunk
- SIP Call
- Intercom Call
- Conference

Hardware

- VOIME, VOIM8 or VOIM24
- VPCM

2.67 IP Trunk

2.67.1 H.323 v4 Service (H.323 Trunk)

Description

When assigned to support H.323 protocol, VoIP channels provide protocol conversion between H.323 v4 and the iPECS protocol or SIP. This permits the VoIP channel to connect to external H.323 networks or terminals and to support H.323v4 supplementary services. In addition, H.323 VoIP channels can register with an external H.323 GateKeeper to support Gatekeeper call routing.

Supplementary services are supported employing H.450.1 ~ H.450.12 standards, which define the supplementary services.

Operation

Conditions

Related WMS Menu

[To set Numbering Plan]
Data Management > Numbering Plan Information > Numbering Plan
[To set options of trunk line]
Data Management > Trunk Information
Trunk Basic Information > Trunk Attribute
[To set Trunk Access Code]
Data Management > Trunk Information > Trunk Basic Information > Trunk Access Code
[To set Incoming Route]
Data Management > Trunk Information > Incoming Route Information > Incoming Route Group
[To set outgoing Route]
Data Management > Trunk Information > Outgoing Route Information > Outgoing Route Group
[To set H323 Trunk]
Data Management > Trunk Information > H323 Trunk Information > H323 Routing Attribute, H323 Incoming Route Setting, Gatekeeper Configuration

Related Features

- System Networking
- SIP Service

Hardware

- VPCM, VOIME, VOIM8 or VOIM24

2.67.2 SIP Trunk

Description

When assigned to support SIP (Session Initiation Protocol), VoIP channels provide protocol conversion between SIP and the iPECS protocol or H.323. 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

Operation

Related WMS Menu

[To set options of trunk line]
Data Management > Trunk Information > Trunk Basic Information > Trunk Attribute
[To set Trunk Access Code]
Data Management > Trunk Information > Trunk Basic Information > Trunk Access Code
[To set Incoming Route]
Data Management > Trunk Information > Incoming Route Information > Incoming Route Group
[To set outgoing Route]
Data Management > Trunk Information > Outgoing Route Information > Outgoing Route Group
[To set SIP Trunk]
Data Management > Trunk Information > SIP Trunk Information > SIP User ID Table, SIP Trunk Configuration

Related Features

- System Networking
- SIP Service

Hardware

- VPCM, VOIM8, VOIM24

2.67.3 IP WAN Dialing After Answer

Description

The system permits sending and receiving DTMF signals after connecting to an external VoIP party. The DTMF signal can be DTMF tone, Text String, or DTMF protocol (H.323 specification) based on the system programming

Operation

Conditions

- The connected VoIP Trunks must transmit DTMF digits in the same way selected in the system database otherwise, digits will not be recognized.

Related WMS Menu

Related Features

Hardware

2.67.4 H.323 Multi-Route Service

Description

The system can have multiple destination IP addresses for a single prefix. If a call attempt to a destination IP address ends in failure, the system can connect the call to another IP address. The system supports only circular routing to select the destination IP address.

Operation

Conditions

- This function is supported for VoIP (H.323) calls only.
- Up to 100 destination IP addresses can be assigned to each outgoing route.

Related WMS Menu

[To set Incoming Route Group]
Data Management > Trunk Information > Incoming Route Information > Incoming Route Group
[To set H323 Incoming Route]
Data Management > Trunk Information > H323 Trunk Information > H323 Incoming Route Setting
[To set Call Setup options]
Data Management > Trunk Information > H323 Trunk Information > H323 Call Setup Table

Related Features

Hardware

2.67.5 SIP Trunk Multi-NAT Routing based on Outgoing Route

Description

SIP NAT routing option can be defined by Trunk Route base.

Operation

- When access to SIP trunk, it checks firewall routing option.
If “Firewall Routing Usage” is set to “Use” and “Firewall IP Address” is define, route to firewall.
If “Firewall Routing Usage” is set to “Use” and “Firewall IP Address” is not defined, search firewall IP in “CCM Server Configuration”.
If firewall IP is defined in “CCM Server Configuration”, route to firewall, if not route to Proxy server.
If “Firewall Routing Usage” is set to “Not Use”, route to Proxy server directly.

Conditions

Related WMS Menu

[To set Firewall Routing option of SIP Trunk based on outgoing route number]
Data Management > Trunk Information > SIP Trunk Information

Related Features

Hardware

2.68 ISDN Supplementary Service

Description

The iPECS-CM system supports PRI (Primary Rate Interface) ISDN circuit. Both T1/PRI (the North American standard 23B + 1D) and E1/PRI (ETSI standard 30B + 1D) channel configurations are supported by setting MDTM and system database.

In many cases, the ISDN service provider offers enhanced subscription services. The system allows access to these ISDN “Supplementary Services” implemented under the ETSI regime as described below.

2.68.1 ISDN AOC (Advice of Charge)

Description

When ISDN Advice of Charge service is provided from the ISDN, the system will deliver charge information to the display of iPECS Multi-button phones and include the AOC in SMDR records. AOC is implemented in accordance with ETSI ISDN AOC Specifications.

Operation

Conditions

- AOC information, which is implemented based on ETSI AOC standard, can be sent during call set-up (AOC-S), during the call (AOC-D) or at the end of call (AOC-E).
- This feature may not be available in the specific ISDN service area or may be a subscription service.
- Meter Count is included in SMDR records but, the cost of charge is not included in SMDR records.

Related WMS Menu

[To set options for Call Charge]

Data Management > Tenant Information > Tenant Basic Information > Tenant Attribute > “Charge Method”
> Tenant Information > Tenant Basic Information > Tenant Attribute > “TIE Call Charge“
> Tenant Information > Tenant Basic Information > Tenant Attribute > “Local Call Charge“,
> Tenant Information > Tenant Basic Information > Tenant Attribute > “CO Call Charge”
> Tenant Information > Tenant Basic Information > Tenant Attribute > “ATD Call Charge”
> Tenant Information > Tenant Basic Information > Tenant Attribute > “DATA Call Charge”
> Tenant Information > Tenant Basic Information > Tenant Attribute > “Extension Call Charge”
> Tenant Information > Tenant Basic Information > Tenant Attribute > “Free Call Charge Report”
> Tenant Information > Tenant Basic Information > Tenant Attribute > “Tenant Prefix Include on Charge Report”
> Tenant Information > Tenant Basic Information > Tenant Attribute > “Tenant Name Display on LCD”
> Tenant Information > Tenant Basic Information > Tenant Attribute > “Call Charge for Attendant through Dialing”
> Tenant Information > Tenant Basic Information > Tenant Attribute > “Failed Call Include on ChargeReport”
> Tenant Information > Tenant Basic Information > Tenant Attribute > “Currency Unit for Charge”
> Tenant Information > Tenant Basic Information > Tenant Attribute > “Fraction for Call Charge”

[To set option for Cost Display]

Data Management > Extension Information > Terminal Information > Terminal Option > “AOC Metering”

[To set option for Charge Report of DN]

Data Management > Extension Information > Number(DN) Information > DN Attribute > “Charge Mode”

[To set option for Charge Report of Trunk]

Data Management > Trunk Information >Trunk Basic Information > Trunk Attribute > "Charge Mode"

[To set Cost of metering, fraction of AOC]

Data Management > Trunk Information > Outgoing Route Information > Outgoing Route Options > "Cost per Metering Pulse"
> Trunk Information > Outgoing Route Information > Outgoing Route Options > "AOC Currency Adjust"

Hardware

- MDTM

2.68.2 ISDN Call Deflection

Description

When the ISDN Supplementary Service "Call Deflection" is supported, users can Off Net Call Forward to forward incoming calls over the ISDN directly without need to establish a connection through the system.

When Call Deflection is used, the system sends a call-deflection request message with the telephone number to receive the call to the ISDN. The ISDN then sends incoming calls to the desired telephone number. The system does not need to set-up a connection between Trunks.

Operation

Conditions

- ISDN must support the Call Deflection Supplementary service defined in ETS300-202/206/207.
- ISDN Trunks that support call deflection must be assigned in the system database.

Related WMS Menu

[To set option for Call Deflection]

Data Management > Trunk Information > Incoming Route Information > Incoming Route Options > "Off-net Forward Usage"

Related Features

- ISDN Supplementary Service

Hardware

- ISDN line

2.68.3 ISDN Malicious Call Id Request

Description

When ISDN supports the Malicious Caller Id service, the system may request CID from the ISDN Trunk. Extension users can request the Malicious Caller ID service to ISDN Trunk during a call then, when system receives the result of request, the system sends error tone or confirmation tone.

Operation

To program an {MCID request} button using the Extension User program;

- [PGM] + {Flex} + 2 + {ISDN MCID Request} feature code + [Save]

To request MCID, while on an incoming ISDN call

1. Press the {ISDN MCID REQUEST} button.
   - If the request is successful, user receives confirmation tone.
   - If the request is fail, user receives error tone.

Conditions

- The ISDN must support the Malicious Caller ID service defined in EN300-130.
- If the MCID request is successful, the user will hear confirmation tone but the CLID information is not shown in the LCD of phone.

Related WMS Menu

[To assign ISDN MCID Request feature code]
Data Management > Numbering Plan Information > Feature Code >"ISDN MCID Request"
[To assign flexible button for MCID Request]
Data Management > Extension Information > Terminal Information > Phone Flexible Button

Related Features

- ISDN Supplementary Service

Hardware

- ISDN Trunk
- iPECS Multi-button Phone

2.69 Key Number Group

Description

A Key Number is a representative number that is used to define the numbers for Hunt groups. If this number is called, a Greeting Announcement can be supplied to the caller during Greeting Tone Duration, or the Queuing Announcement can be supplied to the caller during Queuing Tone Duration when no idle member is available.

The four service types for Key Number Groups are explained below:

- Circular type – Incoming calls are routed to the next Extension in the group. If unavailable or unanswered in the no answer time (“Hunting No Answer Time”), the call will be directed to the next Extension defined in the group. The call will continue to hunt until each Extension in the group has been tried.
- Terminal type – Incoming calls are routed to the first Extension of the Key Number Group. If the Extension is unavailable or unanswered in the no answer time (“Hunting No Answer Time”), the call is routed to the next Extension in the Key Number Group. The call will continue to hunt until reaching the last Extension in the group.
- Ring type – Incoming calls are routed to all available Extensions of the Key Number Group at the same time. When the call is answered, all Extensions stop ringing. When a call is routed to Key Number Group in ringing state, the caller will hear busy tone.
- Longest Idle – Incoming calls are routed to the Extension that has been in an idle state for longest time. If the Extension does not answer the call in “Hunting No Answer Time”, the call will be directed to the next longest idle Extension in the group. The call will continue to hunt until each Extension in the group has been tried.

Operation

Conditions

- A Key Number Group can have up to 250 members.
- Up to 1000 Key Number Groups are supported.
- Calls cannot be routed to an Extension in DND or Call Forward.
- Calls to a Key Number Group receive either a Ring-Back tone or MOH while queued to the group.
- Calls that are not answered in the overflow time, are routed to the defined overflow destination (Extension, group, etc). If no overflow destination is defined, the call is dropped after expiration of the Queuing Tone Duration
- One of seventy (70) VPCM announcements can be assigned as the overflow destination. Callers may route their calls to other destinations while listening to the VPCM announcement.
- After starting greeting or queuing announcement, wait for delay time and send answer signal to incoming trunk for incoming trunk line not to be released.
- Delay time is controlled by “Answer signal sending delay time after starting greeting(Sec)” in
- Key number Group Attribute.
- To provide CCR service during listening queuing or greeting announcement for incoming trunk call, CCR table index should be assigned in Announcement Table.
- Call routing to members register call forwarding follows “Call to the Member Registered Call Forward “ option in Key Number Group Attribute.
  - Deny(All Kind of Call Forward) : Call is not routed to members register any kind of call forward.
  - Allow(Only No Answer Call forward): Call is routed to members register no answer call forward.
  - Allow(call forward): Call is routed to members register call forward, and call forwarding is executed by assigned number.
  - Allow(override): Call is routed to members register call forward, but call forwarding is ignored.

Related WMS Menu

[To assign Key Number Group & Attribute]

Data Management > Extension Information > Group Information > Key Number Group, Key Number Group Attribute

Related Features

- Music-On-Hold (MOH)

Hardware

2.69.1 Key Number Group Call Forward

Description

When a Key Number is called, the call can be routed to another phone or Key Number Group. The Key Number Group Call Forward destination is defined through WMS or by a group member with the {Key No Group Forward(Register)} feature code.

The forward types for Key Number Group Call Forward are below;

- Unconditional – Incoming calls forward unconditionally immediately.
- Queuing Overflow – Incoming calls forward when the number of calls exceeds the designated number of waiting calls.
- Queuing Timeout – Incoming calls forward when the call is not answered during queuing tone duration.
- Queuing Overflow/Queuing Timeout – Incoming calls forward when Queuing Overflow or Queuing Timeout occurs.

The actions for Call to the Member Registered Call Forward are below;

- Deny All Kind of Call Forward – Call is not routed to the member.
- Allow Only No Answer Call Forward – Call is routed to call forward destination only forward type is no answer call forward.
- Allow (Call Forward) – Call is routed to call forward destination.
- Allow (Override) – Call is routed to the group member.

Operation

To set Key Number Call Forward:

1. Lift the handset or press the [Speaker] button.
2. Dial the {Key Number Group Forward (Register)} feature code.
3. Enter the Key Number Group number.
4. Dial *.
5. Enter the forward type (range: 1~4)
6. Enter the destination number.
7. Dial *.

To Cancel Key Number Call Forward:

1. Lift the handset or press the [Speaker] button.
2. Enter {Key Number Group Forward(Cancel)} feature code.
3. Enter Key Number.
4. Dial *.

Conditions

· Only a member of the Key Number Group can set Call Forward for the group.

Related WMS Menu

[To assign Key Number Group & Attribute]

Data Management > Extension Information > Group Information > Key Number Group, Key Number Group Attribute

Related Features

Hardware

2.69.2 Key Number Group No Answer

Description

When a Key Number is called, the member phones of a Key Number Group are ringing.
In this state, If any member phone does not response within a limited time, the call is routed to VMS of Key Number Group.
Now, the caller can leave a message on VMS of Key Number Group.

To Set data

- on WMS :

1. To assign [Voice Mail Access] feature code.
5. WMS > Data Management > Numbering Plan Information > Feature Code – (ex: #0)
2. To assign [Direct VM Transfer] feature code.
6. WMS > Data Management > Numbering Plan Information > Feature Code – (ex: #1)
3. To generate a DN Number to leave a voice mail message.
7. WMS > Data Management > Extension Information > Number(DN) Information > DN Attribute – (ex:3000)
4. To assign a password of the DN Number(ex:3000) to leave a voice mail message.
8. WMS > Data Management > Extension Information > Number(DN) Information > DN Attribute – Phone No:2000 - Extension Password – (ex: 1234)
5. To assign a [Voice Mail Access Password] of the DN Number(ex:3000) to leave a voice mail message.
9. WMS > Data Management > Extension Information > Number(DN) Information > Voice Mail Information - Phone No:(ex:3000) – Voice Mail Access Password – (ex: AUTH Code)
6. To assign a [Group Call Forward Type].

10. WMS > Data Management > Extension Information > Key Number Group Attribute > Group Call Forward Type – (ex: Queueing Timeout)
7. To assign a [Group Call Forward Destination].
11. WMS > Data Management > Extension Information > Key Number Group Attribute > Group Call Forward Destination - (ex: #13000)
8. To assign a [Group No. Answer Time(100 msec).
12. WMS > Data Management > Extension Information > Key Number Group Attribute > Group No. Answer Time - (ex: 100)

- on Member Telephones :
1. Press the [Program] button.
2. Press any one [Flexible] Button. (ex: FB2)
3. Enter a digit "2".
4. Enter {Voice Mail Access} feature code. (ex: #0)
5. Enter VMS DN Number of Key Number Group. (ex: 3000)
6. Press [Hold/Save] button.

Operation

1. Dial the {Key Number Group number} on a telephone which is not a member telephone.
2. Ringing the bell on the Member telephones of the Key Number Group.
3. If any member phone does not response within a limited time(ex:100ms), the call is routed to VMS of Key Number Group(ex:3000).
4. Press "#" button after leave a voice mail message.
5. Place the handset
6. When a voice mail message is saved in a VMS, a Flexible button(ex:FB2) of all member telephones are blinked.
7. Press the blinking FB(ex: FB2) to verify the voice mail message.
8. Enter the password. (ex:1234)
9. Verify the leaved message according to the introduction of VMS.

Related Features

Hardware

2.70 Last Number Redial (LNR)

Description

The system saves the last number (up to 32 digits) dialed by a user, so that the user may simply place a call to the number without having to redial the number manually. With the iPECS Multi-button phones, the Call Log Display function may be used to recall the last incoming/outgoing call.

Operation

iPECS Multi-button Phone

To use last number redial:

1. Lift the handset or press the [Speaker] button.
2. Press the [SPEED] button and dial *.
3. A call is placed to the last dialed number.

Single Line Telephone (SLT)

To use last number redial:

1. Lift the handset.
2. Enter the {Last Number Redial} feature code.
3. A call is placed to the last dialed number.

Conditions

- The last dialed number is saved in volatile memory, so it may be deleted if power is lost.
- This feature can be applied to both Trunk calls and VoIP calls.

Related WMS Menu

[To assign Last Number Redial feature code]

Data Management > Numbering Plan Information > Feature Code > “Last Number Redial”

Related Features

- Station Speed Dial
- System Speed Dial
- Save Number Redial
- Call Log

Hardware

2.71 Line Identification Control

Description

The system receives calling party identification in the ISDN call Set-up message as the CLIP-(Calling Line Identification Presentation). The answering party identification, which may be different from the called party, is received in the ISDN connect message as the COLP (Connected Line Identification Presentation). When provided, the LCD of iPECS Multi-button phones displays the identification.

The system will send calling and answering party identification in the appropriate messages to the ISDN based on the database. The identification message can be restricted and not reported to the other party. Users can control Calling Line Identification Restriction (CLIR) and Connected Line Identification Restriction (COLR) with the system database.

LINE 0001 RINGING
0313358790

Operation

To report MCID to WMS:

1. Press [TRANS] button or, for an SLT, press and release the hook-switch (hook-flash).
2. Enter {MCID Report} feature code.

To deny an Anonymous Call:

1. Lift handset or press [Speaker] button.
2. Enter {Anonymous Call Reject(Register)} feature code.

To allow Anonymous Calls:

1. Lift handset or press [Speaker] button.
2. Enter {Anonymous Call Reject(Cancel)} feature code.

To make a one-time CLIR call:

1. Lift handset or press [Speaker] button.
2. Enter {Caller ID Display Restrict(Call Base)} feature code.
3. Dial a Trunk Access Code and destination number.

To change the CLIR state:

1. Lift handset or press [Speaker] button.
2. Enter {Caller ID Display Restrict(On/Off)} feature code.

To change the COLR state:

1. Lift handset or press [Speaker] button.
2. Enter {Connected ID Display Restrict(On/Off)} feature code.

Conditions

Related WMS Menu

[To set options for CID Display and Generation]

Data Management	> Extension Information > Number (DN) Information > CID Configuration

Related Features

Hardware

- MDTM

2.72 Local survivability

iPECS-CM can be configured as a Central Communication Manager (CCM) or a Local Communication Manger (LCM). The CCM and LCM work as a single system through the IP network and the CCM handles all software features for all devices (gateway modules and phones) registered to CCM and LCM. In the normal condition, the LCM operates in a Bypass Mode and performs limited functions between CCM and devices of LCM.

If the LCM is disconnected from CCM by a WAN or CCM failure, the LCM takes over the CCM function for the local devices and works as the site controller (Local Survival Mode). When LCM is in Local Survival Mode, a call to the CCM or other LCMs can be made through PSTN routing, also known as PSTN Failover.

Specifications for local survivability are as below:

- The maximum number of LCM locations is 255. When all LCMs are redundant, 510 nodes can be achieved.
- The hierarchy between LCM and CCM is a single layer; multi-layer hierarchy is not supported.
- Database modification in the CCM through WMS is synchronized to the database of the LCM.

Mode when the LCM call Server type is CM-S2K/S1K.

- The LCM can be configured with redundancy, Active/Standby server.
- LCM mode change from Local Survival to Bypass is done automatically between the configured CCM and LCM.

Conditions

- Certain features are not supported in Local Survivability mode: Two-Way Recording, Record/Play/Delete Personal Voice Message.
- Call features assigned in Local Survival Mode are not available when the LCM returns to Bypass Mode; DND, Call Forward, Extension User program via the phone, Pre-Defined Text Display, Auto Redial, Button Program, Message Wait, Call Back, Trunk Queuing, Attendant Night Service/Overflow Service, Attendant Billing Registration, Wake Up, etc.
- Call connections are maintained under the following conditions.
  - Conversation between two terminals (phone, gateway) in same LCM is maintained.
  - This is available when CM-S2K-L/CM-S1K-L is used as LCM Call Server.
  - This is not available for the call using VPCM/VOIM/VOIP.
  - This is available once for the same call (Bypass-> Survival, Survival->Bypass).
  - When the CCM is assigned as a backup server for LCM failure, conversation may not be maintained.
- Call traffic between the CCM and LCM can be limited by defining the available bandwidth through WMS.
- Database changes in CCM cannot be updated to the database of LCM in Local Survival mode until LCM returns to the Bypass Mode.

Related WMS Menu

[To configure CCM & LCM Call Server]
System Management > CCM Server Configuration, LCM Server Configuration
[To check LCM Redundancy information]
System Management > LCM Redundancy information
[To change Active and Standby of LCM Call Server]
System Management > LCM Forced Switchover]
[To check LCM operation state & change LCM operation mode manually]

System Management > LCM State Information]
[To set LCM Type, Auto/Manual Bypass Mode]
System Management > LCM Server Configuration > LCM Type

Related Features

- PSTN Failover Routing
- LCM failure Backup Server
- LCM Traffic Limitation by Bandwidth

2.72.1 PSTN Failover Routing

Description

When an LCM is in Local Survival Mode due to a WAN failure, calls between the CCM and LCM or the LCM and other LCMs can be made through PSTN routing. When an Extension of the LCM dials an Extension of the CCM, the system routes the call through the PSTN using Digit Conversion. The Extension number is converted to a PSTN number automatically and the call delivered to the Extension of the CCM through PSTN.

Operation

Conditions

Related WMS Menu

[To enable PSTN Failover Routing]
Data Management > PSTN Failover Information > PSTN Failover Option
[To set PSTN Failover Digit Conversion Table]
Data Management > PSTN Failover Information > PSTN Failover Digit Conversion Table
[To set PSTN Failover Digit Conversion Map]
Data Management > PSTN Failover Information > PSTN Failover Digit Conversion Map

Related Features

- Intercom Call
- Digit Conversion

Hardware

2.72.2 LCM Failure Backup

Description

This feature allows devices (gateways and phones) of LCM to register to CCM when LCM fails. This feature is available when the devices of LCM have a unique IP address and are in the same IP environment with CCM and can communicate with each other. When LCM recovers, the devices will register to LCM automatically and cancel the registration with the CCM.

Operation

Conditions

- Only a CM-S2K/S1K can be used as the LCM; the feature is not available when LIK100/300/600/1200 is used as LCM)
- The iPECS Multi-button phones support this feature.
- The gateways that support this feature are the ASLM, MDTM, VPCM, and MATM.
- Gateways and IP phones should be set in the Remote mode.

Related WMS Menu

[To use CCM for LCM Failure Backup Server]
System Management > System Environment > “Dual CM IP Profile Use”

Related Features

Hardware

- iPECS Multi-button phone
- ASLM
- MDTM
- VPCM
- MATM

2.72.3 LCM Traffic Limitation by Bandwidth

Description

A call from an LCM can be restricted when the call traffic exceeds the bandwidth assigned through WMS. When an Extension associated with the LCM places a call to an Extension in the CCM or an Extension of another LCM, the system disallows the call if the assigned bandwidth is exceeded. When a call from the LCM is restricted by bandwidth, the system can route the call to the destination through the PSTN, this is known as PSTN Failover by LCM Traffic.

Operation

Conditions

- Audio traffic (RTP) is determined by the Codec and packetization interval.
- To conserve bandwidth Video calls can be denied. Bandwidth for a video call is not configurable.
- Users that receive ring signals are not included in the current traffic. Therefore, when a call ringing to multiple users is answered, the designated bandwidth limit can be exceeded. In this case, a reserve ratio may be set so that the users may place calls even if the traffic limit is momentarily exceeded.
- This function is not employed for the call to an Extension in the same LCM.
- If an Extension in another LCM is assigned for call forward, the call forward function is applied, and the LCM Traffic limiting function is not employed.

- If a ring is to be provided to multiple users, all members must be in the same LCM.
- Until a Trunk call is answered, a one-way call is connected for Ring Back tone. Therefore, only half the call bandwidth is required. However, when the call is connected at the expiration of the call digit timer one (1) call bandwidth is required.
- When the LCM Traffic Limitation function is working, the system has the option to route a call to the destination through the PSTN. Therefore, when the option is set, PSTN Failover by LCM Traffic function will work.
- When using PSTN Failover by LCM Traffic, the "Digits before Conversion" of "PSTN Failover Digit Conversion Group" converts digits using the Digit Conversion function. The "Digits after Conversion" assigned in the "PSTN Failover Digit Conversion Group" does not convert digits using the Digit Conversion function.

Related WMS Menu

[To set option & Bandwidth to restrict LCM traffic, PSTN Failover by LCM Traffic]
Data Management > System Feature Information > Call Bandwidth Restriction
[To set PSTN Failover Digit Conversion Table]
Data Management > PSTN Failover Information > PSTN Failover Digit Conversion Table
[To set PSTN Failover Digit Conversion Map]
Data Management > PSTN Failover Information > PSTN Failover Digit Conversion Map

Related Features

- PSTN Failover
- Digit Conversion

Hardware

2.72.4 PSTN Failover Call Forward

Description

IP Phones can select incoming calls to be rerouted automatically to an external number when an LCM failure or the IP phone is disconnected from system.

Operation

To activate PSTN Failover Call Forward

1. Lift the handset or press the [Speaker] button.
2. Enter the {PSTN Failover Forward Register} feature code.
3. Enter destination number.
4. Dial * or press [HOLD/SAVE] button.

To deactivate PSTN Failover Call Forward

1. Lift the handset or press the [Speaker] button.
2. Enter the {PSTN Failover Forward Cancel} feature code.

Conditions

Related WMS Menu

[To assign PSTN Failover Forward Register, Cancel feature code]

Data Management > Numbering Plan Information > Feature Code > "PSTN Failover Forward Register"
> Numbering Plan Information > Feature Code > "PSTN Failover Forward Cancel"

[To set PSTN Failover Forward]

Data Management > PSTN Failover Information > PSTN Failover for Terminal Eject

Related Features

- Call Forward

Hardware

2.73 Message Wait

2.73.1 Short Message Service (SMS)

Description

Short Message Service (SMS) provides the ability to send and receive text messages to and from iPECS phones equipped with a display including the Phontage, UCS Client and WLAN phone. The text is alphanumeric and can be up to 100 characters in length when Latin alphabets are used.

Operation

To send an SMS message to another iPECS display Phone:

1. Press the [PGM] button,
2. Dial 43 (SMS send code),
3. Dial the Extension to receive the message,
4. Press the [Add] soft button and dial any additional Extension to receive the SMS,
5. Press the [Finish] soft button,
6. Enter text message (refer to Character Entry in Station Speed Dial),
7. Press [Send] soft button to send your message.

To retrieve Short messages:

1. Press the [MSG] soft button, the LCD will display the Message Summary display,

MWI(00) VMS(00) SMS(01)
ENTER (MWI:1, VMS 2

2. Dial 3 to view the SMS messages.
3. Press the [NEXT] soft button to view the next SMS message.

To delete a received SMS

1. Press the [Delete] soft button.

Conditions

・Users can send messages from the Extension User Web portal.

Related WMS Menu

[To set SMS access protect]

Data Management > Extension Information > Number(DN) Information > DN Feature Registration > "SMS Access Protect"

Related Features

Hardware

- iPECS Multi-button w/Display
- WLAN Phone (WIT-400H)
- Phontage

2.73.2 Message Wait

Description

When the called Extension does not answer or is in DND, an internal caller can leave a Message Wait Indication to request a return call. The Extension receiving a Message Wait Indication can return calls using the [MSG/CALLBK] button.

Operation

iPECS Multi-button phone

To leave a Message Wait while receiving Ring Back tone or no response:

1. Press the [MSG/CALLBK] button, the user receives confirmation tone.
2. Hang up the call.
3. The [MSG/CALLBK] button LED of the called user flashes.

To leave a Message Wait while receiving DND tone

1. Press the [MSG/CALLBK] button, the user receives confirmation tone.
2. Hang up the call.
3. The [MSG/CALLBK] button LED of the called user flashes.

To respond to Extension Message Wait:

1. Press the flashing [MSG/CALLBK] button. The following message will be displayed.

MWI (01 VMS (00) SMS(00)

2. To retrieve the Message Waiting, dial 1-3 to select the message type.
   - 1 - MWI: Check the missed call log
   - 2- VMS: Check the VPCM messages
   - 3- SMS: Check the SMS messages
3. Check the MWI list with Volume Up/Down.

4. Press the [HOLD] button to select an item.

To cancel a Message Wait
1. Lift handset.
2. Enter the {Message Wait Cancel} feature code.
3. Enter the Extension number with the Message Wait to cancel.

<u>Single Line Telephone (SLT)</u>
To leave a Message Wait while receiving ring back or no response:
1. Press and release the hook-switch (hook-flash).
2. Enter {Message Wait Register} feature code, the user receives a confirmation tone.
3. Hang up the call.

To leave a Message Wait while receiving DND tone
1. Press and release the hook-switch (hook-flash).
2. Enter the {Message Wait Register} feature code, the user receives a confirmation tone.
3. Hang up the call.

To respond to Extension Message Wait:
1. Lift the handset, the user receives Message Wait tone, and then the dial tone.
2. Enter the {Message Wait Answer} feature code.

Conditions
- The [MSG/CALLBK] button LED will stop flashing when the user checks all Message Waits by pressing the [MSG/CALLBK] button.
- When a user attempts to leave a Message Wait, if the Message Wait queue is full, the system deletes oldest message.

Related WMS Menu
[To use Message Wait]
Data Management > Tenant Information > Tenant Basic Information > Tenant Attribute > “Extension Message Wait Use”

Related Features
- Message Wait Reminder Tone
- SMS
- VMS

Hardware

2.73.3 Message Wait Reminder Tone

Description

In addition to the [MSG/CALLBK] button LED, iPECS Multi-button phones can be sent a tone as a periodic remind to the user of Message Waits in queue. This tone is sent to the Extension only while idle and is heard over the speaker.

Operation

Conditions

- The interval between the tones can be 30seconds – 120minutes.
- The reminder tone continues until the message is retrieved.
- The reminder tone is not provided to a busy Extension.

Related WMS Menu

[To set Message Wait Remind Tone]

Data Management	> Tenant Information > Tenant Time Information > Tenant Timer > “Message Wait Remind Tone Time(sec)”

Related Features

- Message wait
- SMS
- VMS

Hardware

- iPECS Multi-button Phone

2.73.4 Message Wait Lamp Indication

Description

All SLT devices receive a “Stutter” dial tone as an audible message wait indication. In addition, industry standard message wait telephones may be connected to the system. The system will cause the lamp to flash when a message is waiting.

Operation

Conditions

- The system switches the lamp On and Off to flash the SLT’s message lamp.
- Activating the lamp in an SLT does not affect the system ability to detect an Off-hook event.
- The SLT must incorporate a 90 VDC neon lamp connected directly across the tip and ring of the line.

Related WMS Menu

[To set SLT type for Message Wait]

Data Management	> Extension Information > Terminal Information> Terminal Attribute > “Terminal Sub Type” -> “MSGWAIT”

Related Features

- Message Wait
- SMS
- VMS

Hardware

- ASLM-MW

2.74 Mobile Extension

Description

A mobile phone may be registered to an Extension allowing the mobile phone to place and receive calls through the system. DID calls are sent to the Extension and the active registered mobile phone simultaneously. If an Extension is a member of a Key Number Group or Hunt Group, calls are routed to the Extension and the mobile phone simultaneously.

Mobile Extensions can use the facilities of iPECS-CM to place internal and external calls. If a mobile phone user dials DID number of a Mobile Extension, the system will check the mobile phone CLI and provide the mobile phone user with the system dial tone. Users can register mobile phones to a Mobile Extension and activate the Mobile Extension features.

Operation

System

If programmed, incoming DID calls are automatically sent to Mobile Extensions.

Multi-button Phone

To register a mobile phone number:

1. Press the [PGM] button.
2. Dial 5 + 2, the Mobile Extension program code.
3. Enter the Mobile Phone List number (1~5).
4. Enter the mobile phone number.
5. Press the [SAVE] button.

To register a mobile Extension CLI number:

1. Press the [PGM] button.
2. Dial 5 + 3, the Mobile Extension CLI program code.
3. Enter the Mobile Extension CLI List number (1~5).
4. Enter the CLID number.
5. Press the [SAVE] button.

To activate a mobile Extension:

1. Press the [PGM] button.
2. Dial 5 + 1 the Mobile Extension Activation program code.
3. Enter the Mobile Extension List number (1~5).
4. Dial 1 to activate the Mobile Extension or dial 0 to deactivate the feature.
5. Press the [SAVE] button.

Single Line Telephone (SLT)

To register a mobile phone number:

1. Lift the handset.
2. Enter the {Mobile Extension Register} feature code.
3. Enter the Mobile Extension List number (1~5).
4. Enter the mobile phone number.
5. Make a hook-flash.

To register a mobile Extension CLI number:

1. Lift the handset.
2. Enter the {Mobile Extension CID Register} feature code.
3. Enter the Mobile Extension List number (1~5).
4. Enter the ICLID number.
5. Press and release the hook-switch (hook-flash).

To activate a mobile Extension:

1. Lift the handset.
2. Enter the {Mobile Extension Activate/Deactivate} feature code.
3. Enter the Mobile Extension List number (1~5).
4. Dial 1 to activate the Mobile Extension or dial 0 to deactivate the feature.
5. Press and release the hook-switch (hook-flash).

To place a call from a mobile phone using IPECS-CM:

1. Enter the DID number of the Extension; the system will check the CLID, answer the incoming call, and provide internal dial tone.
2. Place an internal or external call.

To transfer a call from the mobile phone using IPECS-CM:

1. Enter Flash digit in {Tenant Mobile Feature Code} during the call.
2. Enter the Extension number; the call will be transferred to the Extension, and the mobile phone will return to an idle status.
3. Press the '#' button to reconnect to the call.

Conditions

- When a mobile phone places a call through the system, the CLID of the Extension to which the mobile phone is registered is used.
- This feature is applied to the digital lines (ISDN, T1/E1) only.
- Message Wait and Call Back are not supported for mobile phones.

- Networking features are not supported for mobile phones.
- Hold/Transfer recalls are supported for mobile phones.
- Mobile Extensions may be members of a Key Number Group (UCD/ACD, Ring or Terminal) and can signal (ring) the associated mobile phone
- Up to five mobile phones can be assigned to one Extension.

Related WMS Menu

[To assign Mobile Extension & options]

Data Management > Extension Information > Number (DN) Information > DN Attribute >" Mobile Number"

[To allow DN to use Mobile Extension]

Data Management > Extension Information > Number (DN) Information > DN Feature Allow/Deny > "Mobile Extension"

[To assign feature code for Mobile Extension]

Data Management > Numbering Plan Information > Feature Code > "Mobile Extension Register",
> Numbering Plan Information > Feature Code > "Mobile Extension CID Register"
> Numbering Plan Information > Feature Code > "Mobile Extension Activate/Deactivate"

[To assign Flash Digit for Mobile Extension]

Data Management > Tenant Information > Tenant Basic Information> Tenant Mobile Feature Code > "Flash Digit"

Related Features

Hardware

- iPECS Multi-button Phone

2.75 Multiple Prompt Language Support

Description

With the VPCM, iPECS-CM can support nine (9) languages simultaneously. Prompts in the desired languages are loaded into the VPCM memory along with the Language Selection prompt. To assure the proper language is employed, the Language Selection prompt is played when an incoming call is assigned to be answered by a DID, DISA, Auto Attendant or Hunt Group announcement. The Language Selection prompt is 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 announcements (DID, DISA, etc.) recorded for the selected language.

Operation

Authorized iPECS Multi-button Phone

To record a multiple language selection announcement:

1. Lift the handset.
2. Press the [PGM] button.
3. Dial 9 + 7 (recording code).
4. Enter a VPCM slot number (0001~8000).
5. Enter an announcement number (01~70).
6. Enter a language (1~9); the current recorded announcement will be played.
7. The “Press # to record” prompt is presented.
8. Dial ‘#’.
9. After the recording prompt and beep tone, record the message.
10. At completion of the recording, press the [SAVE] button.

To delete a recorded message:
1. Press the [PGM] button.
2. Dial 9 + 7 (recording code).
3. Enter a VPCM slot number (0001~8000).
4. Enter an announcement number (01~70).
5. Enter a language (1~9); the current recorded announcement will be played.
6. The “Press # to record” prompt will be presented.
7. Press the [SPEED] button, or press the [Delete] soft button to delete the prompt being played, and the system provide the confirmation tone.

<u>Conditions</u>

- Multiple language support is available with VPCM.
- The Language Selection prompt must be recorded by the authorized Extension.
- Separate announcements must be record for each language supported.
- Up to nine (9) Voice Prompt Package files, each with a different language, can be uploaded to VPCM.
- When an Announcement Table is configured with the “language selection” announcement, the system plays the selection announcement to callers first. Once selected by the caller, future announcements are played in the selected language.

<u>Related WMS Menu</u>

[To enable Announcement Recording]
Data Management > Extension Information > Number (DN) Information > DN Feature Allow/Deny> “Privileged User”
[To program Announcement Table for language selection]
Data Management > System Feature Information > Announcement Table > “Slot”
> System Feature Information > Announcement Table > “Token ID”
[To assign language selection announcement to Announcement]
Data Management > System Feature Information > Announcement Table > “Announcement Table for Language Selection”
[To Upload Prompt Language to VPCM]
Version management > Firmware Upgrade > Firmware File Upload
[To set VPCM Prompt Language]
System management > Prompt Language Configuration

[To set the DN Prompt Language Selection]
Data Management > Extension Management > Number (DN) Information > Voicemail Information
[To select Prompt Language for Station & Trunk]
Data Management > Codec Zone Information > Channel Attribute > "Default Prompt Language"
[To set Prompt Language Selection for Room Guest]
Data Management > Hotel Information > Hotel Extension Attribute > "Prompt Language Select"
[To set the Default Prompt Language for Room Guest Check-out]
Data Management > Hotel Information > Hotel General Configuration > "Check-out Default Prompt Language"

Related Features

- DISA
- DID
- Auto Attendant

Hardware

- VPCM

2.76 Multiple Voice Mailbox Indication

Description

Extension users can access any Voice Mailbox using a Voice Mail Access feature code and Authorization code. One or more Flex buttons can be assigned to use a specific Voice Mailbox and indicates the message status for the Voice Mailbox.

Operation

iPECS Multi-button Phone

To assign {Voice Mailbox} to a Flex button using the Extension User program:

- [PGM] + {Flex} + {Voice Mail Access} feature code + VM Extension Number + [SAVE].

To access a Voice Mailbox with the {Voice Mailbox} Flex button:

1. Lift the handset or press the [SPEAKER] button.
2. Press the Voice Mailbox Flex button.
3. Enter the VM password.

Conditions

- A Voice Mailbox Flex button LED will flash indicating new messages have been received in the associated Mailbox.
- Flex buttons cannot be assigned to access the mailbox of a networked Extension Voice Mailbox.

Related WMS Menu

[To set Voice Mail Access feature code]

Data Management > Numbering Plan Information > Feature Code > “Voice Mail Access”

[To assign flexible button]

Data Management > Extension Information > Terminal Information > Phone Flexible Button

Related Features

- Message Storage
- Message Retrieval
- Remote Message Retrieval
- Message Retrieval Option

Hardware

- iPECS Multi-button Phone
- VPCM

2.77 Music On Hold (MOH)

Description

If a call is placed 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 alive.

Messages recorded in VPMC can be used as MOH. Authorized users can record the VPCM announcement for MOH, and the recorded announcement is assigned as the source for MOH in WMS.

Operation

Authorized ExtensionTo record a VPCM message for MOH:

1. Press the [PGM] button.
2. Dial 9 + 7 (recording code).
3. Enter a VPCM slot number (0001~8000).
4. Enter an announcement number (01~70).
5. Enter a language (1~9); the current recorded announcement will be played.
6. The “Press # to record” prompt will be presented.
7. Dial ‘#’.
8. After the recording prompt and beep tone, record the message.
9. After finishing recording, press the [SAVE] button.

Related WMS Menu

[To set Hold Tone]

Data Management > Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > “Normal Hold Tone (Trunk)”

> Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > “System Hold Tone (Trunk)”

[To assign MOH Unicast/Multicast Group]

Data Management	> Extension Information > Terminal Information> Terminal Attribute > "MOH Multicast Group"
	> Extension Information > Terminal Information> Terminal Attribute > "MOH Unicast Group"

[To set MOH Group Configuration]

Data Management	> System Feature Information > MOH Group information> MOH Group Attribute
	> System Feature Information > MOH Group information> MOH Group Configuration

[To assign Announcement Table]

Data Management	> System Feature Information > Announcement Table

Related Features

- Hold

Hardware

- iPECS Multi-button Phone

2.78 Mute

Description

iPECS Multi-button phones can turn off audio transmission through the handset, speakerphone or headset microphone.

Operation

To Mute the active microphone

1. Press the [MUTE] button. The [MUTE] LED is on, and the microphone (handset, speakerphone or headset) is muted; the connected party receives silence.

To activate microphones:

1. Press the flashing [MUTE] button. The [MUTE] LED is off, and the microphone is activated, transmitting audio to the connected party.

Conditions

- Changing from speakerphone to handset or vice versa during while muted, returns the microphone (handset, speakerphone or headset) to the active state.
- Returning to idle will change the Mute status to its normal active condition.

Related WMS Menu

Related Features

- Intercom Call
- Trunk Call

Hardware

- iPECS Multi-button Phone

2.79 Off-Hook Signaling

Description

When an Extension that is busy receives a call, the Extension will receive Off-hook Ring signals through the speaker. When a Call-Wait, Camp-On or Voice Over Announcement is requested to the busy Extension, the Extension receives the tone assigned by system, and, for calls on other DNs, the Extension receives the defined Off-hook ring signal.

Operation

Conditions

- While using the speakerphone, a Camp-On tone is provided over the speaker in place of the assigned Off-hook Ring Signal.
- Pressing the [DND] button activates One-Time DND, rejecting the waiting call, and the off-hook signaling is terminated.
- Off-hook Ring signals terminate when the call is answered, forwarded, or abandoned.
- An Extension that is receiving Off-hook Ring signals, will receive normal ring upon return to idle status.

Related WMS Menu

Related Features

- DN
- Call Wait
- Camp-On
- Voice Over

Hardware

- iPECS Multi-button Phone

2.80 Off-Hook Voice Over (OHVO)

Description

Users on a call may receive a voice announcement through the handset receiver with existing call. The Voice Over is muted so as not to interfere with the existing conversation. The called party may answer the calling party with [HOLD] button.

Operation

iPECS Multi-button phone

To assign {OHVO(Off Hook Voice Over)} to a Flex button:

- [PGM] + {Flex} + 2 (program type for number) + {OHVO{Off Hook Voice Over}} feature code + [SAVE]

Placing a Voice Over (OHVO) while receiving a busy tone:
1. Enter the digit defined for OHVO One Digit Busy Service or press the pre-programmed {OHVO (Off Hook Voice Over)} Flex button.
2. After splash tone, begin announcement.

To answer with Broker Call:
1. Press the [HOLD] button to use the Broker Call.

Conditions

- OHVO calls can be received with iPECS Multi-button phones only.
- The calling party DN must be programmed to allow OHVO calls.

Related WMS Menu

[To enable Voice Over]

Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > “Voice Over”

[To assign One Digit Service for OHVO]

Data Management > Tenant Information > Tenant Basic Information> One Digit Service on Busy

Related Features

- Camp-On
- Call Wait

Hardware

- iPECS Multi-button phone

2.81 Off-net Call Forward Announcement Service

Description

When an Extension or outside caller calls an Extension with Off-net Forward active, the system provides Off-net Call forward tone to the outside caller. The Off-net Call Forward tone can be provided while the call is being processed, or the call may be processed after the Off-net Call Forward tone.

Operation

Conditions

- Ring back Tone service is provided after the Off-net Call Forward tone.
- If tones and calls are processed at the same time, the R2 comfort tone is provided after the Off-net Call Forward tone. If R2 signaling is over, Ring Back tone is provided.

- There are two options for Off-net Call Forward tone service; one is for R2 Trunks and the other is for other Trunk types.
- If the Off-Net Call Forward tone time is set to “0”, the system does not provide this service.
- When a call is transferred to an Extension that is forwarded Off-net, this function is not available to the transferred user.
- When an outside caller on R2 Trunk calls an Extension that is forwarded Off-net, this function is not available.

Related WMS Menu

[To select trunk type for Off-Net Call Forward Announcement]

Data Management	> Tenant Information > Tenant Basic Information > Tenant Attribute > “Announcement Service for Off-Net Call Forward”

[To set Off-net Call Forward Tone]

Data Management	> Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > “Off-Net Call Forward Tone”

[To allow Off-net Call Forward]

Data management	> Extension Information> Number(DN) Information > DN Feature Allow/Deny > “Off-Net Call Forward” > Extension Information> Group Information > Feature Allow/Deny Group > “Off-Net Call Forward”

Related Features

- Call Forward

Hardware

2.82 On-Hook Dialing

Description

iPECS Multi-button phones equipped with a speakerphone allow users to place as well as receive calls while the handset is on-hook. Once the user activates the speakerphone by pressing the [Speaker] button, dial tone is received and the user may place the desired call.

Operation

To dial a number while on-hook:

1. Press the [Speaker] button, dial tone is received and the [Speaker] button LED lights.
2. Place desired call.

Conditions

- If the iPECS Multi-button phone has no speakerphone, the user must lift the handset to converse with the connected party.
- If an outgoing call is not answered, the user must press the [Speaker] button to return to idle.
- When the speakerphone is used, the microphone is active unless the [Mute] button is pressed and the [Mute] button LED is On.

Related WMS Menu

Related Features

- Mute
- Speakerphone

Hardware

- iPECS Multi-button phone

2.83 One Time DND

Description

If a user presses the [DND] button while receiving ring or an off-hook ring, ring is terminated. This feature is applied to the current incoming call only, and is deactivated automatically.

Operation

To activate One Time DND while ringing:

1. Press the [DND] button.

Conditions

- Press the [DND] button during a call, no other call is received until the user hangs up the current call.
- This feature works regardless of DND permission.
- When pressing the [DND] button during the delay ring service on an incoming call, ringing is received.

Related WMS Menu

[To allow DND setting]

Data Management	> Extension Information > Number (DN) Information > DN Feature Allow/Deny > “DND”

Related Features

- Call Wait / Camp-On
- Do Not Disturb
- Directory Number

Hardware

- iPECS Multi-button Phone

2.84 Paging

2.84.1 Internal Page

Description

Extensions are grouped into “zones” to receive paging announcements. An Extension permitted to access the Page facilities, can connect and transmit voice announcements to any or all of the Internal Page zones in the system. Extensions not assigned to any Page zone will not receive page announcements.

When a user accesses a Page zone, all members of the Page zone receive the Page Warning tone, followed by the Page announcement. User are allowed to continue the paging for the defined Paging Time. When the user terminates paging (hangs up), the page zone is returned to idle.

The system supports 100 Page zones per tenant. Page zones 01~99 are used for general-purpose zones and 00 pages to all zones, 01~99.

On an iPECS Multi-button phone, an {Internal Page} button can be assigned to Flex button.

Operation

iPECS Multi-button Phone

To assign an {Internal Page} Flex button using the Extension User program:

- [PGM] + {FLEX} + Button Feature Type (2) + {Internal Page} feature code + [SAVE]

To make a page:

1. Lift the handset.
2. Enter the Internal Page feature code, or press an {Internal Page} Flex button.
3. Enter the Page zone (00 – 99).
4. After the Page Warning tone, make announcement.
5. When finished, hang-up.

Single Line Telephone (SLT)

To make a page:

1. Lift the handset.
2. Enter the Internal Page feature code.
3. Enter the Page zone (00 – 99).
4. After the Page Warning tone, make announcement.
5. When finished, place the handset.

Conditions

- Extensions, which are denied access to paging, will receive error tone when attempting to page.
- An Extension receiving a page or accessing a page is indicated as busy to other callers.
- Extensions in DND or busy will not receive paging.

- Extensions not included in any Page zone cannot receive any page including All Call Page.
- Extensions with a held conference call or Conference Room are not allowed to page. In this case, the user will hear error tone.
- When Page zone 00 is called, all Extensions in Page zones 01-99 receive the page sequentially. If Extensions in more than one zone are called just once, the maximum number of called Extensions cannot exceed 128.

Related WMS Menu

[To assign Internal Page to flexible button]
Data Management > Extension Information > Terminal Information > Phone Flexible Button
[To set options for Page]
Data Management > Extension Information > Number (DN) Information > DN Member > Auto-Answer (Paging, Forced HF)
> Extension Information > Number (DN) Information > DN Feature Registration > "Paging Group Access Table"
[To set Internal Page Group & options]
Data Management > Extension Information > Group Information > Internal Paging Group
> Extension Information > Group Information > Internal Page Group Access Table
[To set max. time for Page]
Data management > Tenant Information > Tenant Basic Information > Tenant Timer > "Paging Time"
[To assign Internal Page feature code]
Data management > Numbering Plan Information > Feature Code>"Internal Page"

Related Features

- Meet Me Page Answer

Hardware

2.84.2 Meet Me Page Answer

Description

Any Extension may respond to a "Meet Me" Page request sent over an Internal Page zone. The user can answer the page from any Extension.

A {Meet Me} Flex button may be assigned on the Multi-button phone.

Operation

iPECS Multi-button Phone

To assign a {Meet Me} Flex button using the Extension User program:

- [PGM] + {Flex} + Button Feature Type (2) + {Internal Page Answer} feature code + [SAVE]

To respond to a page (when receiving a page):
1. Press the {Meet-Me} button or the flashing [HOLD] button.

To respond to a page (when page is not received):
1. Lift the handset or press the [Speaker] button.
2. Enter the {Internal Page Answer} feature code, or press the {Meet-Me} button.
3. Enter the Page group number (00~99).

Single Line Telephone (SLT)
To respond to a page (when page is not received):
1. Lift the handset.
2. Enter the {Internal Page Answer} feature code.
3. Enter the paging group number (00~99).

Conditions

- Page can be answered by any Extension regardless of paging group setting and page access permission.
- Paging parties must remain off-hook until the paged party answers the Meet Me request. The initiator can press the [MUTE] button to eliminate transmitting over the page circuit while waiting for the party to answer.
- The [HOLD] button flashes when the paged users allowed for Meet Me page answer, and the paged users can answer the page by pressing the flashing [HOLD] button.
- When the paging group number is unknown, paged users may answer with group 00. The system will connect the party to the lowest active page group.
- Users in the paging zone of another tenant cannot answer the page with the feature code. However, users can answer the page by pressing the {Meet-Me} button or the flashing [HOLD] button while receiving the page.

Related WMS Menu

[To assign Meet Me to flexible button]
Data management > Extension Information > Terminal Information > Phone Flexible Button
[To allow Page Answer (Meet Me Page)]
Data management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > “Page Answer (Meet Me Page)”
[To assign Internal Page Answer feature code]
Data Management > Numbering Plan Information > Feature Code >”Internal Page Answer”

Related Features

- Internal Page

Hardware

2.84.3 VM Paging

Description

Authorized users can record voice messages in VPCM to be played over designated Page zone(s). This feature enables users to record emergency messages to be used for internal paging during an emergency event.

Operation

Authorized user

To record a message for internal paging:

1. Press the [PGM] button.
2. Dial the recording code (‘9’ + ‘7’).
3. Enter a VPCM slot number (0001~8000).
4. Enter an announcement number (01~70).
5. Enter a language (1~9); the current recorded announcement will be played, and then the “Press # to start record” prompt will play.
6. Dial ‘#’, and after hearing the recording prompt and beep tone, record the message.
7. When finished recording, press the [SAVE] button or dial “#”.

To delete the message:

1. Press the [PGM] button.
2. Dial the recording code (‘9’ + ‘7’).
3. Enter a VPCM slot number (0001~8000).
4. Enter an announcement number (01~70).
5. Enter a language (1~9); the current recorded announcement is played, and then the “Press # to start record” prompt will play.
6. Press the [SPEED] button, or press the [Delete] soft button to delete the prompt being played; the system will provide the confirmation tone.

To assign an {Internal Page (Announcement Table No.)} to the Flex button using the Extension User program:

- [PGM] + {Flex} + Button Feature Type (2) + { Internal Page (Announcement Table No.)} feature code + [SAVE]

To play paging using the Announcement Table:

1. Lift the handset.
2. Enter the {Internal Page (Announcement Table No.)} feature code or press the {Internal Page (Announcement Table No.)} Flex button.
3. Enter the Page zone (00 – 99).
4. Enter the Announcement Table (0001 ~1000).
5. The call is connected after a Ring tone, and the announcement is played over the zone.
6. After the page, hang up.

<u>iPECS Multi-button Phone</u>

To assign {Voice Mail Record for Internal Page} to the Flex button using the Extension User program:

- [PGM] + { Flex} + Button Feature Type (2) + {Voice Mail Record for Internal Page} feature code + [SAVE]

To assign {Internal Page (Recorded Voice Mail)} to the Flex button using the Extension User program:

- [PGM] + {Flex} + Button Feature Type (2) + {Internal Page (Recorded Voice Mail)} feature code + [SAVE]

iPECS Multi-button phone and SLT

To record a message for internal paging:

1. Lift the handset.
2. Enter the {Voice Mail Record for Internal Page} feature code, or press the {Voice Mail Record for Internal Page} Flex button.
3. The current recorded announcement is played, and then, the “Press # to record” prompt.
4. Dial ‘#’.
5. Record messages with the microphone/headset after the recording prompt and the beep tone.
6. At completion of recording, press the [SAVE] button or dial “*”.
7. Dial options 1: Replay, 2: Record, #: End

To delete the message:

1. Lift the handset.
2. Enter the {Voice Mail Record for Internal Page} feature code, or press the {Voice Mail Record for Internal Page} Flex button.
3. The current recorded announcement is played, and then, the “Press # to start record” prompt.
4. Press the [SPEED] button, and the system provides the confirmation tone.

To Page using the recorded message:

1. Lift the handset.
2. Enter the {Internal Page (Recorded Voice Mail)} feature code or press the {Internal Page (Recorded Voice Mail)} Flex button.
3. Enter the paging zone (00 – 99, 00 is all groups)
4. The call is connected after a beep tone, and paging is started.
5. After the paging, hang up the call.

Conditions

- During VM paging, the caller of the paging zone also will hear the message.
- Users may use the Announcement Table or the recorded message for paging.
- If the message being played is stopped, paging will stop as well.
- VM paging can be answered by the same way as general paging; when page is answered, the message will be stopped immediately.

Related WMS Menu

[To assign feature codes for Page]

Data Management > Numbering Plan Information > Feature Code >"Voice Mail Record for Internal Page"
> Numbering Plan Information > Feature Code > "Internal Page (Recorded Voice Mail)"
> Numbering Plan Information > Feature Code > "Internal Page (Announcement Table No.) "

[To assign Page Group Access Index]
Data Management > Extension Information > Number (DN) Information > DN Feature Registration > "Internal Page Access Group Table"

[To assign Authorized User for VPCM announcement recording]
Data Management > Extension Information >Number (DN) Information> DN Feature Allow / Deny > "Privileged User"

[To assign Internal Page Group & access]
Data Management > Extension Information > Group Information > Internal Page Group, Internal Page Group Access Table

[To assign Announcement Table for VM Page]
Data Management > System Feature Information > Announcement Table

Related Features

- Internal Page

Hardware

- VPCM

2.85 Soft Phone Link

Description

A desk phone can be linked to a soft phone device (Phontage or UC client). The user can make calls from soft phone, and, for incoming calls, the caller information pops up on a PC screen. However, audio is only transmitted through the desk phone, while the PC microphone and speaker are disabled. The desk phone is controlled by the soft phone, and the LCD display information of desk phone is displayed to soft phone LCD area.

Operation

Conditions

- Only one hard phone can be linked to a soft phone.
- IP bridge option for soft phone must be set at log in.
- Not only can an iPECS Multi-button phone but also an analog phone can be linked to soft phone.

Related WMS Menu

[To link Soft phone to Hard phone]
Data Management > Extension Information > Terminal Information > Soft Phone Link

Related Features

Hardware

2.86 Pick Up Group

Description

Extensions can be assigned to Pick-Up Groups. This allows calls to other members of the group to be picked up (answered) based on the Pick-Up condition (All/Internal/Trunk Call).

Operation

Conditions

- An Extension can be a member of up to five (5) different Pick-Up groups.
- The maximum members in a Pick-Up Group are 100.

Related WMS Menu

[To assign Pick-up Groups]
Data Management > Extension Information > Group Information > Pick-Up Group

Related Features

- Group Call Pick-Up

Hardware

2.87 Pilot Hunt Group

Description

If an Extension is a member of Pilot Hunt Group, a call to the Extension can be routed to Pilot Hunt Group according to the defined conditions. One of three call types (Always/Trunk/Internal) can be selected for forwarding to Pilot Hunt Group and two hunt types (Circular/Terminal) are available. Call conditions are defined as Unconditional, Busy, No Answer and Busy/No Answer.

Operation

Conditions

- Calls cannot be routed to an Extension in DND or Call Forward.
- A Pilot Hunt Group can have up to 100 members.
- Up to 1000 Pilot Hunt Groups can be supported.

Related WMS Menu

[To assign Pick-up Group]
Data Management > Extension Information > Group Information > Pilot Hunt Call Forward

Related Features

- Do Not Disturb
- Call Forward

Hardware

2.87.1 Pilot Hunt Group Call Forward

Description

A member of Pilot Hunt Group can register Call Forward to reroute incoming calls to any member of the group to another Extension, Key Number Group, or outside telephone number according to the Extension status.

Pilot Hunt Call Forward Options are:

- Code 1 (Unconditional Call Forward) - All incoming calls except recalls forward regardless of the Extension status.
- Code 2 (Busy Call Forward) – Incoming calls forward if the Extension is busy.
- Code 3 (No Answer Call Forward) – All incoming calls not answered by the Extension forward after the no answer timer is expired.
- Code 4 (Busy/No Answer Call Forward) – Calls forward if the Extension is busy or does not answer the calls.

Operation

To activate Call Forward for Unconditional, Busy, No Answer and Busy/No Answer:

6. Lift the handset or press the [Speaker] button.
7. Enter the {Pilot Hunt Group Forward (Register} feature code.
8. Enter a Pilot Hunt Call Forward Option Code (range: 1~4).
9. Enter the destination number. For an external destination, dial the outside telephone number including Trunk Access code.)
10. Dial * or Press [HOLD/SAVE] button to save.

To deactivate Call Forward:

11. Lift the handset.
12. Enter the {Pilot Hunt Group Forward (Cancel)} feature code. If allowed, the DND LED will extinguish and confirmation tone is provided.

Conditions

- To activate the Pilot Hunt Call Forward feature, the DN of the Extension must be included in a specific Pilot Hunt Group.
- Call forward set by an Extension user has priority over Pilot Hunt Call Forward.

Related WMS Menu

[To set Pilot Hunt Group Call Forward]
Data management > Extension Information > Group Information >Pilot Hunt Call Forward

[To assign Pilot Hunt Group Forward feature code]

Data Management	> Numbering Plan Information > Feature Code > "Pilot Hunt Group Forward (Register)" > Numbering Plan Information > Feature Code > "Pilot Hunt Group Forward (Cancel)"

[To set time for No Answer Call Forward]

Data Management	> Tenant Information > Tenant Time Information > Tenant Timer > "No Answer Forward Time"

Related Features

- Call Forward
- Do Not Disturb (DND)
- Preset Call Forward

Hardware

2.88 Pre-defined Text Display

Description

Users can select a message to be displayed in the LCD of the calling party's phone. If the pre-defined text display function is active, for internal calls, the system will send muted ring to the called party and the pre-defined text to the LCD of the calling party.

The ten (10) messages (0~9) are fixed. Users can enter subsidiary information such as time, date and number. Users may be able to enter a user-defined message in addition to the ten (10) fixed messages.

iPECS Multi-button phone users can assign the {Pre-Defined Text register} feature code to a Flex button.

The ten (10) fixed messages include:

Message 1

```
LUNCH RETURN AT hh:mm
```

Message 2

```
ON VACATION
RETURN AT DATE mm-dd
```

Message 3

```
OUT OF OFFICE
RETURN AT TIME hh:mm
```

Message 4

```
OUT OF OFFICE
RETURN AT DATE mm-dd
```

Message 05

OUT OF OFFICE RETURN UNKNOWN

Message 6

CALL: (enter 17 digits)

Message 07

IN OFFICE: STA xxxx

Message 8

IN A MEETING RETURN AT TIME hh:mm

Message 9

AT HOME

Message 0

AT BRANCH OFFICE

Operation

iPECS Multi-button Phone

To assign the {Pre-Defined Text Register} feature code to a Flex button using the Extension User program:

- [PGM] + {FLEX} + Button Feature Type (2) + {Pre-Defined Text Register} feature code + [SAVE]

To activate the Pre-Defined Text Display:

1. Dial [PGM] + 4 + 1 or the {Pre-Defined Text register} feature code.
2. Use [VOL UP]/[VOL DOWN] to view available messages.
3. Enter the message code (1-9, 0, *).
4. Enter auxiliary input, as desired,
5. Press the [SAVE] button.

To deactivate the Pre-Defined Text Display:

1. Press the flashing [DND] button.
   Or
   Press [PGM] and dial 4 + 1 or the {Pre-Defined Text register} feature code.
2. Dial ‘#’.
3. Press the [SAVE] button.

To define the user message:

1. Dial [PGM] + 4 + 2.
2. Enter the user-defined message of up to 16 characters (12 characters for input from phone).
3. Press the [SAVE] button.

4. Dial [PGM] + 4 + 1 or the {Pre-Defined Text register} feature code.
5. Dial '*'.
6. Press the [SAVE] button.

<u>Single Line Telephone (SLT)</u>
To activate the Pre-Defined Text Display feature:

1. Lift the handset.
2. Enter the {Pre-Defined Text register} feature code.
3. Enter the message code (1-9, 0).
4. Enter auxiliary data as desired.
5. Press and release the hook-switch (hook-flash), confirmation tone is provided.

To deactivate the Pre-Defined Text Display feature:

1. Lift the handset.
2. Enter the {Pre-Defined Text register} feature code, or {All Feature Cancel} feature code.
3. Dial '#'.
4. Press and release the hook-switch (hook-flash0, confirmation tone is provided.

Conditions

- The Pre-defined Message Display is deactivated if the user activates DND or Call Forward.
- User messages and the user–defined messages are protected against power loss.
- The Extension user must use an iPECS Multi-button phone with an LCD to receive the message.
- If the feature is active, stutter dial tone is provided to the SLT users. In case of iPECS Multi-button phones, the [DND] button LED blinks at an interval of 15 ipm if the feature is active.
- Activating Pre-defined Text Display does not affect normal operation of the Extension.
- Pre-defined texts 1~4 and 6~8 require additional inputs (time, date, number).
- WIT-400H can display Pre-Defined Text Message.

Related WMS Menu

[To set duplication use of Call Forward, DND or Pre-defined Text Message]
Data Management > Tenant Information > Tenant Attribute > "Call Forward, DND, Absence MSG Duplication Use"
[To assign Pre-defined Text Register feature code]
Data Management > Numbering Plan Information > Feature Code > "Pre-Defined Text Register"
[To use Pre-defined Text display]
Data Management > Extension Management > Number(DN) Information > DN Feature Allow / Deny > "Pre-defined Text Display"
[To set Pre-defined Text]
Data Management > Extension Management > Number(DN) Information > DN Feature Registration > "Pre-defined Text"
> Extension Management > Number(DN) Information > DN Feature Registration > "More Info for Pre-defined Text"

> Extension Management > Number(DN) Information > DN Feature Registration > "User-defined Text"

Related Features

- Do Not Disturb
- Call Forward

Hardware

- iPECS Multi-button Phone with LCD
- WIT-400H

2.89 Prepaid Money

Description

An Extension may be allocated an amount of money for call charges. The Extension is then allowed to place Trunk calls until the call costs consume the “Prepaid Money. If the Prepaid Money is consumed during a conversation, a warning tone is given to the Extension, and after Prepaid Release Time, the call will be disconnected. The phone cannot make outgoing Trunk calls once charges consume the pre-assigned funds.

Operation

To Charge/Delete Prepaid Money

1. Dial {Prepaid Money Register} (Attendant, Front Desk, Authorized Extension),
   Or
   [PGM] + 9 + 8 (Authorized Eension)
   Or,
   [PGM] + 0 + * (Front Desk)
2. Dial an Extension number or range
3. Press the [SAVE] button, or dial digit “*”.
4. Dial Prepaid Money for charging, or dial “0” to delete.
5. Press the [SAVE] button, or dial digit “*”.

Conditions

- Up to nine (9) digits can be entered for the Prepaid Money.
- The Prepaid Money input is added to any remaining money.
- This feature is available for Trunks that receive call-metering signal.
- The calculated meter cost from Trunk is not included in SMDR. The calculated meter cost is shown in for LCD of the Extension.
- Prepaid Money can be registered from the Attendant, Front Desk, or any authorized Extension. Prepaid funds can also be entered through WMS.
- The currency unit for charges and Fraction for Call Charges are based on the tenant configuration.
- The Fraction of AOC of Outgoing Route Options is used for multiplying the cost received from the Trunk.

- When the user dials an Emergency Code, the Prepaid Money is not consumed.
- For proper operation, the "Charge Method of Tenant Attribute" menu should be set to Total Charge.
- If an Extension user enters a User Authorization Code to place an external call, the cost is charged to the user assigned the Authorization not the Extension that places call.

Related WMS Menu

[To set options to use Prepaid Money]

Data Management	> Extension > Number(DN) Information > DN Feature Allow / Deny > "Prepaid Money" > Extension > Group Information > Feature Allow / Deny Group > "Prepaid Money" > Extension > Number(DN) Information > DN Attribute > "Prepaid Money" > Extension > Extension Information > Terminal Information > Terminal Option > "AOC Metering"

[To set Charge options for Prepaid Money]

Data Management	> Tenant Information > Tenant Basic Information > Tenant Attribute > "Charge Method." > Tenant Information > Tenant Basic Information > Tenant Attribute > "Currency Unit for Charge" > Tenant Information > Tenant Basic Information > Tenant Attribute > "Fraction for Call Charge" > Tenant Information > Tenant Time Information > Tenant Timer > "Prepaid Release Time"

[To set Cost options for Charge]

Data Management	> Trunk Information > Outgoing Route Information > Outgoing Route Options > "Cost per Metering Pulse" > Trunk Information > Outgoing Information > Outgoing Route Options > "AOC Currency Adjust"

Hardware

- PRI,
- Analog line (LGCM) with metering Unit

2.90 Preset Call Forward

Description

The system can be configured to forward calls to an Extension automatically. Preset Call Forward can be configured with different destinations depending on System Service Time Mode (Day/Night/Timed), call origination (Internal/External) and the Extension state (Unconditional, Busy, No Answer) as below.

- Unconditional Call Forward – All Extension and Trunk calls forward unconditionally.
- Busy Call Forward for Extension Call – Internal calls forward if the Extension is busy.

- No Answer Call Forward for Extension Call – All the Extension calls not answered by the Extension forward.
- Busy Call Forward for Trunk Call – External calls forward if the Extension is busy.
- No Answer Call Forward for Trunk Call – All trunk calls not answered by the Extension forward.

Operation

Conditions

- Call Forward registered by an Extension has precedence over Preset Call Forward.
- A forwarded call can be transferred to the forwarding Extension.
- Calls will not forward to an Extension in DND.
- If Preset Call Forward is configured in a chain, and the chain includes an Extension in DND, calls bypass to the next Extension, skipping the Extension in DND. If the Extension in DND is the last in the chain, the previous Extension is considered the last Extension in the chain.
- Preset Call Forward is affected by the type of incoming call (internal or external), as well as Extension status.
- Information on Preset Call Forward is not indicated in the LCD of the iPECS Multi-button phones.
- The Station Ring Lock time is employed as the default No-answer time.
- If Time Zone is holiday, Day/Night Mode is always night.

Related WMS Menu

[To define System Time Zone]
System Management > System Time Zone
[To assign Time Zone of Tenant]
Data Management > Tenant Information > Tenant Basic Information > Tenant Attribute > 'Time Zone No"
[To assign Preset Call Forward]
Data Management > Extension Information > Number (DN) Information > DN Feature Registration > "Preset Call Forward"

Related Features

- Call Forward

Hardware

2.91 Push-To-Talk Paging

Description

iPECS Multi-button phones can be assigned to one or more of ten (10) PTT (Push-To-Talk) groups. The users may log in or out of the assigned PTT groups. When logged in, users may send or receive One-way pages to or from other users logged in to the same PTT group. To send a PTT page, users must press and hold the {PTT} Flex button.

Operation

<u>iPECS Multi-button Phone</u>

To assign a {PTT} Flex button using the Extension User program:

- [PGM] + {FLEX} + Button Feature Type(1) + PTT (select the function with the Volume controls) + [SAVE]

To log in to a PTT group:

1. Enter the {PTT Group Login/Logout} feature code.
2. Enter the PTT group number (1~9, 0).

To log out from a PTT group:

1. Enter the PTT login/logout code.
2. Dial ‘*’.

To place a page to the active PTT group:

1. Press the [PTT] Flex button.
2. After confirmation tone, make page announcement.
3. After the page, press the {PTT} Flex button or place the handset.

<u>WIT-400H Phone</u>

To page an active PTT group:

1. Press and hold the [PTT] button.
2. After confirmation tone, make page announcement.
3. Release the [PTT] button.

Conditions

- Conditions associated with Internal Page are also applied to Push to Talk Paging.
- An Extension can log in to a group to send/receive a page to/from one group at a time.
- An Extension may log in to one PTT group at a time.
- An Extension must have the {PTT} Flex button to send and receive PTT page. An iPECS WLAN phone has an assigned PTT button.
- Extensions with a held Conference call or Conference Room are not allowed to use PTT paging. In this case, the user receives error tone when accessing PTT paging.
- A user logged in to PTT group 0 can send or receive all PTT pages. The maximum number of paged users is 128.
- Users cannot log in to PTT groups of other tenants. However, they can be assigned as members of PTT groups of other tenants through WMS. In this case, users cannot page the PTT group but can receive pages.
- For soft phones (UCS Client or Phontage), PTT paging feature code is not left in the call log.

Related WMS Menu

[To assign PTT Group Login/Logout feature code]

Data Management	> Numbering Plan Information > Feature Code > “PTT Group Login/Logout”

[To assign PTT button on flexible button]

Data Management > Extension Information > Terminal Information > Phone Flexible Button
[To assign PTT Group of DN]
Data Management > Extension Information > Number(DN) Information > DN Attribute > PTT Group
[To assign PTT Group]
Data Management > Extension Information > Group Information > PTT Group

Related Features

- Internal Page

Hardware

- Multi-button Phone
- WLAN Phone(WIT-400H)
- Phontage Client

2.92 R2 Comfort Tone

Description

When an Extension or incoming Trunk places an outgoing call over an R2 Trunk, the caller may hear silence for several seconds after dialing due to the R2 signal processing time. For this silence period, the R2 Comfort Tone is used to notify the caller that the call is being processed.

Operation

Conditions

- This function is applied to R2 Trunks only.
- The R2 Comfort Tone is provided after the Outgoing Route Inter digit time expires.
- The R2 Comfort Tone stops at the end of the R2 signaling when Ring Back Tone is received from the remote PSTN.
- The R2 Comfort Tone may not be provided when Off-net Call Forward and its tone service are processed at same time.
- If the tone duration of R2 Comfort Tone is set to “0”, the system does not provide this service.

Related WMS Menu

[To set R2 Comfort Tone]
Data Management > Tenant Information> Tenant Tone/Ring Information > Tenant System Tone > “R2 Comfort Tone”
[To set R2 Connection Method]
Data Management > Trunk Information > Trunk Basic Information > Trunk Access Code > “R2 Transit Signaling Method”
[To set R2 Signal Conversion Group]
Data Management > Trunk Information > Incoming Route Information > Incoming Route Options > “R2 Signal Conversion Group”
Outgoing Route Information > Outgoing Route Options > “R2 Signal Conversion Group”

Outgoing Route Information > Outgoing Route Options > “R2 Comfort Tone (PDD)”
> Outgoing Route Options > R2 signal Conversion Group

[To set Outgoing Route Inter Digit Time]

Data Management > Trunk Information > Outgoing Route Information > Outgoing Route Inter Digit Time

Related Features

- Off-net Call Forward Tone(Announcement) Service
- Transit Service

Hardware

- MDTM

2.93 Redirecting Number

Description

When an incoming Trunk routes to an Extension that has an active Off Net Forward, the system seizes another Trunk and sends the SETUP or IAM message including the information on the Off-net Forward Extension.

Operation

Conditions

- This function is available for ISDN, H.323, SIP Trunks and SS7 only.
- When an Extension calls an Extension with Off-net forward active, the redirecting number is included in the SETUP and IAM message.
- If an incoming Trunk call includes the redirecting number, the redirecting number is transferred to the Trunk transparently when the outgoing Trunk is seized.

Related WMS Menu

[To add Own Code to SETUP message]

Data Management > Trunk Information > Outgoing Route Information > Outgoing Route Options > “Add Own Code to Received CID on Transit”
> Trunk Information > Outgoing Route Information > Outgoing Route Own Code

Related Features

- Transit in-out

Hardware

2.94 Revertible Ring

Description

This feature is employed to assure a phone is ringing properly. The Extension calls the Extension's number and hangs up, and then the Extension will ring normally.

Operation

To use revertible ring:

1. Lift the handset.
2. Dial the Extension number of user's phone; Confirmation tone will be heard.
3. Phone will ring when the user replaces the handset.

Conditions

Related WMS Menu

Related Features

Hardware

2.95 Ring Tone Download

Description

The user can select one of 14 Ring tones so that the iPECS Multi-button phone ring can be distinguished from other nearby phones. Eight tones are stored in the phone memory. The first four tones are fixed and the 5th through 8th Ring tone can be downloaded from a library of ten (10) tones stored in the system's memory.

After downloading the tone from the system memory, it can be selected as the Differential Ring tone.

Operation

IPECS Multi-button phone

To download a ring tone from the system memory to the phone memory:

1. Press the [PGM] button.
2. Dial Extension User program code 23.
3. Enter the desired the ring tone location, '5'~'8', ring tone (0~9) list is displayed on the LCD.
4. When a tone (0~9) is selected, the ring tone can be heard
5. To download, dial 1, or dial 2 to select another ring tone.

To select the downloaded ring tones:

1. Press the [PGM] button.
2. Dial '21' for Extension Ring Tone, or '22' for Trunk Ring Tone.
3. Enter the digit (5~8) to select the Ring Tone.
4. Press the [SAVE] button.

Conditions

- The downloadable Ring Tone files are stored in the system memory as *.wav files with maximum length of 4 seconds. These files can be replaced as desired using the iPECS-CM WMS Upload function.

Related WMS Menu

[To assign Differential Ring]

Data Management > Extension Information > Terminal Information > Terminal Option > "Differential Ring ID for Internal Call"
> Extension Information > Terminal Information > Terminal Option > "Differential Ring ID for External Call"

[To upload Ring Source to Call Server]

Version Management > Tone, Ring Upload

Related Features

- Station Program/Differential Ring

Hardware

- iPECS Multi-button Phone

2.96 Route CLI Service

Description

When a user places an external call, the system can generate the CLI information using the Route Key Number assigned to each route, instead of generating CLI using the Trunk and/or Extension number.

Operation

Conditions

- A single Route Key Number can be assigned to each outgoing route and up to 16 digits can be entered.

Related WMS Menu

[To use Route Key Number for CLI]

Data Management > Trunk Information > Outgoing Route Information> Outgoing Route Key Number
> Trunk Information > Outgoing Route Information> Outgoing Route Options > "Outgoing Route Key Number as CID"

Related Features

Hardware

2.97 Save Number Redial (SNR)

Description

The last dialed internal/external number may be saved in the buffer for future redial. The saved number remains in the memory until the user saves a new number.

Operation

iPECS Multi-button phone

To save a number during a call

1. Press the [SPEED] button twice.

To redial the saved number:

1. Lift the handset or press the [Speaker] button.
2. Press the [SPEED] button.
3. Dial '#'.

Conditions

- Up to 32 digits can be saved for SNR (Save Number Redial).
- If a user redials the saved number, the system automatically assigns the DN that was previously used. If the DN is busy, the prime DN is selected automatically.
- The saved number is protected against power loss.
- SNR is available to the iPECS Multi-button phone only.

Related WMS Menu

Related Features

- Last Number Redial
- Call Log

Hardware

- Multi-button Phone

2.98 SOSM (Police Function)

Description

Designated user's call(voice and tone) is monitored via E1 link. It is used in only Russia

Operation

Conditions

- XMS, MDTM and VPCM are needed for the function.
- User and monitoring connection line should be designated for monitoring.
- General Call, Call Forward, Call Wait, Conference, Call Transfer, Call Pick-up are supported.

Related WMS Menu

Application Information > SOSM Server Information

Related Features

- Call Forward
- Call Wait
- Conference
- Call Transfer
- Call Pick-up

Hardware

- XMS
- MDTM
- VPCM

2.99 Speakerphone

Description

An iPECS Multi-button phone equipped with speakerphone circuitry enables the telephone to be used hands-free in two-way conversations.

Operation

To activate the speakerphone:

1. Press the [Speaker] button, [Speaker] LED lights steady.

To switch from the handset to the speakerphone:

1. Press the [Speaker] button, [Speaker] LED lights steady.
2. Replace Handset, Speakerphone is activated.

To terminate a speakerphone call:

1. Press the [Speaker] button, [Speaker] LED extinguishes.

Conditions

- The Speakerphone is automatically activated by pressing a DN, Trunk Access or a Speed Dial button.
- The [Mute] button LED indicates the status of the microphone.
- 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.

Related WMS Menu

Related Features

- Mute
- Group Listening

- On- Hook Dialing

Hardware

- Multi-button Phone

2.100 Speed Dial

2.100.1 Speed Dial Pause Insertion

Description

A Pause command can be inserted in a Station or System Speed Dial number. When the Pause command is encountered during speed dialing, the system will stop dialing for the defined pause time. The system will restart dialing when the pause time expires. Multiple pauses can be inserted into a Speed Dial.

Operation

Conditions

- A Speed Dial pause may be used on analog Trunks only.

Related WMS Menu

[To assign Station Speed Dial]
Data Management > Extension Information > Terminal Information> Station Speed Dial
[To assign System Speed Dial]
Data Management > System Feature Information > System Speed Dial
[To change Pause Time used for Speed Dial]
Data Management > Tenant Information > Tenant Time Information > Tenant Timer > "DTMF Pause Time"

Related Features

- Station Speed Dial
- System Speed Dial

Hardware

2.100.2 Station Speed Dial

Description

iPECS-CM can store up to 100 Speed Dial numbers bins (00 ~ 99) for each Extension. Each Speed Dial can be programmed with up to 32-digits.

A Speed Dial number can include the following special characters:

- Flash as 1st Digit Activates the dial tone detect function
- [Flash] Activates the trunk flash function

- [Call Back] Activates the Speed Dial pause insertion.

iPECS Multi-button phone users can assign a Station Speed Dial bin to a Flex button. An external telephone number can be assigned directly to the Flex button.

Operation

iPECS Multi-button Phone

To dial using a Station Speed Dial number:

1. Lift the handset or press the [Speaker] button.
2. Press the [Speed] button.
3. Enter the Station Speed Dial bin (00-99).

To program a Station Speed Dial bin:

1. Press the [Speed] button, or press the [DIR] soft button and select “1 Station Speed” menu.
2. Press the [ADD] Soft button.
3. Enter the Station Speed Dial bin (00-99).
4. Enter the number to be stored for an external number, dial the number including the Trunk Access code.
5. Press the [SAVE] button.
6. Enter the name for the Speed Dial number using Character Entry Chart below.
7. Press the [SAVE] button.

To search the Station Speed Dial numbers and place a call:

1. Press the [Speed] button, or press the [DIR] soft button and select “1 Station Speed” menu.
2. Press the [SEARCH] soft button.
3. Enter the key word to search the Speed Dial number.

Single Line Telephone (SLT)

To place a call using a Station Speed Dial number:

1. Lift the handset, dial the {Station Speed Dial} feature code.
2. Enter the Station Speed Dial bin number (00-99).

To program a Station Speed Dial bin:

1. Lift the handset and dial the {Station Speed dial(Register)} feature code
2. Enter the Station Speed Dial bin number (00-99).
3. Enter the number to be stored; for an external number, dial the number including a Trunk access code.
4. Press and release the hook-switch (hook-flash).
5. Enter the name for the Speed Dial number using the Character Entry Chart below.
6. Press and release the hook-switch (hook-flash).

Character Entry Chart (LIP-8000 Series except 8004, 7024LD)

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

Character Entry Chart (LIP-7000 Series except 7024LD, LIP-8004)

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

Conditions

- Accessing an empty Speed Dial bin will return error tone.
- Station Speed Dial numbers are protected against loss of power.
- Special characters (Pause and Flash) can be included by an SLT through the Extension User Web portal.

Related WMS Menu

[To assign Station Speed Dial]
Data Management > Extension Information > Terminal Information > Station Speed Dial
[To assign System Speed Dial]
Data Management > System Feature Information > System Speed Dial
[To assign Station Speed Dial Register/Use feature code]
Data Management > Numbering Plan Information > Feature Code > "Station Speed Dial"
> Numbering Plan Information > Feature Code > "Station Speed dial(Register)"

Related Features

- Last Number Redial
- Save Number Redial
- Speed Dial Pause Insertion

Hardware

2.100.3 System Speed Dial

Description

iPECS-CM can store up to 3,000 System Speed Dial numbers, which can be used by all system users and can be set with WMS. Each System Speed Dial number can be programmed with up to 24-digit numbers.

A Speed Dial number can include the following special characters:

- Flash as 1st Digit Activates the dial tone detect function.
- [Flash] Activates the trunk flash function.
- [Call Back] Activates the Speed Dial pause insertion.

iPECS Multi-button phone users can assign a Speed Dial number or an external number to the Flex button.

Operation

iPECS Multi-button phone

To place a call using a System Speed Dial number:

1. Lift the handset or press the [Speaker] button.
2. Enter the bin number.

To search for a System Speed Dial number and place a call:

1. Press the [DIR] soft button and select "2 System Speed" menu.
2. Press the [SEARCH] soft button.
3. Enter the key word to search the Speed Dial bins using character entry chart in Station Speed Dial.
4. Press the [SAVE] button.

Single Line Telephone (SLT)

To place a call using a System Speed number:

1. Lift the handset.
2. Enter the System Speed Dial bin number.

Conditions

- Accessing an empty Speed Dial bin will return the error tone.
- System Speed Dial numbers are protected against loss of power.
- If a Speed Dial number contains a dial tone detect command, the system must detect the dial tone before dialing the speed dial number.

Related WMS Menu

[To assign speed dial number to use]

Data Management > Numbering Plan Information > Numbering Plan
> System Feature Information > System Speed Dial
> Extension Information > Terminal Information> Station Speed Dial

Related Features

- Trunk Redial
- Last Number Redial
- Save Number Redial
- Speed Dial Pause Insertion
- Station Speed Dial

Hardware

2.100.4 Group Speed Dial

Description

Each Speed Dial Group can store up to 500 Group Speed Dial numbers, which can be used by members of group and can be set with WMS. Each Group Speed Dial number can be programmed with up to 24-digit numbers.

A Speed Dial number can include the following special characters:

- Flash as 1st Digit — Activates the dial tone detect function.
- [Flash] — Activates the trunk flash function.
- [Call Back] — Activates the Speed Dial pause insertion.

iPECS Multi-button phone users can assign a Speed Dial number or an external number to the Flex button.

Operation

iPECS Multi-button phone

To search for a System Speed Dial number and place a call:

1. Press the [DIR] soft button and select “2 Group Speed” menu.
2. Press the [SEARCH] soft button.
3. Enter the key word to search the Speed Dial bins using character entry chart in Group Speed Dial.
4. Press the [SAVE] button.

Conditions

- Accessing an empty Speed Dial bin will return the error tone.
- Group Speed Dial numbers are protected against loss of power.
- If a Speed Dial number contains a dial tone detect command, the system must detect the dial tone before dialing the speed dial number.
- Only authorized user can add new speed dial number.

Related WMS Menu

[To assign speed dial number to use]

Data Management	> Numbering Plan Information > Numbering Plan
	> Tenant Information > Tenant Basic Information > Tenant Attribute > “Group Speed Dial Use”
	> Extension Information > Terminal Information > Terminal Attribute > “Speed Group”
	> Extension Information > Group Information > Group Speed Dial

Related Features

- Trunk Redial
- Last Number Redial
- Save Number Redial
- Speed Dial Pause Insertion
- Station Speed Dial

Hardware

2.100.5 Bell by CID

Description

When an internal or external call is routed to an Extension, the ring signal can be assigned differently according to the received CID. When the incoming Caller Id matches a Speed Dial number, the system will signal the user with the ring type selected for the Speed Dial.

Operation

iPECS Multi-button phone

To select Bell by CID:

1. Press the [DIR] soft button while in idle.
2. Select the Station Speed Dial program.
3. Press the [SEARCH] button.

4. Search the Speed Dial numbers.
5. Press the [EDIT] button when the Speed Dial number is displayed.
6. Press the [BELL] soft menu button.
7. Dial bell type (0~8).
8. Press [SAVE] button.

Conditions

- Bell by CID can be selected for the Station Speed Dials only. It cannot be assigned when installing the program but in the edit mode only. In the edit mode, you can edit the speed dial with the designated Speed Dial digit and name. That is, in order to select a Bell by CID, the speed dial digit and the name must be registered in the Station Speed Dial program.
- If the input value for the selected bell is set to '0', the bell assigned with [PGM] + 21 and with [PGM] + 22 will be selected for internal calls and Trunk calls, respectively.

Related WMS Menu

[To select bell by CID]
Data management > Extension Information > Terminal Information > Station Speed Dial > "Ring Type"

Related Features

Hardware

2.100.6 Dial by Name

Description

A name, up to 16 characters, may be assigned to each Station and System Speed Dial. When assigned, a user may select a Station or System Speed Dial using the name.

The user selects from one of two Dial-by-Name directories and enters characters employing dial pad buttons for each character. The system finds and displays the nearest match to the user entries. The user may continue entering characters or scroll the directory at any point using the [Vol up]/[Vol dwn] button and select a name to call.

Operation

To use Dial by Name:

1. Press DIR soft button.
2. Dial the desired directory
   1: Station Speed Dial
   2: System Speed Dial
   3:Station Name,
   the LCD will display the names in alphabetical order
3. Scroll using the Navigation up/down keys.

4. To enter character, press SEARCH soft button, or press dial pad with the desired character as below

Digit	Search Character	Digit	Search Character
1	@ : /	7	PQRS
2	ABC	8	TUV
3	DEF	9	WXYZ
4	GHI	*	*
5	JKL	0	.
6	MNO		

5. Press the [Hold/Save] to place the call.

To set a Station Name for an iPECS Multi-button phone
1. Press the [PGM] button.
2. Dial 1 + 2 (Station Name Program code)
3. Enter a name.
4. Press the [HOLD] button.

To set a Station Name for Single Line Telephone (SLT)
1. Lift the handset.
2. Enter the {Station Name Register} feature code.
3. Enter a name.
4. Press and release the hook-switch 9hook-flash) to save.

Conditions
- Available characters are A to Z, @, ':', '/', '*', '#' and period.
- The LCD will display multiple names, one per LCD line, up to 16 characters each.
- If a user selects a Directory with no entries or there is no match to the user entry, the "LIST EMPTY" message is displayed.
- Dial-by-Name is only available to iPECS Multi-button phones with a display.
- A user may both scroll and enter characters to search a directory.
- The LCD of opposite user will display name up to 24 characters during call.
- If display area of LCD is shorter than length of name, display of name will be cut.

Related WMS Menu
[To assign Station Speed Dial]
Data Management > Extension Information > Terminal Information > Station Speed Dial
[To assign System Speed Dial]
Data Management > System Feature Information > System Speed Dial

Related Features
- Station Speed Dial
- System Speed Dial

Hardware
- iPECS Phone w/Display

2.101 sRTP Feature

Description

In the past, IPKTS Device which use LG Proprietary protocol for communicating between device and iPECS-CM don't support standard sRTP(AES 128/192/256). So, SIP Phone which support standard sRTP can't connect sRTP session directly with IPKTS device. But, currently, Gateway(MDTM, MATM, DSLM, VPCM, ASLM) and Phone (LIP-80xxE series, LIP-9070) support standard sRTP for connecting sRTP session directly with SIP Phone. For compatibility with old version device which don't support standard sRTP, system can distinguish which device can support standard sRTP. If there is old version Gateway (MDTM, MATM, DSLM, VPCM, ASLM) and phone (LIP-80xxE series), system can distinguish which device can support standard sRTP. Then, if there is device which can't support standard sRTP, system try to connect to sRTP session with LG Proprietary sRTP.

Operation

Conditions

- Currently, only below Gateways and Phones support this feature
- CM gateway : MATM, MDTM, VPCM, DSLM, ASLM
- LIK gateway : VOIM
- Phone : LIP-80xxE Series, LIP-9070
- sRTP connection between IPKTS devices is same way with in the past.
- Don't support sRTP connection directly between SIP Phone and old version Gateway and Phone.
- By considering old CM version and new Gateway, Phone version, new version Gateway and Phone can support old sRTP method with old CM version.
- System can distinguish which device can support standard sRTP when gateway and phone register to system.

Related WMS Menu

Related Feature

Hardware

2.102 Station Feature Allow/Deny

Description

Feature Allow/Deny Groups can be configured for each DN (Directory Number). When any features of Feature Allow/Deny Group are changed, the change is applied to all Extension numbers (DNs) using the same Feature Allow/Deny Group.

Operation

Conditions

- Up to 128 Feature Allow/Deny Groups can be defined.
- If the Feature Allow/Deny Group is not selected in DN Feature Allow/Deny, the feature can be allowed or denied individually.
- If the Feature Allow/Deny Group is assigned in the DN Feature Allow/Deny, the Feature Allow/Deny table of the DN cannot be changed and does not work.

Related WMS Menu

[To use Feature Allow/Deny Group]
Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > "Feature Allow/Deny Group"
[To set Feature Allow/Deny Group]
Data Management > Extension Information > Group Information > Feature Allow/Deny Group

Related Features

Hardware

2.103 Station Lock Out

Description

When an Extension goes Off-hook and does not dial any digit in the 1st Dial Tone time, or delays dialing between digits in excess of the Extension Button Input Time, or stays Off-hook at the completion of activating a feature or program, the Extension will receive Howling tone as an error indication and internal calls are abandoned. In order to make or receive calls, the user must go On-hook.

Operation

Conditions

- Howling tone is sent after any kind of error tone such as Uncompleted Dial tone, Internal No Answer tone, Internal Vacant tone, etc.
- Station Lock-out occurs when the howling tone starts.

Related WMS Menu

[To set option for Lock out auto release of iPECS Multi-button phone]
Data Management > Extension Information > Terminal Information > Terminal Option > "Lockout/Speaker Mode Release Type"
[To set 1st Dial Tone, Howling Tone]
Data Management > Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > "1st Dial Tone", "Howling Tone"
[To set Extension Button Input Time]

Data Management > Tenant Information > Tenant Time Information > Tenant Timer > “Extension Button Input Time”

Related Features

- Internal Call

Hardware

2.104 Step Call

Description

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

Operation

iPECS Multi-button Phone

To make a step call when the dialed Extension is busy:

1. Dial the last digit of the next Extension to call.

Conditions

- After receiving Busy tone, if the user takes no action for the Internal or External Busy tone time, the system will initiate the Station Lock out process.
- This function is available when the “Step Call” option of “One Digit Service on Busy” is allowed.

Related WMS Menu

[To enable Step Call]

Data Management > Tenant Information > Tenant Basic Information > One Digit Service on Busy > "Step Call"

[To set Internal/External Busy Tone time]

Data Management > Tenant Information > Tenant Tone/Ring Information > Tenant Tone Information>“Internal Busy Tone”

> Tenant Information > Tenant Tone/Ring Information > Tenant Tone Information> “External Busy Tone”

Related Features

- Internal Call
- Busy One Digit Service

Hardware

- iPECS Multi-button phone

2.105 System Clock Set

Description

The system Time and Date can be set by an authorized Extension or through WMS.

Operation

Authorized Extension

To set the system time/date:

1. Press the [PGM] button.
2. Dial the Extension User program code, “91” or “92”.
3. Enter four digits for the time (hour/minute) and six (6) digits for date (month/day/year).
4. Press the [SAVE] button, confirmation tone is received and the new setting is applied.

Conditions

- The time data must be entered as the 24-hour format.
- When an NTP Server is assigned in WMS, the system clock is synchronized by the time server.

Related WMS Menu

[To set System Clock]

WMS Management > WMS Configuration > WMS Time Configuration

Related Features

- System Time Zone Setting
- WMS Time Configuration

Hardware

2.106 System Networking

Description

Each iPECS family of systems (iPECS-CM, iPECS-LIK and iPECS MG) as well as the IP enabled digital systems (ipLDK series) can be connected in a distributed control network. In this environment, devices are controlled by the local call processor and have access to resources and certain features of the other systems in the network. The features and functions described in the following subsections can be accessed by users in the distributed network configurations.

The iPECS-CM can request access to the resources of a remote system. The system analyzes and routes calls according to the “Voice Network Numbering Plan” table in order to determine the appropriate destinations.

iPECS-CM supports two types of networking protocols, QSig on ISDN and H.450 on IP, for both basic and advanced networking functions. ISDN PRI interfaces using ESTI standards (ETS

300-237/238/256/257/260 /261 /361/362/363/364) are employed to support QSig. IP connections employ H.450 standards for networking.

Operation

Conditions

- A QSig license is required for the system to access network features.
- With the Unified Dialing Plan (UDP), each Extension can have a unique network ID (up to 8-digits) that is determined by the unique Flexible Numbering Plan of the associated system.

Related WMS Menu

[To set Numbering Plan]
Data Management > Numbering Plan Information > Numbering Plan
[To enable Voice Network]
Data Management > Voice Network > Voice Network Attribute > “Voice Networking”
[To set Voice Network Numbering Plan]
Data Management > Voice Network > Voice Network Numbering Plan

Related Features

Hardware

- VPCM
- MDTM

2.106.1 Network Call

Description

An Extension user can place a call to Extension in other systems in the network by dialing only Extension number as an intercom call over the network.

Operation

To place a network intercom call from an Extension:

1. Lift the handset or press the [Speaker] button.
2. Dial the desired Extension number on another system.
3. The system will access the networking route Trunk based on the “Voice Network Numbering Plan” and send the network call.
4. The system in the destination network receives the call signal from the caller and routes the call to the destination Extension.
5. The network routing Trunk is seized and the network call is connected. The Trunk is returned to idle when the call is terminated by either Extension.

To place a network call from an Attendant console:

1. Dial the Extension; information on the Extension is displayed.

2. Press the [CONNECT] soft button; the system seizes the networking route Trunk based on the “Voice Network Numbering Plan” table, and sends the network call.
3. The network routing Trunk is seized to connect the network call and is returned to idle when the call is terminated by either Extension.

Conditions

- If there are no available networking paths, the user receives the error tone.
- The called party receives the Trunk ring signals for the network call.

Related WMS Menu

Related Features

- System Network

Hardware

2.106.2 Network Transfer

Description

Extension users can transfer calls to an Extension in another networked system by pressing the [Trans] button and dialing the desired Extension number. Calls can be transferred in both screened and unscreened.

In the network there are two transfer mechanisms, Transfer by Join and Transfer by Rerouting. For Transfer by Join, an additional connecting path is needed to transfer calls to another Extension. For Transfer by Rerouting, a new connecting path is used to transfer the call and the old connecting path of transferring Extension is cleared. Transfer by Rerouting is used for iPECS-CM.

Operation

Screened transfer:

1. Press the [TRANS] button while on a trunk call; the call is placed on hold.
2. Dial the number of the Extension in the other system.
3. When the receiving party answers announce the call.
4. Hang-up to complete the Transfer,

Unscreened transfer:

1. Press the [TRANS] button while on a trunk call; the call is placed on hold.
2. Dial the number of the Extension in the other system.
3. Hang-up to complete the Transfer.

Conditions

- The transfer is cancelled when the user presses [TRANS] button.
- A call transferred over the network does not recall the transferring Extension.

- The transferring user receives error tone if no idle networking path is available.
- Network transfer is not activated to a busy Extension.

Related WMS Menu

Related Features

- System Network
- Call Transfer

Hardware

- VPCM
- MDTM

2.106.3 Identification Service

Description

When a user places a networking call, the system includes the name of calling party to the called party between systems.

Operation

Conditions

Related WMS Menu

[To assign name for DN]

Data Management > Extension Information > Number(DN) Information > DN Attribute > "Name"

Related Features

- System Network

Hardware

2.106.4 Call Completion

Description

The iPECS-CM supports the Completion of Calls to a Busy Subscriber (CCBS). With CCBS, when a busy networked Extension is called, the user may request a Call Back when the called Extension returns to idle. When the busy Extension returns to idle the originating Extension is notified and, after answering the Call Back ring, the system places a call to the previously busy Extension.

Operation

To use CCBS (Call Back),

1. Dial an Extension in another networked system.
2. If the Extension is busy (busy tone is received), press the [MSG/CALLBK]
   Or
   Press the [TRANS] button and dial the {Extension Call Back/Trunk Queuing} feature code, or the Busy One Digit Service for Call Back.
3. Hang-up to return to idle.
4. When the busy Extension returns to idle, the originator receivers Call Back ring.
5. When the originator answers the Call Back ring, the system places a call to the previously busy Extension.

Conditions

- Stand-alone IP phones that support H.450 can activate the Call Completion feature.
- If the caller fails to answer the Call Back ring before the network timer expires, the call is disconnected.
- An Extension can leave or have one call completion. If a new call completion is requested, the old one is cancelled.

Related WMS Menu

[To assign Extension Call Back/Trunk Queuing feature code]

Data Management	> Numbering Plan Information> Feature Code > “Extension Call Back/Trunk Queuing)”, “Extension Call Back/Trunk Queuing Cancel”

[To assign One Digit Service for Call Back]

Data Management	> Tenant Information > Tenant Basic Information> One Digit Service on Busy

Related Features

- System Network
- Busy One Digit Service
- Call back

Hardware

2.106.5 Call Offer

Description

A busy user receives a notice that a call is waiting from a user in another networked system. This feature is similar to a Call-Wait.

Operation

To activate Call Offer:

1. Dial an Extension number of another system; the caller will hear a busy tone.

2. Press the [C-WAIT] or [TRANS] button and {Call Wait Request} feature code
   or
   Dial the digit for Busy One Digit Service (Call Wait Request)
3. The busy Extension will hear Off-hook ring, and the calling party will hear a Ring Back tone instead of a Busy tone.

To answer the Call Offer:

1. Press the flashing [HOLD/SAVE] button while hearing Off-hook ring.

Conditions

- Call Offer is only applied to the Extension that is in the talk state, an Extension in the ringing and dialing state cannot receive a 'Call Offer'.
- Call Offer is not available during a conference or paging.

Related WMS Menu

[To assign Call Wait Request feature code]
Data Management > Numbering Plan Information > Feature Code > "Call Wait Request"
[To assign One Digit Service for Call Wait]
Data Management > Tenant Information > Tenant Basic Information> One Digit Service on Busy

Related Features

- System Network

Hardware

2.106.6 Net Conference

Description

Extensions can establish a conference with a parties in another networked system. Up to 128 Extensions are allowed in a conference over the network.

Operation

To establish a Net Conference:

1. Press the [CONF] button while on a call, the existing call is placed on hold and dial tone is provided.
2. Dial an Extension number of another system.
3. When the called party answers, press the [CONF] button; the second call is put on hold, and the dial tone is provided.
4. Press the [CONF] button again; the conference is established between all the parties.

To terminate a Net Conference:

1. Any Extension in the Net Conference hangs up during the conference, then the Net Conference will be canceled and the network path will be cleared.

Conditions

- Standard SIP phones cannot be the originator of a conference of networked users.

Related WMS Menu

Related Features

- System Network

Hardware

2.106.7 Message Waiting Indication (MWI)

Description

Extension users may leave a Message Waiting Indication (MWI) when calling an Extension on another system that does not answer.

Operation

To leave a MWI:

1. While receiving Ring Back tone on a call to an Extension in a networked system, press the [MSG] button
   Or
   Press the [TRANS] button and enter the {Message Wait Register} feature code.
2. The MWI is left at the called Extension, the Msg Wait LED flashes.

To retrieve a MWI:

1. Press the [MSG] button, the Extension number is displayed.

Conditions

- A MWI can be registered only while receiving a Ring Back tone.

Related WMS Menu

[To assign Message Wait Register feature code]
Data Management > Numbering Plan Information> Feature Code > “Message Wait Register”
[To enable Extension Message Wait service]
Data Management > Tenant Information > Tenant Basic Information > Tenant Attribute > “Extension Message Wait Use”
[To set Message Wait Indication Method]
Data Management > Extension Information > Terminal Information > Terminal Option > “Message Wait Indication Method”

Related Features

- System Network

Hardware

2.106.8 Net Call Forward

Description

A remote Extension user can immediately forward calls to other Extensions in the network. The system supports both the Rerouting and Join through system programming.

Operation

To activate Net Call Forward:

1. Lift the handset or press the [Speaker] button
2. Enter the type of Net Call Forward (0=Remote Forward).
3. Enter the phone number to Forward.
4. Enter the password.
5. Dial [*] or [#].
6. Enter the call forward number.
7. Press the [SAVE] button.

To deactivate Net Call Forward:

1. Lift the handset or press the [Speaker].
2. Enter the type of Net Call Forward (0=Remote Forward).
3. Enter the phone number to Forward.
4. Enter the password.
5. Dial [*] or [#].
6. Press the [SAVE] button.

Conditions

- Net Call Forward can be activated on iPECS-CM, LIK and ipLDK.
- Net Call Forward can be deactivated between iPECS-CMs only.
- If the forwarding and the forwarded Extension are located in the same system, the networking path is cleared. That is, the forwarded call will be established as an internal call.
- The system does not check the DND, Call Forward or service status of the forwarded Extension.

Related WMS Menu

Related Features

- System Network
- Call Forward

Hardware

2.106.9 Trunk Transit-In

Description

A DID call can be rerouted to a destination that is in another networked system. The Caller Id is sent to the receiving destination.

Operation

Conditions

• The outside caller hears Busy tone when a networking path is not available for transit.

Related WMS Menu

[To set System Time Zone]
System Management > System Time Zone
[To set Transit options]
Data Management > Trunk Information > Transit Service
[To set Trunk data]
Data Management > Trunk Information > Incoming Route Information > Incoming Route Options
> Trunk Information > Trunk Basic Information > Trunk Attribute
> Trunk Information > Trunk Basic Information > Trunk Access code
[To assign Digit Conversion Map/Group/Option]
Data Management > System Feature Information > Digit Conversion Information > Digit Conversion Map
> System Feature Information > Digit Conversion Information > Digit Conversion Table
> System Feature Information > Digit Conversion Information > > Digit Conversion Option
[To assign Digit Conversion Virtual Tone]
Data Management > System Feature Information > Virtual Dial Tone Digit

Related Features

- System Network
- Digit Conversion

Hardware

2.106.10 Trunk Transit-Out

Description

An Extension can place external calls using the external resources of another system in the network.

Operation

Conditions

・To use the Trunk Transit-Out service, the system user must accurately secure the Trunk.

Related WMS Menu

[To set Transit options]

Data Management	> Trunk Information > Transit Service

[To set Trunk data]

Data Management	> Trunk Information > Outgoing Route Information > Outgoing Route Options
	> Trunk Information > Outgoing Route Information > Trunk Attribute
	> Trunk Information > Trunk Basic Information > Trunk Access Code

Related Features

- System Network

Hardware

2.106.11 Do-Not-Disturb (DND)

Description

A call to an Extension in DND mode will be denied and the calling party receives Busy tone.

Operation

To set DND:

1. Set DND mode at Extension.
2. From an Extension of other system, dial Extension in DND.
3. The calling party hears Busy tone with LCD display through the network.
4. The Extension in DND will not receive any ring signal.

To cancel DND:

1. Press the DND button or enter the DND code; the [DND] button will extinguish.

Conditions

・If BLF Manager is activated when an Extension is in the DND mode, the [DND] button blinks.

Related WMS Menu

[To allow to use DND]

Data Management	> Extension Information > Number(DN) Information > DN Feature Allow/Deny > “DND”

[To assign DND Register/Cancel (Toggle) feature code]

Data Management	> Numbering Plan Information > Feature Code > "DND Register/Cancel (Toggle)"

Related Features

- System Network
- Do Not Disturb

Hardware

2.106.12 Centralized Attendant Service (CAS)

Description

When multiple systems are connected in a network, the Attendants associated with one system can act as the attendant for all of the networked systems receiving Attendant calls.

Operation

1. Dial the Attendant code or Attendant Group Key number, the call is routed to the CAS (Centralized Attendant Service).

Conditions

- A Key number must be programmed for networked systems to call the Attendant night subscriber of the master system.
- If recalls or error signals from the slave systems are programmed to be sent to the Attendant, the calls are routed to the Attendant of the master system. Otherwise, the party receives the recall or error tone.

Related WMS Menu

Related Features

- System Network

Hardware

2.106.13 Centralized Voice Mail

Description

When multiple systems are connected in a network, a Voice Mail system associated with one system can act as the Voice Mail for Extensions in all the networked systems.

Operation

Conditions

- Centralized VMS has the same Numbering Plan as a normal Extension number. Sub-systems can access the VMS of the master system by calling the VMS number from their Extension.
- Internal VM (VPCM) cannot be used as Centralized VMS.

Related WMS Menu

Related Features

- System Network
- Built-In Auto Attendant/Voice Mail
- External Voice Mail System (VMS) Interface

Hardware

2.106.14 Call Intrusion

Description

When a user receives Busy tone on a call to an Extension in another system, the user may intrude on the call establishing a three-party conference by dialing the {Intrude Request} feature code.

Operation

To activate Call Intrusion:

1. Call an Extension in another networked system, the user receives Busy tone.
2. Press the [TRANS] button and the {Intrude Request} feature code
   Or
   Enter the Busy One Digit Service (Intrusion) digit.
3. The user is connected to the Extension's call forming a three-party call.

Conditions

- Call Intrusion is applied to an Extension in the talk state only, Call Intrusion is not available if the Extensions is ringing or dialing.
- Error tone will be provided when the Extension does not support conference or no VPCM is available for the conference.
- This function is supported on iPECS-CM only.

Related WMS Menu

[To assign Intrude Request Feature code]
Data Management > Numbering Plan Information > Feature Code > "Intrude Request"
[To set One Digit Service On Busy]
Data Management > Tenant Information > One Digit Service On Busy > "Intrusion"
[To use Intrusion]
Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > "Intrusion"

Related Features

- System Network

Hardware

2.106.15 ATD Net Station Status Query

Description

An Attendant can query the status of Extensions associated with other networked systems.

Operation

To query the status of other networked Extensions from an Attendant:

1. Dial the desired Extension number in another system.
2. The status of the Extension displays in the LCD of the Attendant.
3. Press the [CONNECT] button at the Attendant to place a call to the Extension.

Conditions

・iPECS 100/300/600 displays IDLE/COS only.
・iPECS-CM displays DND/Call Forward/COS.

Related WMS Menu

Related Features

- System Network

Hardware

2.106.16 Classifying the call type of trunk as Transit and Voice Network type in billing message

Description

1. Added new call type of billing information for voice network call.
   . Name of call type : **SMDA_CALLTYPE_VNET**
   . Value of call type : **0x0E**

2. Classifying the call type of trunk
   . In case of call from trunk to trunk
   1) At first, check if outgoing trunk is voice network, then set as **voice network call type**.
   2) Secondly, check if outgoing trunk is is one of VOIP call, then classified as **data call type**.
   else set as **transit call type**.

   . In case of call from extension to trunk
   1) At first, check if outgoing trunk is voice network, then set as **voice network call type**,,
   2) Secondly, check if trunk is one of VOIP call, then classified as **data call type**.

   else set as other types (local/public, ddd, isd)

Operation

Conditions

Only if country code of system is Russia, then feature is activated.

Related WMS Menu

Related Features

Hardware

2.107 System Service Time Mode

Description

The System Service Time Mode can be controlled automatically based on the system clock, or manually from an Attendant or authorized Extension. There are three service modes: Day, Night and Timed.

The Auto mode controls the mode between Day/Night/Timed services based on the System Time Zone.

The System Service Time mode is used in conjunction with the following features:

- Call Forward
- Preset Call Forward
- Pilot Hunt Call Forward
- Digit Restriction
- Digit Conversion
- Attendant Night Service
- DIL/DISA
- Alternative Incoming Route Service

Operation

Attendant or Authorized Extension

To change the System Service Time Mode by authorized Extension

1. Lift the handset or press the [Speaker] button.
2. Enter a {System Service Time Mode Change} feature code.
3. Enter a tenant number.
4. Enter a System Service Time Mode.
5. Press the [Hold/Save] button.

Conditions

- Day/Night/Timed mode can be changed manually by the Attendant, authorized Extension or automatically based on the System Time Zone.
- System time settings may be changed based on the NTP setting and DST (Daylight Savings Time).

Related WMS Menu

[To use Authorized Extension]
Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > ”Privileged User”
[To set System Time Zone]
System Management > System Time Zone
[To set System Clock]
WMS Management > WMS Configuration > WMS Time Configuration

Related Features

- System Clock Set
- System Time Zone

Hardware

2.108 System Time Zone

Description

The iPECS-CM system employs an IP ”switching” architecture, so there is no restriction of distance or country for installation of phones. If the terminals are installed in a region with a different Time Zone, the time, Day/Night/Timed Service mode, Least Cost Routing time and Holiday information can be different System Time.

The system provides time zone service to account for variation in the local time. Time zone related features are as follow.

- Date/Time and Calendar of Terminal
- Day/Night Call Forward by DN(Directory Number)
- Day/Night Call Forward by Key Number
- Day/Night/Holiday Service by ACD number
- Day/Night Call Forward by Pilot Hunt Group
- Day/Night Digit Restriction
- Day/Night / LCR Time Digit Conversion

Each Time Zone Table consists of the following information:

- Type of standard time zone (GMT-12:00 ~ GMT+14:00)
- Daylight Saving Time
- Day/Night/Timed
- Least Cost Routing Time
- Solar/Lunar Calendar Holidays

Operation

Conditions

- There are up to one-hundred (100) Time Zone Tables.
- If the Standard Time Zone of the Time Zone Table is set to "Server Time zone", the Time Zone of the system is applied.
- Up to fifty (50) holidays can be entered in the Time Zone Table.
- When entering holidays in the Time Zone Table, '99' indicates the end of the month. This allows entering the first day as the New Year holiday.
- If DNs with different Time Zones are registered to a phone, CID and call history are based on the Time Zone of Prime-DN.
- Statistics are based on the "Server Time zone".
- Charging data are based on "Server Time zone".

Related WMS Menu

[To set System Time Zone]
System Management > System Time Zone
[To assign Time Zone Group of DN]
Data Management > Extension Information > Number(DN) Information > DN Attribute > "Time Zone No."
[To assign Time Zone Group of Pilot Hunt Group, Key Number Group, ACD Group, ACD Pilot]
Data Management > Extension Information > Group Information > Pilot Hunt Group > "Time Zone No."
> Extension Information > Group Information > Key Number Group > "Time Zone No."
> ACD/CTI Information > ACD Group Attribute > "Time Zone No."
ACD/CTI Information > ACD Pilot > "Time Zone No."
[To assign Time Zone Group of Tenant]
Data Management > Tenant Information > Tenant Basic Information > Tenant Attribute > "Time Zone No."
[To assign Time Zone Group of Trunk]
Data Management > Trunk Information > Incoming Route Information > Incoming Route Options > "Time Zone No."
Trunk Information > Outgoing Route Information > Outgoing Route Options > "Time Zone No."

Related Features

- Digit conversion
- Key number group
- ACD Group
- Pilot hunt group

Hardware

2.109 Tenant Group

Description

This function enables the system to be used as multiple systems. The iPECS-CM can be divided into up to 100 Tenant groups, and Extensions and Trunks are assigned to the Tenant groups (Groups 01~100). Extensions of Tenants can be restricted in placing calls to other groups. If Tenant group (A) is allowed calling Tenant group (B), the Extensions in the Tenant group (A) can call Extensions in Tenant group (B). Trunks can be assigned for use by specific Tenant groups.

An Attendant can be assigned to each Tenant group to receive Attendant calls from Extensions in the group. The Tenant Attendant can control Day/Night mode for the Tenant group. The Day/Night mode of Tenant group can be controlled automatically by assigning a Time Zone to the Tenant group.

Operation

Conditions

- iPECS-CM Tenant groups can use the system base Numbering Plan or a Tenant based Numbering Plan.
- A call will be restricted between tenants when the Inter Tenant Access is “No Access”, and the user will receive error tone.
- When the Attendant of a Tenant group sets the Day/Night/Timed mode, it will affect the designated Tenant group only.

Related WMS Menu

[To define Number of Tenant in System]
System Management > System Environment > “Number of Tenant in System”
[To define System Numbering Plan type]
System Management > System Environment > “System Numbering Plan”
[To set call access between tenants]
Data Management > Tenant Information > Tenant Basic Information> Inter-Tenant Access

Related Features

- Tenant Prefix
- Internal Call
- Flexible Numbering Plan
- Transfer
- System Service Time Mode
- Attendant

Hardware

2.109.1 Tenant Prefix

Description

An iPECS-CM system can use different Numbering Plans, voice announcements and functional options for each Tenant. The iPECS-CM provides two Numbering Plan methods for a Tenant. One is to use one Numbering Plan for all Tenants (System Base Numbering Plan); the other is to provide individual Numbering Plans for each Tenant (Tenant Base Numbering Plan).

System Base Numbering Plan – The whole system is configured with a single Numbering Plan, and the number used for a specific Tenant may not be used by other Tenants. Therefore, unique numbers are assigned to each Extension in the Tenant, and calls can placed between Tenants.

Tenant Base Numbering Plan - An independent numbering plan is configured for each tenant, so that the numbers can be assigned regardless of whether they are being used by other Tenants. Since the same number can be assigned to multiple Tenants, a Tenant cannot call another Tenant.

A unique prefix can be assigned to each Tenant, when “Tenant base Numbering Plan” is selected, so that users can call the Extension of another Tenant by dialing the Tenant prefix and Extension number.

Operation

To use Tenant Prefix:

1. Lift the handset or press the [Speaker] button.
2. Enter the Tenant prefix.
3. Enter the Extension number of the Tenant.

Conditions

- Tenant prefixes can be assigned by administrator only, and Tenant prefix must be unique in the system. A tenant prefix has up to six (6) digits.
- Tenant prefixes must not be duplicated in the Tenant Numbering Plan.
- A “Tenant Dial Tone” can be provided to an Extension when the Tenant prefix is dialed.
- When an Extension is called using the Tenant prefix and Extension number from another Tenant, the Extension number including the prefix is displayed.
- The Tenant prefix is not included in CID for outgoing calls.
- For CTI/MSI communication or billing information reports, the Tenant prefix and dialed number are given in the all reports.
- If there is a prefix, the telephone number including the Tenant prefix is saved for call history and Station Message Wait.
- A call will be restricted between tenants when the Inter Tenant Access is “No Access” despite of using Tenant Prefix.

Related WMS Menu

[To define System Numbering Plan type]
System Management > System Environment > “System Numbering Plan”

[To assign Tenant Prefix Code]
Data Management > Tenant Information > Tenant Basic Information> Tenant Prefix
[To assign Tenant Prefix Dial tone]
Data Management > Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > “Tenant Dial Tone”
[To set call access between tenants]
Data Management > Tenant Information > Tenant Basic Information> Inter-Tenant Access

Related Features

- Tenant Group

Hardware

2.110 Transit Service

Description

The system supports outgoing Trunk calls placed from an incoming Trunk. In this case, the system also provides the inter-working function between the trunks using different signals. As an example, an incoming call through a Trunk using R2 protocol can seize an outgoing Trunk using PRI. The system supports following Trunk types:

- Analog R2(E&M)
- Analog DTMF (E&M)
- CO
- RD
- LD
- E1/T1 DTMF
- E1/T1 R2
- E1/T1 PRI
- SS No.7
- QSIG(PRI, H323)
- H.323
- SIP

Operation

Conditions

- The system supports inter-working of all Trunks.
- Two protocols, Link-By-Link or End-To-End can be supported for direct transit via an R2 Trunk.
- For transit between SS7 Trunks, all messages are transferred transparently between the Trunks.
- For transit calls between PRIs, all messages are transferred transparently between the Trunks.
- For transit between PRI to SS7 Trunks, some messages are not transferred between the Trunks.

Related WMS Menu

[To set Transit options]
Data Management > Trunk Information > Transit Service
[To set Trunk Access Code]
Data Management > Trunk Information > Trunk Basic Information > Trunk Access Code

Related Features

Hardware

2.111 Trunk Access Code

Description

The system can be configured with flexible Trunk Access codes for each outgoing route. Multiple access codes can be assigned to a single outgoing route. The system provides various services based on the access code. The following services can be provided for each Trunk Access code.

- AND Digit – Digits that are sent to the Trunk automatically.
- R2 Transit Signaling Method – Select Link-By-Link or End-To-End for direct transit via R2 protocol.
- Dialing Method for Outgoing Call – Select En-block or Overlap for digit transmission.
- Max Digit Length – Restricts the number of digits to be transferred to a Trunk.
- En-Block Digit Length – Selects the number of digits to be received from an Extension or an incoming transit call and sent to the outgoing Trunk.
- Trunk Virtual Dial Tone – Determines if dial tone is sent to an Extension or incoming Trunk calls.
- ARS Service – An alternative path can be used when all Trunks assigned to a Trunk Access code are busy.
- ARS Route No. – Designates the alternative route.
- R2MFC Backward value – The digits to be transmitted as a backward signal when R2 operates as an End-to-End. Two methods are supported for Alternative Path routing, Trunk Access code, or Trunk Outgoing Route.

Operation

Conditions

- Multiple Trunk Access codes may be assigned to one outgoing route, but multiple outgoing routes cannot be assigned to an access code. ARS may be used to assign multiple routes to a single access code.
- This function is not supported if the user selects a Trunk using a Flex button without using a Trunk Access code.
- Trunk Access codes may be used with R2 Transit calls.
- R2 Transit Signaling Method and R2MFC Backward Value are available only for an R2 Trunk.

- If the Tenant Use Type of Trunk Access Code WMS menu is configured as Independent Use and system uses the System Base Numbering Plan, each Tenant may seize a different route while using the same Trunk Access code.

Related WMS Menu

[To define System Numbering Plan type]
System Management > System Environment > "System Numbering Plan"
[To set Trunk Access Code]
Data Management > Trunk Information > Trunk Basic Information > Trunk Access Code

Related Features

- ARS Service
- ARS Route No.
- AND Digit
- SMS
- Outgoing Route

Hardware

2.112 Trunk Call

Description

Extensions can be assigned to access Trunks by the system program using Trunk Access codes. iPECS Multi-button phone users may use a flexible button to access a specific Trunk or Trunk Access Code.

Operation

iPECS Multi-button Phone
To place a Trunk call:
1. Lift the handset or press the [Speaker] button.
2. Enter the {Trunk Access Code} or press the {Trunk Access Code} Flex button, or dial an individual Trunk (Private Line) code or Trunk Access code.

To answer to an incoming Trunk call:
1. Lift the handset or press the [Speaker] button.
   Or
   Press the desired [DN] button.

Single Line Telephone (SLT)
To place a Trunk call:
1. Lift the handset.
2. Enter the Trunk access code and dial desired number.

To answer an incoming Trunk call:

1. Lift the handset.

Conditions

- When dialing a Trunk Access code, the system will search and access an idle Trunk in the Trunk Route corresponding to the access code. If there is no idle Trunk in the Trunk Route, the system will provide Busy tone. In this way the user may activate Trunk queuing.
- A user must have access rights to dial a Trunk Access code otherwise, the system will provide error tone. Extensions denied access to the Trunk may receive calls transferred from other Extensions but are not allowed generate a flash on the Trunk.
- When an outgoing call is made, voice is not transmitted to the Trunk until system completes analysis to allow or deny dialing.
- iPECS Multi-button phone users may employ a specific Trunk exclusively by setting the Trunk as an individual Trunk Group (Private Line).

Related WMS Menu

[To set Trunk Access Code]
Data Management > Trunk Information > Trunk Basic Information > Trunk Access Code
[To assign Individual Trunk Access feature]
Data Management > Extension Information > Number(DN) Information > DN Attribute > "Individual Trunk Use"
> Extension Information > Number(DN) Information > DN Individual Trunk Access
> Trunk Information > Trunk Basic Information > Individual Trunk Group

Related Features

- Trunk Group
- Trunk Call Routing Table

Hardware

2.113 Trunk DTMF Duration Service

Description

For Trunk calls, different DTMF durations may be assigned depending on the number of digits. This function is useful when dialing a Speed Dial number or redialing a number. The off duration after sending the first digit can be different from that for the second digit..

Conditions

- The same DTMF transmission on/off duration is applied from the sixth digit.
- When gateway is registered, the duration of DTMF transmission is sent to gateway. Therefore, the duration of DTMF sending is same regardless of sending digit count.

Related WMS Menu

[To set DTMF protocol for Trunk]

Data Management > Trunk Information > Trunk Basic Information > Trunk Attribute > "Dialing Type"

[To set DTMF Sending Time]

Data Management > Trunk Information > Outgoing Route Information > Outgoing Route DTMF Sending Time

Related Features

Hardware

2.114 Trunk Inter-digit Timer Service

Description

When digits are dialed on a Trunk, the Inter-digit timer can be assigned for each digit. In other words, the timeout between the first and the second digit may not be the same as that between the second and the third digit. In case of Trunks with no answer signal, the outgoing call enters the talk state automatically after the Inter digit timer expiry.

Operation

Conditions

- The same Inter digit timer is applied from the sixth digit.
- An Inter digit timer can be assigned to each outgoing route.
- In the En-block mode, the SETUP message transmission point is determined based on this timer.

Related WMS Menu

[To set Dialing Method for Trunk]

Data Management > Trunk Information > Trunk Basic Information > Trunk Access Code > "OG Call Dialing Method"

[To set Inter Digit Time for outgoing]

Data Management > Trunk Information > Outgoing Route Information > Outgoing Route Inter Digit Time

Related Features

Hardware

2.115 Trunk Line Flash

Description

Analog Trunks recognize a brief open 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 within the PBX.

Operation

iPECS Multi-button phone

While connected to an analog Trunk:

1. Press the programmed {Trunk Flash} button. The system generates a flash on the Trunk.

Single Line Telephone (SLT)

While connected to an analog Trunk:

1. Press and release the hook-switch (hook-flash).
2. Enter the “Trunk Flash” feature code.

Conditions

- During a Flash, the LED for the CO line button will remain lit.
- Flash may be stored as a part of a Station or System Speed Dial number.
- Pressing the {Trunk Flash] button during an Intercom Call or while connected to the dial tone will return Intercom Dial tone.
- The flash function is not available for digital Trunk, VOIP, R2 (E1/T1) or ISDN calls.
- SLTs cannot use flash function when dialing, only when connected to an active Trunk.

Related WMS Menu

[To assign Trunk Flash feature code]

Data Management > Numbering Plan Information > Feature Code > “Trunk Flash”

[To assign Trunk Flash Flex Button]

Data Management > Extension Information > Terminal Information > Phone Flexible Button

Related Features

Hardware

2.116 Trunk Line Name Display

Description

If an external call is routed to an Extension or an Extension user seizes a Trunk, the Trunk number is displayed in the LCD. If a name is assigned to a Trunk, the LCD displays the name instead of the Trunk number.

Operation

iPECS Multi-button phone

To display a Trunk name while placing a call:

1. Dial the Trunk Access code, the name of Trunk route is displayed on the LCD.

Operation

Conditions

・Up to 16 characters are allowed as a Trunk name.

Related WMS Menu

[To assign name of Incoming Route]

Data Management > Trunk Information > Incoming Route Information > Incoming Route Group > “Incoming Trunk Name”

[To assign name of Trunk Access Code]

Data Management > Trunk Information > Trunk Basic Information > Trunk Access Code > “Name”

Related Features

Hardware

- iPECS Multi-button Phone with LCD

2.117 Trunk Own Code Service

Description

When an Extension places a Trunk call, the Incoming or Outgoing Route Own Code may be sent with the CLI to Trunk. If an incoming Trunk call directly transits out over a Trunk, the Outgoing Route Own Code may be added to CLI. The Trunk Own Code is added to the Extension number or the received CLI to be transmitted to the outgoing Trunk. When the Connect message that includes the answering user information is sent to Trunk, the Incoming Route Own Code is used for the connected party information.

Operation

Conditions

・This function can be assigned to each outgoing and incoming Trunk route.
・The Own code can be entered for outgoing and incoming routes separately.
・Up to 16 digits can be entered for an Own Code.
・When an incoming call transits directly, the outgoing route Own Code Add service follows the WMS settings.

Related WMS Menu

[To set Add Own Code for Transit]

Data Management > Trunk Information > Incoming Route Information > Incoming Route Options > “Add Own Code to CID on Transit”

> Trunk Information > Outgoing Route Information > Outgoing Route Options > "Add Own Code to received CID on Transit"

[To assign Own Code for incoming/outgoing]

Data Management > Trunk Information > Incoming Route Information > Incoming Route Own Code
> Trunk Information > Outgoing Route Information > Outgoing Route Own Code

Related Features

Hardware

2.118 Trunk Preset Forward

Description

Each Trunk can be configured with No answer Preset Forward. If an incoming Trunk call is not answered in the No answer timer, the call forwards to the Preset Forward destination. The destination can be an Extension, a CCR Table or a Key Number Group.

Operation

Conditions

- Preset Forward is not applied to DID Trunks, when a Trunk call is routed to a Key Number group or a CCR Table.
- Trunk Preset Forward is valid when the Service Type in the Trunk Call Routing Table is set as "Multi Ring" or "First Idle.
- The Trunk Preset Forward is deactivated if the Preset Forward timer is 0.
- Voice Mail groups cannot be the destinations of Trunk Preset Forward.

Related WMS Menu

[To set options for Trunk Preset Forward]

Data Management > Trunk Information > Incoming Route Information > Incoming Route DIL/DISA Service >" Multi Ring No Answer Time (sec)"
> Trunk Information > Incoming Route Information > Incoming Route DIL/DISA Service > Trunk Call Routing Table for Multi Ring No Answer

[To set Trunk Routing Table]

Data Management > System Feature Information > Trunk Call Routing Table

Related Features

- Call Forward
- System Service Time Mode

Hardware

2.119 Trunk Release Guard Time

Description

Analog Trunks require additional time to clear down and return to idle after a call. If the Trunk is accessed prior to clear down, 'glare' may occur. In order to allow sufficient time for the analog Trunk to return to idle, the system holds analog Trunks in the busy state to internal users until the Release Guard time expires. The time between the Extension disconnect and when the system changes the Trunk status to idle is the minimum Trunk Release Guard time required.

Operation

Conditions

Related WMS Menu

[To change Trunk Release Guard Time]

Data Management > Trunk Information > Outgoing Route Information>Outgoing Route Options > "Release Guard Time (sec)"

Related Features

Hardware

2.120 Trunk Route Groups

Description

System Trunks may be seized with the Trunk Access code or directly with the Trunk serial number. Trunk Route groups are used to seize a Trunk with the Trunk Access code. A route group is assigned to the Trunk Access code and Trunk serial numbers are assigned to the route group. The Trunks are seized according the seizing method (Round Robin, Last or First). There are 500 Trunk Route groups in iPECS-CM.

Operation

Conditions

- Up to 500 trunk lines can be assigned to one route group.
- Outgoing route groups are separated from incoming route groups.
- Five hundred (500) Trunk Route groups can be seized with the Trunk Access code.
- One-thousand (1,000) individual Trunk Route groups can be accessed with an individual Trunk Access code.
- A Trunk cannot be included in two or more route groups.

Related WMS Menu

[To assign Route Group for Trunk Access Code]

Data Management > Trunk Information > Trunk Basic Information > Trunk Access Code > "Outgoing Route Group"

[To set options for Incoming/Outgoing Trunk Route Group]

Data Management > Trunk Information > Incoming Route Information > Incoming Route Options
> Trunk Information > Outgoing Route Information > Outgoing Route Options

Related Features

- Individual Trunk Access

Hardware

2.121 Trunk Route Service

Description

Trunks can be assigned to one of the outgoing/incoming routes. The system can define the service type and other options by route.

2.121.1 Incoming Route Options

Description

All Trunks may be assigned to an incoming route. The administrator may select services to be provided for each incoming route.

The following functions may be selected by the administrator for services.

- Incoming Call Charge – Charging data may be generated for incoming Trunks.
- Progress Indication IE – If an incoming Trunk call seizes a non-ISDN trunk, the ISDN progress message may be transmitted to the destination. In addition, ISDN Setup/Call Proceeding/Alerting messages may include in the Progress indication information element.
- R2 ANI Service Request – CLI can be requested for the incoming Trunk calls.
- Add Prefix Digit to CID – If CLI (ANI) received from a Trunk does not include “0”, the designated digit is added to the received CLI.
- Blocking on Clear Forward Wait Time-out – If the Wait Clear Forward Signal is not received for the designated time after transmitting the Clear Backward Signal to the network, the channel is blocked.
- Transit R2 Backward A-3 Sending Delay Time – In transit incoming with R2 signaling, the system sends Backward Signal A-1 (request next digit) and can send Backward Signal A-3 (received complete digit) immediately without receiving the Forward Signal. In this case, the first originating PBX/PX may not detect Backward Signal A-3. Therefore, the system can delay sending Backward Signal A-3 for this time (Transit R2 Backward A-3 Sending Delay Time) after sending Backward Signal A-1 to detect Backward Signal A-3 from the first originating PBX/PX.

- Clear Forward Waiting Time – On a Trunk that does not support both-way recovery, such as DTC, this timer is applied after the called party is disconnected until the calling party disconnect.
- Add Own Code to CID on Transit – This function is applied if CLI of the incoming transit call is the Extension number only. This function inserts a pre-programmed incoming route own code to the CLI of the incoming call.
- Answer for Call Waiting – If an incoming Trunk call is put in Auto Call Wait or Auto Hold by the Extension, the incoming call can be disconnected before the Extension answers the call. This function sends the response signal to the network at the time of Auto Call Wait or Auto Hold.
- BLF Indication for Phone– If a Trunk is assigned to a Flex button of the Extension, the LED indication can be selected.
- Provide Dial Tone – A dial tone can be presented to the outside caller to dial Extension number.
- CID Detect Time – For call routing using CID, the system must guarantee the time to receive CID.
- DISA supervision Time - User can specify the time for answer delay in case of DISA incoming call.

Operation

Conditions

- This function can be assigned to each incoming route.

Related WMS Menu

[To set options for Incoming Trunk Route Group]

Data Management > Trunk Information > Incoming Route Information > Incoming Route Options

Related Features

- BLF
- ICLID
- Transit
- Metering
- R2 ANI Service Request

Hardware

2.121.2 Outgoing Route Options

Description

All the trunk lines of system can be assigned to one of outgoing routes, and the administrator may select the services to be provided for each trunk route.

The following functions may be selected by the administrator for services.

- ISDN Screen Indicator Information – The screen indicator content may be selected when configuring CLI on ISDN.
- Outgoing CID Type – The calling party number content may be selected when configuring CLI on ISDN.
- Outgoing CID inclusion – It is possible to select if the ISDN SETUP message should contain CLI.
- Bearer Capability – When incoming PRI call makes an outgoing call, the assigned bearer capability is included in the SETUP message, ignoring the received bearer capability.
- Sending Complete IE Inclusion – Digit sending complete IE can be included in the ISDN information message.
- CID Insert for Transit Call – If a transit call contains no CLI information, the system can configure CLI information and transmit it to the outgoing Trunk.
- Add Own Code to Received CID on Transit – New CLI can be made by adding Outgoing Route Own code to the CLI of an incoming call.
- Outgoing CID Plan – Calling party numbering plan can be selected in ISDN for CLI in transit call.
- Outgoing Route Key Number as CID – Route Key Number can be used as CLI for outgoing Trunk call instead of CLI of the Extension or the CLI of incoming transit call.
- BLF Indication for Phone – If a Trunk is assigned to a Flex button of phone, the LED indication for the Trunk can be selected.
- Provide Ring Back Tone – If the PSTN does not provide RBT for the trunk call of the Extension user, the system can provide RBT.

Operation

Conditions

- This function may be assigned to each outgoing route.

Related WMS Menu

[To set options for Outgoing Trunk Route Group]

Data Management > Trunk Information > Outgoing Route Information > Outgoing Route Options

Related Features

- BLF
- ICLID
- Transit
- Metering
- R2 ANI Service Request

Hardware

2.121.3 Incoming Route Error Service

Description

If an incoming Trunk cannot be serviced normally, the system may route the call to a service or a preset destination. This function is assigned to each incoming route.

The administrator can select the destination for all incoming Trunk calls for the following cases:

- Busy – Call to busy user; Available options are Release, Attendant, Trunk Call Routing Table, System Tone (Busy tone) and Pilot Hunt Call
- No Answer – No answer by called user; Available options are Release, Attendant, Trunk Call Routing Table, System Tone (Error tone) and Pilot Hunt Call
- Wrong Number – The result of dial analysis of the incoming Trunk call is vacant; available options are Release, Attendant, Trunk Call Routing Table and System tone (Vacant tone).
- Transfer No Answer – No answer by called user after unscreened transfer; available options are Release, Attendant, Trunk Call Routing Table, System tone (Error tone), Transferring Subscriber Call, Continuous Ring and Pilot Hunt Call.
- Recall No Answer – No answer by called party for recall; Available options are Release, Attendant, Trunk Call Routing Table, System Tone (Error tone), Continuous Ring and Pilot Hunt Call.
- DND – A trunk call is placed to the user in DND; Available options are Release, Attendant, Trunk Call Routing Table, System Tone (DND tone) and Pilot Hunt Call
- Extension Lock Out – Call to the user in lock-out; Available options are Release, Attendant, Trunk Call Routing Table, System Tone (Error tone) and Pilot Hunt Call
- ETC – Call to the user in blocking; Available options are Release, Attendant, Trunk Call Routing Table, System Tone (Error tone) and Pilot Hunt Call

Operation

Conditions

- This function may be assigned for each incoming Trunk route.
- If no Attendant is assigned, an Error tone is sent to the outside caller.
- An Error tone is provided if there is no system resource for announcement or if all resources are in use.
- A different destination may be assigned for Day/Night/Timed service modes, respectively.
- All functions assigned in Trunk Call Routing Table are available if the destination is a Trunk Call Routing Table

Related WMS Menu

[To set System Time Zone]
System Management > System Time Zone
[To set alternative destination for unanswered Incoming trunk call]
Data Management > Trunk Information > Incoming Route Information > Incoming Route Error Destination
[To route incoming trunk call to Extension]

Data Management > System Feature Information > Trunk Call Routing Table

Related Features

- System Time Zone
- DND
- Station Lock out

Hardware

2.121.4 Outgoing Route Error Service

Description

If an outgoing Trunk is not answered, the system may route the call to a preset service or destination. This function can be assigned to each outgoing route as follows:

- Recall Answer (No Answer) – No answer by transferred user; available options are Release, Attendant, Trunk Call Routing Table, System Tone (Error tone), Transferring Subscriber Call, Continuous Ring or Pilot Hunt Call.
- Transfer No Answer – No answer by transferred user after unscreened transfer; available options are Release, Attendant, Trunk Call Routing Table, System Tone (Error tone), Continuous Ring and Pilot Hunt Call

Operation

Conditions

- This function may be assigned to each outgoing route.
- If no Attendant is assigned, an error tone is sent to the outside caller.
- An Error tone is provided if there is no resource for announcement or if all resources are in use.
- Different destinations may be assigned for Day/Night/Timed service modes, respectively.
- All functions assigned in the Trunk Call Routing Table are available if the designation is Trunk Call Routing Table

Related WMS Menu

[To set System Time Zone]
System Management > System Time Zone
[To set Alternative Outgoing Route Service]
Data Management > Trunk Information > Outgoing Route Information > Outgoing Route Error Destination
[To set Trunk Call Routing Table]
Data Management > System Feature Information > Trunk Call Routing Table

Related Features

- System Time Zone
- DND

- Station Lock out

Hardware

2.121.5 Route Transit Service

Description

The system can be programmed to allow an incoming Trunk call to seize a Trunk directly for outgoing call (Direct Transit), or an Extension user can transfer an external call to another Trunk. If transit is not allowed, the external user and internal user on an incoming Trunk call will receive error tone. This service can be enabled for incoming/outgoing routes, respectively.

The administrator may assign options for following three cases:

- Direct – Incoming Trunk calls seize a Trunk and place an outgoing call.
- Extension – An Extension user transfers an external call to Trunk.
- ATD – The Attendant transfers an external call to Trunk.

The administrator may select one of the following options for each of the above cases:

- Allow – The service is allowed for all calls.
- Disallow – The service is not allowed for all calls

The system can disconnect the call after the designated timer expires, or inform the Attendant that the time has expired.

- Not process – The call continues until the user terminates the call.
- Release by ATD Monitoring – The Trunk sends the information to the Attendant after a period of time.
- The Attendant can disconnect the Trunk by monitoring the call using this information.
- Release by Transit Call Time – The Trunks are released by the system after a period of time.
- Transit Call Time – This value determine available Transit call duration (30 ~ 3600 seconds).

Operation

Conditions

- This function can be assigned to each outgoing/incoming route.
- The system can be set to allow this function for specific incoming or outgoing routes only.
- The maximum duration of a Trunk-to-Trunk call can be assigned in the system.

Related WMS Menu

[To set Transit options]
Data Management > Trunk Information > Transit Service
[To set Trunk Access Code]
Data Management > Trunk Information > Trunk Basic Information > Trunk Access Code

Related Features

- Trunk Transit - in
- Trunk Transit - Out
- Attendant
- Trunk color ring

Hardware

2.122 DID/DISA Call Restriction

Description

The extension can restrict the incoming trunk calls according to trunk service type.

- Allow both DID/DISA incoming
  All incoming trunk calls are routed to the extension. There is no restriction.
- Restrict both DID/DISA incoming
  All DID and DISA calls are restricted. The extension can receive only DIL calls and internal calls.
- Restrict only DID incoming
  When a DID call is routed to a extension, the extension can refuse that call. The calls of other types are routed to that extension normally Trunk Access through DISA
- Restrict only DISA incoming
  When a DISA call is routed to a extension, the extension can refuse that call. The calls of other types are routed to that extension normally.

Operation

iPECS Multi-button phone

To activate DID/DISA Call Restriction,

1. Register {Trunk DID/DISA Call Restrict/Allow} feature code on Flex button.
2. Press {Trunk DID/DISA Call Restrict/Allow} Flex button.

Conditions

- DIL call is not applied this feature.

Related WMS Menu

Related Features

- Tenant System Ring – Normal Call Ring(Trunk-DID), Normal Call Ring(Trunk-DIL/DISA).

Hardware

2.123 User Authorization Code

Description

User Authorization codes consist of the Extension number and the password for the Extension. The code is used to authorize the user to employ the following services.

- Walking COS
- Trunk Access through DISA
- Remote Call Forward

Operation

iPECS Multi-button phone

To assign an Extension password:

1. Press the [PGM] button.
2. Dial 34, the Extension User Program code.
3. Enter a password (up to 12 digits).
4. Press the [SAVE] button.

Single Line Telephone (SLT)

To assign an Extension password:

1. Lift the handset.
2. Enter the {Password Change} feature code.
3. Enter a password (up to 12 digits).
4. Dial * or make a hook-flash.

All Terminals and DISA users:

To enter the User Authorization Code while the second dial tone is heard

1. Enter the Extension number.
2. Enter the password (up to 12 digits).
3. Dial *.

Conditions

- User Authorization codes for an Extension can be assigned by the Administrator through WMS.
- When User Authorization is assigned to a DISA line, the user must enter the User Authorization code to access a Trunk.
- The user is prompted through an announcement or Authorization dial tone to enter the Authorization code. If the user fails to enter a correct Authorization code, the user may reenter the User Authorization code based on the DISA retry counter.
- If Walking COS is active, the user must enter the User Authorization code at the Extension to place a Trunk call.
- The number of Authorization codes can be up to the number of subscribers.
- User Authorization code include the Extension number and password, so authorization is differentiated by the Extension number. Therefore, passwords for Authorization codes can be duplicated.
- A password may include any digit except '#' and '*'.

- If a terminal is locked due to failure of the Authorization code, the terminal is placed in out of service state. A Terminal that is out of service can be restored by the administrator in "Blocking Management" of the WMS menu.
- Normal users may only assign the password for the Extension.

Related WMS Menu

[To use User Authorization Code for DIL/DISA]

Data Management	>Trunk Information > Incoming Route Information > Incoming Route DIL/DISA Service > "DISA Transit" > Tenant Information > Tenant Basic Information > Tenant Attribute > > Incoming Route Information > Incoming Route DIL/DISA Service > "Authorization for DISA Transit"

[To set Password for DN]

Data Management	> Extension Information > Number (DN) Information > DN Attribute > "Extension Password"

[To assign Authorization Access Digit]

Data Management	> Tenant Information > Tenant Basic Information > Authorization Access Digit.

[To use Terminal Lock for Unauthorized User]

Data Management	> Tenant Information > Tenant Basic Information > Tenant Attribute > Unauthorized Access Retry Count > Tenant Information > Tenant Basic Information > Tenant Attribute > Terminal Lock for Unauthorized Access

Related Features

- Account Code
- DISA
- Trunk Route Groups
- Dialing Restriction / Station Lock
- Dialing Restriction / Walking COS
- Call Forward

Hardware

2.124 VPCM Usage Assignment

Description

The VPCM provides functions as follows:

- Common Use
- Conference Call
- Prompt/Announcement
- IP Trunk
- RTP Relay

One VPCM consists of eight (8) slots and each slot can be set for one of the functions. A VPCM slot assigned a function is used for only the assigned function. Slots assigned for common use may be used to support any of the VPCM functions.

Operation

Conditions

- If the administrator does not assign usage to a VPCM slot, this slot may be used for all four functions.
- If there are slots with assigned usage and those without usage, the slots without usage are used for all functions and seized sequentially.

Related WMS Menu

[To see VPCM Channel Summary]
Data Management > Summary Information > VPCM Channel Summary

Related Features

Hardware

2.125 VPCM Usage Statistic Monitoring

Description

VPCM is used for several purposes (Conference, Prompt/Announcement, VoIP Trunk Media Channel, RTP relay, Voice Mail) in iPECS-CM. So, user should care whether the VPCM channel is enough or not. This feature is referenced when Administrator decide to add VPCM gateway or not in their system. User can retrieve VPCM's traffic information on WMS. To support this feature, system saves VPCM traffic information periodically. The traffic item is like below.

- Utilization of each usage (Conference, Prompt/Announcement/Voice Mail, VoIP Trunk Media Channel, RTP Relay) based on each VPCM Slot Number.
- Total utilization of each usage (Conference, Prompt/Announcement/Voice Mail, VoIP Trunk Media Channel, RTP Relay).

Operation

Conditions

- Number of available channels of VPCM depend on Codec. So, System provide not number of channels but utilization of VPCM to Administrator.
- VPCM traffic information is provided based on each VPCM gateway or total of system.

Related WMS Menu

Related Feature

Hardware

2.126 VPCM GW Detail Information

Description

"VPCM GW Detail Information" feature show the detail information about the VPCM GW.
The detail information of the VPCM are the type of the VPCM, Memory Usage, CFC Filesystem, CFC Mode, and the Prompt Nation code and Prompt version installed in the VPCM.

Operation

1) Normal Operation : When the slot No. is VPCM and the VPCM is connected to CM.
2) Abnormal Condition.
   A. When the slot No. is not a VPCM → "This is not VPCM Slot".
   B. When the Slot No. is a VPCM but this VPCM is not connected to CM.
      "Unregistered VPCM"

Conditions

Related WMS Menu

[To set use of Fax Bridge in System]
Data Management > Summary Information > VPCM Channel Summary
Click "VPCM GW Detail Information"

Related Feature

Hardware

2.127 Multi-NAT of VPCM/VOIM

Description

"Multi-NAT of VPCM/VOIM" feature supports the RTP sending/receiving when multiple VPCM/VOIM in a NAT exist.
Depending on the configuration about the forwarding the RTP packing in the NAT, the CM informs the VPCM/VOIM of the RTP forwarding rules

Operation

1. When the RTP port is changed, the CM sends the command for initialization to VPCM/VOIM
2. When the VPCM/VOIM is registered to CM, the CM sends the RTP forwarding rules to the VPCM/VOIM.

Conditions

Related WMS Menu

Data Management > Summary Information > VPCM/VOIM SLOT/Port Summary

Related Feature

Hardware

2.128 Wake-Up Alarm

Description

This feature allows users to set a Wake-up alarm through phone. When the selected time is reached, the system sends the audible/visual signal to the phone. The user can configure five (5) Wake-up alarms. When the user answer the Wake-up signal, an announcement can be played.

Operation

iPECS Multi-button Phone

To register a wake-up alarm:

1. Enter the {Wake-up Register} feature code or press the [PGM] + '1' + '3.'
2. All registration status of Wake-Up calls can be checked with Volume Up/Down button and the desired index for the Wake-Up alarm can be selected by pressing OK soft button or [SAVE] button.
3. Enter the index number of Wake-up (1~5).
4. Enter the type of Wake-up (1: Once, 2: Daily, 3: Monday~Friday, 4: Monday~Saturday, 5: Date).
5. Enter the hour and minute (HH:MM, 4 digits in 24 Hour Format)
6. If Wake-up type is date, enter six (6) digits (YY/MM/DD) for year, month and day.
7. Press the [SAVE] button.

To cancel the Wake-up alarm:

1. Enter the {Wake-up Cancel} feature code or press the [PGM] + '1' + '4.'
2. Enter the index number of Wake-up (1~5).
3. Press the [SAVE] button.

Single Line Telephone (SLT)

To register a Wake-up alarm:

1. Lift the handset.
2. Enter the {Wake-up Register} feature code; a confirmation tone will be heard.
3. Enter the index number of Wake-up (1~5).
4. Enter the type of Wake-up (1:Once, 2:Daily, 3:Monday~Friday, 4:Monday~Saturday, 5:Date).
5. Enter the hour and minute (HH:MM, 4 digits as 24 Hour Format)
6. If the Wake-up type is date, dial six (6) digits (YY/MM/DD) for year, month and day.
7. Dial '*'; a confirmation tone will be heard.

To cancel the Wake-up alarm:

1. Lift the handset.
2. Enter the {Wake-up Cancel} feature code; a confirmation tone will be heard.
3. Enter the index number of wake-up (1~5).
4. Dial '*'; a confirmation tone will be heard.

Conditions

- If the user goes Off-hook during the Wake-up alarm, the system plays a Wake-Up prompt.

- Wake-up service can be assigned by an LIP Phone, LDP Phone, SLT, Attendant, WMS and external PMS.
- Five (5) Wake-Up options are supported: Once, Daily, Monday~Friday, Monday~Saturday, or a Date.
- Wake-up call registration for Daily, Monday~Friday or Monday~Saturday alarms are maintained until it canceled by the user.
- Wake-up call registration for Once or Date mode is canceled automatically after placing the Wake-up call.
- The Wake-up call is not activated while the user is on conversation.
- The Wake-up call Retry Count can be defined through WMS, the default value is two (2).
- The Wake-up call Retry Interval can be defined through WMS, the default value is 2 minute.
- The following progress of Wake-up call service will be reported through Status Window of WMS;
  - Answer the Wake-Up call
  - No-answer for the Wake-up call

Related WMS Menu

[To assign Wake-up Register/Cancel feature code]

Data Management > Numbering Plan Information > Feature Code > "Wake-up Register"," Wake-up Cancel"

[To allow Wake-up feature of DN]

Data Management > Extension Information > Number(DN) Information > DN Feature Allow/Deny > "Wake-Up"

[To change Wake-up information of DN]

Data Management > Extension Information > Number(DN) Information > DN Feature Registration > "Wake-Up Type/Date/Time"

[To set Wake-up Retry Count, Retry Time]

Data Management > Tenant Information > Tenant Basic Information > Tenant Attribute > "Wake-Up Retry Count"
> Tenant Information > Tenant Basic Information > Tenant Attribute > "Wake-up Retry Interval"

[To assign Wake-up Answer tone]

Data Management > Tenant Information > Tenant Tone/Ring Information > Tenant System Tone > "Wake-Up Answer Tone"

Related Feature

Hardware

- VPCM for Wake Announcement

2.129 IP DECT Interworking

Description

Explanation for installing and operating IP DECT with iPECS-CM system

Outgoing/incoming voice call management: The SME VOIP system can provide multiple priority user classes. Further, up to 3 repeaters can be linked to a Base-station.

Internal handover: User locations are reported to SIP Server in order to provide differentiated services and tariff management. Within a DECT traffic area, established calls can seamlessly be handover between Base-stations using connection handover procedures.

Security: The RTX SME VOIP system also supports robust security functionalities for Base-stations. Most security functionality is intrinsically woven into the SME VOIP network structure so that network connections can be encrypted and terminal authentication can be performed

Operation

Base station Registration

Below procedure should be followed

1. Using handset, find IP feature
2. Using browser, IPDECT (http://ipdect+macaddr)
3. Login to base configuration interface
4. Once you have authenticated, the browser will display front end of the SME configuration interface

Repeater Registration

Adding a repeater menu makes it possible to register the repeater.
Registration is made by select the repeater and pressing register repeater.

Registration handset

Below procedure should be followed

1. Login to a base station
2. Select “Extension” URL and click “Add extension” link
3. Fill out the form and click “Save”
4. Set a check mark on the extension which shall be assigned to the handset you want to register and click “Register handset”

Multi cell setup and management

1. Connect the base station to a private network
2. Use one of the two methods to determine the base station IP address
3. Click on Multi Cell URL link in the SME voip configuration to review the current multi cell settings status of the current base station.
4. On the multi cell setting page, choose Enable option from the drop down menu of the multi cell system parameter
5. Enter the relevant values for system chain ID and synchronization time respectively.
6. On each SME Voip configuration interface for the base station navigate to the home/status page
   And click the reboot button
7. Navigate back to the multi cell settings page by clicking multi-cell URL link at the left column

Conditions

We enumerate some environmental conditions that need to be considered prior to planning, deployment and optimization of network. The considerations are as follows:

- Ensure that the installation area is clean and dry
- Ensure that the floor of the installation area is finished with linoleum, vinyl, ceramic, wooden flooring, computer floor tiles, or polished sealed concrete.
- Ensure that the ceiling of the installation area is finished or treated to prevent particle discharge
- Ensure that the installation area is well lit, and that the light source is uniformly diffused without shadows. Adequate lighting should provide a comfortable reading level and allow the identification of wire insulator colors without undue eye fatigue. Lighting should be comparable to an office work environment, with a minimum level of 21 meter/68.9 feet at each work surface. As a rule of thumb, in a room with a 2.5 meters/8.2 feet ceiling, one 1.2 meters/4 feet fluorescent tube provides sufficient illumination for 1.9-2.4 square meters/20.5-25.9 square feet.
- Ensure that ventilation of the installation area is capable of maintaining an ambient temperature of 0-40°C/32-104°F, and a relative humidity of 20-80% non-condensing, while the system is operating. The maximum power rating of a base station under full load should not exceed 315W/1070 BTU/Hr. These figures are for each cabinet only, and do not take into account heat generated by other equipment. In particular, charging fully-discharged batteries may generate a considerable amount of heat, depending on battery capacity and rate of charge. Refer to the equipment manufacturer data for more information.
- Ensure that the installation area is free of caustic or corrosive liquids, substances, or materials. If batteries will be installed as part of the system, ensure that adequate precautions are taken (such as special ventilation) to prevent corrosive emissions from the batteries. Check local building codes for additional requirements.

Related WMS Menu

[To upgrade firmware]
Version management >SIP Phone Provisioning >SIP Provisioning File Management

[To change or add call feature]
Data Management > Extension information >Number(DN) information >Registration
> Extension information >Number(DN) information >DN Feature Allow/Deny

Related Feature

Hardware

3. SIP PHONE

3.1 SIP Terminal Registration

Description

iPECS-CM supports standard SIP Phones including the Ericsson-LG Enterprise IP-8800 series SIP Phones. The SIP Phone employs the SIP Proxy address, Domain Name and the IP address of iPECS-CM. The SIP User Id is the SIP Extension number with the IP address of the iPECS-CM. For example if the SIP Extension number is 1127 and the iPECS-CM address is '192.168.1.1', the SIP User ID would be 1127@192.168.1.1

Operation

Conditions

- SIP phones must comply with SIP statndards and be proven to interoperate otherwise, the phone may not register properly with the iPECS-CM.
- In order to register 3rd party SIP phones, a license is required in the iPECS-CM.
- Any type of device which supports standard SIP may be used with the iPECS-CM. For example, an IVR, Video Conference Device can be registered to the system.
- SIP terminals, even if they are under NAT, can be registered to iPECS-CM.
- The registered terminals must send a periodic "REGISTER" message to iPECS-CM. Any terminal that fails to send a "REGISTER" message is automatically unregistered after the expiration of the Register Timer.

Related WMS Menu

[To create Number]
Data Management > Extension Information > Terminal Information> Terminal Attribute
[To set User ID, Authentication ID/Password]
Data Management > Extension Information > SIP Extension Information > SIP Terminal Configuration
[To set options of SIP Extension]
Data Management > Extension Information > SIP Extension Information > SIP Extension Attributes

Related Features

Hardware

- SIP Phone

3.2 SIP Name Registration

Description

Normally SIP phone users can program their names to be displayed with the Extension number in the LCD of the calling party.

Operation

To register or delete a name in a SIP phone, refer to the SIP phone User Manual for the "Display Name".

Conditions

- To use this function, the SIP phone must provide the name registration function. If the SIP phone does not provide the Display Name function, the name can be changed through WMS.
- Operation of this feature is reliant on the SIP terminal, verify the SIP terminal operation.

Related WMS Menu

Related Features

Hardware

- SIP Phone

3.3 SIP Call

Description

SIP phones registered in iPECS-CM may place a call to other Extensions (iPECS Multi-button phone, SLT, SIP phone) in iPECS-CM. Users can place Trunk calls using the Trunks in iPECS-CM.

Operation

SIP Phone

To place an intercom call:

1. Dial desired Extension number.
2. Press the SEND button on the SIP phone.

To place an external call:

1. Dial the Trunk access code and desired phone number. For example, if the Trunk Access code is '9' and the phone number is '450-4500', dial "94504500".
2. Press the SEND button on the SIP phone.

Conditions

- Some SIP phones do not have a SEND button; in this case, press '#' or wait for automatic dialing. Refer to the SIP Phone User Manual for additional information.
- In order to place a call, the SIP phone must be registered in iPECS-CM properly.
- Redial/Call logs, when supported by the SIP phone, can be used with the iPECS-CM.
- The following functions are available for outgoing calls from SIP phone:
  - Internal call
  - Trunk call
  - Attendant call
  - Call pick-up

- System speed dial
- Key number group call
- Interphone group call
- Conference
- Page and page answer
- Command call
- Voice mailbox
- Call park answer
- CLIR

Related WMS Menu

Related Features

Hardware

- SIP Phone

3.4 SIP Answering Call

Description

SIP phones registered in iPECS-CM can answer calls from other Extensions (iPECS Multi-button phones, SLTs, SIP phones) or Trunks in iPECS-CM.

Operation

SIP Phone

To answer calls:

1. When a SIP phone receives a call, the iPECS-CM will deliver the caller Id (the calling Extension or CID provided by the Trunk).
2. Lift the handset of the SIP phone or press the [ANSWER] button.

Conditions

- In order to receive calls properly, the SIP phone must be registered in iPECS-CM.
- Received/Missed Call logs, when supported by the SIP phone, can be used with the iPECS-CM.

Related WMS Menu

Related Features

Hardware

- SIP Phone

3.5 SIP Hold Call

Description

SIP terminals can place an active call in a waiting state, hold, where the connected party hears MOH (Music-On-Hold..

Operation

SIP Phone

To place a call on hold:

1. Press the [HOLD] button during a call. The call is placed on exclusive hold; other Extensions cannot seize the call on hold.

To re-seize calls on hold:

1. Press the button with the held call.

Conditions

- For SIP Phones, hold operates as a Transfer hold, so the held party receives “Transfer Hold tone”.
- To use this function, the SIP terminals must support the hold function.
- The operation of this function may be different from this description. Refer to the SIP phone User Manual.

Related WMS Menu

Related Features

Hardware

- SIP Phone

3.6 SIP Transfer Call

Description

SIP phone users can transfer calls to other Extensions or Trunks. To use this function, the SIP phone must support the Call Transfer function.

Operation

SIP Phone

To use Consultation Transfer:

1. Press the [TRANSFER] button
2. Dial destination number
3. After the destination answers the call, announce the call
4. Hang-up to complete the Transfer.

To use Blind Transfer:

1. Press the [TRANSFER] button

2. Dial destination number
3. Hang-up to complete the Transfer.

Conditions

・In order to use this function, the SIP phones must support the call transfer function.
・The operation of this function can be different from this description. Refer to the SIP Phone User Manual.
・The transfer function may not work properly in some SIP phones even if it is supported in the SIP phones. This is because the SIP call transfer message is not compatible between the SIP phones and iPECS-CM.
・iPECS-CM complies with the IETF spec "draft-ietf-sipping-cc-transfer-09.txt".

Related WMS Menu

Related Features

Hardware

- SIP Phone

3.7 SIP Call Forward

Description

SIP phone users can forward incoming calls to other Extensions or Trunks. This function requires the Call Forward function to be supported in the SIP phones.

Operation

SIP Phone

To forward calls:

1. Refer to the manufacturers SIP Phone User Guide.

Conditions

・If the SIP Phone does not have the CFW function, the system Call Forward can be used.
・The operation of this function can be different from this description. Refer to the SIP Phone User Manual.

Related WMS Menu

[To assign Forward Register feature code]
Data Management > Numbering Plan Information > Feature Code > "Forward Register"

Related Features

Hardware

- SIP Phone

3.8 SIP DND

Description

SIP phones can utilize Do Not Disturb if the DND function is supported by SIP phones.

Operation

SIP Phone

To activate DND:

1. Refer to the manufacturers SIP Phone User Guide.

Conditions

- If the SIP Phone does not have the DND function, the system DND can be used.
- The operation of this function can be different from this description. Refer to the SIP Phone User Manual.

Related WMS Menu

[To assign DND Register/Cancel feature cod]

Data Management > Numbering Plan Information > Feature Code > "DND Register/Cancel"

Related Features

Hardware

- SIP Phone

3.9 SIP Conference

Description

SIP Phone with built-in 3-way conference capability can establish a conference call without utilizing the system's conference feature. In addition, SIP Phones can utilize the system's conference feature.

Operation

SIP Phone

To establish a conference using the SIP phone feature while on a call:

1. Press the [CONF] button.
2. Place a call to the desired user.
3. Press the [CONF] button.

To make conference using system feature while on a call:

1. Press the [TRANS] button.
2. Dial {Conference Member Add } feature code
3. Dial the desired Extension number, when the party answers, the conference is connected.

Conditions

- The operation of this function can be different from this description. Refer to the SIP Phone User Manual.

Related WMS Menu

[To assign Conference Member Add feature code]
Data Management > Numbering Plan Information > Feature Code > “Conference Member Add”

Related Features

Hardware

- SIP Phone

3.10 SIP Call Wait/Broker Call

Description

Standard SIP phones enable users to perform Call Wait and Broker Call functions for incoming calls while on busy. This function requires the SIP Phone support Call Wait/Broker Call function.

Operation

SIP Phone

Call wait/Broker call:

1. During a call between User A (SIP phone user) and User B, User C place a call to User A.
2. User C is automatically put on hold.
3. User A can perform broker call between Users C and B.

Conditions

- SIP phones must support the Call Wait/Broker Call function.
- The operation of this function can be different from this description. Refer to the SIP Phone User Manual.

Related WMS Menu

Related Features

Hardware

- SIP Phone

3.11 SIP SMS

Description

SIP phones can exchange SMS messages with other Extensions and Trunks. This function requires the SMS function to be supported in the SIP phones.

Operation

SIP Phone

To send/receive SMS:

1. Refer to the SIP phones User Manual.

Conditions

- To use this function, the SIP phone must support SMS.
- The operation of this function can be different from this description. Refer to the SIP Phone User Manual.

Related WMS Menu

Related Features

Hardware

- SIP Phone

3.12 SIP Voice Mail Notification

Description

When supported by the SIP Phone, the system will notify the phone of a Voice Message Waiting.

Operation

SIP Phone

Indication of voice mail:

1. The [MSG] LED will flash if there is a Voice Mail saved for the user.
2. Press the [MSG] button; the number of new messages and saved messages is provided.

To retrieve voice mail:

1. Dial the {Voice Mail Access} feature code.

Conditions

- To use this function, the SIP terminals must support VM notification.
- The operation of this function can be different from this description. Refer to the SIP Phone User Manual.

Related WMS Menu

[To assign Voice Mail Access feature code]
Data Management > Numbering Plan Information > Feature Code > “Voice Mail Access”

Related Features

Hardware

- SIP Phone

3.13 SIP Video Call

Description

Certain SIP phones, including Ericsson-LG Enterprise’s IP-8850, enable users to perform video calls. This function requires the Video Call function be supported in the terminals of both calling parties and called parties.

Operation

SIP Phone
To place a Video Call from the IP-8850:
1. Dial a SIP phone that supports Video calling.
2. Press the Video Call Send button on the SIP phone.

Conditions

- Video calls may not be supported, depending on the network connection or if the SIP phones are registered in iPECS-CM.
- Video call is not supported if different video codecs are used in the SIP phones. iPECS-CM does not support conversion between video codecs.
- If Video call capability is not available in the SIP phone, only the voice call is established.
- To make a Video conference between three (3) or more video terminals, an external video conference facility is required.
- The operation of this function may be different from this description. Refer to the SIP Phone User Manual.

Related WMS Menu

Related Features

Hardware

- SIP Phone

3.14 One number Multi-phone

Description

Multiple SIP Phones can share one phone number. If a same number is registered to multiple SIP phones, Invites are sent to all phones with the number. If one of the SIP phones answers the call, the SP phone is connected to the caller, and other terminals are notified to stop ringing.

Operation

Conditions

- This function is provided using the MADN (Multi-Assigned Directory Number) Feature.

Related WMS Menu

[To assign DN to Phone]
Data Management > Extension Information > Terminal Information> Phone Flexible Button
[To set DN options]
Data Management > Extension Information > Number(DN) Information > DN Attribute
[To query Extension which use DN & set Ring Option]
Data Management > Extension Information > Number(DN) Information > DN Member

Related Features

- Mobile Extension

Hardware

- SIP Phone

3.15 DSS/BLF

Description

The DSS/BLF function can be assigned to the Flex button of a SIP phone, and then the button on the SIP phone indicates the status of the assigned Extension. The,button also performs Pick-up and Call Transfer features using the DSS/BLF button.

Operation

SIP Phone

Indication of BLF

- Busy: LED on
- Incoming ring: LED flash
- Not used: LED off

Conditions

- This function is only supported for IP8800 Series SIP phones, which comply with the Ericsson-LG Enterprise SIP-Extended I/F specification.

Related WMS Menu

Related Features

Hardware

- Ericsson-LG Enterprise IP-8800 series SIP phone

3.16 Distinctive Ring

Description

Different ring signals can be assigned to the SIP phone depending on whether the call is from an Extension or a Trunk.

Operation

SIP Phone

To program Distinctive Ring based on the call type:

1. Access the Web Program for the SIP phone.
2. Set different rings for internal calls and Trunk calls.

Conditions

- This function is only supported for IP8800 Series, which comply with the Ericsson-LG Enterprise SIP-Extended I/F specification.

Related WMS Menu

Related Features

Hardware

- Ericsson-LG Enterprise IP-8800 series SIP phone

3.17 Intercom

Description

A SIP phone can answers internal calls automatically through the speakerphone without lifting the handset. The SIP Phone must be assigned the Handsfree Answer Mode through the WMS.

Operation

Conditions

- This function is available for the SIP phone, IP8800 Series that support an extended “Alert-Info” message.
- The SIP Phone must support Auto Answer and be assigned for Handsfree ICM Answer.

Related WMS Menu

[To set ICM Answer Mode]

Data Management > Extension Information > Terminal Information > Terminal Options > “ICM Answer Mode”

Related Features

Hardware

- Ericsson-LG Enterprise IP-8800 series SIP phone

3.18 Call-Back

Description

If the called Extension is busy, the caller may request a Call Back when the called party returns to idle.

Operation

SIP Phone

To register Call Back from a SIP phone:

1. While receiving busy, goes on-hook.
2. Dial the “Extension Call Back/Trunk Queuing” feature code + Extension Number
3. Confirmation Tone is received

To answer a Call Back:

1. When the called Extension returns to idle, the caller receives an Invite.
2. When Answered, the call is placed.

Conditions

Related WMS Menu

[To assign Extension Call Back/Trunk Queuing feature code]

Data Management > Numbering Plan Information > Feature Code > “Extension Call Back/Trunk Queuing”

Related Features

Hardware

- SIP Phone

3.19 Call Intrusion

Description

Users can intrude into existing calls. A conference call is established between the call and the intruding Extension.

Operation

SIP Phone

To intrude on an active call:

1. Receive Busy tone.
2. Dial the “Intrude Request” feature code + Extension number.
3. A 3-party conference is established between the parties on initial call and intruding SIP phone.

Conditions

- Allowing intrusion into busy SIP phone is only supported with IP8800 Series that comply with Ericsson-LG Enterprise SIP Extended I/F spec.
- Any SIP phone can attempt intrusion.

Related WMS Menu

[To assign Intrude Request feature code]
Data Management > Numbering Plan Information > Feature Code > “Intrude Request”

Related Features

Hardware

- Ericsson-LG Enterprise IP-8800 series SIP phone

3.20 Call Override

Description

If the called Extension is busy, the SIP phone user may request Call Override; the called party connection is placed on hold and the called Extension is connected.

Operation

SIP Phone

To request a Call Override

1. Place a call to a busy Extension.
2. Receive the busy tone.
3. Dial the “Override Register (SIP)” feature code + Extension number.
4. The existing call of the called party is put on hold, and the called party is connected to SIP phone that requested Call Override.

Conditions

- Allowing override into busy SIP phone is only supported for LIP8800 Series phones, which comply with Ericsson-LG Enterprise SIP Extended I/F specification.

Related WMS Menu

[To assign Override Register feature code]

Data Management > Numbering Plan Information > Feature Code > "Override Register (SIP)"

Related Features

Hardware

- Ericsson-LG Enterprise IP-8800 series SIP phone

3.21 SIP BLA (Bridge Line Appearance)

Description

Multiple numbers can be assigned to one SIP phone using the MADN function. Outgoing and incoming calls, Hold and Conference are available when using the BLA DN.

Operation

Conditions

Related WMS Menu

[To assign DN to Phone]
Data Management > Extension Information > Terminal Information> Phone Flexible Button
[To set DN options]
Data Management > Extension Information > Number(DN) Information > DN Attribute
[To query Extension which use DN & set Ring Option]
Data Management > Extension Information > Number(DN) Information > DN Member

Related features

- DN

Hardware

- Ericsson-LG Enterprise IP-8800 series SIP phone

3.21.1 SIP BLA Registration

SIP Phone Setting

To use BLA feature, the BLA line must be assigned to SIP phone in accordance with the manufacturer's instructions. The User Id should be assigned as the "dn<xxx>at<yyy>, where xxx is the BLA DN and yyy is the Extension's My-DN. For example, to assign DN 1500 as a BLA at SIP Extension number 3003, the User Id is "dn1500at3003.

The Authentication ID and Authentication Password of My-DN should be used by the SIP Phone for BLA line registration.

Operation

Conditions

・A BLA DN is serviced using a MADN.

Related WMS Menu

[To assign DN to SIP Phone]
Data Management > Extension Information > Terminal Information> Phone Flexible Button
[To set DN options]
Data Management > Extension Information > Number(DN) Information > DN Attribute
[To query Extension which use DN & set Ring Option]
Data Management > Extension Information > Number(DN) Information > DN Member

Related Features

- Directory Number (DN)

Hardware

- Ericsson-LG Enterprise IP-8800 series SIP phone

3.21.2 SIP BLA Outgoing Call

Description

A SIP phone can make an outgoing call after seizing a DN button and dialing the desired number in the same manner as the iPECS Multi-button phone. When a DN is seized, the LED of the DN button turns On and dial tone is received. The LED at other phones with the DN will turn On to notify the DN is being used.

Conditions

・If Auto Dialing digits are assigned, the system adds the assigned digit(s) to the number dialed from user and then places the outgoing call.
・If the user takes no action after seizing a DN for pre-determined time, the DN goes to idle and the phone is placed out-of-service automatically.

Related WMS Menu

[To assign Auto Dialing Digit of DN]
Data Management > Extension Information > Number(DN) Information > DN Attribute > "Auto Dialing Digit"

Related Features

Hardware

- Ericsson-LG Enterprise IP-8800 series SIP phone

3.21.3 SIP BLA Incoming call

Description

When a DN assigned as BLA is called, all SIP phones which have the DN will receive an Invite message. When one of SIP Phones answers the call, the other SIP phones are notified.

Operation

Conditions

- When an Extension answers a call, no Call Logs are left at other Extensions that received ring signals for the call.
- Delay ring can be assigned by WMS and the SIP phone. A delay ring signal by system is applied then delay ring signal by SIP phone is applied.

Related WMS Menu

Related Features

Hardware

- Ericsson-LG Enterprise IP-8800 series SIP phone

3.21.4 SIP BLA Hold

Description

When a DN assigned as BLA is placed on hold, the LED of DN button of all SIP phones that have an appearance of the DN will blink to notify users that the DN is on hold. Any SIP phone with the DN can answer the held DN.

Operation

Conditions

Related WMS Menu

Related Features

Hardware

- Ericsson-LG Enterprise IP-8800 series SIP phone

3.21.5 SIP BLA Barge-in

Description

When a SIP phone is using a DN assigned as BLA, any phone with an appearance of the DN can Barge-in on the call, establishing a 3-party conference.

Operation

Conditions

・This is same as the Branch Line DN feature.

Related WMS Menu

[To set Branch Line of DN]

Data Management > Extension Information > Number(DN) Information > DN Feature Registration > “Branch Line / Bridge Line”

Related Features

Hardware

- Ericsson-LG Enterprise IP-8800 series SIP phone

3.22 Catch Call

Description

CM can inform you of the incoming call information when the status of user is ‘Busy’ or ‘DND’ or ‘Forward’ or the incoming call number is registered to spam. In this case, the user would set the ‘Catch Call Log Save Option’ menu to ‘Busy’ or ‘Busy’, ‘DND’, ‘Forward’, ‘Spam’.

Operation

To use Catch Call Log Save Option only (Busy)

1. Subscriber A is calling status.
2. Other subscriber B makes call to A.
3. The subscriber A can check incoming call information of subscriber B.

To use Catch Call Log Save Option (Busy, DND, Forward, Spam)

1. Subscriber A is calling or DND or forwarding status.
2. Other subscriber B makes call to A.
3. The subscriber A can check incoming call information of subscriber B.
4. If the call number of subscriber B is set to spam number, A can also check the call information from spam number.

Conditions

・SIP user can check by SMS.

Related WMS Menu

[To set use of Catch Call in System]

Data Management > Extension Information > Terminal Option > Catch Call Log Save Option

Related Features

Hardware

Thanks for purchasing iPECS CM

iPECS is an Ericsson-LG Brand