iPECS SBG-1000 User Manual

(IP-PBX Features)

Regulatory Information

Before connecting the Smart Business gateway to the telephone network, you may be required to notify your local serving telephone company of your intention to use "customer provided equipment." You may further be required to provide any or all of the following information:

PSTN line Telephone numbers to be connected to the system	
Model name:	Smart Business gateway
Local regulatory agency registration number:	locally provided
Registered jack:	RJ-11 with Desk Holder/Wall Mount
	RJ-21X with Main Cabinet

The required regulatory agency registration number is available from your local Ericsson-LG representative.

This equipment complies with the following regulatory standards, FCC Part 15 and 68, IC (Industry Canada) CS03, TBR03, and TBR04. Also, this equipment complies with the safety requirements of UL60950, CSA60950, EN60950, EN55022 and EN55024

The Smart Business gateway has been designed to comply with the Hearing Aid Compatibility requirements as defined in Section 68.316 of Part 68 FCC Rules.

If the telephone company determines that customer provided equipment is faulty and may possibly cause harm or interruption in service to the telephone network, it should be disconnected until repair can be affected. If this is not done, the telephone company may temporarily disconnect your service.

The local telephone company may make changes in its communications facilities or procedures. If these changes could reasonably be expected to affect the use of the Smart Business gateway or compatibility with the network, the telephone company is required to give advanced written notice to the user, allowing the user to take appropriate steps to maintain telephone service.

The Smart Business gateway complies with rules regarding radiation and radio frequency emission as defined by local regulatory agencies. In accordance with these agencies, you may be required to provide information such as the following to the end user:

WARNING

This equipment generates and uses R.F. energy, and if not installed and used in accordance with the Instruction Manual, it may cause interference to radio communications. It has been tested and found to comply with the appropriate limits for a telecommunication device. The limits are designed to provide reasonable protection against such interference, when operated in a commercial environment.
Operation of this equipment in a residential area could cause interference, in which case the user, at his own expense, will be required to take whatever measures may be required to correct the interference.

CAUTION

This system employs a Lithium battery as back-up power for the real-time clock and memory. The battery is not replaceable in the field. RISK OF EXPLOSION IF BATTERY IS REPLACED BY AN INCORRECT TYPE. Dispose of used batteries accordance with the manufacturer's instructions.

Revision History

ISSUE	DATE	DESCRIPTION OF CHANGES
1.0	2011, August	Initial Release
1.1	2012, February	Update for SBG S/W version A.0Au
1.2	2012, October	Update for SBG S/W version 1.0Bb
1.3	2013, April	Update for SBG S/W version 1.0Cd

Table Of Contents

1. INTRODUCTION

1.1 OVERVIEW

Smart Business gateway (iPECS SBG-1000) is Ericsson-LG's internet Protocol (iP) Enterprise Communications Solution designed to meet the telecommunication needs of small-sized business. Smart Business gateway uses advanced packet voice and IP switching technology, which is combined with a rich feature content, to set a new standard in Voice over IP (VoIP) systems.

The system consists of basic one FXS port, eight 10/100 Base-T LAN Ethernet ports (including four POE ports) and one Wan Ethernet port, one USB port. iPECS SBG-1000 is installed on the desk and powered from an AC/DC adapter, which converts 100~240VAC to 48VDC. With optional board, the system can have one/two FXO ports or one BRI port additionally and FXS can be increased two ports.

iPECS SBG-1000 supports a variety of Phones; LIP Phones using iPECS Protocol, SIP Terminals (WIT-400H, 88xx), and analog single-line devices. With the LIP Phones, commonly-used features are activated with the touch of a single button. Additionally, most functions can be accessed from any telephone by dialing specific codes.

Providing an environment rich in features, in addition to a fully featured voice intercom, the iPECS SBG-1000 incorporates built-In Auto-Attendant (AA) and Voice Mail (VM), Remote Management such as Web Based Admin.

By employing packet voice and IP switching, the iPECS SBG-1000 infrastructure can be employed for, or can share the enterprise data network. Further, since all terminals have a unique IP address, they can be moved anywhere with access to a network that can connect to iPECS SBG-1000 and function without the need for "re-programming". The use of the single common infrastructure and ability to easily install or relocate telephones results in significant savings from the time of installation and throughout the life of the system.

1.2 HARDWARE COMPONENTS

iPECS SBG-1000 is shipped with the iPECS SBG-1000 module, a power adaptor and a power cord as shown in Figure 1.2-1.

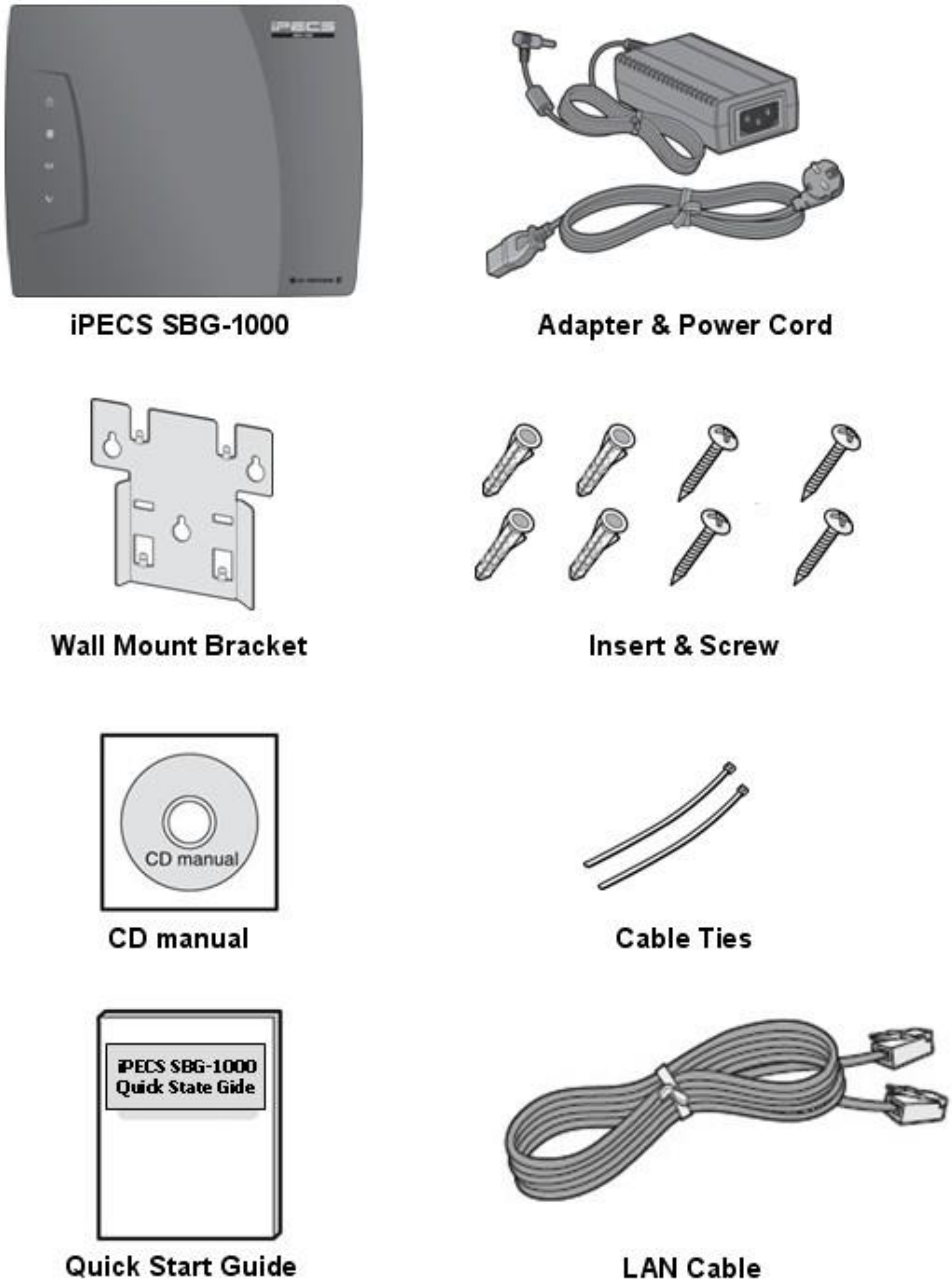


Figure 1.2-1 Components in iPECS SBG-1000 package

To obtain terminal options with iPECS SBG-1000, contact an authorized agent of Ericsson-LG Co., Ltd.

Table 1.2-1 iPECS SBG-1000 products

No.	Product	Description	Remark
1	Smart Business gateway	Smart Business gateway Gateway Module	Basic
2	AC/DC Adaptor	AC/DC Adaptor for module, (48VDC, 0.8A)	Basic
3	AC Power Cord	AC power cord for an Adaptor	Basic
4	LIP Phones	Ericsson-LG LIP Phones using iPECS protocol	Option
5	SIP Terminals	LIP-8002, IP-DECT, Dual Mode Wireless Phones	Option
6	POTS Terminals	analogue single line devices	Option
7	DECT Phones	Ericsson-LG System DECT Phones	Option

1.3 MANUAL APPLICATION

This document provides detailed information covering description and operation of the numerous features available in the iPECS SBG-1000 system software. The document is written assuming the system employs the default numbering plan.

1.3.1 Organization

Features are arranged in two different major groupings as follows:

- Section 2 Call Features
- Section 3 Web Administration

1.3.2 Feature Information

Each section is an alphabetical listing of features with the description and operation of each. The structure is divided into 6 parts as below:

- **Description:** explains the nature of the feature.
- **Operation:** gives detailed step-by-step operation of the feature for Keysets and SLTs.
- **Conditions:** explains known feature interactions and constraints related to the feature.
- **Programming:** lists database entries that may be required for proper feature operation.
- **Reference:** lists related topical information to aid in understanding the feature.
- **Hardware:** lists hardware required for proper feature operation.

1.4 SYSTEM CAPACITIES

The iPECS SBG-1000 is presently available in one configuration as shown in the Table 1.4-1.

Table 1-4-1 Smart Business gateway Capacity Chart

DESCRIPTION		CAPACITY
		Smart Business gateway – iPECS SBG-1000
Stations	IP Extension	11 / 23[1)]
	FXS (FAX, SLT)	1 / 2[2)]
	SIP Extension	6
	DECT	6[3)]
	Total	12 / 24[1)]

DESCRIPTION		CAPACITY
		Smart Business gateway – iPECS SBG-1000
VoIP channel		3 / 4 / 6 1)
FXO port (Analog trunk)		0 / 1 / 2 / 4 2)
BRI port		0 / 1 / 2 2)
Auto Attendant channels		4
Attendants		1
PFT		1 4)
USB Host Port		1
Paging Zones		10
Common Speed Dial		800 (23 digits)
Individual Speed Dial		20 (23 digits)
Last Number Redial		15 (23 digits)
Save Number Redial		1 (23 digits)
SMDR Buffer		5000
Station Groups		10
Station Group Members		12 / 24 1)
Authorization Codes	Station	12 / 24 1)
	System	100
Voice Mail Box		240 / 480 minutes 5)

1) Capacity S/W license decides the number of total stations and the maximum available number of VoIP channel.

Capacity S/W License	DECT Usage	Total Stations	VoIP Channel
Normal	ON	12	3
	OFF		
Extended	ON	24	4
	OFF		6

2) Basic iPECS SBG-1000 has only 1 FXS port and FXS, FXO and BRI port capacity depends on the optional board as follows.

Optional Board	FXS	FXO	BRI
CSIU	Basic 1 + 1 = 2	1	0
CIU1	Basic 1	1	0
CIU2	Basic 1	2	0
CIU4	Basic 1	4	0
BRIU	Basic 1	0	1
BRIU2	Basic 1	0	2

3) Up to six (6) DECT stations can be registered but only four (4) DECT stations can place a call or get a ring simultaneously.
4) PFT is connected to FXO LINE1 and it works only with CSIU, CIU1, CIU2 and CIU4.

5) The capacity for Voice Mail Box depends on the lock key for the VSF Memory Extend.

1.5 Hardware Description

iPECS SBG-1000 can be mounted on any flat surface itself. The external AC/DC adaptor (48Vdc /0.8A) feeds power to the System. iPECS SBG-1000 includes battery back-up circuitry using a long-life Lithium battery to maintain the real-time clock and prevent loss of system database during power fail.

Connection ports

- In the right side
  - One WAN port (RJ-45: 10/100/1000 Base-T Ethernet port)
  - 8 LAN ports (RJ-45: 10/100 Base-T Ethernet port)
  - One basic FXS port (RJ-45)
  - One MISC port for Relay Contact and Alarm Detection
  - 48 VDC Power Input Jack
- In the left side
  - One USB Port in the left side

Buttons

- In the right side
  - Reset to Default button
- In the left side
  - WPS (WiFi Protect Setup) button
  - Reset button

Indicators

The following LAN LEDs provide visual representation of iPECS SBG-1000 LAN ports activities and status in normal state.

Table 1.5-1 iPECS SBG-1000 LAN Activities and Status

Name	LED Color	Status	Description
LINK/ACT	Green	ON	Valid LAN Link
		OFF	Link Fail
		FLASH	TX/RX Activity
10/100	Orange	ON	100 Mbps
		OFF	10 Mbps

The following WAN LEDs provide visual representation of iPECS SBG-1000 WAN port activities and status in normal state.

Table 1.5-2 iPECS SBG-1000 WAN Activities and Status

Name	LED Color	Status	Description
LINK	Green	ON	Valid LAN Link
		OFF	Link Fail
ACT	Orange	ON	100 Mbps
		OFF	10 Mbps

Status LEDs on the top panel indicate operation states of the iPECS SBG-1000 as shown in below Table 1.5-3.

Table 1.5-3 iPECS SBG-1000 Operation States

Icon	Name	LED Color	Status	Description
	POWER	Blue	ON	Power On
			OFF	No Power
	WAN	Pure Green	ON	Valid WAN Link
			OFF	WAN Link Fail
	WLAN	Pure Green	ON	WLAN is Running
			OFF	WLAN is Not Initialized
			BLINK	WPS is in progress
	PBX	Pure Green	BLINK	Call Task is Running
			ON/OFF	Call Task Not Initialized

1.6 SPECIFICATIONS

Environmental Specification		
	Degrees (℃)	**Degrees (℉)**
Operation Temperature	0 ~ 40	32 ~ 104
Optimum Operation Temperature	20 ~ 26	68 ~ 78
Storage Temperature	0 ~ 70	32 ~ 158
Relative Humidity	0~80% RH non-condensing	

Power Adaptor Specification*	
AC Input	AC100-240V, 50/60Hz, 1A max.
DC Output	DC48V, 0.8A max

** A Power adaptor is delivered with Smart Business gateway.*

FXS Specification (Basic SLT or CSIU)		
Connector	RJ-45	
Loop Distance	1.5 Km	AWG #24 (0.5mm)
Ring Capacity	60Vrms (up to 3 REN)	
Ring Frequency	25Hz	

FXO Specification (CIU1, CIU2, CIU4 or CSIU)	
Connector	RJ-45
REN (Ringer Equivalent Number)	0.7B

ISDN BRI Specification (BRIU, BRIU2)	
Connector	RJ-45
Network Type	T interface

Ethernet Specification (LAN port 1 ~ LAN port 8)	
Connector	RJ-45 shielded
LAN Interface	10/100 BASE-T (Auto-Negotiation), 10 Mbps or 100 Mbps, IEEE 802.3
Maximum Wiring Distance Cable	100 m/ 0.328kft, Category 5 UTP cable

Ethernet Specification (WAN port)	
Connector	RJ-45 shielded
WAN Interface	10/100/1000 BASE-T (Auto-Negotiation), 10 Mbps or 100 Mbps or 1000 Mbps, IEEE 802.3/ IEEE 802.3ab/ IEEE 802.3az
Maximum Wiring Distance Cable	100 m/ 0.328kft, Category 5e UTP cable for 1000 Mbps

PoE Specification (LAN port 1 ~ LAN port 4 only, LAN port 5 ~ LAN port 8 are not supported)	
Interface specification	IEEE 802.3af (Total budgets : 20 W)

WiFi Specification	
Interface specification/ Frequency	IEEE 802.11 b/g/n (Draft 2.0), 2x2 MIMO, 2.412GHz ~ 2.472GHz

DECT Specification (WiFi Version – not supported)	
Interface specification	CAT-iq 2.0, DECT 6.0
Frequency	1,880MHz ~ 1,900MHz for Europe, 1,920MHz ~ 1,930MHz for US

USB Specification	
Connector	USB Female Plug type A
Mode	Host V 1.1 / 2.0

Physical Specification		
W x D x H	278 x 233 x 34 mm	10.94 x 9.17 x 1.34 in
Weight (with CIU1 back-up)	860.5g	1.90 lbs

2. CALL FEATURES

2.1 SYSTEM TIME

2.1.1 LCD Date/Time Format Control

Description

The Attendant can select the format of the time and date provided to the LCD of all LIP Phones in the system.

The Attendant can select (toggle between) two formats for both time and date. The formats are:

- Date: Month/day/year or Year/month/date
- Time: 12 hour or 24 hour (military)

Operation

Attendant

To Change LCD Date Format (toggle):

1. Press the **[PGM]** button.
2. Dial '021' (Date Display Format program code).

To Change LCD Time Format (toggle):

1. Press the **[PGM]** button.
2. Dial '022' (Time Display Format program code).

Conditions

Programming

Related Features

Attendant Position

Hardware

2.1.2 Auto Service Mode Control

Description

The service mode defines different ring assignments and answering privileges for the system. The service mode can be controlled automatically through definitions in the Day/Night/Timed Mode

Table, which defines the time of day for the Day, Night and Timed shift modes. The Attendant may change the system mode selection from automatic to manual.

Operation

System

Operation of this feature is automatic.

Conditions

Programming

VOICE CONFIG	System Data – Day/Night/Timed Schedule

Related Features

Off-Hook Signaling
Day/Night/Timed Ring Mode
CO Ring Assignment

Hardware

2.1.3 Day/Night/Timed Ring Mode

Description

The Ring Mode is controlled automatically by the system clock. Ring assignments are applied based on the time of day and day of week. Three modes of ring (Ring Assignments) are provided, Day, Night and Timed.

The Attendant controls the system Ring Service mode changing from Auto Service Mode to Day, Night or Timed service mode. Based on the service mode selected, different ring assignments, answering privileges are invoked for system users.

Operation

Attendant

To change Day/Timed/Night Ring Mode manually;

1. Press the **[DND]** button.
2. Dial 1~4. (1: Day mode, 2: Night mode, 3: Timed mode, 4: Auto Service mode)
3. Press **[HOLD/SAVE]** button.

Conditions

1. Only Attendants can change Day/Timed/Night Ring Mode for the system manually and program the Auto Ring Mode Selection Table.

2. Stations receive incoming ring for CO lines based on database assignment and the system mode (Day/Night/Timed) when the call arrives.
3. When the Day/Night/Timed Mode Table is programmed, the ring is changed automatically based on the times assigned in the table.
4. The Attendant always has manual control of System mode by enabling/disabling the Auto Service Mode Control.

Programming

VOICE CONFIG CO Line Data – Call Routing by Line
CO Line Data – Ring Assignment Table
System Data – Day/Night/Timed Schedule

Related Features

CO Ring Assignment

Hardware

2.2 CALL FORWARD

Description

Users may have selected incoming calls re-routed to other stations, station groups, the VSF, or over a system CO line (Off-Net).

The user selects the type and condition under which calls are to be forwarded by entering a Call Forward code as follows:

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

- Code 7: Off Net Busy; forwards all calls, except recalls, to the selected station when station is busy (SLT only).
- Code 8: Off Net No Answer; forwards all calls, except recalls, to the selected station when the station does not answer within the No Answer timer (SLT only).
- Code 9: Off Net Busy/No Answer; forwards calls if the selected station is busy or does not answer within the No Answer timer (SLT only).

Operation

LIP Phone

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

1. Lift the handset or press the **[SPEAKER]** button to receive dial tone.
2. Press the **[FWD]** button.
3. Dial desired Call Forward code (1-4).
4. Dial the station or station group to receive calls.
5. Replace the handset, return to idle.

To activate Call Forward, Off Premise (to an external number):

1. Lift the handset or press the **[SPEAKER]** button to receive dial tone.
2. Press the **[FWD]** button.
3. Dial Forward condition (1-4)
4. Press **[SPEED]** button and desired bin number.
5. Replace the handset, return to idle.

To activate Call Forward, Remote (Follow-me):

1. Lift the handset or press **[SPEAKER]** button to receive dial-tone
2. Press the **[FWD]** button,
3. Dial Call Forward code ‘0’,
4. Dial the station’s Authorization Code (Station number + password),
5. Dial Forward condition (1-4),
6. Dial the destination station or station group,
7. Replace the handset, return to idle.

To deactivate Call forward:

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

SLT

To activate Call Forward, Unconditional, Busy/No Answer to an internal number:

1. Lift the handset to receive dial tone.
2. Dial 54 (Call Forward code).
3. Dial desired Call Forward code (1-4).
4. Dial station or station group to receive the calls.
5. Replace the handset, return to idle.

To activate Call Forward, to an external number:

1. Lift the handset to receive dial tone.

2. Dial 54 (Call Forward code).
3. Dial Call Forward code (6-9),
4. Dial Speed Dial bin number.
5. Replace handset to return to idle.

To activate Call Forward, Remote (Follow-me):
1. Lift the handset.
2. Dial 54 (Call Forward code).
3. Dial Remote Forward code '0'
4. Dial the station's Authorization Code (Station + Password),
5. Dial Forward condition (1-4)
6. Dial the destination station or station group.
7. Replace handset return to idle.

To deactivate the Call forward:
1. Lift the handset, receive stutter dial-tone,
2. Dial 54 (Call Forward code).
3. Dial '#' to cancel Call Forward.

Conditions

1. A station receiving a forwarded call can transfer the call to the forwarding station.
2. A forwarded intercom call will signal the receiving station in the Tone Signaling mode, regardless of the Intercom Signaling Mode at the station.
3. Calls cannot be forwarded to a station in DND; if attempted, an error tone is returned.
4. Active Call Back or Queue requests are not canceled when attempting to activate Call Forward.
5. When Call Forward is active, a station can make outgoing calls (internal or external) but cannot activate a Call back or Queue request.
6. For CO calls, manually activated Call Forward will override any Preset Call Forward assigned for the station or CO line.
7. Call Forward status is maintained in the system's non-volatile memory for protection from power outage.
8. A station in a Station Hunt Group (Circular or Terminal) can be assigned to receive incoming hunt calls, overriding any Call Forward (the system either recognizes the Forward condition and bypasses hunt calls around the station, or routes hunt calls to the station based on the system database; Member Forward).
9. Off-Net Call Forward of incoming CO calls is essentially an automated DISA call that will establish an Unsupervised Conference; these calls are subject to the conditions of a DISA call and Unsupervised Conference and may require entry of an Authorization Code.
10. Off-Net forward calls are not answered until the system completes dialing of the external call; the call, regardless of internal or external, is then connected to the Off-Premise call.
11. An unlimited number of stations may be set-up in a Call Forward chain, forwarding calls from one station to the next; a station cannot forward calls to a station already a part of the chain.

12. Calls to a Call Forward chain will progress as appropriate through the chain to the last station; if the last station enters DND, CO calls revert to the previous station while intercom calls receive a DND tone.
13. No Answer forward will employ the Station No Answer Forward Timer unless it is set to zero in which case the System No Answer Timer is used.
14. If the Attendant activates Unconditional Call Forward, the receiving station will receive Attendant calls and recall ring; if the receiving station is an LIP Phone, the user will be able to activate features normally reserved for a Main Attendant.

Programming

VOICE CONFIG	System Data – Call Feature Timer – Call Forward No Answer Timer

Related Features

DND (Do Not Disturb)
Station Groups
Individual Speed Dial
Common Speed Dial
Intercom Answer Mode
Call Forward, Preset

Hardware

2.3 CALL FORWARD, PRESET

Description

With Preset Call Forward, calls to a station forward to a pre-determined destination assigned in the system database. Preset Station Call Forward can define separate treatments for CO calls and intercom calls. In addition, separate busy and no-answer treatments are defined, and calls can be directly forward to the users Voice Mail box. Call treatments available include:

- Unconditional; all calls are immediately forwarded
- Internal Busy; Intercom calls encountering a busy signal are forwarded immediately
- ICM No-Answer; Intercom calls not answered in the No-Answer time are forwarded (Note: calls to a busy station are also forward after the No-Answer time)
- External Busy; external calls that encounter busy are forwarded immediately
- External No-Answer; external calls, not answered in the No-Answer time are forward (Note: calls to a busy station also are forward after the No-Answer time)

Operation

System
Operation of Preset Call Forward is automatic.

Conditions

1. A station receiving a forwarded call can transfer the call to the forwarding station.
2. Calls cannot be forwarded to a station in DND; if attempted, an error tone is returned.
3. Manual Forward has higher priority than Preset Forward and overrides any Preset Forward setting.
4. Calls to a Preset Call Forward chain will progress as appropriate through the chain to the last station. If a station in manual Call Forward or DND is encountered, it is bypassed and the next station in the chain is signaled. If the last station has entered DND, CO calls revert to the previous station, signaling until answered or abandoned.
5. Internal Busy or No Answer will only operate when the internal call encounters a busy state or no answer, respectively. External Busy or External No Answer will only operate when the external call encounters a busy state or no answer, respectively.
6. Preset call forward status is not shown in the station's LCD display.
7. A station in a Station Hunt Group (Circular or Terminal) can be assigned to receive incoming hunt calls, overriding any Call Forward. That is, either the system recognizes the Forward condition and bypasses hunt calls around the station or routes hunt calls to the station based on the system database.
8. No Answer forward will employ the Station No Answer Forward Timer unless it is set to zero in which case the System No Answer Timer is used.

Programming

VOICE CONFIG	Station Data – Preset Call Forward System Data – Call Feature Timer – Call Forward No Answer Timer

Related Features

Call Forward
Off-Hook Signaling
VSF Integrated Auto Attd/Voice Mail
DND (Do Not Disturb)

Hardware

2.4 CALL PARK

Description

A user may place an active CO call in a special holding location (Call Park/Park Orbit) for easy access from any station in the system (default=601-610).

Operation

LIP Phone

To park an active external call:

1. Press the **[TRANS]** button.
2. Dial the Call Park/Park Orbit code (601-610).

3. Return to idle.

To retrieve a parked call:
1. Lift the handset or press the **[SPEAKER]** button,
2. Dial the Call Park/Park Orbit code (601-610).

SLT
To park an active external call:
1. Momentarily press the hook-switch.
2. Dial the Call Park/Park Orbit code (601-610).
3. Return to idle.

To retrieve a parked call:
1. Lift the handset.
2. Dial the Call Park/Park Orbit code (601-610).

Conditions

1. If the selected Call Park/Park Orbit returns a busy signal, the user may dial another Call Park/Park Orbit without the need to disconnect.
2. Intercom calls cannot be placed in a Call Park/Park Orbit location.
3. A Parked call will recall to the station that parked the call should the Call Park Timer expire; the normal Hold Recall process is then initiated.
4. A Parked call will indicate busy at all appearances.

Programming

VOICE CONFIG System Data – Call Feature Timer – Call Park Recall Timer

Related Features

Hold Recall
Attendant Recall

Hardware

2.5 CALL PICK-UP

2.5.1 Directed Call Pick-Up

Description

A station may answer incoming and transferred intercom, CO and IP calls ringing at another station (Call Pick-Up). All ringing calls are subject to Directed Call Pick-up except Queue Callbacks.

LIP phone users may assign a Flex button as a **{DIRECTED CALL PICK-UP}** button.

Operation

LIP Phone

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

1. Lift the handset or press **[SPEAKER]**.
2. Dial **[PGM]** + **{FLEX}** + ‘7’ + **[SAVE].**

To Pick-up a call ringing at another station:

1. Lift the handset or press **[SPEAKER]**.
2. Dial 7 (Directed Call Pick-up code).
3. Dial the ringing station’s intercom number.

OR

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

SLT

To Pick-up a call ringing at another station:

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

Conditions

1. To pick-up a CO call, the station must have an idle appearance button available.
2. When several calls are ringing at a station simultaneously, Call Pick-up will connect the first-in, highest priority call. Call priority order is: CO transferred call > CO hold-recalled call > CO incoming call > queued call.
3. Queue callback and Private Line calls are not subject to Call Pick-up; any attempts receive an error tone.
4. Only ringing intercom calls are subject to Call Pick-up; handsfree announced calls cannot be picked up by another station.

Programming

Related Features

Intercom Answer Mode
Ringing Line Preference
Group Call Pick-Up

Hardware

2.5.2 Group Call Pick-Up

Description

A station can answer (Call Pick-Up) incoming and transferred intercom, CO and IP calls ringing at another station. All ringing calls, except Private Line and Queue Callbacks, are subject to Pick-up by other stations in the same group.
LIP phone users may assign a Flex button as a **{GROUP CALL PICK-UP}** button.

Operation

LIP Phone

To assign a **{GROUP CALL PICK-UP}** button:

1. Lift the handset, press **[PGM]** + **{FLEX}** + '**' + **[SAVE].**

To Pick-up a call ringing at another station:

1. Lift the handset or press **[SPEAKER]**.
2. Dial '** (Group Call Pick-up code).

OR, 2. Press the **{GROUP CALL PICK-UP}** button.

SLT

To Pick-up a call ringing at another station:

1. Lift the handset.
2. Dial '** (Group Call Pick-up code).

Conditions

1. To pick-up a CO call, the station must have an idle appearance button available.
2. When several calls are ringing simultaneously, Call Pick-up will connect the first-in, highest priority call. Call priority order is: CO transferred call > CO hold-recalled call > CO incoming call > queued call.
3. Queue callback calls are not subject to Call Pick-up; any attempt will receive an error tone.
4. Only ringing intercom calls are subject to Call Pick-up; handsfree announced calls can not be picked up by another station
5. When a station belongs to multiple groups, calls to the station group with lowest sequential group number are answered first. For example, if an incoming call is ringing at both Station Groups 620 and 621; when Station 100 (member to both Station Groups) responds to the ringing and picks up, the call to Station Group 620 will be answered first (by default).

Programming

VOICE CONFIG Station Group Data

Related Features

Intercom Answer Mode
Directed Call Pick-Up

Station Groups

Hardware

2.6 CALL TRANSFER

2.6.1 Call Transfer, Station

Description

CO calls can be transferred to other stations in the Smart Business gateway (iPECS SBG-1000) system. Calls can be transferred announcing the call (screened) or without an announcement (unscreened).

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

Users can transfer an active Intercom call to other stations in the iPECS SBG-1000 system, using either screened or unscreened transfer. When used, the Intercom station is placed on Exclusive Hold, and the Transfer Recall timer is initiated. if the timer expires before the Intercom call is answered, the call will bounce back (recall) to the transferring station until answered or abandoned.

Operation

LIP Phone

While on a CO call, to perform a Screened Call Transfer:

1. Press **[TRANS]**.
2. Dial the station to receive the transfer.
3. At answer or splash tone announce the call.
4. Hang-up to complete the transfer.

OR

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

While on a CO call, to perform an Unscreened Call Transfer:

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

OR

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

To perform an Screened Transfer while on an ICM call:

1. Press **[TRANS]** button.
2. Dial Station to receive call.
3. At answer or Splash tone, announce call.

4. Hang-up to return to idle.

OR

1. Press **{DSS/BLF}** button for the desired station.
2. At answer or Splash tone, announce call.
3. Hang-up to return to idle.

To perform an Unscreened Transfer while on an ICM call:

1. Press **[TRANS]** button.
2. Dial Station to receive call.
3. Hang-up to return to idle.

OR

1. Press **{DSS/BLF}** button for the desired station.
2. Hang-up to return to idle.

SLT

While on a CO call, to perform a Screened Call Transfer:

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

While on a CO call, to perform an Unscreened Call Transfer:

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

To perform a Screened transfer of an active Intercom call:

1. Momentarily depress the Hook-switch.
2. Dial Station to receive call.
3. At answer or Splash tone, announce call.
4. Hang-up to return to idle.

While on an Intercom call, Unscreened call transfer:

1. Momentarily depress the Hook-switch.
2. Dial Station to receive call.
3. Hang-up to return to idle.

Conditions

1. The transferring station may camp a call at a busy station (refer to Camp-On).
2. The LED of a **{LOOP}** button will display the status of a call until the station no longer has call supervision (ex., the call is successfully transferred).
3. To prevent Toll abuse, CO lines without an active call (either incoming or dialed digits on outgoing) cannot be transferred.
4. For outgoing CO Line calls, the system will monitor the CO Line for dial-tone to prevent Toll abuse; when an IP Line is seized, the system does not monitor for dial-tone.

5. While on intercom call transfer, the **[ICM]** button provides an appearance for the transferred station; LED indicates status and pressing the button connects to the station.
6. A station in DND or out-of-service can not receive a transfer; any attempt will result in an error tone.

Programming

VOICE CONFIG System Data – Call Feature Timer – Transfer Recall Timer

Related Features

Hold Recall
Call Transfer,
Call Waiting/Camp-On
Station Flexible Buttons
DND (Do Not Disturb)

Hardware

2.6.2 Call Transfer, CO

Description

A station may be permitted to transfer a CO call to another CO line, establishing an Unsupervised Conference between the two external parties.

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

Operation

LIP Phone

While on a CO call, to perform a Screened Call Transfer:

1. Press **[TRANS]**.
2. Place CO call to forward party.
3. At answer, announce the call.
4. Hang-up to complete the transfer.

While on a CO call, to perform an Unscreened Call Transfer:

1. Press **[TRANS]**.
2. Place CO call to forward party.
3. Hang-up to complete the transfer.

SLT

While on a CO call, to perform a Screened Call Transfer:

1. Momentarily depress the hook-switch.
2. Place CO call to forward party.

3. At answer, announce the call.
4. Hang-up to complete the transfer.

While on a CO call, to perform an Unscreened Call Transfer:
1. Momentarily depress the hook-switch.
2. Place CO call to forward party.
3. Hang-up to complete the transfer.

Conditions

1. For this feature, at least one of the two CO lines (transferred or receiving) must provide detection of disconnect supervision and lost loop condition.
2. If during transfer to an external party, the user presses the CO line of the original call, the outgoing call is disconnected and the original call is connected to the user.

Programming

VOICE CONFIG	Station Data – Authorization Code & COS – Offnet FWD System Data – Call Feature Timer – Transfer Recall Timer

Related Features

Hold Recall
Call Transfer, Station

Hardware

2.6.3 Call Transfer, Voice Mail

Description

CO calls can be directly transferred to a station's VSF voice mail-box.

Operation

LIP Phone

While on a CO call, to perform a Call Transfer:
1. Press **[TRANS]**.
2. Press **[MSG/CALLBK]** button.
3. Dial the number or press the **{DSS/BLF}** button for the desired station.
4. Hang-up to complete the transfer.

Conditions

1. The LED of a **{LOOP}** button will display the status of a call until the station no longer has call supervision (ex., the call is successfully transferred).

Programming

VOICE CONFIG	Station Data – Preset Call Forward System Data – Call Feature Timer – Transfer Recall Timer

Related Features

Hold Recall
Call Waiting/Camp-On
VSF Voice Mail

Hardware

LIP Phone

2.7 CALL WAITING/CAMP-ON

Description

Call Waiting is used to notify a busy station that a call is waiting. The busy station is notified of the waiting call with a Camp-On tone. For users of an LIP Phone, the LED of the **[HOLD]** button will flash.

After receiving a busy signal, the calling station camps on to the called station. The called station can respond by:

- answering the waiting call, which places the active call on hold,
- activating One-Time DND, or
- ignoring the Camp-On tone.

Operation

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

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

Conditions

1. The user may only Camp-On to a station in busy mode; a user may not Camp-On to a station in DND, a conference, receiving a Page, etc.
2. The Camp-On procedure can be employed by an Attendant to activate DND Override.
3. If the calling station disconnects from the call after activating Camp-On, Camp-On is cancelled.
4. A Camp-On tone is sent each time the calling user presses the '*' button.

Programming

Related Features

DND (Do Not Disturb)

Intercom Call (ICM Call)
Voice Over

Hardware

2.8 CO ACCESS

Description

Stations can access outgoing CO lines based on CO Group Access programming. LIP Phones may use flexible buttons assigned to access a specific **{CO}** line button for outgoing calls or a **{LOOP}** button.

Individual users may be allowed to assign CO access flexible buttons.

Operation

LIP Phone

To place an outgoing CO call:

1. Lift the handset or press the **[SPEAKER]** button.
2. Press desired **{CO}** line, **{LOOP}** button or dial the CO line or Group access code.

To answer an incoming CO call:

1. Lift the handset or press the **[SPEAKER]** button.

OR

1. Press flashing **{CO}** line, **{LOOP}** button and lift the handset to speak privately

SLT

To place an outgoing CO call:

1. Lift handset.
2. Dial the CO line or Group access code.

To answer an incoming CO call:

1. Lift handset.

Conditions

1. When a user dials Access Random CO Line code, the system will search for an idle CO line; the system may continue to search through all CO lines for an available line.
2. A telephone user not allowed access to a CO line will receive an error tone when access is attempted. The station may receive transferred calls on such denied access lines but will not be able to flash or use the CO line for an outgoing call.
3. A station denied access to a CO line but assigned to ring for the CO line will receive ring; the user may transfer the call using a flashing LED **{CO}** line button but cannot make an outgoing call on the CO line.
4. CO lines placed on hold may be retrieved by dialing the 8# (retrieve held CO code) and the CO line number.

5. The Tx path to a station will be muted until the system has verified the Toll Restriction for the CO line.

Programming

VOICE CONFIG	Station Data – Common Attributes – CO Group Access
	CO Line Data – Call Routing by Line
	CO Line Data – Call Routing by Caller Number
	CO Line Data – Ring Assignment Table

Related Features

CO Ring Assignment

Hardware

2.9 CO QUEUING

Description

When CO lines are busy, permitted users can request to be placed in queue awaiting the CO line, or a CO line in the same group to become available. When an appropriate CO line becomes available, the system calls the waiting station on a first-in, first-out basis.

Operation

LIP Phone

To request to be placed in queue for a busy CO line:

1. Press busy **{CO}** or **{CO GRP}** button.
2. Press the **[MSG/CALLBK]** button; confirmation tone is received.
3. Hang-up, the **[MSG/CALLBK]** LED flashes.

To cancel the queue from the queued station:

1. Press the **[MSG/CALLBK]** button, the **[MSG/CALLBK]** LED extinguishes.

SLT

To request to be placed in queue while receiving an All Lines Busy signal:

1. Momentarily press the hook-switch.
2. Dial 56 (Callback feature code).

To cancel the queue from the queued station:

1. Lift the handset.
2. Dial 56 (Callback feature code).

System

When a CO line becomes available:

1. The System will send a distinctive Queue recall to the station that was first-in queue, the appropriate **{CO}** LED button will flash
2. The CO line and station will appear busy to all other users.

Conditions

1. A CO line can have any number of simultaneous queue requests.
2. A station may only have a single active CO queue request; activating a new queue request will replace, and cancel an existing queue.
3. A Queue recall will always signal the station with a tone ring, ignoring the station's assigned Intercom Signaling mode.
4. Queue recall will bypass a busy station, and place the station at the bottom of the queue list.
5. Queue recall will signal a station for 15 seconds, after which, the station is removed from the queue (the queue is cancelled).

Programming

Related Features

CO Access

Hardware

2.10 THREE-PARTY VOICE CONFERENCE

Description

The system will allow three internal and external parties to be connected on a call, conference. An unlimited number of 3-party conferences may be established.

Operation

LIP Phone

To establish an ad-hoc conference:

1. Establish first call.
2. Press the **[CONF]** button; the LED will light, and the connected party is placed on exclusive hold (the user receives dial-tone).
3. Place second call.
4. When connected, press **[CONF]**; the new call is placed on exclusive hold.
5. Press **[CONF]** button to establish 3-party conference.

To place a conference on hold:

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

To retrieve held conference:

1. Lift the handset
2. Press **[CONF]** button; all parties will be reconnected.

Conditions

1. The **[CONF]** button remains illuminated at the initiators phone for the duration of the conference.
2. There is no limit on the number of 3-way conferences the system will support.
3. If the system receives a disconnect signal and no internal parties remain in the conference, the conference is terminated and all parties are disconnected. If an internal party is still connected when a disconnect signal is received, the connection to remaining parties is maintained.
4. The normal Hold Recall process is applied to a conference on hold using the Unsupervised Conference recall Timer for recall timing.
5. If while setting up a conference, system error tone is received, the initiator must press the **[CONF]** button to obtain the Intercom dial-tone.
6. A station that is busy, in DND or other non-idle state, cannot be added to a conference.
7. SLT can join a conference call, but can not make a conference call

Programming

Related Features

Automatic Speaker Select
Hold Recall

Hardware

2.11 CUSTOMER SITE NAME

Description

A Customer Name, up to 24 characters, may be entered into the system database. The name is displayed on the SMDR and database outputs as well as during an Admin session.

Operation

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

Conditions

Programming

VOICE INSTALL	System – Identification – Site Name
SYSTEM ID	Customer Site Name (PGM 100-Btn 2)

Related Features

Hardware

2.12 FAX

Description

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

Operation

System

System tones are automatically blocked when FAX utilization is set to "ON".

Conditions

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

Programming

VOICE INSTALL FAX – FAX Configuration – FAX Utilization

Related Features

Call Waiting/Camp-On

Hardware

2.13 DELAYED CO RING

Description

Ring signals for an incoming CO call can be sent to stations immediately upon detection or after an assigned ring cycle delay. The delay can be up to 9 system ring cycles, thus allowing other stations to answer the call.

Operation

System

Delay Ring operation is automatic when assigned:

Conditions

1. Delay Ring can be assigned for a station.
2. If no delay is entered when programming Ring assignments, the station will immediately ring when a call is received.
3. Private Lines may be assigned with delayed ring.

Programming

VOICE CONFIG	CO Line Data – Call Routing by Line CO Line Data – Ring Assignment Table

Related Features

CO Ring Assignment

Hardware

2.14 DELAYED AUTO ATTENDANT

Description

An incoming CO call can be routed to the VSF Auto Attendant either immediately upon detection or after a delay of up to 30 seconds. This allows other stations assigned immediate ring the opportunity to answer before the call is routed to the Auto Attendant.

Operation

System

Operation of this feature is automatic when assigned.

Conditions

1. When Delayed Auto Attendant Ring is assigned, after the delay, the call will no longer ring assigned stations and will only ring to the VSF Auto Attendant.
2. If no delay is entered, the call immediately will ring to the VSF Auto Attendant.
3. To assign Delayed Attendant ring, at least one station or Station Group must be assigned for immediate ring.
4. Ring is assigned to a VSF Auto Attendant announcement (01-70) as a “station type” with a delay from 00 to 30 seconds.

Programming

VOICE CONFIG	CO Line Data – Call Routing by Line CO Line Data – Ring Assignment Table

Related Features

CO Ring Assignment

Hardware

2.15 DIAGNOSTIC/MAINTENANCE

Description

The system software incorporates various diagnostic and maintenance routines that may be "called" remotely or locally through the systems RS-232 serial ports, a TCP/IP connection using a Web browser or telnet terminal established over IP networks. Routines that can be accessed include trace functions at the device level, commands for diagnostics and maintenance, and tools for manipulation at the OS level.

Operation

Conditions

Programming

Related Features

Hardware

2.16 DIAL-BY-NAME

Description

A name, up to 16 characters, may be assigned to each Individual and Common Speed Dial. In addition, each station may be assigned a 12-character name. When assigned, a user may place an intercom call to another station or select a Station or Common Speed Dial using the name.

The user selects from one of three Dial-by-Name directories and enters characters employing 2 dial pad buttons for each character (refer to Character Entry Chart Table). 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▲]/[VOL▼]** button and select a name to call. The number associated with a selected name may be displayed by using the **[TRANS]** button.

Operation

LIP Phone

To use Dial by Name:

1. Press soft key **[DIR]**.
2. Dial the desired directory.
3. Search the directory using the **[VOL▲]/[VOL▼]** button or by entering the number .

Table 2.16.1 Character Entry Chart

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

4. Press the **[SAVE]** button to place the call.

To toggle between the name and number display:
1. Press the **[NAME/TEL]** soft button or **[TRANS/PGM]** button.

To program the station user name:
1. Press the **[PGM]** button.
2. Dial 34 (User Name Program code).
3. Dial the name, up to 12 characters.
4. Press **[SAVE]**.

Attendant
To program a name for another station:
1. Press the **[PGM]** button.
2. Dial 031 (Attendant User Name Program code).
3. Dial the station number.
4. Dial the name, up to 12 characters.
5. Press **[SAVE]**.

Conditions

1. Available characters are A to Z, space and period.
2. The LCD will display multiple names (one per LCD line, up to 16 characters).
3. If a user selects a directory with no entries or there is no match to the user entry, error tone is provided.

4. Dial-by-Name is only available to LIP Phones with a display; other users will receive error tone if an attempt is made to access Dial-by-Name.
5. A user may both scroll and enter characters to search a directory using the **[VOL▲]/[VOL▼]** buttons.

Programming

Related Features

Individual Speed Dial
Common Speed Dial

Hardware

LIP Phone w/Display

2.17 DND (DO NOT DISTURB)

Description

A station can be placed in DND to block incoming CO and Intercom calls, transfers and paging announcements.

Operation

LIP Phone

To activate DND:

1. Press the **[DND]** button; the **[DND]** button LED illuminates.

To remove DND:

1. Press the **[DND]** button; the **[DND]** button LED extinguishes.

SLT

To activate DND:

1. Dial 53 (DND feature code); confirmation tone is received.

To remove DND:

1. Dial 53 (DND feature code), confirmation tone is received.

Conditions

1. Pressing the **[DND]** button while ringing will activate One-Time DND.
2. An Attendant may cancel DND for other stations.
3. DND service is not available to Attendants.
4. Recalls for CO calls will override the DND feature.
5. A station in DND is out-of-service for all incoming calls including Station Group calls.

6. A station in DND is bypassed by calls forwarded to the station; if the last station in a Call Forward chain is in DND, the call will ring to the previous station in the chain.
7. When calling a station in DND, the LIP Phone display will indicate the DND status.

Programming

Related Features

Feature Cancel
Call Forward
Station Groups

Hardware

2.18 EMERGENCY CALL

Description

Regardless of a station's CO accessibility, the user may dial assigned Emergency numbers.

Operation

System

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

Conditions

1. If there is no idle CO line for emergency call, one of busy CO line will be forced to release the current call and used for emergency call.
2. Emergency call can be placed without any CO access code, except Australia. But if emergency call is same as station number or starts with station number, then emergency call cannot be used without CO access code. For example, if emergency code is "112" and station number "112" is used, user should dial CO access code first in order to make emergency call. Otherwise station "112" receives a ring.

Programming

VOICE CONFIG	System Data – Emergency Dialing

Related Features

Hardware

2.19 FLEXIBLE NUMBERING PLAN

Description

User access to the iPECS SBG-1000 system resources and features is accomplished through feature codes or LIP Phone buttons. The Administrator, if desired, assigns codes for individual functions in the Flexible Numbering Plan. The feature codes are defined in the system's Flexible Numbering Plan.

Operation

System

System implements feature activation based on the Flexible Numbering Plan.

Conditions

1. Feature codes can be 1-3 digits in length.
2. During programming, conflicts in the Numbering Plan are not allowed; the existing non-conflicting Numbering Plan is used until correctly updated.

Programming

VOICE INSTALL	Numbering Plan

Related Features

Hardware

2.20 HEADSET COMPATIBILITY

Description

An industry standard headset can be connected to an LIP Phone in place of or in addition to the handset. The station 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

LIP Phone

To change operation from Speakerphone to Headset:

1. Press the **[PGM]** button.
2. Dial 12 (Headset select code).
3. Dial '0' to select Headset, '1' to select Speakerphone.

OR,

1. Press **[HEADSET]** button.

2. Dial ‘0’ to select Headset, ‘1’ to select Speakerphone.

To change the device to receive ring signals:
1. Press the **[PGM]** button.
2. Dial 13 (Ring select code).
3. Dial ‘1’ for Speaker, ‘2’ for Headset or ‘3’ for both.

To place/answer calls using the headset:
1. Press the **[SPEAKER]** with the phone in Headset mode.

Conditions

1. The Intercom Signaling Mode can be set in the Headset mode as with the Speakerphone mode.
2. The station always receives Page announcements over the speaker of the LIP Phone.
3. Although the phone is in the headset mode, the system will monitor the hook-switch status. If a user lifts the handset to go off-hook, audio is delivered to the handset.

Programming

VOICE CONFIG	Station Data – Common Attributes – Headset Ring Station Data – Common Attributes – Headset or Speaker Mode

Related Features

Speakerphone
Paging

Hardware

2.21 HOLD

2.21.1 Hold

Description

CO lines may be placed in a waiting state such that other stations in the system are able to access the CO line. Stations must have System database access to the CO line to access the held call.

If the call remains on hold at expiration of the System Hold Recall Timer, normal Hold Recall will be activated.

Operation

LIP Phone

To place a call on System Hold:

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

To access a call from System Hold:
1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **{CO}** line button.

OR
1. Lift the handset or press the **[SPEAKER]** button.
2. Dial 8# (Held CO Call Access code).
3. Dial the CO line number

SLT

To place a call on System Hold:
1. Momentarily press the hook-switch.
2. Dial 67 (System Hold feature code).

To access a call from System Hold:
1. Lift the handset.
2. Dial 8# (Held CO Call Access code).
3. Dial the CO line number.

Conditions

1. When a CO line is placed on System Hold, the button LED will flash at 30 ipm (it will wink at the holding station and will flash at all other stations).
2. A call on System Hold can be retrieved from any station allowed access to the CO line in the system database using the CO line button or the Held CO call access code.
3. The LED of **{LOOP}** buttons will display the CO line status.

Programming

VOICE CONFIG	System Data – Call Feature Timer – Attendant Recall Timer System Data – Call Feature Timer – System Hold Recall Timer

Related Features

Call Transfer,
Hold Recall

Hardware

2.21.2 Hold Recall

Description

When a user places a CO call on hold, a hold timer is activated. If the timer expires, the held call will recall at the station for the I-Hold Recall time. If the call remains unanswered, the call is placed on System Hold and the Attendant also receives a recall for the Attendant Recall time. If still unanswered after the Attendant Recall time, the CO call is disconnected and the appropriate circuits are returned to idle.

Operation

Hold Recall operation is automatic.

Conditions

1. Separate timers are assigned for the various types of hold: System, Transfer, etc.
2. If the I-Hold timer is set to zero, the station will not receive a recall; if the Attendant Recall timer is set to zero, the also Attendant will not receive a recall.
3. If the specific Hold timer is set to zero, recall is disabled.

Programming

VOICE CONFIG	System Data – Call Feature Timer – Attendant Recall Timer System Data – Call Feature Timer – System Hold Recall Timer System Data – Call Feature Timer – Transfer Recall Timer

Related Features

Call Transfer,
Hold

Hardware

2.21.3 Automatic Hold

Description

While on an active CO call, the system will place the call on hold automatically if the user presses the **[FLASH]**, **[CONF]**, **{DSS/BLF}** or other feature buttons. In addition, the station can be programmed to support CO to CO Automatic Hold. In this case, pressing a CO button while on a CO call will place the active call on hold and access the selected CO line.

Operation

LIP Phone

To use Automatic Hold while on an active CO call:

1. Press the desired feature button or **{CO}**; the active call is placed on Hold.

Conditions

1. There is no limit on the number of calls that can be placed on Hold using Automatic Hold.

Programming

Related Features

Hold Recall

Hardware

LIP Phone

2.22 CALL ROUTING BY CALLER NUMBER

Description

The system can employ caller number to determine the routing of incoming external calls. Each CO Line may be assigned to employ call routing by caller number. The system will compare the received caller number to entries in the Call Routing by Caller Number Table, and if a match is found, will route the call to the destination defined in the Ring Assignment Table. Destinations can be the VSF, a station or a station group.

Operation

System

System implements routing automatically based on database entries and the received caller number.

Conditions

1. For analog CO Lines, the system will await receipt of valid ICLID for the ICLID Ring Timer. At expiration of the timer, if ICLID is not received, the call is routed based on the type and other programming (Ring assignments, etc.).
2. If the received caller number does not match an entry in the Call Routing by Caller Number Table, the call is routed based on the type and other programming (Ring assignments, etc.) for CO Line.
3. The caller number received from the CO Line may be a telephone number or name that must match an Call Routing by Caller Number Table entry.

Programming

VOICE CONFIG	CO Line Data – Call Routing by Line
	CO Line Data – Call Routing by Caller Number
	CO Line Data – Ring Assignment Table

Related Features

CO Ring Assignment

Hardware

2.23 IP FAX RELAY, T.38 SUPPORT

Because of their nature, Fax tones do not transmit well through IP networks, particularly when compression is employed. To address this, iPECS SBG-1000 supports the T.38 protocol that defines the translation of fax tones to digital signals. When Fax tone is detected on a port of an iPECS SBG-1000, the system will activate a T.38 Fax relay channel to the appropriate Line or SLT module.

Operation

Operation of this feature is automatic.

Conditions

Programming

VOICE INSTALL FAX – FAX – T.38

Related Features

Hardware

2.24 LNR (LAST NUMBER REDIAL)

Description

The last number dialed is stored (up to 23 digits) in the station's Last Number Redial buffer. The user may request the system redial the last dialed number without the need to dial the number.

For LIP Phones with displays, the last 15 numbers are stored in the LNR buffer. The user may view the numbers using the **[VOL▲]/[VOL▼]** button and select the number to dial from the list.

Operation

LIP Phone

To assign a Flex button as an {redial} button:

1. Press **[PGM]** + **{FLEX}** + **[PGM]** + '54' + **[SAVE]**

To use Last Number Redial:

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **{REDIAL}** button.
3. Press the **[VOL▲]/[VOL▼]** button to highlight the desired number.
4. Press **[SAVE]** or **{REDIAL}** to dial the number highlighted.

SLT

To use Last Number Redial:

1. Lift the handset.

2. Dial ‘52’, the Last Number Redial code.

Conditions

1. For LIP Phones with display, the redial buffer will store duplicate numbers unless dialed consecutively.
2. When the CO line used for the original call is busy, the system will select an idle line from the same CO line Group to place the call.
3. Using Last Number Redial will cancel Automatic Called Number Redial if active.
4. The LNR buffer is not stored in non-volatile memory and will be erased if power to the system is lost.
5. Manually dialing a Flash during an outgoing call will cause only those digits dialed after the Flash to be stored in the LNR buffer.

Programming

Related Features

Save Number Redial (SNR)
Individual Speed Dial
Common Speed Dial

Hardware

2.25 MOH (MUSIC-ON-HOLD)

Description

When a call is placed on hold, the system will deliver audio from the defined MOH source. In this way, the connected user can determine that the connection is still active.

A message or music recorded in the VSF can be employed as MOH. The Attendant records the VSF announcement for MOH and VSF MOH is assigned as the MOH source.

Operation

System
Operation of MOH is automatic.

Attendant
To record a VSF announcement for MOH:
1. Press the **[PGM]** button.
2. Dial 05 (VSF Record code).
3. Dial 071 (VSF MOH Announcement number).
4. The current announcement is played followed by the “Press # to record” prompt.
5. Dial ‘#’.
6. After the record prompt and beep-tone, record message.
7. Press the **[SAVE]** button to stop recording and save the message.

Conditions

1. Only VSF announcement number 71 may be used for the MOH message.

Programming

VOICE CONFIG	Station Data – Station Hold Music CO Line Data – CO Hold Music

Related Features

Hold

Hardware

2.26 REGISTRATION & REGISTRATION TABLE

Description

To eliminate the potential for unintended device registration, the system can be programmed to allow local device registration employing MAC addresses. Using the defined MAC address registration, the system allows devices with matching MAC addresses to register.

The Registration Table permits entry of up to 12 MAC addresses to be registered for the device; entering the MAC address permits the device to register with the system. If the device which has the matching MAC address is successfully registered, the MAC address is removed from the Registration Table.

Operation

Operation of registration is automatic based on the system database.

Conditions

Programming

VOICE INSTALL	Station Registration – Registration Table

Related Features

Hardware

2.27 RINGING LINE PREFERENCE

Description

A station is automatically connected to incoming calls by lifting the handset or pressing the **[SPEAKER]** button when assigned Ringing Line Preference (RLP).

Operation

LIP Phone

To answer a call while the station is ringing:

1. Lift the handset or press the **[SPEAKER]** button.

Conditions

1. When multiple calls are ringing at the station, a priority defines the order in which calls are answered. The priority is:
   Transfer > recalls > incoming calls > queued calls
2. Intercom calls are always given the lowest answering priority.
3. SLTs operate only in the RLP mode; when ringing, lifting the handset connects the SLT to the ringing call.

Programming

Related Features

Automatic Speaker Select

Hardware

2.28 SPEED DIAL

2.28.1 Display Security

Description

Individual and Common Speed Dial numbers may be programmed so that the digits are not displayed on the LCD of LIP phones.

Operation

To assign Display Security to a Speed Dial number:

1. Dial "*" as the first digit of the Speed Dial number.

Conditions

1. The number is displayed when programming a Speed Dial number.
2. Display Security does not affect the SMDR output.
3. Display Security is provided on all CO calls including calls that are transferred or recall.

Programming

VOICE CONFIG	Station Data – Individual Speed Dial CO Line Data – Common Speed Dial

Related Features

Speed Dial

Hardware

2.28.2 Individual Speed Dial

Description

Each user can store commonly dialed numbers for easy access using Individual Speed Dial bins. Each station has access to 20 Speed Dial numbers. Each Speed Dial number can be up to 23 characters in length and may include special instruction codes.

Special instruction codes available are:

'*' as 1st digit	Activate Display Security, do not display number.

LIP Phone users may assign a Flex button for One-Touch access to a specific Speed Dial bin. In addition, the LIP Phone user may assign a Telephone number directly to a Flex button. In this case, the telephone number is allocated to the highest numbered available Individual Speed Dial bin.

Operation

LIP Phone

To assign a Flex button as an {individual speed dial} button:

1. Press **[PGM]** + **{FLEX}** + **[SPD/DEL]** + Individual Speed Dial bin number + **[SAVE]**

To dial using an Individual Speed Dial:

1. Lift handset or press the **[SPEAKER]** button.
2. Press the **[DIR]** soft button.
3. Press the **[SPEED]** soft button.
4. Dial the desired bin number (00–19).

To program an Individual Speed Dial number:

1. Press the **[DIR]** soft button.
2. Press the **[SPEED]** soft button.
3. Press the **[ADD]** soft button.
4. Dial the Speed Dial bin number (00-19).
5. Dial the number to be stored.
6. Press the **[SAVE]** button.

7. If desired, enter a name.
8. Press the **[SAVE]** button.

SLT

To dial using an Individual Speed Dial:

1. Lift handset.
2. Dial 58 (SLT Speed Dial access code).
3. Dial the desired bin number ('00' – '19').

To program an Individual Speed Dial number:

1. Dial 55 (SLT Speed Programming code).
2. Dial the Speed Dial bin number (00-19).
3. Dial the number to be stored.
4. Momentarily press the hook-switch.
5. If desired, enter a name (refer to Character Entry Chart Table).
6. Momentarily press the hook-switch.

Alpha-numeric characters may be entered to name the Speed Dial number using the chart below.

Table 2.72-1 Character Entry Chart

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

Conditions

1. Accessing an empty Speed Dial bin will return an error tone.
2. All Speed Dial numbers stored in memory are protected from power loss.
3. A name can be entered for a Speed Dial number to permit access from the Dial-by-Name directory.
4. Stored speed dial number should not include CO access code.

Programming

VOICE CONFIG Station Data – Individual Speed Dial

Related Features

Dial-by-Name
Display Security
LNR (Last Number Redial)
Save Number Redial (SNR)
Common Speed Dial
Flex Button Direct Speed Dial Assignment

Hardware

2.28.3 Common Speed Dial

Description

Commonly dialed numbers can be stored by the Attendant or by the Administrator in Web Admin for easy access by stations. Up to 800 Common Speed Dial numbers are available. Each Speed Dial number can be up to 23 characters in length and may include special instruction codes.

Special instruction codes available are:

‘*’ as 1st digit Activate Display Security.

LIP Phone users may assign a Flex button for One-Touch access to a specific Common Speed Dial bin.

Operation

LIP Phone

To assign a Flex button as a {common speed dial} button:

[PGM] + **{FLEX}** + **[SPD/DEL]** + Common Speed Dial bin number + **[SAVE]**

To dial using a Common Speed Dial:

1. Lift handset or press the **[SPEAKER]** button.
2. Press the **[DIR]** soft button.
3. Press the **[SPEED]** soft button.
4. Dial the desired bin number (‘200’-‘999’).

SLT

To dial using a Common Speed Dial:

1. Lift handset.
2. Dial ‘58’, the SLT Speed Dial access code.
3. Dial the desired bin number (‘200’-‘999’).

Attendant

To program a Common Speed Dial number:

1. Press the **[DIR]** soft button.
2. Press the **[SPEED]** soft button.
3. Press the **[ADD]** soft button.
4. Dial the Speed Dial bin number ('200'-'999').
5. Dial the number to be stored.
6. Press the **[SAVE]** button.
7. If desired, enter a name, see Alpha-numeric entry chart under Individual Speed Dial.
8. Press the **[SAVE]** button.

Conditions

1. Accessing an empty Speed Dial bin will return error tone.
2. All Speed Dial numbers are stored in memory protected from power loss.
3. A name can be entered for a Speed Dial number to permit access from the Dial-by-Name directory.
4. Stored speed dial number should not include CO access code.

Programming

VOICE CONFIG CO Line Data – Common Speed Dial

Related Features

Dial-by-Name
Display Security
LNR (Last Number Redial)
Save Number Redial (SNR)
Individual Speed Dial

Hardware

2.29 STATION GROUPS

Description

Stations can be grouped for incoming call routing and Call Pick-up purposes. Up to 12 Station Groups can be defined with up to 24 stations in a group. Seven types of groups can be defined:

- Circular
- Terminal
- Ring
- Pick-Up
- VSF-Voice Mail
- IPCR
- Net VM (Centralized External VM)

Circular Station Group

In Circular Hunt, calls to a station in the group will go to the station, if unavailable or unanswered in the hunt no answer time; the call will be directed to the next station defined in the group. The call will continue to hunt until each station in the group has been tried. The call remains at the last station or passes to a designated overflow station or group.

A Circular Station Group can be assigned with a pilot number (the Station Group Number) so that calls to the pilot number will hunt. In this case, the call will be directed to the first station in the group, and if needed, hunt through each station in the group until reaching the last station. The call may remain at the last station, passed to an overflow destination or sent to a voice mail-box.

Terminal Station Group

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

A Terminal Hunt Group can be assigned with a pilot number (the Station Group number) so that calls to the pilot number will hunt. In this case, the call will route as described for Circular Pilot Number hunting.

Station Ring Group

A call to any station in the Group will cause all stations in the group to ring and any station in the group may answer the call. If the call remains unanswered beyond the Overflow timer, the call is sent to the Overflow destination, which can be a Station, Station Group or Voice Mail-box.

Multiple calls can be received by a Station Ring Group and can be serviced in any order.

Pick-Up Station Group

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

VSF AA/VM Group

The VSF memory is employed by the integrated Smart Business gateway AA/VM application. Incoming calls can be directed to one of 70 user-recorded announcements, which may request further routing instructions from the user in the form of caller dialed digits. These digits are employed to route the caller as defined in the system CCR (Customer Controlled Routing) Tables.

The VSF AA/VM Group Voice Mail application receives calls forwarded or recalling from a station. Such calls will receive the user's pre-recorded greeting and may leave voice messages. The user may call the VSF AA/VM Group to review and manage the integrated Voice Mail application.

IPCR

This group is defined to support IP Call Recording service.

Net VM

This group is defined to support a Centralized Voice Mail system for a networked environment. At supported systems, the group is used to handle the AA/VM requirements from the central iPECS. The Net VM group may be an external VM system or the iPECS Feature Server.

Group Announcements
Station Group routing can be augmented with announcements recorded in the VSF AA/VM. Callers can be routed to one of several user-recorded announcements. The system answers the call and plays the defined 1st announcement to the caller. The announcement may provide the caller with routing options for Caller Controlled Routing. The 1st announcement may be “Guaranteed” meaning it will play in full before routing the call. A 2nd announcement can be provided to the caller when queue timers expire.

Operation

Conditions

1. Station Group calls are not routed to member stations that are in DND.
2. When a member of a Circular, Terminal, or Ring Group activates Call Forward, calls to the group may still route to the member based on the Member Forward option.
3. A call transferred to a Station Group will follow the routing for the group and will not initiate the Transfer Recall process.
4. Calls to a Station Group receive either a ring-back tone or MOH while queued to the group.
5. Calls which are not answered in the Overflow time, are routed to the defined Overflow destination, station, group, etc. If no Overflow destination is defined, the call is dropped after expiration of the Overflow timer.
6. One of the 70 VSF announcements may be assigned as the Overflow destination. These announcements allow for Caller Controlled Routing.
7. iPECS SBG-1000 has two default station groups. Group 631 is default Ring group which includes all stations in member list. Group 630 is default VSF-Voice Mail group.

Programming

VOICE CONFIG	Station Group Data – Station Group Assignment
	Station Group Data – Station Group Attributes

Related Features

Group Call Pick-Up
MOH (Music-On-Hold)
VSF Integrated Auto Attd/Voice Mail

Hardware

2.30 SMDR (STATION MESSAGE DETAIL RECORDING)

2.30.1 Call Cost Display

Description

Each SMDR call record includes a "Cost" field, which is a calculated estimate of the cost of the call. The call cost updates in real-time and displays on the LIP Phone LCD in place of the call duration.

The cost is determined by:

- Fixed charge per "Call Meter Pulse",
- ISDN Advice of Charge, or
- Estimated cost updated based on Elapsed Call Timer and assigned costing.

The technique selected to determine cost is based on the type of facility (analog CO, ISDN, or VoIP), services provided by the carrier and the system database.

Analog CO

Where "Call Metering Pulse" service is available from the carrier, the system will apply the "SMDR Cost per Unit Pulse" and the "SMDR Decimal" to the Call Metering received to estimate the call cost.

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

ISDN

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

VoIP

For VoIP calls, the system uses the call duration, cost/pulse and decimal values to establish the call cost estimate. The cost is updated periodically according to the "Elapsed Call Timer".

Operation

System

Call cost is estimated automatically and output to LIP Phone displays and the SMDR TCP port

Conditions

1. The call cost display begins after the "SMDR Start Timer" expires, if enabled, or at receipt of the first Call Meter Pulse.
2. Once connected to the system, the call duration includes the total time the call is connected including periods when the call is on hold, in queue, etc.
3. To enable Call Cost Display, the "SMDR Cost per Unit Pulse" and "SMDR Decimal" must be assigned; when not assigned, the call duration is provided by the system.

Programming

VOICE CONFIG	System Data – SMDR Attributes – Call Metering System Data – SMDR Attributes – SMDR Cost Per Metering Pulse System Data – SMDR Attributes – SMDR Decimal Position System Data – SMDR Attributes – Record Start Guaranteed Time

Related Features

SMDR (Station Message Detail Recording)
Lost Call Recording
Traffic Analysis

Hardware

2.30.2 SMDR Call Records

Description

SMDR (Station Message Detail Recording) provides detailed information on incoming and outgoing calls. Assignable options in the system database permit recording of all calls, all outgoing calls or toll calls and calls that exceed a fixed duration. Call records are output either upon completion of the call (real-time) or in response to a request from the Attendant.

The SMDR record output is as shown in the figure below. There are two flexible fields, Field I and Field II. Each Field is defined to show Ring duration, CLI (Caller Id) or CPN (Called Party Number).

```
STA  CO  TIME     START        DIAL/CLI/CPN NUM-1      COST    ACCOUNT CODE DIAL/CLI/CPN NUM II
SSSS BBB DD:DD:DD EE:EE FF/FF/FF HCCCCCCCCCCCCCCCCCCC sssssssss aaaaaaaaaaaa hcccccccccccccccccc
```

The various fields or items for a Call Record are:

- STA: 2~4 digit station number.
- CO: 3 digit CO Line number
- Time: Call duration in hours, minutes and seconds
- Start: Date and time call was placed/received
- NUM I: Flex Field I outgoing call dialed number & incoming call Ring duration, CLI or CPN
- Cost: Cost of Call
- Account Code:Account code entered for call (Not used in iPECS SBG-1000), MSN CLI
- NUM II: Flex Field II incoming call Ring duration, CLI or CPN

Operation

System

For real-time SMDR, records are output after completion of the call as shown in the figure above:

Attendant

To print SMDR records:

1. Press the **[PGM]** button.
2. Dial 0111 (SMDR print code).
3. Enter the desired station range.
4. Press the **[SAVE]** button.

To delete stored records:

1. Press the **[PGM]** button.
2. Dial 0112 (SMDR delete code).
3. Enter the desired station range.
4. Press the **[SAVE]** button.

To abort SMDR printing:

1. Press the **[PGM]** button.
2. Dial 0114 (SMDR abort code).
3. Press the **[SAVE]** button.

Conditions

1. For SMDR, if the first dialed digit(s) match the programmed LD code or the number of dialed digits exceeds the LD digit count, the call is considered an LD call. When behind a PBX, LD determination is made only if a PBX Trunk Access code is dialed as the first digit(s).
2. Except for DISA calls, the duration of ring for an incoming call is provided in the Dialed number field.
3. A header, including the assigned “Customer Site Id” is output after two blank lines and is repeated every 66th line.
4. The SMDR output is a simple ASCII stream of up to 80 characters per line.
5. When enabled, SMDR call record timing begins after the “SMDR Start Timer” expires and ends at call completion.
6. For incoming calls, the “NUM I” and “NUM II” fields will display the assigned data item – Ring Service time, CLI, or CPN. For outgoing calls, the NUM I field will always display the dialed number, user or system.
7. For outgoing calls which are starting with MSN CLI button, “Account Code” fields will display MSN CLI, if print MSN is configured to “ON”.

Programming

VOICE CONFIG	System Data – SMDR Attributes System Data – SMDR Attributes – SMDR Ring/CLI/CPN Service-I System Data – SMDR Attributes – SMDR Ring/CLI/CPN Service-II System Data – SMDR Attributes – Print MSN

Related Features

Call Cost Display
Lost Call Recording
Traffic Analysis

Hardware

2.30.3 Lost Call Recording

Description

Incoming calls where the caller hangs up before answer or while in a hold state are considered Abandoned or Lost calls. Special SMDR call records are provided for lost calls in real-time, as they occur, and a summary Lost Call count report is available on demand.

The real-time Lost Call records provide details on the called party, when and how long the call rang or was on hold before being abandoned, etc. Description of the record details is provided in the following charts. As noted in the charts, the dialed number field indicates the type of call and the ring or hold duration before the call was abandoned. The first character in the NUM I field is the status of the call when abandoned:

- R: normal ring to a station,
- G: ring to a station group and
- H: call placed in a hold state, including Transfer hold.

```
STA  CO  TIME     START         DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
EXT  001 00:00:00 14/05/02 15:45 R RING 01:35
```

- Incoming call on CO Line 1 received on May 14, 2002 at 3:45 pm, rang the assigned stations for 1 minute and 35 seconds.

```
STA  CO  TIME     START         DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
101  002 00:00:00 14/05/02 16:45 R RING 02:03
```

- Station 101 rang for an incoming call on CO Line 2 on May 14, 2002 at 4:45 pm, rang for 2 minutes and 3 seconds.

```
STA  CO  TIME     START         DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
101  001 00:00:00 15/05/02 09:35 R 100 RING 00:49
```

- Incoming call on CO Line 1 on May 15, 2002 at 9:35 am forward from station 101 to station 100 and rang for 49 seconds.

```
STA  CO  TIME     START         DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
104  002 00:00:00 16/05/02 11:06 G621 RING 01:32
```

- Incoming call on CO Line 2 on May 16, 2002 at 11:06 am routed to station 104 of Station Group 620 and rang for 1 minute and 49 seconds.

```
STA  CO   TIME     START         DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
621  001  00:00:00 16/05/02 14:03 G621 RING 00:39
```

- Incoming call on CO Line 1 on May 16, 2002 at 2:03 pm routed to Station Group 621 and rang for 39 seconds.

```
STA  CO   TIME     START         DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
100  002  00:03:32 16/05/02 15:30 H100 03:02
```

- Call on CO Line 2 on May 16, 2002 at 3:30 pm placed on hold by station 100 for 3 minutes and 2 seconds had total duration of 3 minutes and 32 seconds.

```
STA  CO   TIME     START         DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
129  001  00:00:45 18/05/02 08:40 H100 RING 00:33
```

- Call on CO Line 1 on May, 18, 2002 at 8:40 am was transferred by station 100 to station 129 was on hold for 33 seconds.

The output for the Lost Call summary count report is shown in the figure below:

```
Lost call count start time: 05/01/02 09:31
Current time 26/04/02 16:32
Total Lost call count until now: 121
```

Operation

Attendant

To print the summary Lost Call Count report:

1. Press the **[PGM]** button.
2. Dial 0115 (Lost Call Count report code).
3. Press the **[SAVE]** button.

To reset the Lost Call summary Count:

1. Press the **[PGM]** button.
2. Dial ‘0116’, the Lost call Count reset code.
3. Press the **[SAVE]** button.

Conditions

1. When the Lost Call Count is reset, the SMDR port will provide a “count reset” message.
2. Individual Lost Call records are only available real-time and not on-demand.
3. “Print Incoming Calls” and “Print Lost Calls” must be enabled in the SMDR Attributes for the system to output real-time Lost Call records and for the Lost Call Count summary report.
4. The fields of a Lost Call Record are the same as a normal SMDR Call Record.

Programming

VOICE CONFIG System Data – SMDR Attributes

Related Features

Call Cost Display
SMDR Call Records
Traffic Analysis

Hardware

RS-323 device to capture SMDR

2.31 SYSTEM ADMIN PROGRAMMING

2.31.1 Keyset Administration

Description

The system database can be accessed and modified using the keypad and Flex buttons of an LIP Phone. The display of the LIP Phone can be used to view items in the iPECS SBG-1000 database. The user may be required to enter a password for access to Keyset Admin. operation.

Operation

Attendant

To program in Keyset Administration:

1. Press the **[PGM]** button.
2. Dial '*#', Enter Admin code
3. Enter Password; confirmation tone will be heard.
4. Enter PGM code (100 or 102)
5. Press the desired **{FLEX}** button.
6. Enter new value
7. Press the **[SAVE]** button; a confirmation tone is heard

Conditions

1. Only an Attendant can enter and change system database items.
2. System ID (PGM100), and System IP Address Plan (PGM102) can be programmed using Keyset Admin. operation.
3. If new value is invalid, an error tone is heard and the old value is displayed again.

Programming

Related Features

Web Administration

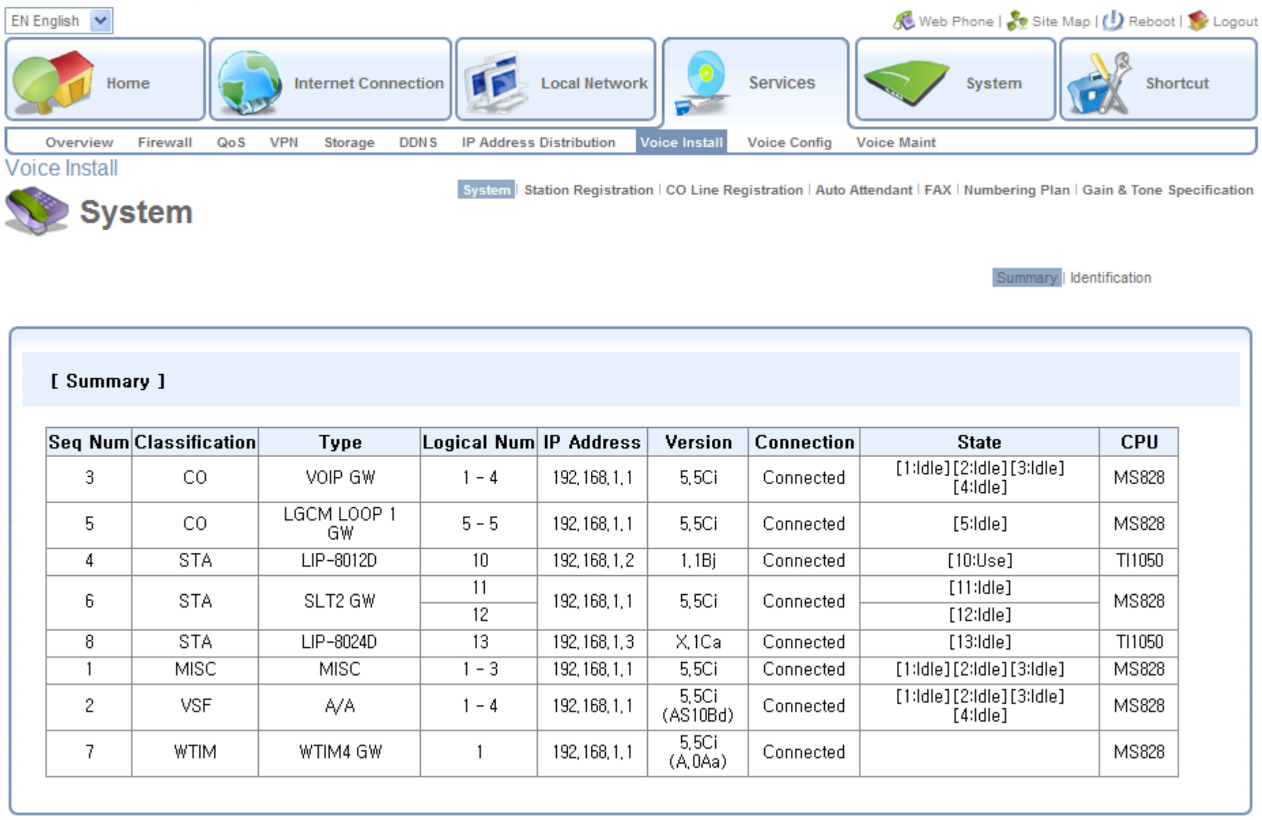
Hardware

2.31.2 Web Administration

Description

The system database is accessed and modified via a LIP Phone or the Network interface. The Network accesses the system's Web server, using the user's Web browser. When properly configured, the user can remotely access the System database.

Web administration consists of 3 Tabs across the top of the screen; each tab has multiple menu items.



Seq Num	Classification	Type	Logical Num	IP Address	Version	Connection	State	CPU
3	CO	VOIP GW	1 - 4	192.168.1.1	5.5Ci	Connected	[1:Idle][2:Idle][3:Idle][4:Idle]	MS828
5	CO	LGCM LOOP 1 GW	5 - 5	192.168.1.1	5.5Ci	Connected	[5:Idle]	MS828
4	STA	LIP-8012D	10	192.168.1.2	1.1Bj	Connected	[10:Use]	TI1050
6	STA	SLT2 GW	11	192.168.1.1	5.5Ci	Connected	[11:Idle]	MS828
			12				[12:Idle]	
8	STA	LIP-8024D	13	192.168.1.3	X.1Ca	Connected	[13:Idle]	TI1050
1	MISC	MISC	1 - 3	192.168.1.1	5.5Ci	Connected	[1:Idle][2:Idle][3:Idle]	MS828
2	VSF	A/A	1 - 4	192.168.1.1	5.5Ci (AS10Bd)	Connected	[1:Idle][2:Idle][3:Idle][4:Idle]	MS828
7	WTIM	WTIM4 GW	1	192.168.1.1	5.5Ci (A.0Aa)	Connected		MS828

Figure 2.72-1 iPECS SBG-1000 Web Admin. Voice Install View

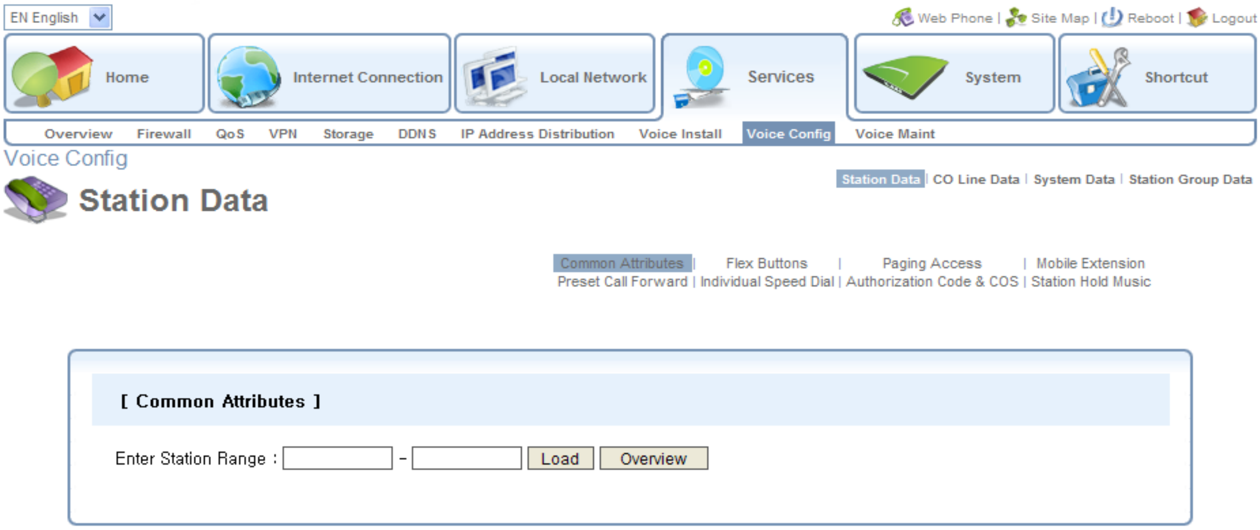


Figure 2.72-2 iPECS SBG-1000 Web Admin. Voice Config View

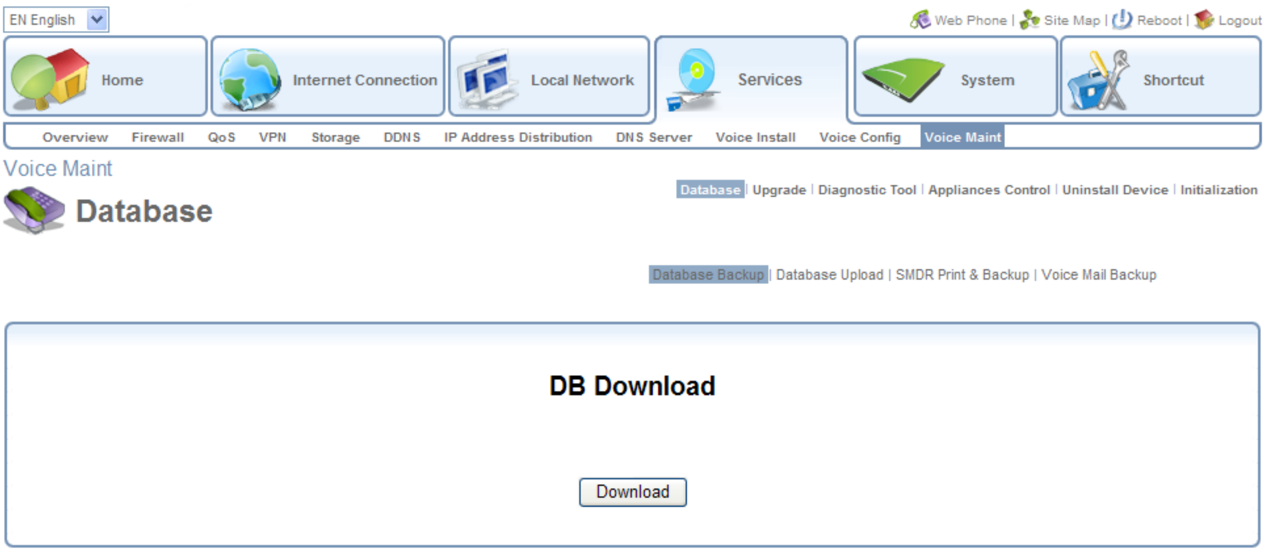


Figure 2.72-3 iPECS SBG-1000 Web Admin. Voice Maint View

For more information on database administration and maintenance, refer to the detailed feature description and operation.

Operation

Operation is detailed in the each feature description and operation.

Conditions

Programming

Related Features

Keyset Administration

Hardware

2.32 TRAFFIC ANALYSIS

Description

The iPECS SBG-1000 monitors, stores and periodically or upon request outputs various traffic statistics covering system resources. The output from the system can be used to:

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

Attendants enable Periodic Reporting. Once enabled, the system continues to monitor and output the requested report until the Periodic Report is disabled. On demand reports selected by the Attendant. The Traffic Report is sent to the defined TCP port.

System resources covered by Traffic Reports are:

- Attendant Traffic Report
- Call Summary Report
- Hourly Call Report
- H/W Unit Usage Summary Report
- CO Summary Report
- Hourly CO Report

Summary Traffic Reports cover one of five Analysis periods selected at time of print:

- Today's peak activity hour (within 24 hours)
- Yesterday's peak activity hour (24 hours prior to Today's activity)
- Last hour activity
- Today's total activity
- Yesterday's total activity.

Operation

Attendant

To print the All Summary Traffic Report periodically:

1. Press the **[PGM]** button.
2. Dial 0122 (All Summary report code).
3. Select hour for print (00-23).
4. Select minute for print (00-59).
5. Select Analysis Period (1-5).
6. Press the **[SAVE]** button.

To cancel the periodic All Summary Report:

1. Press the **[PGM]** button.
2. Dial 0123 (Cancel All Summary report code).

To print a traffic report:

1. Press the **[PGM]** button.
2. Dial 0121, or 0124-0129 (report code),
   - 0121 All Summary Traffic Reports
   - 0124 Attendant Traffic Report
   - 0125 Call Summary Report
   - 0126 Hourly Call Report
   - 0127 Hardware Usage Summary Report
   - 0128 CO Summary Report
   - 0129 Hourly CO Report
3. Press the **[SAVE]** button.

Conditions

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

Programming

Related Features

SMDR Call Records

Hardware

device to capture reports

2.32.1 Traffic Analysis, Attendant

Description

The Attendant Traffic Report covers operational statistics for the Attendants. The report outputs periodically or the Attendant requests output of the report for a defined Analysis period. The following is a sample report and description of the report fields.

```
==========================================================================
Site Name   : abc co
Report Type : Attendant Traffic Report - Today Peak
Date        : 19/01/11 15:03
==========================================================================


 Atd  Meas  --------- Calls ---------   ------ Time ------   Time   Speed  Atd
 No   Hour  Total  Ans Abnd H-Abd Held  Avail   Talk   Held  NoAns  Ans    Type
  10  13:00   104   82   22     3   18  10:12  14:21  01:23  00:52  00:23  Sys

============================== print completed =============================
```

Field	Description
ATD No	Attendant Station Number
Meas Hour	(Measurement Hour) Hour data accumulation began
Calls Total	Total number of calls, except group & recalls, routed to the Attendant
Calls Ans	(Calls Answered) Calls answered during the Analysis period
Calls Abdn	(Calls Abandoned) Calls abandoned before answer by the Attendant, does not include calls abandoned while on hold.
Call H-Abdn	(Calls Abandoned from Hold) Calls abandoned while on hold
Calls Held	Number of calls placed on hold by the Attendant
Time Avail	(Time Available) Time attendant was available to handle new calls
Time Talk	Total time the Attendant was active on calls
Time Held	Time Attendant had calls on hold
Time NoAns	(Time No Answer) Average time calls were ringing or in queue for attendant before abandoned
Speed Ans	(Speed of Answer) Average time calls rang before answer by Attendant
ATD type	(Attendant Type) System or Main

Operation

Attendant

To print the Attendant Traffic Report:

1. Press the **[PGM]** button.
2. Dial 0124 (Attendant Traffic report code).
3. Select Analysis Period (1-5).
4. Press the **[SAVE]** button.

Conditions

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

Programming

Related Features

SMDR (Station Message Detail Recording)

Hardware

Device to capture reports

2.32.2 Traffic Analysis, Call Reports

Description

Call activity statistics are provided in the Hourly Call Reports.

Hourly Call Report

The Hourly Call Report covers hourly completed call activity for the selected Analysis period. The report indicates the number of completed calls for each hour during the Analysis period as shown:

```
==============================================================================
Site Name   :
Report Type : Call Hourly Report
Date        : 19/01/11 15:38
==============================================================================

 Anal Hour          # Calls Completed
  15:00                  0
  14:00                  0
.....
.....
  17:00                211
  16:00                543
  Total Calls    :       754

============================== print completed ==============================
```

Operation

Attendant

To print the Hourly Call Report

1. Press the **[PGM]** button.
2. Dial 0126 (Hourly Call report code).

Conditions

Programming

Related Features

SMDR (Station Message Detail Recording)

Hardware

Device to capture reports

2.32.3 Traffic Analysis, H/W Usage

Description

The Hardware Usage report provides statistics for the system's special Hardware resources such as the VSF as shown in the following sample report.

```
==============================================================================
Site Name   : abc co
Report Type : H/W Unit Usage Summary Report - Today Peak
Date        : 19/01/11 14:52
==============================================================================

 Unit  Num    Anal   Total  Total
 Type  Unit   Hour   Req    Denied
```

```
  VSF    4    00:00     0      0

=============================== print completed ==============================
```

Operation

Attendant

To print the Hardware Usage Summary Report:

1. Press the **[PGM]** button.
2. Dial 0127 (H/W Usage Summary report code).
3. Select Analysis Period (1-5).
4. Press the **[SAVE]** button.

Conditions

Programming

Related Features

SMDR (Station Message Detail Recording)

Hardware

Device to capture reports

2.32.4 Traffic Analysis, CO Reports

Description

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

```
==============================================================================
Site Name   : abc co
Report Type : CO Group Summary Report - Today Peak
Date        : 19/01/11 19:43
==============================================================================

 Peak Hour For All CO: 10:00
 Grp Num  Anal   Total Total Inc.  Out.   Grp    %    %
 No  COs  Hour   Usage Seize Seize Seize  Ovfl  ACB  FAO
   1   6 10:00      1     3     0     3     0    0  ---
   2   2 00:00      0     0     0     0     0    0  ---

=============================== print completed ==============================
```

Field	Description
Grp No.	CO Group number

Num COs	The number of CO lines in the group
Anal Hour	(Analysis hour) hour during the analysis period with peak usage.
Total Usage	Total number of call attempts on CO lines in the Group
Total Seize	Total number of times CO lines in the group were used for any call
Inc Seize	(Incoming Seizures) Total number of incoming calls answered for CO lines in the group.
Out Seize	(Outgoing Seizures) Total number of outgoing calls attempted on CO lines in the group.
ACB	(All COs Busy) Percentage of the time that all CO lines in the group were simultaneously busy.
FAO	(Failed Attempts Outgoing) Percentage of outgoing calls offered to the CO lines in the group that were denied due to All Trunks Busy condition.

Operation

Attendant

To print the CO Traffic Summary Report:

1. Press the **[PGM]** button.
2. Dial 0128 (CO Traffic Summary report code).
3. Select Analysis Period (1-5).
4. Press the **[SAVE]** button.

To print the CO Hourly Traffic Report:

1. Press the **[PGM]** button.
2. Dial 0129 (CO Hourly Traffic report code).
3. Select CO Group (00-05).

Conditions

Programming

Related Features

SMDR (Station Message Detail Recording)

Hardware

Device to capture reports

2.33 VSF INTEGRATED AUTO ATTD/VOICE MAIL

2.33.1 VSF

Description

The Voice Store & Forward (VSF) unit, which is equipped in iPECS SBG-1000, provides the system memory to support the integrated Auto Attendant, Voice Mail and system announcement applications available in the System. 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 used by the Auto Attendant and Voice Mail applications as well as other system features. The VSF has a storage capacity of up to 240/480 minutes of announcement and message storage; approximately 10 minutes of storage is generally used for fixed system prompts. The capacity of VSF storage depends on the lock key for the VSF Memory Extend.

2.33.2 VSF-Auto Attendant

Description

When a call comes into the system through a CO line, the call may be routed to one of 70 user recorded VSF Announcements. An announcement is assigned as a Station Group announcement or as Auto Attd announcement with an CCR Table that permits Caller Controlled Routing (CCR). Station Group announcements are played when a call is routed to the group based on definitions in the Station Group Attributes.

For an Auto Attd Announcement the system will play the announcement and monitor for digits from the connected external party. A CCR Table defines a dialed digit (0 – 9) to a route. Each single digit is defined a corresponding route:

- Station
- Station group
- Speed Dial number
- Page Zone
- Voice Mail
- VSF Announcement

In addition, the system will monitor digits for a station number. If the user dials a station number, the Auto Attd will complete an unsupervised call transfer to the station.

Operation

Attendant

To record an Auto Attd Announcement:

1. Press the **[PGM]** button.
2. Dial 05 (Message Record code).
3. Dial the appropriate number from 001-072 (Announcement number).
4. The current announcement is played followed by the Press # to record prompt.
5. Dial ‘#’.
6. After the record prompt and beep-tone, record message.
7. Press the **[SAVE]** button to stop recording and save the message.

To delete a recording:

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

2. Dial 05 (Message Record code).
3. Dial the appropriate number from 001-072 (Announcement number).
4. The current announcement is played followed by the “Press # to record” prompt.
5. Dial ‘#’.
6. Press the **[SPEED]** button during playback to erase message

System

Operation of the CCR Tables and Auto Attendant are automatic.

Conditions

1. There are no individual time limits on an Auto Attendant announcement.
2. The external caller may receive a ring-back tone before playback of a VSF announcement.
3. The Attendant must “Save” a recording before returning to the on-hook state, otherwise, the existing recording is used.
4. To record or delete an Auto Attendant message, all of the VSF channels must be idle.
5. The external caller may dial at any time during an Auto Attendant announcement and must dial prior to the expiration of the CCR Analysis timer.
6. If the external caller dials an invalid selection or station, the system will play the ‘Invalid Entry’ prompt and allow re-entry using the DISA Retry Counter.
7. If the external caller dials more than a single digit, the call is routed based on the System Numbering Plan.
8. Calls routing by an Auto Attendant (CCR) Announcement are interactive DISA calls and are subject to conditions of a DISA call.
9. The ‘*’ digit is reserved in the CCR Tables to repeat the current or previous Auto Attd announcement.
10. The ‘#’ digit is reserved for callers to access their Voice Mail-box remotely.
11. A CCR Announcement may be programmed to disconnect after playing.
12. The Auto Attd Announcement feature is supported for DISA and DID calls.
13. System announce number 71 is for the MOH.

Programming

VOICE CONFIG	Station Group Data CO Line Data – Call Routing by Line CO Line Data – Ring Assignment Table CO Line Data – Call Routing by Auto Attendant System Data – Call Feature Timer – VSF User Maximum Record Timer System Data – Call Feature Timer – VSF Valid User Message Timer

Related Features

Station Groups
Remote Message Retrieval

Hardware

VSF

2.33.3 VSF Voice Mail

2.33.3.1 Message Storage

Description

When a station activates Call Forward to the VSF Group, a call is transferred to a VSF mail box or a transferred call recalls to the VSF, the call is handled by the iPECS SBG-1000 Mail application. The caller connects to the called station's User Greeting followed by a beep tone.

The remote caller can record a message and hang-up or dial '*' for further options. When disconnect occurs, the VM application stores the message in the called user's voice mail-box and activates the Message Waiting indication at the user's station.

Operation

Remote Caller

To leave a voice message after hearing announcement:

1. Wait for the beep, then leave a message.
2. Hang up to quit recording,

Or,

2. Dial '*' for more options.

Conditions

1. Two timers are provided to control voice message length. The Valid User Message Timer establishes the minimum voice message length; voice messages shorter than this timer are not stored. The VSF User Maximum Record Timer establishes the maximum voice message length; when the VSF User Maximum Record Timer expires while recording a voice message, confirmation tone is heard and the message is saved for the destination station.
2. If all the VSF channels are in use, the Ring Back tone is provided until a VSF channel is available.
3. All stations including, SLTs can leave and receive voice messages.
4. Individual User Greetings and Voice Mails are protected from AC power loss.

Programming

VOICE CONFIG	Station Data – Preset Call Forward – Transfer Mail Box System Data – Call Feature Timer – Call Forward No Answer Timer System Data – Call Feature Timer – VSF User Maximum Record Timer System Data – Call Feature Timer – VSF Valid User Message Timer

Related Features

Call Forward
Station Message Wait/Call Back
VSF Voice Mail
Call Transfer, Voice Mail

Hardware

VSF

2.33.3.2 Message Retrieval

Description

A user can access their Mail Box locally by placing a call to the VSF Voice Mail group, or from an LIP Phone, by pressing the **[MSG]** button, or by pressing a **{VMAIL-BOX}** button while off-hook receiving Intercom dial tone.

Prompts are then received to guide the user in the Voice Mail Box operation. The user must enter a Mail Box number, generally the station number, and a corresponding password in response to the "Request for Password" *("Please enter your password code."*) prompts.

If the user enters a valid and matching Mail box and password, 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 is played *("To replay message, press one, to listen to the next message, press two, to delete message, press three, to forward message, press four, to call the sender, press five, to skip message, press six, to return to main menu, press nine.")*. 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 first-saved message is played. At the end of the message, the "Saved Message option" prompt is played *("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."*). 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 Smart Business gateway 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

***LIP* Phone**

To assign a {vmail-box} Flex button:

1. **[PGM]** + **{FLEX}** + VM group + Mail-box (station) number + **[SAVE]**

To retrieve Voice Mail locally:

1. Lift the handset or press the **[SPEAKER]** button.
2. Press **[MSG]** button; the message contents summary is shown.

1. ICM MWI(001)
2. VSF MSG(002)

3. Dial digit ‘2’ to select VSF Messages
4. Dial the Mail Box and corresponding password to receive the “Number of Messages” prompt.
5. Dial desired option code.
6. At completion of session, hang-up to return to idle.

Or,

1. Lift the handset or press the **[SPEAKER]** button
2. Press **{VMAIL-BOX}** button.
3. Dial the Mail Box and corresponding password to receive the “Number of Messages” prompt.
4. Dial desired option code.
5. At completion of session, hang-up to return to idle.

To attach a memo to the current voice message:

1. During or after the New or Old Message option prompt, dial ‘7’.
2. At the beep, record the memo.
3. Dial ‘*’ to stop recording and store the memo.
4. During or after the New/Old option prompt, dial 4 to forward the message and memo.

SLT

To retrieve Voice Mail locally:

1. Lift the handset.
2. Dial the Voice Mail Group to receive the “Mail Box & Password” prompts sequentially.
3. Dial the Mail Box and corresponding password to receive the “Number of Messages” prompt.
4. Dial desired option code.
5. At completion of session, hang-up to return to idle.

To attach a memo to the current voice message:

1. During or after the New or Old Message option prompt, dial ‘7’.
2. At the beep, record the memo.
3. Dial * to stop and store the memo.
4. During or after the New Old option prompt, dial ‘4’ to forward the message and memo.

Conditions

1. If no new/old messages are available, pressing ‘1’ or ‘2’, is an invalid operation and the user receives the “Invalid Entry” prompt or “No Message” prompt.
2. If the dialed number is not recognized, the "Invalid Entry" prompt is played. After the second invalid entry, the user is disconnected.
3. The user may dial digits at any time during a voice mail playback, system prompt or silence; the user must dial a digit in response to a system prompt within the CCR Analysis timer or the system will disconnect and return error tone.

4. Messages are retrieved in LIFO (Last in First out) order.

Programming

Related Features

Message Retrieval Options
Remote Message Retrieval

Hardware

VSF

2.33.3.3 Remote Message Retrieval

Description

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

Operation

Remote Caller

To access Voice Mailbox from a remote location

1. Lift the handset.
2. Dial the telephone number of a DISA assigned CO line assigned for answer by a VSF Auto Attd.
3. Upon answer, dial '#' to receive the "Request for Mail Box number" prompt.
4. Follow local access procedures.

Conditions

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

Programming

Related Features

Message Retrieval Options
VSF-Auto Attendant
Message Retrieval

Hardware

VSF

2.33.3.4 Message Retrieval Options

Description

The user may dial the digit 9 to receive the “VM Long Options” prompt while in the Voice Mail Box, including during or after a voice message or system prompt, except when an option has been selected that requires user dialing. Some options involving user dialing include the Message Retrieval Option 1/2 (Play New/Saved Message), 7 (Cancel or Forward message, Remote Access Only) or 8 (Mail Box settings), refer to Table. The “VM long Options” prompt is:

"To play new messages, press one, to play saved messages, press two, to set station forwarding, press seven (This option is available only for remote access), to set greeting or password, press eight, to disconnect, press pound, Press 0 for the operator, Press nine to hear this message again."

The VSF Voice Mail will respond to incoming digits as shown in the following table.

Digit	Function
1	Play New Msg
2	Play Saved Msg
7	Set Cancel/Fwd, available only for remote access
8	Mail Box Setting, "Mailbox Settings" prompt
9	VM Long options
#	Drop, "Goodbye”
0	Attd Call, Call to Attendant.

Operation

LIP Phone

To access a Message Retrieval option

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

SLT

To access a Message Retrieval option

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

Conditions

1. 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 an error tone.
2. 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 Mail Box. If the user hangs up, the VSF will recall at the user Station, and upon answer will play “Request Mail Box Number” prompt.

Programming

Related Features

Message Retrieval
Remote Message Retrieval
Voice Mailbox Settings

Hardware

VSF

•

2.33.3.5 E-Mail Notification

Description

With the VSF, the system stores the voice message and sends an e-mail to the e-mail address associated with the station as notification of the new voice message. The voice message can be attached to the e-mail as a wav file.

Operation

System

System automatically sends e-mail to notify user of new voice message.

Conditions

1. The voice message is stored in the VSF as well as being attached to the e-mail. The voice message must be expressly deleted from the VSF even if the e-mail is deleted.
2. The e-mail is sent to the address assigned for the station with the “sender” address defined for the VSF. Note the latter is required, as many e-mail servers will reject anonymous e-mails.
3. The e-mail address for the VSF and the station is defined under the Web Admin.
4. The Voice message can be attached to the e-mail notification as a .wav file, if the Attach Message option is enabled. If disabled, the notification e-mail does not include an attached wav file.

Programming

VOICE CONFIG	Station Data – Station Voice Mail Attributes – VSF MSG

Related Features

VSF-Auto Attendant

Hardware

VSF

2.33.3.6 Voice Mail Back-up Station

Description

With the VSF, an iPECS soft-phone (Phontage) user receives notification of new messages for assigned stations. The soft-phone will indicate the total messages for the assigned stations and the soft-phone. The soft-phone user can download the messages for other stations to the hard drive of the soft-phone PC and, using the soft-phone application, manage the messages on the hard drive. If enabled, the soft-phone user may delete voice messages from the VSF memory.

Operation

System
System automatically notifies back-up station of new messages.

Phontage
See Phontage Guide.

Conditions

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

Programming

VOICE CONFIG Station Data – Station Voice Mail Attributes – VSF Backup

Related Features

VSF-Auto Attendant

Hardware

VSF

2.33.3.7 Voice Mailbox Settings

Description

The user can program the Mail Box 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”)*.

Operation

To program Mail Box settings while “in” the Voice Mail Box:

1. Press ‘8’, for Mail Box settings; the “Mail Box Setting” prompt is received.

To enter a new password:

1. Dial ‘2’ to receive the “Password Entry” prompt (“Please enter your new password and press pound when finished.").
2. Dial new password.
3. Press ‘#’; the “Reenter Password” prompt will be heard ("Please re-enter your password to confirm and press pound when finished.").
4. Dial new password again.
5. Press ‘#’ and the “Password Confirmation” prompt will be heard *("Your password is saved.")*.

To create a new greeting:

1. Dial ‘1’ to hear the “Greeting Option” prompt ("To listen to your current greeting, press five to record a new greeting, press seven, to return to the main menu, press nine.”).
2. Dial ‘5’, to hear your greeting.

OR

3. Dial ‘7’ to hear the “Record Greeting” prompt ("At the tone, record your new greeting, press # when done.").
4. After the beep, record your desired greeting speaking in a normal voice.
5. Press ‘#’ and receive the “Greeting Confirmation” prompt ("Your greeting is saved.").

Conditions

1. If the user is external, the user must begin dialing within the CCR Analysis time, if not the call is released.
2. If the dialed number is not recognized, the "Invalid Entry" prompt is played.
3. The user must assign a password (Authentication code, up to 12 digits) before access to the mailbox will be allowed.

__NOTE:__ A greeting does not have to be recorded.

Programming

Related Features

Message Storage
Message Retrieval
Remote Message Retrieval
Message Retrieval Options

Hardware

VSF

2.33.3.8 Call Forward from VM

Description

External users can activate or deactivate Call Forward for their station. Pressing ‘7’ while retrieving messages will return the "Mailbox Set Forward" prompt, (“To forward calls to another extension, press one. to cancel forwarding, press 2 to return to the main menu, press nine.”).

Operation

To activate Call Forward while in the VM:

1. Press ‘7’, for Mail Box set forward, the “Mail Box Set Forward” prompt is received.

To activate Call Forward:

1. Dial ‘1’ and receive the “Password Entry” prompt (“Please enter the number to forward to ...").
2. Dial the Station Number as follows:
   - To forward to another station, dial the station number.

- To forward calls Off-net, dial '*' and enter Individual Speed number. If the Individual Speed bin is valid, the confirmation announcement "forwarded to station ('xxx')" or "forwarded to speed bin number (yyyy)" is played.

To deactivate Call Forward:
1. Dial '2' and receive the "Station forwarding is canceled" prompt.

To return to the Main menu:
1. Dial '9' and receive the "Mail Box Settings" prompt.

Conditions

1. If the user is external, the user must begin dialing within and dial subsequent digits within the VSF Inter-Digit Timer; if not, the call is released.
2. This Mail Box Set Forward is only available for external users.

Programming

Related Features

Message Storage
Message Retrieval
Remote Message Retrieval
Message Retrieval Options

Hardware

VSF

2.33.3.9 Outbound Message Notification

Description

The VSF is able to dial an external number to notify a user of a new voice message. The system employs the mobile extension number registered for the station receiving the message. When a caller leaves a message with notification configured, the system places a call to the registered mobile extension. When the user answers, the extension prompt is played followed by the new message prompt, ("You have xx new messages."). The new message prompt indicates the number of unheard messages.

The user must listen to the new message to confirm the notification. If the user takes no action within the CCR Inter-digit timer or hangs-up, the call is disconnected and the system will retry the call after the retry timer expires, until the user listens to the message or the number of attempts reaches the retry counter. If the user does not answer, the ISDN or VoIP connection times out or disconnects before answer, or is busy, the system disconnects the notification and will retry the call after the retry timer. The system will retry the notification until the notification is successful or the number of call attempts reaches the Retry count.

Operation

Operation of message notification is automatic when configured.

Condition

1. Outbound notification over a PSTN line is not available.
2. Caller Id will be the external caller who left the message or, for messages from another station, Caller Id will be the station receiving the message.
3. If VSF Notify is changed to ‘Not Use’, any existing notification will be terminated after the initial notification call.
4. For proper operation, the Station COS and CO Group access for the station must be such as to allow the notification call.
5. The destination of the notification is the Mobile telephone number assigned in Mobile Extension table.
6. If all lines in the assigned CO group are busy when the system attempts to place the notification call, the System will continuously try to seize CO line until a line is successfully seized.
7. The Retry counter is incremented after the system access the CO line for notification.
8. The Retry count is from 1 to 9; the retry interval is from 1 to 3 minutes.
9. If a new message is logged before answer of the notification call, the message will be available to the user and a new notification is not invoked. If a new message is received after answering the notification call, the System will invoke another notification call. The user will receive the notification after returning to idle.

Programming

VOICE CONFIG Station Data – Mobile Extension

Related Features

Mobile Extension
Message Retrieval
Remote Message Retrieval
Message Retrieval Options

Hardware

VSF

2.33.4 Company Directory

Description

This feature allows a caller reaching Call Routing destination to utilize the DTMF keys to “spell” the name of a subscriber and be directed to their extension.
If system checks and finds subscriber stations, Company Directory First Name and Last Name has to be preprogrammed and station’s Subscriber Name has to be prerecorded in each station. When station users record his subscriber name, users can record by dialing Company Directory Name code.

Operation

To record Company Directory Name:

1. Dial the Company Directory Name code.
2. Choose the Recording menu of Subscriber Name, 1.
3. After recording announcement, record subscriber name.

To delete Company Directory Name:

1. Dial the [Company Directory Name] flexible number.
2. Choose the Deleting menu of Subscriber Name, 2.

To use Company Directory Feature:

When outside caller reaches Company Directory as destination, the system will prompt "Press one to search by first name", "Press two to search by last name", then "Enter the first 3 characters of the person's first name" or "Enter the first 3 characters of the person's last name" Once 3 characters have been entered, the system will check the entered characters for a match to the programmed name in Station Voice Mail Attributes menu (Company Directory First Name and Last Name). System will search an entry only among those who have recorded the subscriber name.

If a 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.

If more than one match is found, the system will play:

"for [subscriber name], press 1"

"for [subscriber name], press 2"

…

"for [subscriber name], press 9"

When the caller enters the desired digit, the system will route the caller to that station

Condition

1. To be searched in company directory, subscriber name must be recorded. And the first and last name also should be programmed properly.
2. This feature works only when Station VM Feature Usage is set to "ON".

Programming

VOICE INSTALL	CO Line Registration – SIP ID Configuration
VOICE CONFIG	Station Data – Station Voice Mail Attributes – Company Directory
	CO Line Data – Call Routing by Called Number
	CO Line Data – Call Routing by Auto Attendant
	System Data – System Attributes – Station VM Feature Usage

Related Features

Hardware

VSF

2.33.5 Record VM Greeting using Call Routing

Description

This feature provides an option to record a System (001-072) greeting by reaching Call Routing destination.
Administrator can assign VM Greeting option at Call Routing destination.

Operation

To use Record VM Greeting Feature:

1. When outside user reaches recording VM greeting as destination, system prompts "enter password" prompt.
2. User enters system attendant's station number and password.
3. "Enter system greeting number" is played.
4. User enters 3 digits system greeting number 001 ~ 072.
5. "To record a new greeting, press pound" is played. And then if there is a recorded greeting, the greeting is played.
6. User dials '#'.
7. "Begin recording" is played and confirm tone is heard.
8. User records the greeting and dials '#' to finish.
9. System goes to step 5 above. So, "Enter system greeting number" is played.
10. User can hang up.

Condition

1. This feature works only when Station VM Feature Usage is set to "ON".

Programming

VOICE INSTALL	CO Line Registration – SIP ID Configuration
VOICE CONFIG	CO Line Data – Call Routing by Called Number
	CO Line Data – Call Routing by Auto Attendant
	System Data – System Attributes – Station VM Feature Usage

Related Features

Hardware

VSF

2.33.6 Administrator Mailbox

Description

This feature provides a mailbox that has administrative interface via telephone commands to perform common tasks associated with the VM. In addition, administrator mailboxes may be used to record a broadcast message which is delivered to all mailboxes.

Operation

Station Voice Mail Attributes menu in Web Admin has “Administrator MailBox” option to mark the mailbox with administrative options.
If a mailbox is marked as an administrator, the main prompt will offer “to access administrative options, press six”
After pressing 6, the options that can be performed from the mailbox are:
Administrator Mailbox
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

- Add mailbox [1]
  “Please enter the mailbox number”
  “Enter COS 1-5”
  “Press 1 to confirm or # to cancel and go back to administration main”
  When confirmed, the specified station can access to VSF.
- Delete mailbox [2]
  “Please enter the mailbox number”
  “Press 1 to confirm or # to cancel and go back to administration main”
  When confirmed, the specified station cannot access to VSF.
- Reset mailbox password [3]
  “Please enter the mailbox number”
  “Press 1 to confirm or # to cancel and go back to administration main”
  The password for station is defaulted to be empty. [Authorization Code & COS]
  Station password is used for mailbox access in SBG.
- Record Mailbox greeting [4]
  “Please enter the mailbox number”
  “to listen press 1, to record press 2, to delete press 3.”
  “Press # to cancel and go back to administration main”
- Record a broadcast message [5]
  “Please enter the mailbox number” (01-10)
  “to listen press 1, to record press 2, to delete press 3, to send press 4.”
  “Press # to cancel and go back to administration main”
- Record Mailbox Name [6]
  “Please enter the mailbox number”
  “to listen press 1, to record press 2, to delete press 3.”
  “Press # to cancel and go back to administration main”

Condition

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

2. The broadcast message is the shared message. System provides broadcast message first regardless message retrieve option.
3. A station can delete broadcast message without confirmation.
4. Only administrator can delete a recorded broadcast message.
5. This feature works only when Station VM Feature Usage is set to “ON”.

Programming

VOICE CONFIG	Station Data – Station Voice Mail Attributes – Administrator MailBox System Data – System Attributes – Station VM Feature Usage

Related Features

Hardware

VSF

2.33.7 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 or hangs up.

Operation

An administration option shall enable/disable the announcement only flag.
An administration option shall allow the selection of previous menu or hang up after the mailbox greeting. Previous menu is for Call Routing by Auto Attendant feature.

Condition

1. There are no limits on the number of mailboxes that can be marked as announcement only.

Programming

VOICE CONFIG	Station Data – Station Voice Mail Attributes – Announce only MailBox Station Data – Station Voice Mail Attributes – Announce only Option

Related Features

Hardware

VSF

2.33.8 Message Cascade

Description

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

Operation

A mailbox has a cascade field in administration. That field is another mailbox number.
A mailbox has a cascade field type in administration. One of the following types may be selected:

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

Conditions

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

Programming

VOICE CONFIG	Station Data – Station Voice Mail Attributes – Cascade MailBox Station Data – Station Voice Mail Attributes – Cascade Type

Related Features

Hardware

VSF

2.33.9 Class of Service Settings

Description

The system shall provide administrative options to create COS. These COS can then be assigned to stations/mailboxes on an individual basis.

Operation

The system shall provide 5 COS settings, 1-5.
Each COS has the following items that can be programmed:

1. Greeting length: 0-99 seconds (60 seconds default)
2. Message length: 0- 600 seconds (0 default)
3. Number of messages: 0-250 (0 default) (0- Number of message is the same as system capacity)
4. Retention time: 0-99 days (0 days default)

5. E-mail notification: Disable/Notification only/ Notification and Delete (default: Notification and Delete)
6. Future delivery messages: On/Off (Off default)
7. Confirm message receipt: On/Off (Off default)
8. Private message mark: On/Off (Off default)

Conditions

1. The default class of service for all mailboxes is 1. COS 1 contains all default settings.
2. Future delivery message feature is not supported.
3. This feature works only when Station VM Feature Usage is set to "ON".

Programming

VOICE CONFIG	Station Data – Station Voice Mail Attributes – VM COS System Data – System Attributes – Station VM Feature Usage

Related Features

Hardware

VSF

2.33.10 Send Message

Description

A user can record the message and send it to other mail box with several options or distribution list. The VM long option prompt is modified to add "To send a message press 4".

Operation

To send a message:

1. Record Voice Message
   a. User press '4' in the main menu and record a message.
   b. The prompt will state "At the tone, please leave a message and to stop recording press pound key".
2. Decision of Destination
   a. The prompt will state "Enter the mail box number or distribution list number followed by pound. To spell the name, press *"
   b. To Mail Box
      i. Enter the mail box number
      ii. Refer to Step 3.
   c. To Distribution List
      i. Enter distribution list number and # key.
      ii. The prompt will state "Your message has been sent".
   d. To Spell Name
      i. Press '*'.

ii. The prompt will state "Press one to search by first name, Press two to search by last name", then "Enter the first 3 characters of the person's first name" or "Enter the first 3 characters of the person's last name"
iii. Once 3 characters have been entered, the system will check the entered characters for a match to the programmed name in Station Voice Mail Attributes menu (Company Directory First Name and Last Name). System will search an entry only among those who have recorded the subscriber name.
iv. If a match is found, the system will play:
"for [subscriber name], press 1"
"for [subscriber name], press 2"
…
"for [subscriber name], press 9"
v. When the caller enters the desired digit, the system will select the destination station.
vi. Refer to Step 3.

3. Delivery Selection
   a. After Step 2-b or 2-d, the prompt will state "For regular delivery, press one. To mark urgent, press two. To mark private, press 3. To mark urgent and private, press 4. To request delivery receipt of the message for future, press 5".
   b. Press 1~5, the prompt will state "Your message has been sent"
   c. If press 1~4, then refer to "Mark a message private". Or if press 5, then refer to "Mark a message for delivery confirmation"

Conditions

1. This option is only played to the mailbox owner if the mailbox COS allows it.
2. This feature works only when Station VM Feature Usage is set to "ON".

Programming

VOICE CONFIG	Station Data – Station Voice Mail Attributes – VM COS System Data – System Attributes – Station VM Feature Usage

Related Features

Hardware

VSF

2.33.11 Distribution Lists

Description

This allows a mailbox owner to setup a group of mailboxes and send a message using one number instead of having to enter each mailbox individually.
A maximum of 5 distribution lists can be set up per mailbox.
The VM long option prompt is modified to add "for personal options press 5".

Operation

To edit a distribution list:

1. Press ‘5’ from the main menu, then the prompt will state “to edit a list press 1”.
2. Press ‘1’, then the prompt will state “Enter list number 1~5”.
3. Press 1~5, then the prompt will state “to edit a list, press 1, to delete a list, press 2”.
4. Press ‘1’, then the prompt will state “to add a mailbox, press 1, to delete a mailbox, press 2, to listen to mailboxes in list, press 3”.
    a. Add mailbox to list
        i. Press ‘1’, then the prompt will state “Please enter the mailbox number”.
        ii. Enter the station number, then the prompt will state “mailbox XXX added”.
    b. Delete mailbox from list
        i. Press ‘2’, then the prompt will state “Please enter the mailbox number”.
        ii. Enter the station number, then the prompt will state “mailbox XXX deleted”
    c. Listen to mailboxes in list
        i. Press ‘3’, then the prompt will state “mailbox XXX” and repeat for all members.

Conditions

1. The maximum number of mailboxes in a distribution list is 25.
2. This feature works only when Station VM Feature Usage is set to “ON”.

Programming

VOICE CONFIG System Data – System Attributes – Station VM Feature Usage

Related Features

Hardware

VSF

2.33.12 Mark a message private

Description

This allows a mailbox owner to mark a message as private, replying to, or creating a message to another user. These messages cannot be forwarded.

Operation

To mark a message private:

1. After replying to a message, recording a message, the following options will be played:
   For regular delivery, press 1.
   To mark urgent, press 2.
   To mark private, press 3.
   To mark urgent and private, press 4

To request delivery receipt of the message for future, press 5
2. Press ‘3’ or ‘4’.

Conditions

1. Private message cannot be forwarded.
2. This option is only played to the mailbox owner if the mailbox COS allows it.
3. This feature works only when Station VM Feature Usage is set to “ON”.

Programming

VOICE CONFIG	Station Data – Station Voice Mail Attributes – VM COS System Data – System Attributes – Station VM Feature Usage

Related Features

Hardware

VSF

2.33.13 Mark a message for delivery confirmation

Description

This allows a mailbox owner to mark a message for confirmation of delivery. When the user has listened to the sent message, a message is dropped in the sender mailbox confirming listen receipt.

Operation

To mark a message private:

1. After replying to a message, recording a message, the following options will be played:
   For regular delivery, press 1.
   To mark urgent, press 2.
   To mark private, press 3.
   To mark urgent and private, press 4
   To request delivery receipt of the message for future, press 5
2. Press ‘5’

Once the user has listened to a message that has been marked for receipt, the user that marked the message will get a new message in their mailbox.
When they access this message a prompt will state: “message for XXXX (mailbox #) was listened to on HH:MM MM/DD”
This message is then treated a normal VM and all options apply.

Conditions

1. This option is only played to the mailbox owner if the mailbox COS allows it.

2. This feature works only when Station VM Feature Usage is set to “ON”.

Programming

VOICE CONFIG	Station Data – Station Voice Mail Attributes – VM COS System Data – System Attributes – Station VM Feature Usage

Related Features

Hardware

VSF

2.34 WAKE-UP ALARM

Description

This feature allows a user or Attendant to set a wake-up time or desired time to be alerted. When the time is reached, the system will signal with an audible and visual signal.

Operation

Attendant

To register Wake-Up:

1. Press the **[PGM]** button.
2. Dial 023 (Attendant Station Program code).
3. Dial the desired station range, for a single station, enter an ‘*’ in place of the second station number.
4. Dial 2-digit hour and 2-digit minute for alerting.
5. For a daily (repeating alarm), dial ‘#’.
6. Press **[SAVE]** button.

To erase Wake-Up:

1. Press the **[PGM]** button.
2. Dial 024 (Attendant Station Program code).
3. Dial the desired station range, for a single station, enter an ‘*’ in place of the second station number.
4. Press **[SAVE]** button.

LIP Phone:

To register Wake-Up:

1. Press the **[PGM]** button.
2. Dial 21 (Set Wake-up code).
3. Dial 2-digit hour and 2-digit minute for alerting (hh:mm).
4. For a daily (repeating alarm), press ‘#’.
5. Press **[SAVE]** button.

To stop the alarm notification:

1. Lift the handset or press **[SPEAKER]**.

To erase Wake-Up:

1. Press the **[PGM]** button.
2. Dial 22 (Erase Wake-up code).
3. Press **[SAVE]** button.

SLT

To stop the alarm notification:

1. Lift the handset.

Conditions

1. When receiving a wake up signal, lifting the handset will be heard Wake-up alarm prompt.
2. The Wake-up alarm Ring signal is 30 seconds, On/90 seconds, Off (3 times). If no action is taken by the user, the ring signal is given to the Attendant with a display designating the station number that did not respond.
3. Time (hh:mm) must be entered in the Military 24-hour format.
4. The daily alarm will reset and repeat each day until erased (cancelled)' the One-time alarm will reset and cancel automatically.

Programming

Related Features

Hardware

2.35 DIRECT STATION SELECT/BUSY LAMP FIELD (DSS/BLF)

Description

When a Flex button on an LIP Phone is assigned as a **{DSS}** button it also serves as a Busy Lamp Field; the LED indicates the status of the associated station or system facility.

Operation

LIP Phone

Operation of this feature is automatic for assigned Flex buttons.

Conditions

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

Programming

Related Features

Intercom Call (ICM Call)
Station User Programming & Codes

Hardware

2.36 INTERCOM CALL (ICM CALL)

Description

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

Operation

LIP Phone

To place an intercom call:

1. Lift the handset or press the **[SPEAKER]** button to receive ICM dial tone.
2. Dial station number or press the **{DSS/BLF}** button.
3. For ring-back tone, await answer or
   For Intercom splash-tone, speak and await answer.

SLT

To place an intercom call:

1. Lift the handset to receive ICM dial tone.
2. Dial station number.
3. For ring-back tone, await answer or,
   For Intercom splash-tone, speak and await answer.

Conditions

1. Intercom Dial tone will time-out if action is not taken within Dial-Tone Time or, if the time between digits exceeds the Inter-digit Timer; error tone is received on dial tone time-out.

2. ICM Dial tone is halted after dialing the first digit.
3. If the called station is busy, Intercom Busy tone is provided for the Busy Tone time (7 sec.) then, Error tone is sent by the system; the caller may disconnect or activate a feature such as Message Wait/Callback.
4. For LIP Phone users, consecutive Intercom calls can be placed without the need to regain ICM dial tone (no need to hang-up) between calls; the user simply presses another **{DSS/BLF}** button.
5. An Intercom call to a station in the HF answerback or Voice Announce mode (H or P Intercom Signaling Mode) is not considered answered unless the called user lifts the handset or presses the **[SPEAKER]** button (goes off-hook).

Programming

VOICE CONFIG	System Data – Call Feature Timer - ICM Dial Tone Timer System Data – Call Feature Timer – Inter Digit Timer

Related Features

Intercom Answer Mode
Speakerphone

Hardware

2.37 INTERCOM CALL HOLD

Description

While on an active ICM Call, LIP Phone users can place the ICM Call on hold; the held station will receive the assigned Music-on-Hold. The call is placed on Exclusive Hold and recalls at the holding station when the Exclusive Hold Recall timer expires.

Operation

LIP Phone

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

1. **[PGM]** + **{FLEX}** + **[PGM]** + '53' + **[SAVE]**

To place an active ICM call on hold:

1. Press the **[HOLD]** button; the ICM dial tone is received and the **{ICM}** button LED will flash at the exclusive hold rate and the ICM dial tone is received.

To retrieve the held ICM call:

1. Press the **{ICM}** button or the **{DSS/BLF}** button associated with the held station; the **{ICM}** button LED will be On and the ICM call connected.

Conditions

1. Only one ICM call may be placed on hold at a time.

Programming

Related Features

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

Hardware

LIP Phone

2.38 INTERCOM CALLER CONTROLLED ICM SIGNALING

Description

A user can change the signaling mode of an Intercom call from Tone ring to Voice announce, or Voice announce to Tone ring.

Operation

LIP Phone

To change the ICM Signaling mode:

1. Place intercom call.
2. Dial '#', ICM Signaling mode will change from Voice announce to Tone ring or Tone ring to Voice announce.

SLT

To change the ICM Signaling mode:

1. Place intercom call as normal.
2. Dial '#', ICM Signaling mode will change from Voice announce to Tone ring, or Tone ring to Voice announce.

Conditions

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

Programming

Related Features

Intercom Answer Mode

Hardware

2.39 INTERCOM LOCK-OUT

Description

If the user takes no action after going off-hook for the Dial Tone timer or fails to dial an additional digit within the Inter-digit timer, the station will receive an error tone for 30 seconds and be placed out-of-service (locked-out). The LED of associated **{DSS/BLF}** buttons will flutter (flash) rapidly to indicate the out-of-service status.

For LIP Phone users, if the **[SPEAKER]** is used, the station will receive an error tone for 30 sec. and then automatically return to idle.

Operation

System

Operation of Intercom Lock-out is automatic based on the Dial Tone & Inter-digit timers.

Conditions

1. Error tone is presented for 30 sec. followed by 30 sec. of Howler tone followed by lock-out and silence.

Programming

VOICE CONFIG	System Data – Call Feature Timer - ICM Dial Tone Timer System Data – Call Feature Timer – Inter Digit Timer

Related Features

Intercom Call (ICM Call)

Hardware

2.40 INTERCOM STEP CALL

Description

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

Operation

LIP Phone

To activate step call, while receiving busy on a dialed Intercom call:P
1. Dial a digit other than the last digit of the busy station's intercom number.

Conditions

1. If the user dials the last digit of the busy station, Camp-On will be activated.
2. After receiving busy tone, if the user takes no action for the Busy Tone timer, the system will start the Intercom Lock-out procedure.
3. For Step Call to work, the ICM Station called must have the same digits except for the last digit.

Programming

Related Features

Intercom Lock-Out
Intercom Call (ICM Call)

Hardware

2.41 MESSAGE WAIT/CALL BACK

2.41.1 Station Message Wait/Call Back

Description

A station can activate a Message Wait indication requesting a Call Back when the called station does not answer or is in DND. A station may receive a Message Wait from any number of other stations in the system. The station receiving the Message Wait can return the calls to the station using the **[MSG/CALLBK]** button.

When a busy station is called, the calling user may request to be placed in a queue to receive a Call Back. When the called station returns to idle, the system signals the initiating station with Callback ring. When the user answers, the now idle station is called.

Operation

LIP Phone

To leave a Message Wait, while receiving ring back tone or no response on a call announce (H or P mode):
1. Press the **[MSG/CALLBK]** button; confirmation tone received.
2. Hang up, Message Wait is activated.

To leave a Message Wait, while receiving DND tone:
1. Press the **[MSG/CALLBK]** button; confirmation tone received.

2. Hang-up, Message Wait is activated.

To leave a Call Back (queue for a station), while receiving busy notification:
1. Press the **[MSG/CALLBK]** button; the user receives confirmation tone.
2. Hang up, to return to idle.

To respond to a Call back recall received when the busy station becomes available:
1. Lift the handset, or press the **[SPEAKER]** button.
2. Previously busy station is called.

To retrieve Station Messages Waiting:
1. Press **[MSG/CALLBK]** button; either the message contents summary will be shown as below.

1. ICM MWI(001)
2. VSF MSG(002)

2. Dial ‘1’ to select ICM MWI (Station Message Wait)
   - ‘1’ – ICM MWI, Station Message Wait,
   - ‘2’ – VSF MSG, VSF Message Wait

To return a call from the current Station Message:
1. Press the **[SAVE]** button.

To delete the first Message Wait from the list:
1. Press ‘*’ button
2. Press ‘1’ button to confirm the deletion, the list is updated removing the first station number in the list.

To delete all Message Waits:
1. Press ‘*’ button.
2. Press ‘3’ button.

SLT

To leave a Message Wait, while receiving ring back tone or no response on a call announce (H or P mode):
1. Momentarily press the hook switch.
2. Dial 56 (Message Wait/Call Back code).
3. Hang up, Message Wait is activated.

To leave a Message Wait, while receiving DND tone:
1. Momentarily press the hook switch.
2. Dial 56 (Message Wait/Call Back code).
3. Hang up, Message Wait activated.

To retrieve a Station Message Wait:
1. Dial 57 (Message Wait/Call Back Answer code).

To leave a Call Back (queue for a station), while receiving busy:

1. Momentarily press the hook switch.
2. Dial 56 (Message Wait/Call Back code).
3. Hang up, return to idle.

To respond to a Call back recall, received when the busy station becomes available:

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

Conditions

1. A Message Wait/Call Back return call will always ring at the receiving station overriding the Intercom signaling mode selected.
2. A station can leave only one callback request at a time.
3. If a station is attempting to leave a message and the system Message Wait queue is full, the station will receive ICM busy tone.
4. A Message Wait reminder tone can be enabled to remind the user of messages waiting.
5. A station in Call Forward can leave a message wait.
6. A Message Wait indication is left at the originally-called station even if the call is forwarded.
7. An LIP Phone with LCD may call back to the station(s) that left messages in any desired order, or the normal ("oldest first") order.
8. Placing an Intercom call to a station will cancel any existing Message Wait from that station.

Programming

VOICE CONFIG	System Data – Call Feature Timer – MSG Wait Reminder Tone Timer

Related Features

Message Wait Reminder Tone

Hardware

2.41.2 Message Wait Reminder Tone

Description

LIP Phones can be sent a tone as a periodic reminder to the user of message waits in queue. This tone is sent to the station only while idle and is heard over the speaker.

Operation

System

Reminder tone is sent to stations automatically when assigned.

Conditions

1. Interval set between tones can be 00 to 60 minutes; the 00 setting disables the reminder tone.

2. The reminder tone will continue until all messages have been retrieved.
3. A station that is busy or in DND will not receive the Message Wait Reminder tone until it returns to idle.

Programming

VOICE CONFIG System Data – Call Feature Timer – MSG Wait Reminder Tone Timer

Related Features

Message Wait/Call Back

Hardware

LIP Phone

2.42 PAGING

2.42.1 Paging & All Call Paging

Description

A station can connect and transmit voice announcements to any or all of the system Paging zones. Stations are grouped into “zones” to receive pages to the zone. Stations not assigned to any zone will not receive a page including All Call pages.

A page warning tone will be provided to the Paging Zone(s) prior to the audio connection. The user is allowed to continue the page for the specified Page Time-out timer after which the user is disconnected and the Paging Zone(s) is returned to idle.

The default Paging Zone dial access codes are as follows:

Paging Zones	501~510
All Call Page	500

Flexible buttons of an LIP Phone may be assigned to access a Paging Zone as a **{PAGING ZONE}** button.

Operation

LIP Phone

To assign a Flex button as a {paging zone} button:

1. Lift the handset, and press **[PGM]** + **{FLEX}** + Paging Zone number + **[SAVE]**

To make a page:

1. Lift the handset.
2. Dial the desired paging code or press a **{PAGING ZONE}** button.
4. After the Page Warning Tone, make announcement.

5. Replace the handset to return to idle.

To queue for a page when busy is received:
1. Press the **[MSG/CALLBK]** button.
2. Replace the handset to return to idle.

SLT

To make a page:
1. Lift the handset.
2. Dial the desired paging code.
3. After the Page Warning Tone, make announcement.
4. Replace the handset, to return to idle.

To queue for a page when busy is received:
1. Dial 56 (Call Back code).
2. Replace the handset returning to idle.

Conditions

1. Stations dialing a Page Code will be queued when any of the other Paging zones are busy.
2. If an LIP Phone user attempts to page using the speakerphone, pre-selection will be activated and display will show "LIFT THE HANDSET TO PAGE".
3. Stations receiving a page are considered idle for other incoming calls and ring will override Page announcements over an LIP Phone speaker.
4. Stations in DND or busy will not receive Page announcements.
5. A station accessing a Paging Zone is considered busy.
6. Stations which are not included in a Paging Zone will not receive any page, including All Call.
7. A station is permitted only one Paging Zone queue request at a time; if a station attempts another Paging Zone queue, only the last-received queue request is honored.
8. When a busy Paging Zone becomes idle, the system will select the oldest paging queue, and signal the appropriate station; the signaled station will have an audible ring (distinctive ring) indicating the queue callback.
9. The All Call Paging, while signaling the queued station, is considered busy; additionally, All Call Paging is considered busy when any paging zone is active.
10. The queue recall is always in tone ring mode regardless of the station's ICM signaling mode.
11. If the waiting station is idle, the Call Back ring signals the station for 15 sec., after which the queue is canceled and the next station in the queue is signaled.
12. If the waiting station is busy, and the Paging zone becomes available, the next idle station in the Paging Queue list is signaled and the busy waiting station is placed at the bottom of the Paging Queue list. If there is no idle next station in the Paging Queue, the Paging Queue is canceled.
13. When the waiting station goes to idle, and both a "Paging Queue" and "CO Call back Queue" exist, the Paging Queue is given priority.

Programming

VOICE CONFIG	Station Data – Paging Access System Data – Call Feature Timer – Paging Timeout Timer

Related Features

Meet Me Page Answer

Hardware

2.42.2 Meet Me Page Answer

Description

Any station may respond to a “Meet Me” Page request over a Paging Zone; the user can answer the page from any station and be connected to the paging party.

Flexible buttons of an LIP Phone may be assigned as a **{MEET ME}** button.

Operation

LIP Phone

To assign a Flex button as a {meet me} button:

1. Lift the handset, press **[PGM]** + **{FLEX}** + ‘511’ + **[SAVE].**

To answer a page with Meet Me Page:

1. Lift the handset, or press the **[SPEAKER]** button.
2. Dial 511 (Meet Me Page code) or press the **{MEET-ME}** button.
   Or
3. Press the [HOLD] button.

SLT

To answer a page with Meet Me Page:

1. Lift the handset to receive intercom dial tone.
2. Dial 511 (Meet Me Page code).

Conditions

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

Programming

VOICE CONFIG	System Data – Call Feature Timer – Paging Timeout Timer

Related Features

Paging & All Call Pag

Hardware

2.43 CO RING ASSIGNMENT

Description

Each station in the system can be programmed to provide an audible signal when the system detects an incoming call on specified CO lines. Separate ring assignments are made for Day, Night and Timed Ring operation mode. In addition, the audible signal at the station can be delayed by 1 to 9 ring cycles allowing other stations to answer the call first.

Operation

System

Operation of this feature is automatic.

Conditions

1. Separate assignments are made for stations to ring in the Day, Night, and/or Timed Ring mode.
2. A busy station receives the Muted ring or Call Waiting tones (as appropriate) for the station's off-hook ring assignment.
3. The system Ring mode can be selected manually or automatically. In Automatic mode, Day/Night selection is determined based on the Automatic Ring Mode Selection table; the Attendant has manual control over the Ring mode selection.
4. The Attendant's LCD displays Night and Timed Ring Mode and the **[DND]** button LED will flash.
5. If a CO line is not assigned to ring at any station, incoming calls on the CO line will ring the first available Attendant.

Programming

VOICE CONFIG	CO Line Data – Call Routing by Line CO Line Data – Ring Assignment Table System Data – Day/Night/Timed Schedule

Related Features

Day/Night/Timed Ring Mode
Off-Hook Signaling

Hardware

2.44 CO LINE RELEASE GUARD TIME

Description

To assure that the PSTN switching equipment has sufficient time to restore the idle status, the system will hold CO lines in a busy state to users after release of a CO line by a station. The time between station disconnect and when the system changes the CO line status from busy to idle is the CO Line Release Guard time.

Operation

System

Operation of this feature is automatic.

Conditions

Programming

VOICE CONFIG	System Data – Call Feature Timer – CO Release Guard Timer

Related Features

Hardware

2.45 IP TRUNKING

2.45.1 SIP Service

Description

When assigned to support Session Initiation Protocol (SIP), VoIP channels provide protocol conversion between SIP and the iPECS protocol. This permits the VoIP channel to connect to external SIP networks for call services. In addition, to the IETF RFC-3261 SIP draft standard, iPECS SBG-1000 VoIP channels support other SIP-related RFCs including:

RFC-2617	HTTP Authentication, Basic & Digest
RFC-3515	Refer Method
RFC-3264	Offer/Answer Model
RFC-3265	SIP Basic Call Flow Examples
RFC-3891	SIP “Replaces” Header

Using the SIP database assignments, the system will register and authenticate with the SIP proxy server permitting the system to interoperate employing SIP to establish, manage and terminate real-time voice sessions with external parties.

Operation

System

Operation of SIP Service is automatic.

Conditions

Programming

VOICE INSTALL	CO Line Registration – Server Information CO Line Registration – SIP ID Configuration

Related Features

Hardware

2.46 CALLING/CALLED PARTY IDENTIFICATION

Description

The iPECS SBG-1000 system receives calling party identification in SIP INVITE message or the ISDN call Set-up message, Calling Line Identification Presentation (CLIP). The answering party identification, which may be different from the called party, is received in SIP 200 OK message or the ISDN connect message, Connected Line Identification Presentation (COLP). When provided, the LCD of LIP Phones displays the identification, which is included in call records.

LIP Phone Display is shown:

```
LINE RINGING
CLI 03438502821
```

The system will also compare the identification to programmed Speed Dial bins; when a match exists, the Name of the Speed Dial bin displays in place of the number, CO Name display.

The system will send calling and answering party identification in the appropriate messages to SIP or the ISDN based on the database. Identification messages may be restricted, not reported, to the far-end user. Calling Line Identification Restriction and Connected Line Identification Restriction may be enabled in the system database or by **{CLIR}** and **{COLR}** Flex buttons.

Operation

System

Operation of this feature is automatic.

LIP Phone

To program **{CLIR}** button:

[PGM] + **{FLEX}** + **[PGM]** + ‘43’ + **[SAVE]**

To program **{COLR}** button:

[PGM] + **{FLEX}** + **[PGM]** + ‘44’ + **[SAVE]**

To activate CLIR or COLR, before placing or answering a SIP call or an ISDN call

1. Press the **{CLIR}** or **{COLR}** Flex button.

Conditions

1. This feature may not be available in the specific SIP/ISDN service area or may be a subscription service.

Programming

Related Features

Hardware

2.47 ANSWERING MACHINE EMULATION

Description

When a call is sent to a voice mail-box, the associated station can be assigned to notify the user and allow the user to screen the call. Two methods of notification and call screening are:

- Ring mode – the user is notified by the Answering Machine Emulation (AME) Flex button (if programmed), which will flash; the user may press the Flex button to screen the caller as the voice message is stored.
- Speaker mode - when the call is sent to the Voice Mail-box, the caller’s voice is automatically broadcast over the speaker of the user’s LIP Phone.

The user may terminate screening, and either leave the caller in voice mail to record a message, talk with the caller and record the conversation in the mail-box, or answer the call and disconnect the Voice Mail.

The user’s LIP Phone must be assigned with an AME Flex button for proper operation.

Operation

LIP Phone

To assign an {AME} button:

Ring Mode

1. Lift the handset, and press **[PGM]** + **{FLEX}** + ‘64’ + ‘0’ + **[SAVE]**

Speaker Mode

1. Lift the handset, and press **[PGM]** + **{FLEX}** + ‘64’ + ‘1’ + **[SAVE]**

To screen a call in the Ring mode:
1. Press the flashing **{AME}** button, the caller’s voice is broadcast over the station speaker and simultaneously stored in the Voice Mail-box.

To stop the voice broadcast/screening and leave the caller in Voice Mail:
1. Press the illuminated **[SPEAKER]** button.

To talk with the caller and record the conversation in Voice Mail:
1. Press the illuminated **[MUTE]** button.

To answer the call and cancel the voice message:
1. Press the illuminated **{AME}** button, the caller is connected and Voice Mail is disconnected.

Conditions

1. AME is supported only on an LIP Phone (an **{AME}** Flex button must be assigned on the phone).
2. If the user answers the call using the **{AME}** button, the caller is connected in the normal manner, the Voice Mail is disconnected, and any message recorded by the caller is not stored (when VSF is in use).

Programming

Related Features

VSF Integrated Auto Attd/Voice Mail

Hardware

LIP Phone

2.48 AUTO CALLED NUMBER REDIAL (ACNR)

Description

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

Operation

LIP Phone

To assign a Flex button as an {redial} button:
1. Press **[PGM]** + **{FLEX}** + **[PGM]** + ‘54’ + **[SAVE]**

To activate ACNR while receiving busy, no answer:

1. Press the **{REDIAL}** button or **[ACNR]** soft button.
2. Hang-up handset, or press **[SPEAKER]**.

To cancel ACNR while idle:

1. Press flashing **{REDIAL}** button or **[STOP]** soft button.

To cancel ACNR during an ACNR attempt:

1. Lift the handset or press the **[MUTE]** or flashing **{REDIAL}** button.

System

1. The system initiates the ACNR process, starting the ACNR Pause Timer.
2. At expiration of the timer, the system attempts the previous call.
3. When the called party answers, the calling user may answer by lifting the handset or using speakerphone to communicate with called party.

Conditions

1. Four timers and a retry counter can be programmed.
   - ACNR Pause Timer – Time allowed between ACNR attempts.
   - ACNR Delay Timer – At expiration of Pause Timer, if no line is available, the system will wait for delay timer before retry attempt.
   - ACNR Tone Detect – After dialing, the system will abandon retry if no tone or answer is detected within the Tone Detect time.
   - ACNR Retry Count – Count determines the number of times system will retry before ACNR is automatically cancelled.
2. The call will be placed on the same path as originally used; if the path is busy, an available CO line in the same group will be seized.
3. The ACNR Retry Counter decreases by one when the system completes the dialed number.
4. When the ACNR Pause Timer expires, if the station is in a busy state, the ACNR Delay Timer is invoked.
5. Upon completion of dialing, the system will monitor the call for progress signals.

Programming

VOICE CONFIG	System Data – Call Feature Timer – ACNR Delay Timer
	System Data – Call Feature Timer – ACNR Pause Timer
	System Data – Call Feature Timer – ACNR Retry Counter
	System Data – Call Feature Timer – ACNR Tone Detect Timer

Related Features

LNR (Last Number Redial)
Speakerphone
Mute

Hardware

LIP Phone

2.49 AUTO RELEASE OF [SPEAKER]

Description

After completion of certain features, the **[SPEAKER]** turns off automatically, returning the LIP Phone to idle.

Operation

System
Auto Release of [speaker] operation is automatic for supported features.

Conditions

1. This feature applies to all User and Attendant Programming except CO line Disable and Version Display.
2. Auto Release of **[SPEAKER]** also applies to features including Call Park, Call Back, Call Forward and CO Queuing.
3. If, during Station User Programming, erroneous data is entered, error tone is received and the user must correct the error before the station will return to idle automatically.

Programming

Related Features

Hardware

LIP Phone

2.50 AUTOMATIC SPEAKER SELECT

Description

LIP Phones can access a CO line or an internal circuit by pressing the appropriate button without the need to lift the handset or press the **[SPEAKER]** button. Audio from the CO line or called station is sent to the speaker as if the user pressed the **[SPEAKER]** button and the speakerphone's MIC is activated.

Operation

LIP Phone

To access an internal or external system resource:

1. Press an assigned **{FLEX}** button.

Conditions

1. For LIP Phones not equipped/assigned with speakerphone, the user must lift the handset to be heard.
2. Paging while on the speakerphone may cause feedback from the paging equipment; if Auto Speaker is enabled and a **{PAGING ZONE}** button is pressed, the display will show "LIFT THE HANDSET". To complete the page, the user must lift the handset within the predefined 5-second period or the Station will return to idle.

Programming

Related Features

Hardware

LIP Phone

2.51 CALL LOG DISPLAY

Description

Users with LIP Phones that have Soft keys (8012D and 8024D) can view a log of incoming, outgoing and missed calls on the display.

Operation

LIP Phone

To access the Call Log menu:

1. Press the **{LOG}** soft button; the following will display,

```
01. ► CO2  12345678
02. ◄ CO2  23456789
  BACK    SELECT     SEND
```

2. Use the Up/Down Navigation keys to view the other log contents.

Conditions

Related Features

Hardware

LIP 8012D, 8024D Phone

2.52 CALL WAIT

Description

When a busy LIP Phone receives a incoming CO call, the muted ring is heard, giving a audible indication of the call; DID Call Wait must be enabled in Station User Programming.

Operation

When programmed, operation of this feature is automatic.

Conditions

1. The incoming CO call will follow the call routing defined in Exceptional Call Routing after the expiration of the DID/DISA no answer timer expires.
2. The LIP Phone must have an appearance button programmed for the CO line.
3. Assigning the CO line with ICLID routing automatically disables DID Call Wait.

Programming

VOICE CONFIG	Station Data – Common Attributes – DID Call Wait System Data – Call Utility Timer – DID/DISA No Answer Timer

Related Features

Call Routing

Hardware

LIP Phone

2.53 DND - ONE TIME DND

Description

A station can reject and terminate a ringing or off-hook muted ringing call by pressing the **[DND]** button. When the station returns to the idle status, DND is cancelled and the **[DND]** LED extinguishes.

Operation

LIP Phone

To activate One Time DND while on a call:

1. When receiving a call, press the **[DND]** button; the **[DND]** LED illuminates, and station enters DND status.

System

To deactivate DND:

1. When the station returns to idle, DND is disabled and the **[DND]** LED extinguishes.

Conditions

1. CO recalls override One Time DND.
2. The Attendant can override stations in One Time DND using Camp-On or Intrusion; the Attendant Station does not have One Time DND service.
3. One Time DND cancels existing Callback requests.
4. When DND is activated, the DND message "DO NOT DISTURB STA [XXX]" will display to a Station attempting Camp-on at the called Station.
5. DND LED illumination is only applied while muted ringing.

Programming

Related Features

Call Waiting/Camp-On
DND (Do Not Disturb)

Hardware

LIP Phone

2.54 FLEX BUTTON DIRECT SPEED DIAL ASSIGNMENT

Description

A user may program a telephone number directly to a Flex button without the need to assign the number to a Speed Dial bin. In this case, the telephone number is allocated to the highest numbered Individual Speed Dial bin available.

Operation

LIP Phone

To assign a telephone number to a Flex button:

1. Press the **[PGM]** button.
2. Press the desired Flex button.

3. Press the soft button below the “TEL NUM” display selection.
4. Dial the telephone number.
5. Press the **[HOLD/SAVE]** button
6. Dial the name to be associated with the number (optional).
7. Press the **[HOLD/SAVE]** button.

To place a call using the Flex button:
1. Lift the handset or press the **[SPEAKER]** button.
2. Press the assigned Flex button.

Conditions

1. This feature is available to LIP phone users only.
2. When a Flex button is assigned with a telephone number, the system will allocate the number to the highest available Individual Speed Dial bin number; if no bin is available, the user will receive an error tone when attempting to assign the telephone number.
3. The telephone number may include any of the special Speed Dial instructions **(**display security, etc.).

Programming

Related Features

Individual Speed Dial

Hardware

LIP 8012D, 8024D phone

2.55 INTERCOM ANSWER MODE

Description

Each LIP Phone can select the answer mode used for incoming ICM calls while the station is idle. There are three answer modes available:

- Handsfree (H) – When an ICM call is received, the user receives splash tone followed by the ICM caller’s voice; the user may respond to the caller without lifting the handset or pressing the **[SPEAKER]** button.
- Privacy (P) – When an ICM call is received, the user receives splash tone followed by the ICM caller’s voice; to respond, the user must lift the handset or press the **[SPEAKER]** button.
- Tone (T) – An ICM call will cause the LIP Phone to provide an audible ICM ring tone; the user must lift the handset or press **[SPEAKER]** to answer.

📖 ***NOTE:*** *A SLT always functions in Tone ring mode.*

Operation

LIP Phone

To change ICM Answer Mode:

1. Press **[PGM]** button; the **[SPEAKER]** button LED lights steady.
2. Dial 11 (Station User Program code); confirmation tone is received.
3. Dial the desired ICM Answer Mode code (1=H, 2=T, 3=P).
4. Press the **[SAVE]** button.

Conditions

1. Regardless of ICM Answer Mode selected by the user, Message Wait, Callback, Call Forward and Attendant Override will ring in Tone mode.
2. Page announcements are not affected by ICM Answer Mode Selection.
3. The default ICM Answer Mode is Tone ring; the active mode is stored in battery-protected memory.

Programming

Related Features

Intercom Call (ICM Call)
Paging
Message Wait/Call Back
Call Forward

Hardware

LIP Phone

2.56 MUTE

Description

An LIP Phone can turn off audio transmission from the handset, speakerphone or headset microphone (Mic Mute).

Operation

LIP Phone

To Mute the Microphone:

1. Press the **[MUTE]** button; the **[MUTE]** button LED illuminates, the microphone (Handset, Speakerphone, Headset) is muted, and the connected party receives silence.

To activate the microphone:

1. Press the illuminated **[MUTE]** button; the **[MUTE]** button LED extinguishes, and the microphone is activated, transmitting audio to the connected party.

Conditions

1. Changing from speakerphone to handset or vice versa during a mute condition will eliminate mute status.
2. Returning to idle or placing another CO or intercom call will deactivate mute status and return to the normal (active microphone) status.

Programming

VOICE CONFIG Station Data – Common Attributes – Headset Ring

Related Features

Speakerphone
Headset Compatibility

Hardware

LIP Phone

2.57 OFF-HOOK SIGNALING

Description

Off-hook Signal is a muted ring signal delivered to the LIP Phone speaker. When an off-hook station receives a call, or a CO call rings into the system for the off-hook station, the station will receive the assigned Off-hook Signal (ring), or a Camp-On in the case of ICM calls (Voice-Over Announcement or Off-hook ring signal may be received).

Operation

System

Operation of Off-hook ring signals is automatic.

Conditions

1. While using the speakerphone, a Camp-On tone is provided over the speaker in place of the assigned Off-hook ring Signal.
2. DND overrides and terminates any Off-hook signaling.
3. Off-hook ring signals terminate when the call is answered, forwarded, or abandoned.

4. A Station that receives an off-hook signal will receive normal ring signaling once the Station returns to idle.

Programming

Related Features

Call Waiting/Camp-On
CO Ring Assignment
DND (Do Not Disturb)
DND - One Time DND

Hardware

LIP Phone

2.58 ON-HOOK DIALING

Description

LIP 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 or Automatic Speaker Select, a dial tone is received and the user may dial the desired number.

Operation

LIP Phone

To activate On-Hook Dialing:

1. Press the **[SPEAKER]** button; dial tone is received and the **[SPEAKER]** button LED lights.
2. Place desired call (dial station ICM number, or select CO path and dial).

Conditions

1. If the outgoing call is not answered, the user must press the illuminated **[SPEAKER]** button to return to idle.
2. When the speakerphone is used, the microphone is active unless the **[MUTE]** button is pressed (**[MUTE]** button LED is illuminated).

Programming

Related Features

Mute
Speakerphone

Automatic Speaker Select
Headset Compatibility

Hardware

LIP Phone

2.59 SAVE NUMBER REDIAL (SNR)

Description

The last dialed number on a CO call may be stored (up to 23 digits) in a buffer for future use. This number is saved in memory until the user requests a new number be stored. Numbers dialed for subsequent calls do not affect the Save Number buffer.

Operation

LIP Phone

To save a dialed number, while on a CO call:

1. After dialing (before hanging up), press the right navigation button.
2. Press the soft **[SAVE]** button.

To dial a saved number:

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the soft **[DIR]** button.
3. Press the soft **[SPEED]** button.
4. Press '#'.

Conditions

1. The saved number can be a maximum of 23 digits.
2. Dialing the saved number will automatically seize the CO line that was used for the original call. If the CO line is busy, a CO line from the same group will be selected and the saved number dialed. If all CO lines from the group are busy, the user will receive All Lines busy tone and may queue.
3. If there is no **{CO}** button, the call will be presented on a **{LOOP}** button.
4. Save Number Redial is protected from power failure.

Programming

Related Features

Individual Speed Dial
Common Speed Dial
LNR (Last Number Redial)

Hardware

LIP Phone

2.60 SPEAKERPHONE

Description

LIP Phones equipped with speakerphone circuitry enable the telephone to be used hands-free in two-way conversations.

Operation

LIP Phone

To activate the Speakerphone:

1. Press the **[SPEAKER]** button; **[SPEAKER]** LED lights steady.

To switch from Handset to Speakerphone:

1. When Handset is in use, press the **[SPEAKER]** button; **[SPEAKER]** LED lights steady.
2. Replace Handset, and Speakerphone is activated.

To terminate a Speakerphone call:

1. When Speakerphone is in use, press the **[SPEAKER]** button; **[SPEAKER]** LED extinguishes.

Conditions

1. If Automatic Speaker Select is enabled for the station, pressing a DSS, CO/LOOP or Speed Dial button will automatically activate the speakerphone.
2. The **[MUTE]** button LED indicates the status of the Microphone; when lit, the Microphone is inactive.
3. When Headset operation is assigned for the station, the Speakerphone is disabled and the **[SPEAKER]** button activates the Headset audio path instead of the speaker.

Programming

VOICE CONFIG	Station Data – Common Attributes – Headset Ring

Related Features

Mute
Automatic Speaker Select
Headset Compatibility

Hardware

LIP Phone

2.61 STATION FLEXIBLE BUTTONS

Description

The LIP Phone incorporates a field of "Flex" buttons as well as the fixed feature buttons. The Flex buttons are assigned in the system database to access features, functions or resources of the system. Specifically, Flex buttons can be assigned as:

- Empty button – has no system database assignment.
- **{DSS/BLF}** button – used to place One-touch ICM calls to a designated station and display Station status.
- Flex Numbering Plan button – activates the feature associated with the assigned digits from the Flexible Numbering Plan.
- Speed Dial bin button – accesses and dials the number from the assigned Speed Dial bin.
- Loop button – provides an appearance for incoming CO calls when a direct CO appearance is not available; the Loop button LED provides the status for the duration of the call (must be programmed using Web. Admin.).
- Station User Program Code button – accesses or activates the special features available with Station User Program Codes (refer to Section 3.67).
- CO Line Appearance button – provides access to the individual CO Line assigned to the Flex button. The CO Line button LED provides the status of the CO Line. This button is only available in an attendant station.

With the exception of CO Line buttons and Loop button, Flex buttons can be assigned at the station by the end-user.

Operation

LIP Phone

To assign a Flex button at the station:

1. Press the **[PGM]** button.
2. Press the desired Flex button.
3. Dial the digits from the Flexible Numbering Plan.
4. Press the **[SAVE]** button.

OR

1. Press the **[PGM]** button.
2. Press the desired Flex button.
3. Press the **[PGM]** button.
4. Dial the digits from the Station User Program Code (refer to Section 3.67) or Fixed Number Plan.
5. Press the **[SAVE]** button.

Conditions

1. The **{LOOP}** buttons provide a status indication for the call as long as the station has supervisory control.
2. A station may have multiple **{LOOP}** buttons.
3. The priority for the appearance of a CO call transfer is first a direct CO Line appearance (**{CO}** button), if not available, a **{LOOP}** appearance is used. If there is no appearance available, the transferring station recalls immediately.

Programming

VOICE INSTALL	Numbering Plan
VOICE CONFIG	Station Data – Flex Buttons

Related Features

Flexible Numbering Plan
Station User Programming & Codes

Hardware

LIP Phone

2.62 STATION USER PROGRAMMING & CODES

Description

LIP Phone users can program an array of functions and features, access status information and assign special features codes to Flex buttons. The Station User Program Codes used for these purposes are fixed as listed, and shown below.

Table 2.72-2 Station User Programming

Code	Description	Entries
11	ICM Answer Mode	1: H, 2: T, 3: P
12	Headset/Speakerphone mode	0:H, 1:S
13	Select Headset Ring type	1:S, 2:H, 3:Both
14	Register Bluetooth device	
15	Activate Bluetooth device	
16	Intercom Differential Ring	1~8
17	CO Differential Ring	1~8
18	Station Ring Download	5~8 + 0~9
21	Set Wake-Up Time	Once/Permanent & Hour/Min
22	Erase Wake-Up Time	
31	LCD Display Language	Domestic/English

Code	Description	Entries
32	Sys version display	
33	Select BGM source	(0~1)
34	User Name registration	'name'
35	Display Phone IP Address	
36	Display Phone MAC Address	
37	Display Phone Version	
38	Network Configuration	
41	Forced Forward to Destination	Station Group Number
42	**{Call Log Display}** button	
43	CLIR Service	
44	COLR Service	
45	Register Authorization Code	Auth Code
4*	LOOP button	
50	CALLBACK button	Button PGM only
51	CONF button	Button PGM only
52	MUTE button	Button PGM only
53	ICM button	Button PGM only
54	REDIAL button	Button PGM only
56	Activate Mobile Extension button	Button PGM only
57	DID Call Wait	Button PGM only

In addition, a Station User Program Menu display is provided by the LIP Phone display to assist the user in programming Station User Program Code features and functions.

- **[VOL▲]**/**[VOL▼]** buttons – used to scroll through menu items and the dial pad is used to enter a selection.
- Program Codes – also used to assign a function/feature to a Flex button.

USER PROGRAM MENU Displays:
First top-level Menu selection

[1] KEYSET [2] WAKE UP TIME

Under selection [1] Keyset, select 1~8 as below

[1] ANSWER MODE [2] HEADSET OR SPK MODE

[3] HEADSET RING MODE [4] REGISTER BLUETOOTH

[5] BLUETOOTH USAGE [6] STA RING TYPE

[7] CO RING TYPE [8] STA RING DOWNLOAD

Under selection [2] Wake Up Time, select 1~2 as below

[1] SET WAKE UP TIME
[2] WAKE UP DISABLE

Next top-level Menu selection

[3] SUPPLEMENTARY
[4] SERVICES

Under selection [3] Supplementary, select 1~8 as below

[1] LCD DISPLAY LANGUAGE
[2] iPECS SBG-1000 INFO

[3] BGM
[4] REGISTER STA NAME

[5] DISP PHONE IP ADDR
[6] DISPLAY MAC ADDR

[7] DISP PHONE VERSION
[8] NETWORK CONFIG

Under selection [4] Services select 1~5 as below

[1] FORCED FWD TO DEST
[2] CALL LOG DISPLAY

[3] CLIR SERVICE
[4] COLR SERVICE

[5] USER AUTH REGIST
[1] FORCED FWD TO DEST

Operation

LIP Phone

To assign a Station User Program Code to a Flex button:

1. Press the **[PGM]** button; the Station User Program Menu is displayed.
2. Press the desired Flex button.
3. Dial the desired Station User Program Code and additional inputs that may be required.
4. Press the **[SAVE]** button.

To activate a Station User Program Code feature or function:

1. Press the **[PGM]** button; the Station User Program Menu is displayed.
2. Use the **[VOL▲]/[VOL▼]** buttons (as needed) to display the desired menu item, or dial the desired Station User Program Code and additional inputs as required.

Programming

Conditions

Related Features

Station Flexible Buttons
Station Message Wait/Call Back
Wake-Up Alarm
Headset Compatibility
Attendant Station Program Codes

Hardware

LIP Phone w/Display

2.63 VOICE OVER

Description

Voice Over allows LIP Phones to receive a voice announcement through the handset receiver while off-hook on a call (CO or Intercom). The Voice Over is muted so as not to interfere with the existing conversation. The called station user may then respond to the calling party using Camp-On response or DND.

Operation

LIP Phone

Placing a Voice Over (OHVO) while receiving busy:

1. Dial '#'.
2. After splash tone, begin announcement.

Responding to a Voice Over announcement:

1. Use Camp-On response procedure or One-Time DND.

SLT

Placing a Voice Over (OHVO) while receiving busy:

1. Dial '#'.
2. After splash tone, begin announcement.

Conditions

1. When the called station responds via Camp-On, all conditions and options available to Camp-On apply.
2. OHVO may be used to notify the called party of a transferred call (CO Line or Intercom) by announcing the call then releasing to complete the transfer.
3. When a call is transferred via OHVO the receiving station will receive a ringing after the transfer is complete.
4. If the receiving station is in conference or using the Speakerphone, Voice Over is not available; Camp-On will be activated and a Camp-On tone sent to the receiving station.
5. If the receiving station is SLT or SIP extension, Voice Over is not available.

Programming

VOICE CONFIG	Station Data – Common Attributes – Voice Over

Related Features

Call Waiting/Camp-On

Hardware

LIP Phone to receive Voice Over

2.64 ATTENDANT POSITION

Description

By default, Station 10 is the Attendant on the iPECS SBG-1000 system. The Attendant position must be equipped with an LIP multi-button Phone.

Operation

Condition

1. Attendant is assigned as Station 10 (default, logical number).
2. Attendant calls and recalls are always routed to the attendant.

Programming

Related Features

Hardware

LIP Phone

2.65 ATTENDANT RECALL

Description

Unanswered or abandoned CO calls that remain unanswered for the Hold or Transfer Hold timer (when appropriate), will recall the station placing the call on hold. If the call remains unanswered for the assigned Recall time, the Attendant will receive a recall. Both the Attendant and station will receive the recall signal for the Attendant Recall Timer period after which the system will disconnect and return the CO line to idle.

Operation

System
Attendant recall operation is automatic.

Conditions

Programming

VOICE CONFIG	System Data – Call Feature Timer – Attendant Recall Timer System Data – Call Feature Timer – I-Hold Recall Timer

Related Features

Hold
Call Transfer

Hardware

2.66 ATTENDANT STATION PROGRAM CODES

Description

Using the Attendant Station Program Codes, the Attendant can print SMDR and Traffic reports on-demand, control certain user features, record VSF announcements, and enable/disable Auto Service Mode Control, etc. Items are available using the Program Code directly or scrolling through the multi-level display menu. The following indicates the menu displays, including the digit for selecting the item, the item description and further required entries. The various levels of the display menu are indicated by indentation.

Operation

Attendant
To activate an Attendant Station Program Code feature or function:

1. Press the **[PGM]** button, the Attendant Station Program Menu is displayed.

2. Dial '0' to access Attendant Station Program codes (refer to Attendant Station Program Codes below).
3. Enter desired code, or use the **[VOL▲]/[VOL▼]** buttons to display the desired menu item and enter the desired code.
4. Dial additional inputs, as necessary.

Table 2.72-3 Attendant Station Programming

[1] PRINT
- [1] SMDR
  - [1] PRINT SMDR STA BASE
    - station range input
  - [2] DELETE STATION BASE
    - station range input
  - [3] DISPLY CALL CHARGE
  - [4] ABORT PRINTING
  - [5] PRINT LOST CALL
  - [6] DELETE LOST CALL
- [2] TRAFFIC
  - [1] PRINT ALL SUMMARY
    - enter Analysis time & type
  - [2] PRINT ALL PERIDICLLY
    - enter Analysis time, type & Print time
  - [3] ABORT PERIDIC PRINTING
  - [4] PRINT ATD TRAFFIC
    - enter Analysis time & type
  - [5] PRINT CALL SUMMARY
  - [6] PRINT CALL HOURLY
    - enter Analysis time & type
  - [7] PRINT H/W USAGE
    - enter Analysis time & type
  - [8] PRINT CO SUMMARY
    - enter Analysis time & type
  - [9] PRINT CO HOURLY
    - enter CO Group

[2] CLOCK/WAKEUP
- [1] LCD DATE MODE CHANGE
- [2] LCD TIME MODE CHANGE
- [3] ATD SET WAKE UP TIME
  - enter station range
- [4] ATD WAKE UP DISABLE
  - enter station range
- [5] SYSTEM DATE TIME SET

[3] STATION SET

[1] REG STATION NAME
enter station number
[2] DND/FWD CANCEL
enter station range
[3] LCD Display Language
select language
[4] SET ICM ONLY MODE
enter station range
[5] RESTORE COS
enter station range

[4] ISOLATE CO FAULT

[5] REC VSF ANNCEMENT
enter VSF Announcement (001~072)

[9] USB UPGRADE

[#] WHTU SUBSCRIBE

Condition

Programming

Related Features

SMDR (Station Message Detail Recording)
Traffic Analysis
VSF Integrated Auto Attd/Voice Mail
Dial-by-Name
Station User Programming & Codes

Hardware

LIP Phone

2.67 ATTENDANT CALL/QUEUING

Description

Any station can call the Attendant by dialing the Attendant Call code ‘0’. When an Attendant call encounters a busy signal, the call is queued to the Attendant; the call will be delivered to the Attendant.

Operation

To call the Attendant:
1. Dial Attendant Call Code.

Condition

1. The ICM calling party will receive ring-back tone or MOH (as specified) while queuing.
2. Calls to the Attendant's station intercom number are sent to the Attendant station dialed as with any intercom call.

Programming

VOICE CONFIG Station Data – Station Hold Music

Related Features

Attendant Position
Intercom Call (ICM Call)

Hardware

2.68 DISABLE OUTGOING CO ACCESS

Description

The Attendant can place CO lines out-of service, disabling outgoing calls on the CO path. This is normally done in the event of an undetected fault interrupt service on a CO path. Incoming calls continue to be processed normally.

Operation

Attendant

To disable/enable Outgoing CO access (toggle):
1. Press the **[PGM]** button.
2. Dial 04 (Attendant Station Program code).
3. Press the **{CO}** button of the line(s) to be disabled; confirmation tone is heard and the selected line(s) changed to inactive status.

Conditions

1. If the desired CO line is in use, the Attendant may still disable the CO line; the feature will take effect after the desired CO line returns to idle.
2. Once the line is disabled, the Attendant appearances for the disabled CO line will flutter at 240 ipm (inactive status), while other stations will show the CO line as busy (LED solid On).

3. The CO line outgoing access status is stored in battery-protected memory in case of a power failure.
4. Multiple CO lines may be enabled/disabled without redialing the Attendant Station Program code; confirmation tone is heard after each CO line is enabled/disabled.
5. Incoming calls on a disabled CO line will continue to operate normally.

Programming

Related Features

Attendant Position

Hardware

2.69 FEATURE CANCEL

Description

The Attendant can cancel features such as DND and Call Forwarding that are active at other stations.

Operation

Attendant

To deactivate DND/Call Forward at other stations:

1. Press the **[PGM]** button.
2. Dial 032 (Attendant Station Program code).
3. Dial the desired station range, or the same station number twice for a single station.
4. Press the **[SAVE]** button; a confirmation tone is heard, and the Attendant station returns to idle status.

Conditions

Programming

Related Features

Call Forward
DND (Do Not Disturb)
Attendant Position

Hardware

2.70 SLT BROKER CALL

Description

Broker Call allows the SLT user to engage in 2 calls at once, alternating between the two parties, so that the conversation with each party is private.

There are two types of Broker Call, Transfer and Camped On:

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

Operation

SLT

To activate a Transfer Broker Call:

1. While on an active call (external or intercom), momentarily press the hook-switch, intercom dial tone received and call is placed in Exclusive Hold state.
2. Place second call.
3. To alternate between calls momentarily press the hook-switch.

To activate a Camp-On Broker Call:

1. While on an active call (external or intercom).
2. Receive a Call Waiting/Camp-On tone.
3. Momentarily press the hook-switch, intercom dial tone received and call is placed in Exclusive Hold state.
4. Dial 66 (Camp-On Answer feature code); camped-on call is connected.

To alternate between the calls:

1. Momentarily press the hook-switch.
2. Dial 66 (Camp-On Answer feature code).

Conditions

1. After performing the hook-switch flash, if the call results in an error, busy, no answer or an abnormal state, the SLT user may momentarily press hook-switch to retrieve the held call.
2. During a Transfer Broker Call, if the SLT user goes on-hook, the Broker Call parties are connected completing a Call Transfer.
3. During a Transfer Broker Call, if the active caller disconnects from the SLT user, the held party (if another station), is connected to the SLT.
4. If the held party is a CO call, the SLT user receives error tone and may go on-hook to receive recall and retrieve the held call
5. During a Camp-On Broker Call, if the SLT user goes on-hook, the active call is disconnected and the held call recalls to the SLT.

6. During a Camp-On Broker Call, if the active party disconnects from the SLT, the SLT user receives error tone; the SLT user may momentarily press the hook-switch to retrieve the held party or go on-hook and receive recall.
7. If after the hook-switch flash, the user takes no action for the dial tone timer, the SLT will receive an error tone; if the SLT returns to an on-hook state, the SLT automatically will receive a recall ring.

Programming

Related Features

Message Wait/Call Back
Call Waiting/Camp-On
Call Transfer

Hardware

2.71 SLT HOWLER TONE

Description

When a SLT station goes off-hook and does not initiate dialing in the Dial tone timer duration, delays dialing between digits in excess of the inter-digit time, or stays off-hook at the completion of activating a feature or program, the station will receive the howler tone as an error indication and the call attempt will be abandoned. In order to complete the call, the user must return to on-hook and restart the call.

Operation

System

The system will deliver howler tone automatically, as required

Conditions

1. Howler Tone is sent after a period, of about 30 sec. of error tone.
2. Lock-out occurs when howler tone starts.

Programming

Related Features

Intercom Lock-Out

Hardware

2.72 DIALING RESTRICTIONS

2.72.1 Class of Service

Description

Dialing privileges can be assigned for each station; the dialing privileges are designated according to the Station Class of Service (COS) assignments as shown in the following tables. Users placing an outgoing call or dialing after answering a call will be allowed one of the four Station COS privileges assigned.

Table 2.72-1 Station Class of Service

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

- Toll Exception Tables – Each Toll Exception Table permits entry of 50 Allow codes and 50 Deny codes. Each code can contain up to 20 digits including digits 0-9, "#" as a wild card (any digit) and "*" as the end of entry mark (refer to Station Class of Service table to determine application of Toll Exception).
- Exception Table process – As digits are dialed, they are compared to entries in the appropriate Exception Table. Based on the Allow and Deny entries, the system applies the following rules to allow or deny calls.
  - Rule 1 – If a table has no entries, no restrictions are applied.
  - Rule 2 – If there are only Deny entries, restrictions are provided as Deny only.
  - Rule 3 – If there are only Allow entries, restrictions are provided as Allow only.
  - Rule 4 – If there are both Allow and Deny entries, the Deny entries are searched. If the dialed number matches a Deny entry, the call is restricted; if no match is found the call is allowed.

Operation

System

The system automatically applies the assigned COS.

Conditions

1. Dialing privileges are based on Station COS.

Programming

VOICE CONFIG	Station Registration – Authorization Code & COS System Data – Toll Exception Table

Related Features

Day, Night & Timed Station COS
Temporary Station COS/Lock

Hardware

2.72.2 Day, Night & Timed Station COS

Description

Each station is assigned a COS for three modes: Day, Night and Timed service modes. The service mode is generally controlled by the System Attendant. Based on the mode, appropriate dialing privileges are established.

Operation

System

Dialing restrictions are automatically applied based on COS assignments:

Conditions

Programming

VOICE CONFIG	Station Registration – Authorization Code & COS System Data – Toll Exception Table

Related Features

Class of Service
Temporary Station COS/Lock
Day/Night/Timed Ring Mode

Hardware

2.72.3 Temporary Station COS/Lock

Description

The Attendant can change the Station's Class of Service to temporarily preventing unauthorized toll dialing from the station (ex., lock the station). The station is still allowed to place internal calls and Emergency number calls.

Operation

System Attendant

To activate Temporary COS:

1. Press the **[PGM]** button.
2. Dial 034 (Temp COS code).
3. Dial the Station range.
4. Press the **[SAVE]** button.

To restore the assigned COS:

1. Press the **[PGM]** button.
2. Dial 035 (Restore COS code).
3. Dial station range
4. Press the **[SAVE]** button.

Conditions

1. The station is restored to the Station COS as appropriate for the active service mode, Day, Night, or Timed.

Programming

VOICE CONFIG	Station Data – Authorization Code & COS System Data – Toll Exception Table

Related Features

Class of Service
Day/Night/Timed Ring Mode

Hardware

2.73 SIP EXTENSION SERVICE

Description

The iPECS SBG-1000 system supports standard SIP phones; compatible SIP phones support the IETF standard RFC3261 for real-time communications over the internet. Once registered, iPECS SBG-1000 will deliver services to the SIP Phone.

Three steps should be followed for SIP phones to be registered to iPECS SBG-1000 and receive services from the system: SIP phone Lock key installation, Station User Login configuration for SIP phone, and SIP phone configuration.

Operation

Web Admin

To install the Lock key:

1. Enter Web Admin.
2. Select Voice Maintenance>Appliances Control>Lock Key Install.
3. Enter the proper Lock key in SIP Phone field.
4. Click **[Save]** button

To configure Station User Login for SIP phone:

1. Enter the Web Admin.
2. Select Voice Install>Station Registration>Station User Login.
3. Enter ID, Password and Desired Number.
4. Click **[Save]** button

WIT-400H

1. Press Menu + 8.Settings + 1.Profile Settings.
2. Edit System Default profile or add new profile about WLAN and network configuration.
3. Press Menu + 8.Settings + 2.SIP Setting.
4. Edit Phone Number, Display Name, Password, SIP Domain, Proxy IP and Proxy Port. The Phone Number must be the desired station number and also the SIP Domain and the Proxy IP should be the iPECS SBG-1000 LAN IP address.
5. Press Menu + 8.Settings + 3.Provisioning Setting.
6. Edit Address. The address should be the iPECS SBG-1000 LAN IP address.
7. Press Menu + 8.Settings + 1.Profile Settings.
8. Select the profile which you want to connect.

Please refer to the WIT-400H User Guide for the details.

SIP Phone

To set-up the SIP Phone:

1. Configure SIP Phone settings (ex. IP address, Subnet mask, Gateway, Telephone number, Proxy address, Expiration timer, Domain address etc.). The Telephone number must be the desired unused station number and also the proxy address and the domain address should be the iPECS SBG-1000 IP address (refer to the SIP phone User Guide for further information).
2. Boot the SIP Phone, which will register it with iPECS SBG-1000.

Conditions

1. Up to 6 SIP phones will be supported with Lock key. Ericsson-LG SIP extensions such as WIT-400H and LIP-8002 can be used without Lock key.
2. Desired Number for SIP phone should exist in station number list and also should be not assigned for other extension.
3. The Station User name will be overwritten by the SIP Phone Display Name setting.

4. If the Station number is changed in the iPECS SBG-1000 database, the SIP Phone should be reconfigured and re-registered with the changed telephone number.
5. When SIP Phones are used with iPECS SBG-1000, service tones from the system will not be heard (Confirmation tone, etc.).
6. iPECS SBG-1000 can not support full system feature with SIP extension such as WIT-400H, LIP-8002 and etc. For example, tones and LCD display messages for SIP extension can not be fully controlled by iPECS SBG-1000.

Programming

VOICE MAINT	Appliances Control – Lock Key Install
VOICE INSTALL	Station Registration – Registration Table

Related features

Hardware

2.74 PRIME LINE IMMEDIATELY/DELAYED

Description

When a user goes to an off-hook state, the system normally provides ICM dial tone. If desired, a station can be assigned to access a pre-assigned system resource (Prime Line). The Prime Line can be any of the Idle Line Settings:

- Seize a CO Line,
- Call another station,
- Activate a Flex button feature.

Prime Line access can be defined as immediate or delayed. When assigned immediate, upon an off-hook event, the system provides access to the Prime Line. With Delayed Prime Line, the station user receives normal Intercom dial tone for the Prime Line Delay timer and after the delay, the Prime Line is accessed.

Operation

LIP Phone

To access the station’s Prime Line

1. Lift the handset or press **[SPEAKER]** button and take no action, Prime Line as assigned will be accessed.

Conditions

1. Any of the station’s Flex buttons may be assigned as the Prime Line. When the user lifts the handset or presses the **[SPEAKER]** button, the system will act as if the user had pre-selected the button prior to going off-hook.

2. Selection of another Flex button or Feature button just prior to an off-hook event will override the Prime Line assignment.
3. When Delayed Prime Line is set, the user must wait, taking no action until the Prime Line is accessed. The user receives ICM dial tone during this period and may dial any valid numbering plan digit(s) or select a Flex button or feature button.
4. If the Prime Line Delay Timer is greater than Dial tone timer, the Delayed Prime Line will not activate. It will be necessary to reduce the delay timer or extend the Dial tone timer.

Programming

VOICE CONFIG	Station Data – Common Attributes – Prime Line Station Data – Common Attributes – Idle Line Selection System Data – Call Feature Timer – Prime Line Delay Timer

Related Features

Speakerphone
Intercom Call (ICM Call)
Station Flexible Buttons

Hardware

LIP Phone

2.75 INTERNATIONAL CALL RESTRICTION

Description

Outgoing international call is normally very expensive. International call can be restricted by program.
When international call restriction is programmed, iPECS SBG-1000 tries to compare dialed digits to CO with programmed international prefix. If digits are matched, call should be released and access deny message will be displayed on LCD with error tone.
When CO-CO international call is programmed to be restricted, following 3 cases should be restricted.

1) CO call transfer to international call
2) international outgoing call transfer to CO call
3) CO call forward to international call

Operation

When programmed, operation of this feature is automatic.

Conditions

1. When all international call is programmed to be restricted, all outgoing international call is restricted. Incoming CO call can not restricted by this rule.
2. If call transfer is restricted, transferee call should be recalled after transferor hang up.

Programming

VOICE CONFIG System Data – International Call

Related Features

Call Transfer,
Call Forward
Call Forward, Preset

Hardware

2.76 IP SYSTEM DECT

Description

iPECS SBG-1000 supports office building mobility employing Digital European Cordless Technology (DECT). iPECS SBG-1000 has the internal Wireless Telephone Interface Module (WTIM). The internal WTIM manages up to six (6) DECT phones (GDC-450H or LWS-WK) and maintain an uninterrupted communications link to iPECS features and resources.

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

Operation

System Attendant

To register DECT phone to iPECS SBG-1000:

1. Press the **[PGM]** button.
2. Dial 0# (WHTU SUBSCRIBE code).
3. Press the **[FLEX 1]** button.
4. Dial the Station number to be used for DECT phone.
5. Dial the phone type. 3 is for GDC-34x/4xx and 4 is for LWS-WK.
6. Press the **[SAVE]** button.
7. Proceed to instruction below for GDC-450H or LWS-WK.
8. When the registration is completed below message is shown on LCD of attendant.

STATION : 14 SUBSCRIBED: SUCCESS

GDC-450H

To register GDC-450H to iPECS SBG-1000

1. Press the **Menu** button to display the menu.
2. From the menu use the **Navigation** button to highlight Phone Register.
3. Press the **OK** button; this displays the Phone Register menu.

4. Select “LWS Subscription” using the up and down arrows of the **Navigation** button and press the **OK** button.
5. The GDC-450H searches for the iPECS/iPECS SBG-1000, displaying and “Searching..1”. When a iPECS/iPECS SBG-1000 is found, its RFPI, is displayed. The RFPI of your iPECS/iPECS SBG-1000 is available from your System Administrator, or perhaps the attendant.
6. Press **OK** button while highlighting the RFPI to continue the registration to the system, or Press **No** button to continue the search.
7. Press **OK** button; on successful registration, a confirmation tone is received at the GDC-450H and the iPECS/iPECS SBG-1000.
8. If the registration fails, repeat procedure from Step 1 to 7 at the System Attendant and Step 1 to 8 from the GDC-450H.

LWS-WK

To register LWS-WK to iPECS SBG-1000

1. Press **[MENU]** button to display the menu.
2. Highlight “Phone Register” using the Navigation up/down key, and then press **[OK]** soft button or Navigation ‘OK’ key.
3. Select “Subscription” using the Navigation up/down key, and then press **[OK]** soft button or Navigation ‘OK’ key.
4. Display [Searching..1].
5. The system [RFPI : eg. 01234567890123] will be displayed when a system is found.
6. Press **[OK]** soft button or Navigation ‘OK’ key. In a few second, a confirmation tone is received at the LWS-WK.
7. If the registration fails, repeat procedure from Step 1 to 7 at the System Attendant and Step 1 to 6 from the LWS-WK.

Conditions

1. Up to six (6) DECT phones can be registered and maximum 4 DECT calls can be placed simultaneously.
2. During the registration of DECT phone, Monitor or Speaker button at the iPECS/iPECS SBG-1000 attendant phone should not be pressed until the DECT phone completes the registration and registration confirmation tone is heard.

Programming

VOICE INSTALL Station Registration – DECT Registration

Related Features

Hardware

GDC-450H handsets

2.77 ALARM SIGNAL/DOOR BELL

Description

The system can be configured to recognize the status of an external contact (normally open or closed). The system will signal to the Attendant Station when the contact activates. This capability is commonly employed to provide remote Alarm or Door Bell signals to the user.

The Attendant Station receives the Alarm Signal, either a single tone burst repeated at 1-minute intervals or a continuous tone. The Alarm Signal may be terminated at the user's phone by dialing the Alarm Stop code or, if assigned, pressing the **{ALARM STOP}** button. To rearm the Alarm function, the alarm condition must be cleared and the Alarm signal terminated.

When used as a Door Bell, the Attendant Station receives a single tone burst each time the external contact is activated and no reset is required.

Operation

System

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

LIP Phone

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

[PGM] + **{FLEX}** + '65' + **[SAVE]**

To terminate an Alarm Signal while idle:

1. Dial the Flex Numbering Plan code 65, confirmation tone is received and the Alarm Signal is terminated. If the alarm condition is cleared, the system will automatically rearm the alarm monitoring.

Or,

2. Press the **{ALARM STOP}** button.

Conditions

1. The Alarm contacts must be "dry", no voltage or current source connected.
2. The Attendant Station will show "ALARM" or "DOOR BELL" as appropriate.

Programming

VOICE CONFIG System Data – Alarm Attributes

Related Features

Door Open

Hardware

LIP Phone

External contact connected to Alarm input of iPECS SBG-1000, refer to iPECS SBG-1000 **Quick Start Guide.**

2.78 DOOR OPEN

Description

The iPECS SBG-1000 hardware is equipped with relays that activate External Control Contacts. The contacts can be assigned to one of several functions including a Door Open Contact. When used as a Door Open Contact, the contact is connected to a door-lock release mechanism. When the Attendant Station receives the Door Bell signal, the user may dial the Door Open code to activate the contact.

LIP Phone users may assign a Flex button as a **{DOOR OPEN}** button.

Operation

LIP Phone

To assign a Flex button as an **{DOOR OPEN}** button to terminate the Alarm Signal:

[PGM] + **{FLEX}** + Door Open code ('#*') + **[SAVE]**

To activate the relay contact

1. Lift handset or press **[SPEAKER]** button.
5. Dial Door Open code, '#*'.
6. Hang-up to return to idle.

Or,

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

Conditions

1. The contacts are rated at 1 amp, 24 VDC.

Programming

VOICE CONFIG	System Data – Alarm Attributes

Related Features

Alarm Signal/Door Bell

Hardware

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

2.79 MOBILE EXTENSION

Description

A mobile phone may be registered to a station allowing the mobile phone to place and receive calls through the system. SIP or ISDN DID calls are sent to the user's LIP Phone and the active registered mobile phone simultaneously. If the mobile phone is paired with a Hunt group station, Hunt group calls routed to the station also ring to the active mobile phone when enabled.

The mobile phone users can access the facilities of the iPECS SBG-1000 to place internal and external calls as well as activate/access features. To access system facilities and resources, the mobile user calls the SIP number or the ISDN DID number of the corresponding LIP Phone. When the call is received, the system matches the CLI to the mobile phone and provides the mobile user with system dial tone.

Operation

System

Incoming SIP or ISDN DID calls are sent to active mobile phones automatically.

Mobile Phone

To place a call from the mobile extension using the iPECS SBG-1000:

1. Dial the SIP number or the ISDN DID number of the station, the system will check the CLID, answer the call and the user will receive intercom dial tone.
2. Place internal or external iPECS SBG-1000 call as normal.

To Transfer a call from the mobile extension using the iPECS SBG-1000:

1. Dial '*' while on an iPECS SBG-1000 call.
2. Dial the desired extension, the call is transferred and the mobile phone returns to idle.
   Note: the mobile may reconnect by dialing '#'.

Conditions

1. When the mobile phone places an external call through iPECS, the CLI of the corresponding station is used.
2. The Mobile Extension features are supported via system digital (SIP and ISDN) lines only.
3. Message Wait and Callback cannot be activated to a mobile phone.
4. The Mobile Extension feature is not supported over a distributed networked environment.
5. When an incoming SIP or ISDN call is received, the system will access an SIP or ISDN line and place a call to the mobile phone. Thus, an SIP or ISDN line must be available for the system to notify the mobile user of the incoming call.
6. Hold and Transfer Recalls to the mobile phone are sent to mobile phone and the associated station.
7. Circular and Terminal Hunt Group calls can be routed to the active Mobile Extension.

Programming

VOICE CONFIG	Station Data – Mobile Extension

Related Features

DND (Do Not Disturb)
Station Message Wait/Call Back
Attendant Recall
Distributed Control Network

Hardware

LIP Phone

2.80 SYSTEM NETWORKING

2.80.1 Distributed Control Network

Description

In the Distributed Control Network, each iPECS SBG-1000 system maintains control over the devices registered to it. The networked systems communicate allowing other networked systems access to resources over the network. In addition, other features and functions as detailed in the following sections of this manual are available to users in a distributed network environment. The iPECS SBG-1000 permits remote access to various resources through registered gateway Modules and terminals.

In addition, iPECS SBG-1000 will request access to resources of remote systems. The user-dialed number is analyzed and the call routed according to the Net numbering table. Should the main path fail to respond, the iPECS SBG-1000 routes the call employing the alternative Speed Dial route assigned.

iPECS SBG-1000 supports H.450 over IP, for the basic networking functions and the proprietary iPECS protocol for the advanced networking features.

Operation

Operation of Distributed Networking is automatic when configured & defined

Conditions

1. To use the networking features, the software lock-key installation is required. Each iPECS SBG-1000 system has a unique software lock-key. To get the software lock-key, contact the distributor of iPECS SBG-1000 system.
2. Unified Dialing Plan (UDP): Each station can have a unique number up to 7 digits in the networked systems, but it depends on their own numbering plan.
3. The alternative route employs a Speed Dial number to place a call and is not a Networked call. Thus, the Distributed Control Network features are not available.

Programming

Related Features

Hardware

2.80.1.1 Net Call

Description

A station user can make a call to a station in other systems by dialing only a station number just as an intercom call within the same system.

Operation

1. Lift Handset or press the **[SPEAKER]** button. The system provides a user with a dial tone.
2. Dial the station number of other systems, or press the {NET DSS} button of other systems.
3. The station seizes the network CO line according to the net routing table, and the system sends a digit stream that is modified by the net routing table.
4. The called party receives a digit stream that is sent by calling party, and analyzes it using the net routing table to determine
5. The right destination. The called station receives a ringing signal.
6. The LED of [Network CO] button will be extinguished when the Net Call is cleared.

Conditions

1. Net call must be used without seizing a CO line.
2. User hears an error tone if there is no idle networking path.
3. In spite of ICM mode, the called party receives a ringing signal for the networking call.
4. When system detects the fatal error from the network, system sends the digit stream to the network using the alternate speed dial bin. In this case, the call is not a networking call.

Programming

VOICE INSTALL	CO Line Registration – Net Basic Attributes CO Line Registration – Net CO Line Attribute CO Line Registration – Net Numbering Plan

Related Features

Hardware

2.80.1.2 Net Transfer

Description

A station user can transfer any kind of CO line to a station in other systems by pressing **[TRANS]** button and dialing a transferred station such as a call transfer within the same system. There are two kinds of transfer, screened and unscreened transfer.

There are two kinds of standard transfer method in H.450; Transfer by join and Transfer by rerouting. The main difference is how control the connecting path between transferring and transferred station. In case of Transfer by join, additional connecting path will be needed to transfer the call to another station. In case of Transfer by rerouting, new connecting path is used to transfer the call and old connecting path of transferring station will be cleared.

Operation

Screened transfer

1. Press the **[TRANS]** button at a station during conversation with a CO line. The CO line is placed on Exclusive Hold.
2. Dial the station number of another system to transfer the call. The transferred station of another system receives a ring signal.
3. Announce when the transferred station answers. Both stations can make a conversation each other, but the held CO is still in waiting on Transfer hold.
4. Hang-up to complete the transfer.

Unscreened transfer

1. Press the **[TRANS]** button at a station during conversation with a CO line. The CO line is placed on Exclusive Hold.
2. Dial the station number of another system to transfer the call.
3. Hang-up to complete the transfer.

Conditions

1. If both of transferred and transferred-to stations are located in the same system, the networking path will be cleared. That is, the transfer call will be setup as intercom call.
2. The transfer will be canceled when user presses the flashing [CO] or **[TRANS]** button.
3. Net Transfer call does not recall to the origination.
4. User hears an error tone if there is no idle networking path.
5. Net transfer is not activated to a busy station.

Programming

VOICE INSTALL	CO Line Registration – Net Basic Attributes
	CO Line Registration – Net Supplementary Attr.
	CO Line Registration – Net CO Line Attribute
	CO Line Registration – Net Numbering Plan

Related Features

Hardware

2.80.1.3 Identification Service

Description

Calling Name Identification Presentation (CNIP): When a user makes a net call and a name of station is programmed in the Station Name field, the system includes the name of calling party to the called party between systems.

Operation

1. A Net Call is arrived a station with LCD display.
   While ringing, the CNI will be displayed if they are included in the Setup message.

Conditions

VOICE INSTALL	CO Line Registration – Net Basic Attributes

Programming

VOICE INSTALL	CO Line Registration – Net Basic Attributes

Related Features

Hardware

2.80.1.4 Call Completion

Description

There are two kinds of call completion as follows;

Completion of Calls to Busy Subscribers (CCBS):

After calling a user in another system using basic call and encountering a busy tone. A station user can be notified when the busy destination of another system becomes idle. If the user wants to make a call to the destination on that notification, the call can be reinitiated to the destination of another system again.

Completion of Calls on No Reply (CCNR):

After calling a user in another system using basic call and encountering no reply. The caller can be notified when the destination becomes an idle status after some actions. If the caller wants to make a call to the destination, the call can be reinitiated to the destination again.

Operation

To make CCBS (Call Back)

1. Dial the station of another system that is a busy.
2. Press the [CALLBK] button while a busy tone is provided.
3. The call is cleared after a confirmation tone.
4. The busy station goes to Idle; the originator receives a call-back ring.
5. When the originator answers to the call-back ring, a new call will be activated to the calling station.

Conditions

1. Stand-alone IP Phone that supports H.450 can activate the Call Completion feature.
2. A station can leave or have only one callback message, and a new request will be left message wait indication message on busy station.
3. A voice message cannot be left even though the VSF is installed in a local system.
4. When the originator does not answer the call back ring within net timer, the call will be cleared.
5. There are two modes: One is connection mode and the other is connectionless mode. This can be selectable at Net Basic Attributes.

Programming

VOICE INSTALL	CO Line Registration – Net Basic Attributes
	CO Line Registration – Net CO Line Attribute
	CO Line Registration – Net Numbering Plan

Related Features

Hardware

2.80.1.5 Call Offer

Description

A busy user on one node is given notification that another call is waiting from another node. It is similar to a Camp-On function.

Operation

To activate Call Offer

1. Dial a busy station number of another system. The caller hears a busy tone.
2. Press the [CAMP ON] button or '*' during hearing a busy tone.
   The busy station receives an off-hook muted ring.
   The calling station hears a ring-back tone instead of a busy-tone.

To answer the Call Offer

1. Press the flashing CO line button while receiving a muted ring.

Or,

2. The muted ring is changed to normal CO ring when you go on-hook state. Then you can answer the offered call.

Conditions

1. Call Offer is only applied to a station that is in talk status.
2. During a conference or paging, call offer is not activated.

Programming

VOICE INSTALL	CO Line Registration – Net Basic Attributes

Related Features

Hardware

2.80.2 Centralized Control TNET

Description

In a Centralized Control TNET (Transparent Network), a central MFIM controls all remote modules and terminals providing transparent networked access to all the features and functions of the central iPECS as well as the resources connected to the iPECS.

Where the remote device is not directly reachable by the iPECS, RTP packets must be relayed through a local VoIP channel. A remote device may not be reachable when WAN access for the device is through a firewall or NAPT server. In this case, the remote devices are assigned a zone to manage RTP traffic between other devices connected in the TNET. The zone defines when an individual device requires use of the local VoIP channel. Zones are used to identify other group characteristics as outlined in section Remote Device Zone Management.

Remote sites may include an MFIM operating in the local mode as a live back up to the remote central system. Under normal circumstances, the central MFIM controls remote devices (gateway Modules and terminals) including any local MFIM VoIP channels. However, should the WAN connection between the central system and the remote devices fail, the local MFIM will assume the call server responsibility for the local devices. The local MFIM thus provides local survivability.

Under certain operating conditions, this equipment cannot be relied upon to make emergency calls. Alternative arrangements should be made for access to the emergency services.

Operation

Operation of Centralized Network is automatic when configured & defined.

Conditions

1. iPECS SBG-1000 can operate in local MFM mode only, not central MFIM mode.
2. In a Centralized Network, the maximum number of channels available is the maximum number of channels supported by the central MFIM.
3. In TNET, Centralized Miscellaneous functions (Relay support, MOH, BGM, Alarms and External Page) are not supported but, all terminals in the TNET can make and receive pages.
4. When NAPT or other firewall functions are implemented, packet relay for RTP packets is required. Packet relay requires VoIP channels for each simultaneous call desired.
5. The local MFIM will take over operation of registered devices if the central controlling MFIM does not respond to three consecutive poll attempts over a 10-second period. The central MFIM will gain control automatically upon return of the WAN connection.
6. iPECS can be installed behind a NAPT however, Fixed Nat – port forwarding is required for the host to be reachable by remote devices.
7. You can set TNET register enable ON/OFF for using TNET in local MFIM.
8. If TNET Register is enabled, all devices are automatically registered to the central MFIM, except DECT devices.

9. If any new device is registered to iPECS SBG-1000, TNET Register must be disabled and enabled again in order to register new device to the central MFIM.

Programming

VOICE INSTALL System – Tnet Attributes

Related Features

Hardware

2.81 STATION CALL COVERAGE

Description

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

Operation of this feature requires a **{CALL COVERAGE}** button at the covering LIP Phone and the covered station must activate call coverage. A station can have multiple Call Coverage buttons each covering a different station and multiple stations can have a Call Coverage button for a given station.

Operation

LIP Phone

To assign a **{CALL COVERAGE}** button at the covering station

[PGM] + **{FLEX}** + '*#' + covered Station number + **[SAVE]**

When a covered station receives a call, the covering station will receive the following display:

CALL FOR STA xx MAY 06 11 04:30 pm

Conditions

1. A LIP Phone user may cover for an SLT or other stations. However, since a Flex button is required, an SLT cannot provide coverage for other stations.
2. When off-hook or in DND, the covering station will only receive a visual indication of the call from the LED of the **{CALL COVERAGE}** button and display, no off-hook ring is provided.

3. The **{CALL COVERAGE}** button will provide an appearance for CO lines that do not appear on the covering station except for Private Lines. To cover for Private Lines, the covering station must have an appearance and be allowed access to the Private Line.

Programming

Related Features

Hardware

2.82 IP CALL RECORDING

Description

System can record automatically or manually using IPCR server. IPCR(IP call recording) Server can be registered to iPECS SBG-1000 system. The station with agent ID is automatically recorded about call, external call.

Operation

Registration

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

1. IPCR setting before registration to system.
   1.1) PBX registration(system IP, SIP ID, SIP Password)
   1.2) IPCR Server registration
   1.3) User registration
   1.4) Channel registration
2. System should set register MAC table for IPCR's MAC or set to "All Stations" in Registration Table page.
3. If SIP ID is not allowed, SIP ID and password should be set in PGM 443.

Programming Agent ID

1. Enter the number of IPCR's order in IPCR Agent page.
2. Match Agent ID to favorite station.
3. You can see the ACR(Auto-call recording) or ODR(On Demand Recording).
4. You should choice STN Type for station. But DID Type is for the future.

Two way recording

1. You should set IPCR group(ex: 620) and the SIP number of the IPCR as the member of IPCR Group.
2. Automatic Recording Destination should be set the IPCR Group(ex: 620).

Operation

1. IP-Phone(S100) without agent ID answered from IP-phone(S101) with agent ID(A500).
2. If S101 can conference 3 way, S101 connects IPCR with the agent ID(A500).
3. If you have flex button with two-way record of IPCR Group (620) and agent ID is ACR, it's

flashed during two way recording. But it's ODR, first time, it's not flashed. After pressing the flex button (two way record), it's flashed. ACR is unconditionally recorded after connection and ODR is conditionally by user's choice.

4. Even though it's ODR, it can be recorded during talking. If users don't press the two-way recording button within talking, it's erased.

Conditions

1. You can search the recorded using Web Admin of IPCR.
2. IPCR server can be registered up to 10 servers in a system.

Programming

VOICE CONFIG	Station Group Data – IPCR Agent
	Station Group Data – Station Group Assignment
	Station Data – Common Attributes - Automatic Talk Recording Dest

Related features

Hardware

2.83 AUTHORIZATION CODES (PASSWORD)

Description

Authorization Codes provide a means to control access to Off Premise Call Forward or DISA and may be required for outgoing CO Line access based on configuration of the iPECS SBG-1000 database. When users dial an Authorization Code that matches an Authorization Code stored in the database, the system invokes the Station COS or the COS assigned to Authorization code. Each Authorization code has separate Day/Night mode COS assignments.

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

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

Operation

LIP Phone

To enter an Authorization Code when second dial tone is received

1. Dial the station number for the Station Authorization code or, for a System Authorization Code, dial '*' or '*' and the Authorization table index.

2. Dial the corresponding Authorization Code.
3. Place call as normal.

SLT

To enter an Authorization Code when second dial tone is received

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

DISA

To enter an Authorization Code when second dial tone is received

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

Conditions

1. When a DISA Line is marked for Authorization Code entry, the caller will hear DND Warning tone and must input a valid Authorization Code to continue. In case of an entry error, the user may retry entry of the code. In case of multiple entry errors, the user may retry entry based on the DISA Retry counter.
2. A user must enter a valid code within the number of attempts assigned as the Auth Retry Count.
3. The default Station Authorization code is the station number and "*".
4. The total number of Authorization codes is provided in Table 1.4-1.
5. An Authorization code may include any dial pad digit except '#'.
6. Duplicate or conflicting System Authorization codes are not allowed when using the older "*" and code operation. For example, code '1234' conflicts with code '123' and cannot be recognized as a unique code. Since the index operation employs the table index and the station number forms part of the Station code, conflicts will not occur and duplicate codes are allowed for these types of Authorization code.
7. Use of Authorization codes varies based on the system nation code. In some regions, particularly the US and UK, a System Authorization code may be required for DISA access. Entering a Station code on a DISA line will fail in these areas.

Programming

VOICE CONFIG	Station Data – Authorization Code & COS System Data – System Attributes – DISA Retry Count System Data – System Attributes – Auth Retry Count System Data – System Attributes – Old Auth Code Usage

Related features

Auto Service Mode Control

CO Access
Temporary Station COS/Lock
Call Forward
Station User Programming & Codes

Hardware

2.84 USB UPGRADE

Description

The Attendant can upgrade iPECS SBG-1000 via USB memory. Before upgrading, a user must save the iPECS SBG-1000 Rom file (GS87MXXXX.rom) in the top directory of USB memory.

Operation

Attendant

To upgrade iPECS SBG-1000 in Attendant phone

1. Save the iPECS SBG-1000 Rom file (GS87MXXXX.rom) in the top directory of USB memory.
2. Insert the USB memory into the USB port on iPECS SBG-1000.
3. Press the **[PGM]** button.
4. Dial 09 (Attendant Station Program code).
5. Number of iPECS SBG-1000Rom file in USB memory is displayed.

ROM FILE NUM : TOTAL 2 PRESS 0-1 TO VIEW FILE

6. Dial Number of iPECS SBG-1000 Rom file to display iPECS SBG-1000 Rom file name.

0 : GS87M10Ar.rom PRESS [SAVE] TO UPGRADE

7. Press the **[SAVE]** button to upgrade iPECS SBG-1000.

0 : GS87M10Ar.rom COPY IN PROGRESS

0 : GS87M10Ar.rom BURNING IN PROGRESS

8. Following upgrade, result confirmation is displayed.

0 : GS91MA0Af.rom PRESS [SAVE] TO RESTART

9. Press the **[SAVE]** button to restart iPECS SBG-1000.

Conditions

1. USB upgrade using the Attendant Station programming will support up to 10 ROM image files.

Programming

Related Features

Hardware

2.85 AUTO CALL RECORDING

Description

LIP Phone users can be configured in the system to record all calls to a mailbox or the hard disk drive of an iPECS Phontage. When recorded to a mailbox, users manage the recording through voice mail. For recordings to the Phontage, recordings are managed directly by with the ability to listen to, delete or send the recording to others via e-mail.

Operation

Recording of calls is automatic when assigned.
To manage the recordings, use the procedures outlined in the Phontage User Guide.

Conditions

1. The Phontage can record one call at a time and must be idle. While recording, if the Phontage places or receives a call, recording terminates.
2. When call recording begins, the station will receive a Call Recording confirmation tone.
3. A remote Phontage will not support call recording.
4. Conference call cannot be recorded.

Programming

VOICE CONFIG Station Data – Common Attributes – Automatic Talk Recording Dest.

Related Features

VSF Integrated Auto Attd/Voice Mail
Two-way Record

Hardware

LIP Phone
VSF
Phontage

2.86 TWO-WAY RECORD

Description

A LIP Phone user can record any active conversation to the station user's mailbox or to hard disk drive of an iPECS Phontage. All calls including incoming, outgoing, internal, and external calls can be recorded. A **{RECORD}** button must be assigned to access this feature.

Operation

LIP Phone

To assign a flexible button as a **{RECORD}** button

1. **[PGM]** + **{FLEX}** + **[PGM]** + 80 + recording destination (optional) + **[SAVE]**

To activate Two-Way Record while on a CO call

1. Press the **{RECORD}** button, record warning tone heard, and recording starts.

To stop Two-Way Record while on an CO call

1. Press the **{RECORD}** again.

Or,

1. Hang-up, return to idle.

To manage the recordings, use the procedures outlined in the Phontage Guide.

Conditions

1. The **{RECORD}** button LED will flash at 240 ipm while recording.
2. Use of this feature when the Two-Way Recording Warn Tone is disabled may be interpreted as a violation of federal, state, or local laws, and an invasion of privacy. Check applicable laws in your area before recording calls using this feature.
3. If a destination for the recording is not defined for the **{RECORD}** button, the Automatic Talk Record Destination defined in Station Data - Common Attributes is employed.
4. Conference call cannot be recorded.

Programming

VOICE CONFIG	Station Data – Common Attributes – Automatic Talk Recording Dest.

Related Features

VSF Integrated Auto Attd/Voice Mail
Auto Call Recording

Hardware

LIP Phone
VSF
Phontage

2.87 EXECUTIVE/SECRETARY FORWARD

Description

LIP Phones can be assigned as Executive/Secretary pairs. By activating DND, the Executive also activates Unconditional Call Forward to the Secretary, which will forward Executive calls to the

Secretary. With the “CO Call to Secretary” option enabled, all CO calls to the Executive forward to the Secretary regardless of the Executive’s station status. In addition, if the Secretary is in DND, Executive calls sent to the Secretary route back to the Executive if the “Call Exec If Sec in DND” option is enabled.
Each Executive can be assigned a “Grade” (01, highest ~12, lowest). Executives with a higher grade can call lower grade Executives overriding the Executive/Secretary forward.
With the “Icm Call to Secretary” option enabled, all internal calls to the Executive (except for calls from higher or same grade executive) forward to the Secretary regardless of the Executive’s station status.

Operation

LIP Phone

To activate/deactivate Executive/Secretary forward from the Executive iPECS Phone

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

Conditions

5. The maximum number of Executive/Secretary pairs is 10.
6. An Executive may have multiple Secretaries and a Secretary may have multiple Executives. Each forms a separate Executive/Secretary pair.
7. If the Secretary is busy when a call is received for the Executive, the caller will receive busy tone.
8. The Secretary may override the DND status of the Executive to Camp-On and transfer calls to the Executive.
9. A chain can be constructed by assigning the Secretary of one pair as an Executive of another. Although a chain may be constructed, a loop back is not allowed.
10. If an Executive has multiple Secretaries, calls will automatically route to the Executive’s first idle Secretary.
11. The Executive may use Call Forward to send calls to stations other than the Secretary.
12. The Executive Grade can be assigned only for Country Code “82”, Korea.

Programming

VOICE CONFIG	Station Data – Executive / Secretary Table

Related Features

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

Hardware

LIP Phone

2.88 LCR (LEAST COST ROUTING)

Description

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

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

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

1. Internal LCR

   If the user dialed digits match an Internal LCR code in the LDT (Leading Digit Table), a CO path is selected and digits are modified using the DMT (Digit Modification Table).

2. Loop LCR

   If the user dials 9 (the 1st available CO line access code) or presses a **{LOOP}** button to place a call, the Loop LCR mode is accessed. Dialed digits are compared to the Loop LCR codes in the LDT; the system will seize a CO line from the assigned CO Group and sends digits from the DMT.

3. Direct CO LCR

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

Operation

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

Conditions

1. There are a total of 6 LCR access mode combinations that can be defined as below:

   1) LCR Access Mode 00 (M00) – LCR is disabled.

   2) LCR Access Mode 01 (M01) – Loop LCR access is active.

   3) LCR Access Mode 02 (M02) – Loop and Internal LCR access are active.

   4) LCR Access Mode 11 (M11) – Direct CO and Loop access are active.

   5) LCR Access Mode 12 (M12) – Direct CO, Loop and Internal LCR access are active, seize Co line according to the LCR attributes.

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

2. Leading digit entries may be duplicated in the Leading Digit Table. Using a different "LDT Index" will make each entry unique. The system will use the lowest matching entry.
3. When Direct CO LCR is used on an ISDN line, an ISDN information message with the Called party Info Element, which includes only the numbering plan and numbering type, is sent to

the ISDN to maintain the ISDN connection while the user finishes dialing and the system modifies the digits.
4. For Direct CO LCR, the number of digits for the LDT Table should be programmed considering the dial-tone time-out of the network.
5. Since a CO path is connected, Direct CO LCR does not support the Alternative DMT index, which allows the system to select a second or alternative CO path to place the call.
6. If the LCR CO group is not assigned, the system will not seize a CO line call and make internal call.
7. If the LCR CO group is assigned as the unused group, the system it will seize a CO line according to the station attributes.

Programming

VOICE INSTALL	CO Line Registration – LCR Attribute CO Line Registration – LCR LDT CO Line Registration – LCR DMT CO Line Registration – LCR Initialization

Related Features

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

Hardware

2.89 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.

Operation

Operation of this feature is automatic when programmed

Conditions

Programming

VOICE INSTALL	CO Line Registration – CO Channel List

Related Features

Hardware

2.90 SYSTEM CLOCK SET

Description

The attendant can set the system Time/Date.

Operation

System Attendant

To set the system clock:

1. Press the **[PGM]** button.
2. Dial 025 (System Date Time Set code).
3. Dial six (6) digits for the Date (MM/DD/YY) or **[SAVE]** button to skip the Date.
4. Dial six (4) digits for the Time (HH/MM) or **[SAVE]** button to skip the Time.
5. Press the **[SAVE]** button.

Conditions

1. The time is entered as a 24-hour (military) clock, 24-hour mode.
2. If Automatic Time Update is enabled, the system will check the time periodically as configured and time/date can be reset automatically. This will override the attendant setting.
3. iPECS SBG-1000 can support the year from 1970 to 2037. YY from 70 to 99 is treated as 19YY and YY from 00 to 37 is converted to 20YY. YY from 38 to 69 is invalid and error tone will be heard.

Programming

Related Features

Day/Night/Timed Ring Mode

Hardware

LIP Phone

2.91 CALL FORWARD CO OFF-NET

Description

The attendant can forward incoming CO calls to a remote “Off-Net” location. Calls are forward via a Speed Dial bin. When a call is received, the system will automatically place a call using the Speed Dial number, dialing the number and connecting the incoming call in an Unsupervised conference.

Operation

System Attendant

To activate CO Off-Net Call Forward:

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **[FWD]** button.
3. Dial “5”, the Off-Net Forward feature code.
4. Dial CO access code to forward, the incoming call.
5. Dial the Speed Dial bin to be used to place the outgoing call, the LED of any Off Net forwarded **{CO}** button at the Attendant stations will flutter at a rate of 240 ipm.

To deactivate CO Off-Net Forward:

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **[FWD]** button.
3. Dial “5”, the Off-Net Forward feature code.
4. Dial CO access code to forward, the incoming call.
5. Dial “#”.

Conditions

1. The attendant can forward any CO line, CO Group or all CO calls using the appropriate CO dial access code. In addition, a Flex button assigned to a CO line or group may be forwarded using the Flex button.
2. The conditions of the Unsupervised conference feature apply.
3. The conditions of the Call Forward Off-Net feature apply.
4. When enabled, calls forwarded Off-Net will receive the Off-Net forward prompt.
5. When enabled, calls forwarded Off-Net will receive the DTMF repeat tone.
6. To utilize Off-Net call forward of incoming analog CO line to other analog CO line, a valid Open Loop Detect Timer should be assigned by programming for the analog CO lines to prevent CO line lock up when analog CO lines are in use.

Programming

VOICE CONFIG	System Data – Call Feature Timer – Unsupervised Conference Timer System Data – Call Feature Timer – Open Loop

Related Features

Call Forward

Hardware

LIP Phone

3. WEB ADMINISTRATION

Smart Business gateway (iPECS SBG-1000) incorporates a Web Server. Using a Web browser the system's Web Server can be accessed and the database managed in a user-friendly environment. In addition to modifying the system database, Web Admin provides for system file upload, remote upgrade, and database download.

http://192.168.1.1

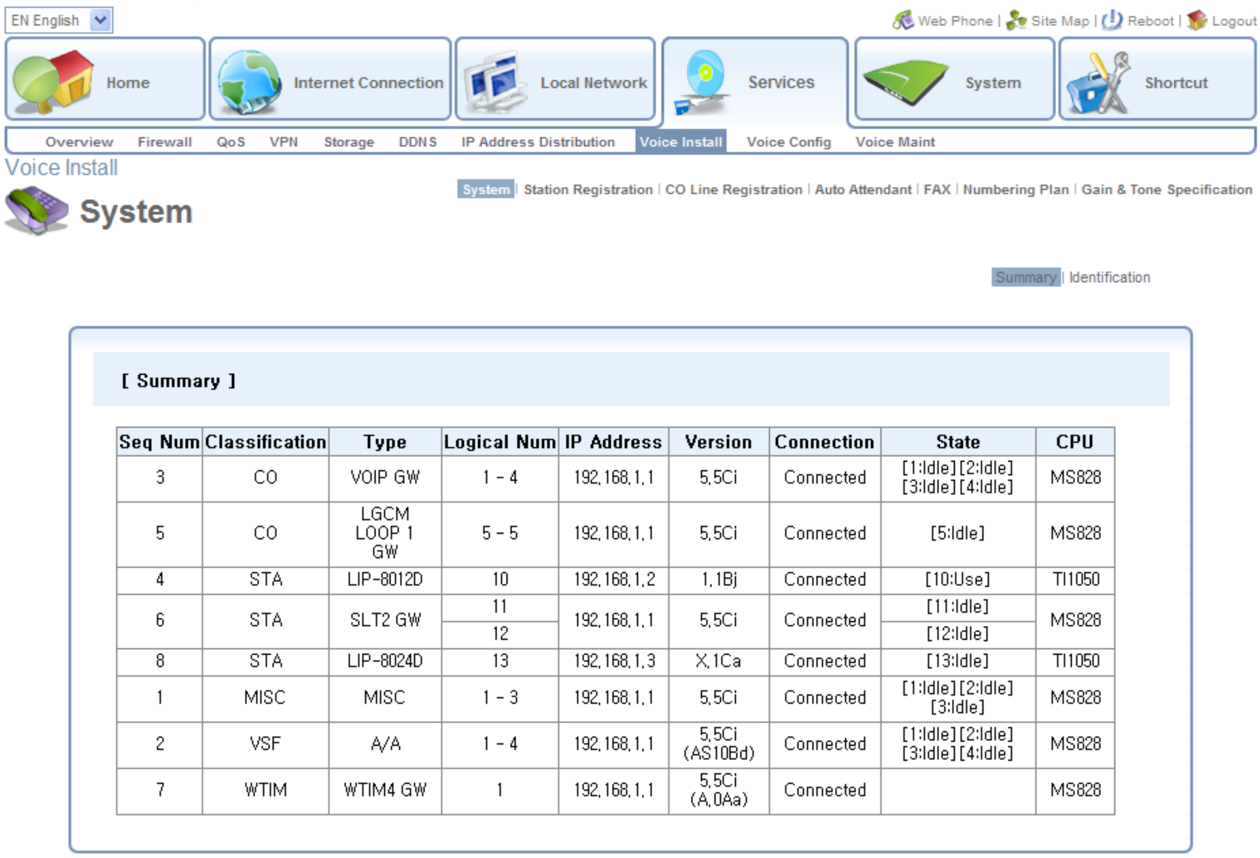
When accessed, the system will display the iPECS SBG-1000 Web Admin. Log In screen where the user must enter an assigned ID and password.

Items in the Menu bar can be clicked to display the items listed which are described further in the following sections:

- Voice Installation – access to database for system installation
- Voice Configuration – access to database for system configuration including Station, SIP Line data
- Voice Maintenance – permits database download and system or device upgrade

3.1 VOICE INSTALLATION

In this section, the user can see or change the database for system installation including the nation code, Station Registration, Station/CO Line Registration, Auto Attendant, FAX, numbering plan, Admin Authorization. Gain & Tone specification and System Tone Frequency can be modified.



Seq Num	Classification	Type	Logical Num	IP Address	Version	Connection	State	CPU
3	CO	VOIP GW	1 - 4	192.168.1.1	5.5Ci	Connected	[1:Idle][2:Idle] [3:Idle][4:Idle]	MS828
5	CO	LGCM LOOP 1 GW	5 - 5	192.168.1.1	5.5Ci	Connected	[5:Idle]	MS828
4	STA	LIP-8012D	10	192.168.1.2	1.1Bj	Connected	[10:Use]	TI1050
6	STA	SLT2 GW	11	192.168.1.1	5.5Ci	Connected	[11:Idle]	MS828
			12				[12:Idle]	
8	STA	LIP-8024D	13	192.168.1.3	X.1Ca	Connected	[13:Idle]	TI1050
1	MISC	MISC	1 - 3	192.168.1.1	5.5Ci	Connected	[1:Idle][2:Idle] [3:Idle]	MS828
2	VSF	A/A	1 - 4	192.168.1.1	5.5Ci (AS10Bd)	Connected	[1:Idle][2:Idle] [3:Idle][4:Idle]	MS828
7	WTIM	WTIM4 GW	1	192.168.1.1	5.5Ci (A.0Aa)	Connected		MS828

3.1.1 System

3.1.1.1 Summary

This page displays information of registered devices including the device type, logical number, IP address, version of device, connection status and current state of each devices, and also known CPU type.

3.1.1.2 Identification

Under Identification, the country is identified using International Dial codes (Nation Code). A Site Name (up to 24 characters) and My Area Code (local area code) maybe defined. The Site Name is primarily useful for the installer/programmer as a reference to customer. This information is used to set gain, frequencies and other system characteristics specific to the country and regional regulatory requirements.

In addition, the system can be programmed to select the base Flexible Number Plans, which are the numbering plans for the normal case and for the networking case. Individual items from the selected base Numbering Plan can be changed under Flexible Numbering Plan in section 3.1.6.

When the system is installed behind a NAPT server, the fixed IP Address provided by the NAPT server must be assigned to firewall IP address. Also, use this firewall IP address to identify the iPECS SBG-1000 in remote devices.

3.1.1.3 Tnet Attributes

Each LM (Local MFIM), which is part of a Central Control Network, must be defined with the IP Address of the CM (Central MFIM) as well as the LM configuration data that will be sent to the CM at the time the LM registers with the CM. Total port counts define the ports, which are allocated in the CM database for use by devices registered to the LM. The number of ports defined in the database of each LM must be equal or less than the ports defined in the CM for the LM in order to register properly.

Table 3.1.1.3 Tnet Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Register Enable	This field informs the LM to attempt registration with the CM. This field must be set to ON for proper registration.	0: OFF 1: ON	OFF
IP Address	This field defines the IP address of the CM that will be used by the LM.	IPv4 address	
IPKTS Port number	In the TNET environment, the IP KTS protocol signaling UDP port is defined. At present this field is not used, do not change this port number.	0000-9999	5588
Total no of port	This field defines the total number of ports the LM will request be allocated by the CM for devices attached to the LM. This value must be equal to or less than the port count in the CM for the LM devices.	000-999	0
Polling Count	This field defines the maximum polling failures an LM considers a WAN fault	00-99	05
Polling interval	This field defines the interval time between LM to CM polling attempts.	00-99	02

3.1.2 Station Registration

3.1.2.1 Station List & Replacement

Registered station list is displayed on this page; a user can see IP address, device type, version of device, connect status of registered devices. In addition, a user can change the logical number of station, station name, remark and so on.

3.1.2.2 Registration Table

By entering device MAC addresses, the system will allow the device to register.

While initial installation, you may want to register all connected stations without MAC address programming. Then you should select “All Stations” in front of Save button on the “Registration Table” web page and click Save button. After all connected stations are registered, you should select back to “MAC matched Station Only” to prevent unintended device registration. While “All Stations” can be registered, 4th LED – IP-PBX LED will be flashing more rapidly as warning sign.

3.1.2.3 Station User Login

Station User Login is primarily intended for Phontage and SIP extension registration. A station must register with the system each time it is connected to the system. A user may register the Phone employing a Login code (User ID) and password. Once registered, the station number is assigned. Once registered, this User ID must match the password for future registrations. The ID and password can be pre-assigned along with desired station number and a remark. A link-paired station can be assigned or pre-assigned by assign the same Desired-Number as a Master station.

Table 3.1.2.3 Station User Login

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Registered Number	Station number registered to the station, displayed only after registration	Station number	
Linked	Indicates Linked pair status and station number, displayed only	M or S	
ID	User Login ID.	12 Characters	
Password	User Login password	12 digits	
Desired Number	Station number desired for the device.	Station number	

3.1.2.4 DECT Registration

On this page, the DECT id and authorization codes are defined. In addition, a pull down menu selects one of four subscription events, subscribe, (de)subscription, mobility or display registered stations. A separate password box permits password entry to terminate (erase) all DECT subscriptions.

Table 3.1.2.4 DECT Registration

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Park Code	PARK (Portable Access Rights Key) Code : Unique System Id entered at DECT handset subscription to identify the system. To assign a PARK code, enter code and click [SAVE].	14 digits	
AC Code	Authentication Code entered at DECT handset to verify subscription. To assign AC Code, enter AC value and click [SAVE].	Up to 8 digits	
DECT Subscribe Enable	Enables the system to accept subscription from a DECT handset.		
Desired Station	Desired station number for the wireless DECT handset		
Type of Phone	Three types of phones may be selected including type for GDC-450H, type for LWS-WK and type for GDC-500H.	GDC-34x/4xx LWS-WK GDC-5xx	GDC-34x/4xx
	Press [SEND] after entering the number and type.		
DECT Unsubscribe	Terminates the subscription for a DECT handset.		
Station Number	Enter the registered station number and click [SEND], the subscription is terminated and the wireless DECT handset will no longer be serviced.	Station number	

DECT Mobility	When a DECT handset is registered at multiple systems that are networked, calls can be routed over the network to the DECT handset location.		
Station Number	Enter the registered station number, select Mobility ON or OFF and click [SEND].	Station number	
Registered Stations	Displays all registered DECT handsets.		

3.1.3 CO Line Registration

3.1.3.1 CO Channel List

CO channel list is displayed on this page; a user can view CO line numbers, and connect status of each CO channel. In addition, a user can change the usage of each CO channel or CO line name and jump to routing program for each CO line types.
Call Type, Calling Num Plan, Called Num Plan and TEI Type can be configured for BRI line. And Signal type (DTMF or Pulse) can be configured for analog CO line.

3.1.3.2 MSN Configuration

This page is enabled only when iPECS SBG-1000 has BRI port (BRIU option board is installed).

MSNs can be assigned in iPECS SBG-1000 up to 24 Calling/Called IDs. Those values are used for CLI of outgoing call and for incoming call routing. Last 3 digits of IDs are treated as flexible DID table bin while incoming call routing.

3.1.3.3 SIP ID Configuration

Various parameters must be entered for proper operation of SIP Trunking including SIP registration related information and the program for incoming call routing.

Table 3.1 SIP ID Configuration Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Table Utility	If registration is enabled (refer to Registration Option) the iPECS SBG-1000 can send the User ID to the SIP Proxy for registering the ID. Otherwise, the Authentication user ID and password are used.	USE/NOT USE	NOT USE
Registration Option	In some situations (during provisioning of the SIP Server or Proxy), it may be desirable not to attempt registration. This field may be used to determine if registration should occur.	Register Provision	Provision
Register User Name	User ID@Domain.	64 characters	
Authentication User ID	Authentication name assigned in SIP Proxy when required for registration.	64 characters	
Authentication User Password	User password as assigned in SIP Proxy when required for registration.	128 characters	
Authorized Representative ID Table Index	Representative ID table used for ID individuality		0
Contact Number	User ID.		
Name of Called Number	Name of Called Number		

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Route To (Day-Mode Period)	Call routing destination in Day-Mode		Station Group 631
Route To (Night-Mode Period)	Call routing destination in Night-Mode		Station Group 631
Route To (Timed-Mode Period)	Call routing destination in Timed-Mode		Station Group 631
See Caller Number First	Call is routing according to the 'Call Routing by Caller Number' if caller number is matched	ON/OFF	OFF

3.1.3.4 Server Information

Various parameters must be entered for proper operation of SIP Trunking including SIP proxy and Registration.

Table 3.1 Server Information Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Soft Switch Type	Allows identification of soft-switch to support extended soft-switch capabilities. KT, SK TELINK : Service Provider	Normal BroadSoft KT SK TELINK KT IMS MS OCS SKYPE CONNECT SIP-CC TI PK0 ERICSSON IMS	Normal
Proxy Server Address	SIP Proxy server IP address.	IP address	
Proxy Server Port	Default port for SIP messages to proxy.	Port	5060
Proxy Registration Timer	Time-out for registration.		3600
Domain	Domain name associated with VOIP channels. Is used in SIP "TO: header message" to SIP Server. Required when the Proxy uses a port other than 5060.	Max 32 Characters	
SIP Pound Use	ON: Send digit '#' when user press '#' OFF: The '#' is used for sending complete.	OFF ON	ON
SIP Trunk DID	ENABLE: Incoming SIP call treats as DID conversion type Use "as is" (no treatment) when SIP ID is not matched.	DISABLE ENABLE	DISABLE
DTMF Type	DTMF sending mode	INBAND 2833 INFO(DTMF) INFO(DTMF RELAY) INFO(TELEPHONE EVENT) INFO(NORTEL NETWORKS)	2833
IP AUTH USAGE	ON: Discard SIP Request (INVITE, REGISTER, NOTIFY, OPTIONS, MESSAGE …) if VIA IP and From IP are neither the server IP nor SIP Extension IP.	OFF ON	OFF

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CODEC Priority Configuration	1st . priority 2nd. priority 3rd. priority 4th. priority 5th. Priority 1) If priorities are specified CODECs then it will work for negotiation RTP data. 2) If only 1st. priority is specified and the others are none, then it will work as single CODEC only does.	none g.711-u g.711-a g.723.1 g.729 g.729-a	none
Fail Over Usage	SIP Module Service Down ON or OFF when in Registration Fail or Link Down	0: OFF 1: ON	ON
No Response Time to Fail Over	Call Setup No Response : no response timer after send outgoing setup message to SIP proxy server - 0 or [Empty] : do not use 'no response timer' 3~10 : wait for 3 to 10 second	0, 3 ~ 10 sec	5 sec
Fail Over CO Group Number	FailOver CO Group Number : Case #1 - when SIP CO line is in connected/alive state : after no response time, setup message will be re-sent using this failover CO line group Case #2 – when SIP CO line is in disconnected/OOS state : setup message will be sent using this failover CO line group	1 ~ Max Number of CO Group	none

3.1.3.5 CO Gateway Order

Each gateway Module is assigned a Sequence Number. The system uses the Sequence Number to assign logical (software) port numbers. This Sequence Number relates the hardware and software port numbers for each gateway Module.

During initial installation, VOIP module is registered before additional PSTN module is registered. It means that VOIP channel always starts with CO line 1 and the additional PSTN lines are followed. You can change this by switching orders in this menu. If you change CO gateway order, system will be restarted after your confirmation. And after restarting, you should re-configure for all CO line attributes. So, we strongly recommend this menu MUST be used only for initial installation.

3.1.3.6 Network Basic Attributes

Selecting Network Basic Attributes will display the Network Basic Attributes entry page.

Table 3.1.3.6 Network Basic Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Net Enable	Enable Networking function	1:ON 0:OFF	OFF
Net Retry Count	Reserved for future usage	00-99	00
Net CNIP Enable	The name of calling station is sent to the called system between iPECS systems. CNIP is displayed at called party stations display based on the programming	1:ON 0:OFF	ON
Net CONP Enable	Reserved for future usage	1:ON 0:OFF	OFF

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Net Signal Method	Select the information element type for QSIG supplementary service message.	1:FAC 0:UUS	FAC
Net Cas Enable	It is not used.	1:ON 0:OFF	OFF
Net VPN Enable	Reserved for future usage	1:ON 0:OFF	OFF
Net CC Retain Mode	It is not used.	1:ON 0:OFF	OFF

3.1.3.7 Network Supplementary Attributes

Selecting Network Supplementary Attributes will display the Network Supplementary Attributes entry page.

Table 3.1.3.7 Network Supplementary Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Net Transfer Mode	Select type for Transfer and Call forward – Rerouting or Join	1:RERT 0:JOIN	RERT
TCP Port for Blf	TCP port for sending BLF message to BLF Manager	0000-9999	9500
UDP Port for Blf	UDP port for sending BLF message to BLF Manager	0000-9999	9501
Blf Manager IP	IP Address of BLF Server used only when iPECS is configured with LDK systems for Voice Networking.		0.0.0.0
Duration of BLF Status	Duration of BLF status message sending to BLF Server	01-99 (100 msec)	10
Multicast IP	IP address of Multicast for BLF service		0.0.0.0
Net Trans Rcl timer	Network transfer fault recall timer to be used when no responses from other systems.	001-300 (msec)	10
NET Reroute CO Group	The start times for Day, Night and start and end times for Timed modes are entered for each day of week. After Timed end time the mode goes to day if time is less than Night mode.	MFIM & MFIM100:& IPECS- Micro& IPECS-50 00-20 Other MFIM: 00-72	0

3.1.3.8 Network CO Line Attributes

Selecting Network CO Line Attributes will display the Network CO Line Group entry page. Enter the desired data and click Load to display the Network CO Line Group.

Table 3.1.3.8 Network CO Line Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Net CO Group	Networking CO group programming for Networking call.	00-24	00
Net CO Line Type	Select network CO Line Type	1:NET 0:PSTN	PSTN

3.1.3.9 Network Numbering Plan Table

Selecting Network Numbering Plan Table will display the Network Numbering Plan Table data entry page.

Table 3.1.3.9 Network Numbering Plan Table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
System Usage	Select system usage	0:NET 1:PSTN	NET
Numbering Plan Code	'*' means any digits can be inserted between 0 ~ 9. The digits followed by '#' is a internal station number	16 digits	
Numbering Plan CO Group	'00' means an internal net station number.	00-24	. .
CPN Information	Flex 1: ISDN CPN INFORMATION Flex 2: (FLEX BTN 1- 4) 1: 00 CPN INFORMATION 01 2: 00 CPN INFORMATION 02 3: 00 CPN INFORMATION 03 4: 00 CPN INFORMATION 04	16 digits	
Alt Speed Bin	Alternative Dial Number (System SPD Bin) when the networking path has a fatal problem.	200-999 or 2000~4999	
MFIM (E) IP Address	IP Address of destination MFIM/E system only when iPECS systems are configured for Voice Networking		0.0.0.0
MFIM(E) Port	Port Number of destination system for Networking.	0000-9999	5588
Digit Repeat	When the number plan code, see above, is for PSTN call or transit-call, this number code can be enveloped in SETUP message or not whether if this field is set or not.	Yes No	No
Net PSTN Enblock	Choose "Transit-out Public Line" as Enblock or Over-lap.	Yes No	No
PSTN CLI Method	NET: Send network station number for CLI PSTN: Send full CLI (eg, 02-450-1000)	NET PSTN	NET
CO Attendant Code CLI	Determine whether if Centralized ATD CLI is sent or not when slave system makes transit call.	On Off	Off
Firewall Routing	Select IP address (Firewall IP address or Non-firewall IP address). If the destination system(VOIM) is in same VPN then Non-firewall IP address should be sent. Otherwise the firewall IP address should be sent. ON : Send firewall IP address OFF : Send Non-firewall(Internal) IP address	On Off	ON
Transit Out Auth COS	When there's a transit out call request from slave system user by seizing CO line, apply COS according to the authorization code.	Yes No	No
SMDR Dgt Hide	Determine to display dialed digit of transit out call or not at the slave system ; it can contain authorization code.	Yes No	No

3.1.3.10 Network Feature Code Table

Selecting Network Feature Code Table returns the data entry page.

Table 3.1.3.10 Network Feature Code Table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Net Feature Code	Networking Feature Code programming for Networking paging call.	16 digits	None
Net Feature Destination	INT PAGE ZONE : 1-10 ALL CALL PAGE ZONE : INT(1), ALL(3)	3 digits	N/A

3.1.3.11 LCR Control Attribute

Selecting LCR Control Attributes will display the LCR Control Attributes data entry page.
The LCR Tables provide a mechanism to define the database, which will route outgoing calls, particularly long distance, using the most cost effective route. User dialed digits are compared to table entries and modified appropriately based on time of day, day of week, and assigned routes. There are four LCR Tables, LCR Control Attributes, LCR Leading Digit Table, LCR Digit Modification Table, and LCR Initialization Table.
The LCR Control Attributes Table, among other items, allows assignment of the LCR Access Modes. The LCR Access Modes defines the user operations that will access the LCR feature. The LCR Access Modes are:

- LCR Disabled.
- Loop (user dials ‘9’ or CO Group code (8xx), or presses a Loop button).
- Loop and Internal (user dials digits without a CO Access Code prefix).
- Loop and Direct CO Line (user dials CO Line Access Code (88xx) or pressing a {CO line} button).
- Loop, Direct CO Line, and Internal.
- Loop, Direct CO Line, and Internal and Direct Loop.

In addition, days of the week are grouped into zones (Day Zones) and the time of day can be set into three groups (Time Zones). Table 3.1.3.11 provides general descriptive information and input ranges.

Table 3.1.3.11 LCR Assignment

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
LCR Access Mode	This entry defines the effective LCR modes, the modes by which the user can access LCR.		
Day Zone	Each day of the week is assigned to a Day Zone (1~3). The active Day Zone is the Zone assigned to the current day of the week.	days of the week	
Time Zone1	This entry defines the hours of the day during which Time Zone 1 is active. Note hours not defined in Time Zone 2 and 3 are automatically part of Time Zone 1.	00~24	00~24
Time Zone2	This entry defines the hours of the day during which Time Zone 2 is active.	00~24	
Time Zone3	This entry defines the hours of the day during which Time Zone 3 is active.	00~24	

3.1.3.12 LCR LDT (Leading Digit Table)

Selecting LCR-LDT (Leading Digit Table) will display the LCR-LDT data entry page.
The Leading Digits Table is used to analyze the user-dialed digits to determine an appropriate Digit Modification Table Index. The Table is divided into bins. The digits (up to the first 12) dialed by the

user are compared with the entries in the Leading Digit Table. In addition, indices to the Digit Modification Table are defined for each Time Zone of each Day Zone. Table 3.1.3.12 provides a brief description and entries for the Leading Digit Table.

Table 3.1.3.12 LCR Leading Digits Table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Compared Digits	Up to 6 digits that, if matched by the user dialed digits, will access the DMT Indices of the associated Leading Digit Table bin.	12 digits	
DMT1	This entry defines the Digit Modification Table index (00~99) for each Time Zone for Day Zone 1. The appropriate index will be selected for the current Day and Time Zone. One entry (DMT index) is made for each Time Zone, six (6) digits.	Must be 6 digits 3 DMT indices	
DMT2	This entry defines the Digit Modification Table index (00~99) for each Time Zone for Day Zone 2. The appropriate index will be selected for the current Day and Time Zone. One entry (DMT index) is made for each Time Zone, six (6) digits.	Must be 6 digits 3 DMT indices	
DMT3	This entry defines the Digit Modification Table index (00~99) for each Time Zone for Day Zone 3. The appropriate index will be selected for the current Day and Time Zone. One entry (DMT index) is made for each Time Zone, six (6) digits.	Must be 6 digits 3 DMT indices	

3.1.3.13 LCR DMT (Digit Modification Table)

Selecting LCR-DMT (Digit Modification Table) will display the LCR-DMT data entry page.

Using the index determined from the analysis of the LCR Leading Digits Table, the dialed number is modified in accordance with the Digit Modification Table and sent over the CO group assigned for the index.

Digits of the dialed number can be deleted based on the "Removal Position" and "Number of digits to be removed" entries and a digit stream can be inserted in the resulting number. Counting from the first dialed digit, the Removal Position defines the location of the digit where removal begins and, the Number of digits to be removed defines the number of digits to remove. The "Add Digits" are then inserted in the resulting number at the digit position assigned by the Add Position entry. The resulting number is then dialed over the CO path assigned. If the assigned path is not available, the "Alternate DMT index" is used to determine the number and CO path to be used.

Table 3.1.3.13 LCR Digit Modification Table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Add Digits	This entry defines the digit stream to insert in the number after digit removal. Digits 0~9, '*', '#', and special characters, P: timed Pause D: Dial tone detect F: Billing station number	25 digits	
Removal Position	This entry defines the position of the digit where removal is to begin, starting with the 1st dialed digit (01).	01~12	1
Number of digits to be removed	This entry defines the number of digits to remove starting at the "Removal Position	00~12	0
Add Position	This entry defines the position in the number, after digit removal, where the Add Digits are inserted.	01~13	1

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CO Group	This entry defines the CO Group that the system will attempt to use for the call.	1~5, 21	1
Alternative DMT Index	This entry defines an Alternate Digit Modification Table Index to use if no path is available in the assigned CO Group.	00~99	

3.1.3.14 LCR Table Initialization

Selecting LCR Table Initialization will display the LCR Table Initialization data entry page.
The LCR Table Initialization allows global values to be assigned to the various Digit Modification Table entries. In addition, the LCR Leading Digits and LCR Digit Modification Tables can be initialized to the default (no entries) state.

3.1.4 Auto Attendant

3.1.4.1 Voice Mail Group Number

A user can program Voice Mail Group Numbers using this page.

3.1.4.2 System Prompt Upload

A user can upload system prompt information using this page; system prompt can be uploaded with .rom format or .wav format.

1. Prompt Upload menu: upload rom file
   ??96Wxxxx.rom (?? is nation, i.e., PM, IT, GS, KR, etc.; xxxx indicates the version)
2. Individual Upload menu: upload wave file
   1.wav – 999.wav (system prompt should be G.711 u-law wave file format)

3.1.4.3 Announcement Upload & Download

A user can upload and download announcements with .rom format or .wav format.

1. Individual upload/download: upload .wav file
   1.wav – 72.wav
2. SysGreeting upload/download: upload .rom file
   SGTYPE1.rom

3.1.5 FAX

FAX Configuration page displays the logical station number of SLT port; a user can change FAX utilization of each SLT port and also can make T.38 service enabled for FAX.

3.1.6 Numbering Plan

3.1.6.1 Numbering Plan

Feature dial codes for the system can be assigned using the System Flexible Number Plan. Feature codes should be one (1) to four (4) digits in length and must not conflict. For example, Feature codes 53 and 536 represent a conflict. The system will not update the database until correct data is entered. The Table provides a brief description for each feature and default codes.

Table 3.1.6.1 Flexible Numbering Plan

ATTRIBUTE	DESCRIPTION	DEFAULT
Attendant Call	Dial code to call Main Attendant	0
Alarm Reset	Code to reset Alarm contacts	65
Direct Call Pick-Up	Dial code to activate Directed Call Pick-up	7
Group Call Pick-Up	Group Call Pick-up dial code	**
Answering Machine Emulation	Dial code to assign an Answering Machine Emulation Flex button	64
Call Forward	Code to activate Call Forward.	54
Door Open	Dial code to activate Door contact (open door)	#*
Call Coverage Ring	Code for Call Coverage button	*#
Access Random CO Line	Dial code to access the 1st available CO Line in any accessible group	9
CO Line Group Number	Dial code to access a CO Line from a group	801-802
Access Individual CO Line	Dial code to access a specific CO Line/IP Channel	8801-8805
Paging Zones	Page Zone access dial codes	501-510
All Call Paging	All Call Page access dial code	500
Answer Paging (Meet Me)	Meet-Me-Page answer dial code	511
Call Park Locations	Dial code to place/retrieve a call in system Park Orbit	601-610
Group Pilot Number	Station group pilot number	620-631
Do-Not-Disturb (DND)	Dial code to activate Do-Not-Disturb	53
DND/FWD Cancel	Code to cancel DND/FWD	59
Leave Call-Back	Code to activate Message Wait/Call Back	56
Answer Call-Back	Code to return Message Wait/Call Back	57
Camp-On Answer	Dial code to answer a Camped-On call	66
Last Number Redial (LNR)	SLT Last number redial feature access dial code	52
Speed Dial Program	SLT Speed Dial programming access code	55
Speed Dial Access	SLT Speed Dial access code	58
CO Line System Hold	Code to place a CO Line call on System Hold	67
Retrieve Any Of Held CO Line	Dial code to access last CO Line or IP channel from Hold	8*
Retrieve Specific Held CO Line	Dial code to access a specific CO Line/IP channel from Hold	8#
Company Directory Name	Code to check and change recording station subscribe name of Company Directory feature.	68

3.1.6.2 E.164 Management

SIP extension (ex. iPECS Communicator) may have own phone-book and it can be written in E.164 format. E.164 format telephone number is needed to be converted to make it go out through PSTN. And an incoming external calling party number that is written in normal PSTN format is needed to be converted to E.164 format to match SIP extension’s phone-book telephone number that is written in E.164 format. E.164 management can support the conversion from E.164 format to normal PSTN format for outgoing call and from normal PSTN format to E.164 format for incoming CLI.

Table 3.1.6.2 E.164 Management

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Area Code	My Area Code (decision of Long Distance or Local call). Usage> Digit Conversion between E.164 and normal format.	Max 5 digits	
International Access Code	00XXXX (to access International call service provider) Usage> Digit Conversion between E.164 and normal format.	Max 5 digits	
Local Number Digit Count	0 - 30, To distinguish Local or Long distance call if incoming calling party number of local PSTN call is possibly presented without area code. Usage> Digit Conversion from normal to E.164 format.	00-30	
Leading Zero Insertion For Area Code	'YES' will be required if Area Codes have Leading 'Zero' prefix. Usage> Digit Conversion from E.164 to normal format.	NO/YES	Depends on Nation
My Area Code Insertion	Insert my area code if outgoing called number is decided as a local PSTN call. It depends on national requirement. Usage> Digit Conversion from E.164 to normal format.	NO/YES	NO
Exceptional Conversion : for Outgoing Dial Number	Convert outgoing called number that is written in E.164 format starting with my nation code followed by special/exceptional telephone number that does not include area code. Usage> Digit comparison goes first with this table than area code comparison.		
Exceptional Conversion : for Incoming CLI Number	Convert incoming calling number that is written in normal format starting with other kind of international access code or special/exceptional telephone number that does not include area code. Usage> Digit comparison goes first with this table than international, nation and area code comparison.		

Example case 1: Phone-book dialing that is written in E.164 format from an SIP extension to PSTN

- E.164 format telephone number is needed to be converted to make it go out through PSTN.
- Basic condition : My Nation=82, My Area=031, International Access Code=001
- Phone-Book Dialing (E.164) ➔ PSTN (Normal)

My Area Code Insertion	**NO**		**YES**	
Leading Zero Insertion For Area Code	**NO**	**YES**	**NO**	**YES**
Local Call +82314501234	4501234		314501234	0314501234
Long Distance Call +8223471234	23471234	023471234	23471234	023471234
International Call +61416221234	00161416221234			

- Exceptional Conversion Table : for Outgoing Dial Number
  - Without Exceptional Table Conversion
    1. ‘15’ is no my Area Code, Leading Zero Insertion For Area Code=YES
       +8215881234 ➔ 015881234

2. ‘15’ is my Area Code, My Area Code Insertion=NO
   +8215881234 ➔ 881234
- With Exceptional Table Conversion : 1588 ➔ 1588
  1. Regardless of condition
     +8215881234 ➔ 15881234
- Exceptional Conversion Table implementation rule
  1. This is normally used for special service prefix.
  2. Exceptional Table Conversion goes first than other conditions.
  3. Exceptional Table Comparison is implemented only for a National call (when leading '+Nation Code' is my nation).
     <example>
     +1588xxxx : is not exceptional converted because it is regarded as an other international call
     +821588xxxx : is exceptional converted because '82' is my nation and it is a national call

Example case 2: Incoming CLI conversion from normal telephone number to E.164 format

- An incoming external calling party number that is written in normal PSTN format is needed to be converted to E.164 format to match SIP extension’s phone-book telephone number that is written in E.164 format.
- Mandatory condition : E.164 Call Log CLI=ON for SIP extension
- Basic condition : My Nation=82, My Area=031, International Access Code=001
- First condition : calling party type 'National' from PSTN
- Normal PSTN CLI ➔ E.164 format to give CLI to SIP extension
  - Local Call
    1. 0314501234 => +82314501234
       additional condition : N/A
    2. 314501234 => +82314501234
       additional condition : N/A
    3. 4501234 => +82314501234
       additional condition : Local Number Digit Count = '7'
       This is local call that is not presented with area code. In this case 'Local Number Digit Count' is required.
  - Long Distance Call
    1. 023471234 => +8223471234
       additional condition : N/A
    2. 23471234 => +8223471234
       additional condition : N/A
  - International Call
    1. 00161416221234 => +61416221234
       additional condition : my International Access Code = '001'
    2. 001+61416221234 => +61416221234
       additional condition : N/A
  - Exceptional Conversion Table : for Incoming CLI Number
    1. usage #1 : for other kinds of International Access Code
       1) table conversion

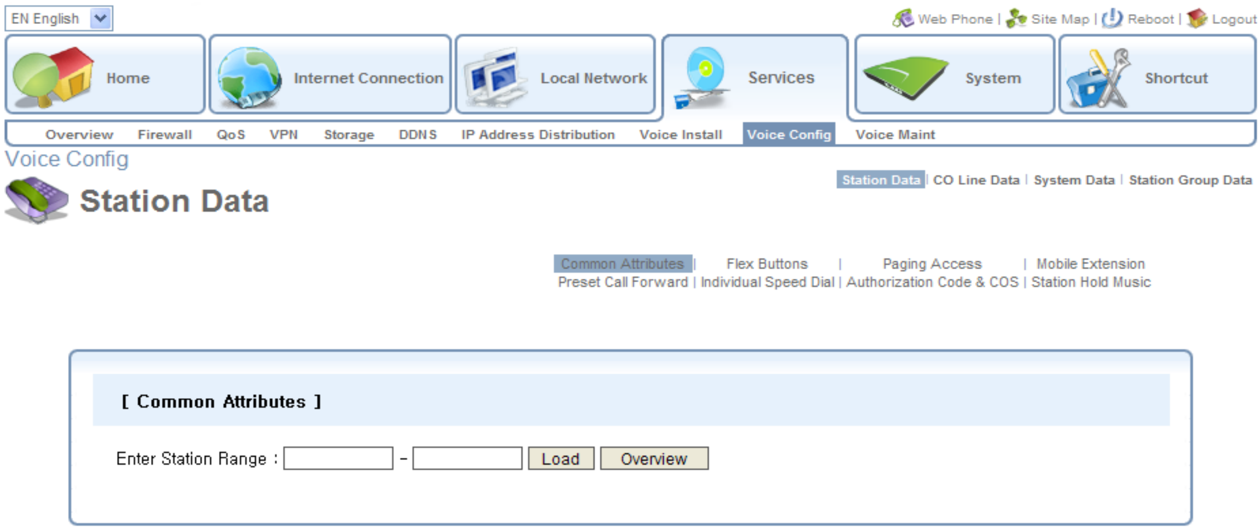
002 ➔ '<empty>'
00700 ➔ '<empty>'

2) then,
00261416221234 ➔ +61416221234
0070061416221234 ➔ +61416221234

2. usage #2 : for a exceptional number for a special service
   1) table conversion
   1588 ➔ 821588
   2) then,
   15881234 ➔ +8215881234

3.2 VOICE CONFIGURATION

In this section, user can see or change database for Station, CO Line, System, Station Group (shown).



3.2.1 Station Data

3.2.1.1 Common Attributes

The Common Attributes Table defines features and functions available to the station.

Table 3.2.1.1 Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CLI Table Index	Default outgoing CLI Table Index.	0-5	1
CLIR Service	CLIR (Calling Line Identification Restriction), an ISDN service, removes the calling party ID sent from the ISDN to the called party with a RESTRICT instruction in the SETUP message. If enabled, the system will send the RESTRICT instruction to the CO when an outgoing ISDN call is placed.	ON OFF	OFF

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
COLR Service	COLR (Connected Line Id Restriction), an ISDN service, removes the connected party ID sent from the ISDN to the calling party with a RESTRICT instruction in the CONNECT message. If enabled, the system will send the restrict instruction to the CO when the station places an ISDN call.	ON OFF	OFF
CO Group Access	Stations can be allowed or denied access to CO group.	1, 2, 3, 4, 5	1, 2, 3, 4, 5
Station Forward No Answer Timer	This timer determines the duration the station will ring prior to Ring-No-Answer Forward. This setting affects both manual and Preset Call Forward and overrides the Call Forward No Answer Timer in System Call Feature Timer	000-600 seconds	000
Active in OOS	If a station is Out-of-Service and has previously forwarded calls, the system will forward the calls if enabled.	ON OFF	OFF
DID Call Wait	When a busy station receives an external call, the call may queue to the station instead of receiving a busy tone. With Call Wait, the caller will hear a Ring-back tone and the CO line LED flashes.	ON OFF	ON
Voice Over	Enables use of Voice Over by station	ENABLE DISABLE	DISABLE
Prime Line	Enables Delayed Prime Line (Idle Line) activation, see Idle Line Selection.	HOT WARM	WARM
Idle Line Selection	When a station goes to an off-hook condition (lifts handset or presses **[speaker]** button), the system normally provides intercom dial tone. In place of dial tone, the station can be programmed to activate Flex button as if pressed, access a CO Line, access CO Group or call a Station	No Selection Flex Button (1-24) CO Line (1-8) CO Group (1-5) Station	No Selection
CO PGM	A station can be permitted to change the CO Line numbers (ports) associated with a CO Line button.	Disable Enable	Disable
Automatic Talk Recording Dest.	When Auto Call recording is defined for a station, the recording Phontage station number or station group number is defined here.	Station Station Grp	
Headset Ring	Selects device to receive incoming ring signals (Speaker, Headset or Both).	Speaker Headset Both	Speaker
Headset or Speaker Mode	Select Speaker or Headset mode for the IP Phone.	Headset Speaker	Speaker
LCD Display LED	The LCD LED, upper left of LCD, may be used for Intercom Call ring Indication or Message Wait Indication.	MWI Ring	Ring
Back Light Usage	The backlight of the IP phones is assigned to stay off, light only when the station is busy, or light constantly.	Always Off Busy Only Always On	Busy Only
Send SLT CLI Info	When allowed, the system sends CLI information to the SLT	ON OFF	ON

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Video Show on Calling	When voip video door phone rings to a video-enabled IPKTS handset, the video streaming commences immediately while the IPKTS handset is in the ringing and the video stream continues when answered. - OFF : normal implementation (video starts after answer) - ON : video stream from this Video Door Phone to the ringing video-enabled LIP Phone even though this is alerting stage. - Condition : A VOIP channel for RTP-Packet-Relay purpose is required to serve ring-back-tone generation via a DSP channel That is because, system does pre-answer to the Video Door Phone even though the receiving station is on alerting state.	ON OFF	OFF
E.164 Call Log CLI	Give E.164 format CLI to SIP extension for an incoming CO call.	ON OFF	OFF

3.2.1.2 Flex Buttons

Each Flex button on a LIP Phone can be assigned a function (TYPE) as listed:

- Empty
- Number Plan
- User Program code
- Station Speed Dial
- System Speed Dial

After selecting the Type for a button, enter the appropriate value (Where applicable).

3.2.1.3 Paging Access

Each LIP Phone is assigned to be able to receive announcements from each Page Zone. A station can be assigned to any, all, or no page zones. The iPECS SBG-1000 system supports up to 10 Internal Page Zones. By default, all stations are assigned to Zone 1.

📖 ***NOTE:*** *A station not assigned to any page zone will not receive any page announcements.*

3.2.1.4 Executive / Secretary Table

Stations can be paired as Executive/Secretary pairs so that when the Executive enters DND, the call to the Executive is automatically routed to the Secretary. Up to 10 Executive/Secretary pairs can be defined. An Executive may have multiple Secretaries and a Secretary can be assigned to multiple Executives. A Secretary of one pair may be the Executive of another however, assignments that form a loop-back are not allowed.

The “CO Call to Sec” option will route all CO calls to the Executive to the defined Secretary‘s station regardless of the Executive‘s station status. The “Call Exec if Sec DND” option will route Executive calls back to the Executive if the Secretary is in DND. The Exec Grade permits higher grade Executives to override the Executive/Secretary Forward feature to call a lower grade

Executive (Korea only). The highest grade is 1 and the lowest grade is 12. The “ICM Call to Sec” option will route all internal calls to the Executive(except for calls from higher or same grade executive) to the defined Secretary‘s station regardless of the Executive‘s station status. The “Sec Auto Ans” option will make the Secretary answered forcedly with speaker phone as soon as the Executive calls to the Secretary.

3.2.1.5 Mobile Extension Table

Selecting Mobile Extension Table will display the Mobile Extension data entry page.
A mobile phone can be used in conjunction with an LIP Phone. The Mobile phone can access system resources available to the user’s wired phone and will receive ring for incoming iPECS SBG-1000 calls. The user may be allowed to enable the Mobile extension and define the mobile number. The system can be defined to employ a specific CO Line Group to place calls to the Mobile phone. In addition, the mobile phone can be assigned to receive hunt group calls to the primary extension. Also, parameters for notification of new VSF voice mails can be defined.

Table 3.2.1.55 Mobile Extension Table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Internal Access	The user may be allowed to access internal call from the mobile extension.	Disable/Enable	Disable
Usage	Mobile extension feature can be enabled and Fail Over to Mobile extension can be included.	Disable Mobile Ext Fail Over	Disable
Hunt Enable	When the paired station is a member of a hunt group (Circular or Terminal), group calls can be sent to the active mobile extension.	Disable Enable	Disable
VSF/VMIM Notify	Enables outbound notification by the system when the VSF has unheard messages.	Not Use Use	Not Use
Notify Retry	Defines the number of attempts the system will make to complete a notification when receiving busy/no-answer.	1 – 9 Times	3 Times
Retry Interval	Defines the time between notification attempts. If a notification fails, the system will retry after the timer expires.	1 – 3 Minute	3 Minute
Notify CLI	When the system sends CLI to the mobile extension, the CLI can be either the original caller’s CLI or the CLI of station.	Caller My Ext.	Caller
Call Back	If it is set to “ON”, incoming mobile extension call will be released before answered and system places a call to mobile extension. After mobile extension answers, the dial tone is provided and mobile extension can make internal or external call.	OFF ON	OFF
Delay Timer	Mobile extension call will be placed after delay time.	0 ~ 255	0
CO Group	CO group used to call the mobile extension.	0 ~ 5	01
Telephone Number	Telephone number or CLI of the Mobile extension.		Not assigned
CLI Number	When the mobile Telephone number and CLI do not match, the CLI entered here is used to authorize incoming calls from the mobile.		Not assigned

3.2.1.6 Station Voice Mail Attributes

Various Station Voice Mail attributes can be configured in this menu including voice mail to e-mail notification and voice mail backup.
Some of these options can be worked only when Station VM Feature Usage is set to "ON".
Clicking "VM COS" link in Station Voice Mail Attributes will display VM COS Attributes window.

Table 3.2.1.66.1 Station Voice Mail Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
VM Message No	Number of Voice Mail that is left to the station.		
VM COS	A station has Voice Mail COS from 1 to 5. By COS grade, Voice Mail COS attribute can be assigned.	1-5	1
Administrator Mailbox	Administrator Voice Mail feature can be activated.	Disable Enable	Disable
Announce Only Mailbox	If this option is set to ON, caller cannot leave a message to the station, only can hear the station's user greeting.	Disable Enable	Disable
Announce Only Option	When Announcement Only Mailbox is ON, after hearing a station's user greeting, next step is decided by this option – go to previous menu or call is disconnected.	Previous Menu Hang Up	Previous menu
Company Directory – First Name	First Name of Company Directory is assigned.	Max 12 Char	N/A
Company Directory – Last Name	Last Name of Company Directory is assigned.	Max 12 Char	N/A
Cascade Mailbox	A destination station number of message cascading feature is assigned.	Station No	N/A
Cascade Type	Message cascading type is programmed.	Disable Copy Immediate Copy Urgent Move Immediate Move Urgent	Disable
Message RW/FF Time	Rewind / Fast Forward time of left message is defined .	03 - 99	04
VSF MSG – SMTP Mail Server Address	The VSF includes notification of new messages to the user's voice mail. This field defines the user's e-mail mail server for the notification.	IP v4 address Or Mail server name	
VSF MSG – User Mail Address	The VSF includes notification of new messages to the user's voice mail. This field defines the e-mail address to notify when a new message is received at the VSF or VMIM.	e-mail address	
VSF MSG – SMTP Mail Server ID	The VSF includes notification of new messages to the user's voice mail. This field defines the e-mail address to notify when a new message is received at the VSF or VMIM.		
VSF MSG – SMTP Mail Server Password	Unified Mail server password		
VSF MSG – Attached Message	When e-mail notification of a new VSF message is enabled, the e-mail may include the voice mail as a wav file attachment.	OFF ON	ON
VSF MSG – Date/Time	When ON, play the data/time stamp of VSF message	OFF ON	ON
VSF MSG – Delete Message	After sending VSF mail, attached VSF message is deleted, when this option is ON.	OFF ON	OFF

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
VSF MSG – SMTP Security	Choose SMTP Security.	No Security SSL TLS	No Security
VSF MSG – SMTP Port	Choose SMTP Port	1 - 65535	25
VSF MSG – Sender Mail Address	Program VSF mail Sender mail Address.	e-mail address	
VSF Backup Delete Option	A Phontage may monitor voice messages for another station as a backup. The Phontage will include the message count for the station in the Voice message count. When enabled here, the Phontage may delete messages for the station.	OFF ON	OFF
VSF Backup Station	A Phontage may monitor voice messages for another station as a backup. The Phontage will include the message count for the station in the Voice message count. This field defines the Phontage station number that will be used as the VSF back up.	station	
VSF Backup Prompt	A Phontage may backup VSF Prompts.	OFF ON	OFF

Table 3.2.1.66.2 VM COS Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Greeting Length	This defines maximum user greeting length.	0-99	60
Message Length	This defines maximum user message recording time.	0-999	0
Number Of Messages	This defines maximum number of voice mail message.	0-250	0
Retention Time	Voice mail messages will be automatically deleted after this amount of days.	0-99	0
E-Mail Notification	E-mail notification can be enabled or disabled.	Disable Notification Only Notification and Delete	Notification and Delete
Future Delivery Message	Future Delivery message can be enabled or disabled.	0: OFF 1: ON	OFF
Confirm Message Receipt	Confirm message receipt can be enabled or disabled.	0: OFF 1: ON	OFF
Private Message Mark	Private message mark can be enabled or disabled.	0: OFF 1: ON	OFF

3.2.1.7 Preset Call Forward

Stations can be programmed so that incoming CO and Intercom calls are forwarded to a preset station or station group. This allows an external call or internal call to initially ring at a station and forward to a pre-determined destination. Preset Call Forward can be separately assigned Unconditional, Internal Busy, Internal No Answer, External Busy or External No Answer preset forwarding to any station, hunt group or system speed dial bin (off-net). As a default, no Preset Call Forward is assigned.

For Transfer Mail-Box, enter the Station Group number of the Voice Mail group; this will permit LIP Phone users to forward calls directly to the desired user’s Voice Mail-Box.

3.2.1.8 Individual Speed Dial

Each user can store commonly dialed numbers for easy access using Individual Speed Dial bins. Each station has access to 20 Speed Dial numbers; each Speed Dial number can be up to 23 characters in length and may include special instruction codes.

Special instruction codes available are:

‘*’ as 1st digit — Activate Display Security, do not display number.

LIP Phone users may assign a Flex button for One-Touch access to a specific Speed Dial bin. In addition, the LIP Phone user may assign a Telephone number directly to a Flex button. In this case, the telephone number is allocated to the highest numbered available Individual Speed Dial bin.
Stored speed dial number should not include CO access code.

3.2.1.9 Authorization Code & COS

Authorization codes (up to 12 digits) are used to control access to system resources and facilities. Voice Mail-box and certain Call Forward types may require the input of a valid Authorization code. The Station entries are associated with individual stations.

All stations are assigned a Class-of-Service (COS), which determines the user’s ability to dial certain types of calls (refer to Station COS Table). Separate COS assignments are made for Day, Timed and Night Mode operation. As a default, all stations are assigned with a Station COS of 1 for all modes (No restrictions).

Table 3.3.1.7 STATION COS

STATION COS	RESTRICTIONS
1	No restrictions are placed on dialing from the station.
2	The assignments in Exception Table A are monitored for allow and deny numbers.
3	The assignments in Exception Table B are monitored for allow and deny numbers.
4	The assignments in both Exception Tables A & B are monitored for allow and deny numbers.
5	The leading digit cannot be a Long Distance code and assignments in Exception Table C apply.
6	Number of digits cannot exceed LD digit count and assignments in Exception Table C apply.
7	Intercom and Emergency number calls are allowed. Incoming and transferred calls are allowed.

A station must be allowed Off Net Fwd to forward external incoming calls outside the system or establish a CO-to-CO connection.
If Station Account is set to “ON”, the station user must enter an authorization code to access CO lines.

3.2.1.10 Station Hold Music

iPECS SBG-1000 supports two types of Hold Music for Intercom calls. One type of MOH is Hold Tone, and the other type is Record Play. A VSF announcement may be recorded and played as MOH to the connected caller.

3.2.2 CO Line Data

3.2.2.1 Call Routing by Line

This page is enabled only when iPECS SBG-1000 has FXO ports (CSIU, CIU1, CIU2 or CIU4 option board is installed).

Each CO line is assigned to signal a station or group for an incoming call (Ring). Separate ring assignments are made for Day, Night, and Timed Ring modes. When assigned to ring to a VSF announcement, the call can be dropped automatically after the assigned announcement by checking the “Auto Drop”.

When CO Lines are programmed to Ring to a VSF Group as an Automated Attendant, the Ring signal can be on an immediate or delayed basis allowing other stations/groups to be assigned Ring and answer prior to signaling the AA. The delay is defined in seconds from 00 to 30.

When a call is received, the system may use the ICLID (Incoming Caller ID) to route the call. The system will delay routing a call for ICLID ring timer while awaiting ICLID. If ICLID ring timer is 0, ICLID routing is disabled.

3.2.2.2 Call Routing by Called Number

This page is enabled only when iPECS SBG-1000 has BRI port (BRIU or BRIU2 option board is installed).

These characteristics are required for proper operation of the system and BRI incoming call destination selection.

Table 3.2.2.2 Call Routing by Called Number

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Name of Called Number	Name of Called Number		
Route To (Day-Mode Period)	Call routing destination in Day-Mode		Station Group 631
Route To (Night-Mode Period)	Call routing destination in Night-Mode		Station Group 631
Route To (Timed-Mode Period)	Call routing destination in Timed-Mode		Station Group 631
See Caller Number First	Call is routing according to the ‘Call Routing by Caller Number’ if caller number is matched	ON/OFF	OFF

3.2.2.3 Call Routing by Caller Number

The system can employ Incoming Calling Line ID (ICLID) to determine the routing of incoming external calls. The system will compare the received ICLID, and if a match is found, will route the call to the destination defined in the Ring Assignment Table index.

Table 3.2.2.3 Call Routing by Caller Number

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Caller Number	ICLID (Incoming Caller ID) to match for the index. If the Caller ID matches the Table entry, the index is used to select the route.	23 Digits	None
Caller Name	ICLID name that is sent by the system to the destination for the ICLID routed call.	12. Character	None

3.2.2.4 Ring Assignment Table

If the Incoming Caller ID matches an entry in Call Routing by Caller Number, the index from the Table is used to determine the call routing from the Ring Assignment Table. Separate ring assignments are made for Day, Night, and Timed Ring mode for each index, 001 to 250, in this table. When assigned to ring to a VSF announcement, the call can be dropped automatically after the assigned announcement by checking the “Auto Drop”.

When CO Lines are programmed to Ring to a VSF Group as an Automated Attendant, the ring signal can be on an immediate or delayed basis (00-30 sec.) allowing other stations/groups to be assigned ringing and answer prior to signaling the AA.

3.2.2.5 Exceptional Call Routing

When a DID line or DISA user dials an invalid/vacant or busy station number the caller will be sent to the assigned destination. The destination is separately defined for invalid, busy, and No Answer conditions and can be defined as the Attendant, busy tone or a Station Group. For calls on a DID line to a busy station, DID Call Wait can be assigned, refer to Station Common Attributes in section 3.2.1.1.

Also, for DID calls only, announcements (prompts) can be sent from the VSF to the caller for various conditions, busy, error, DND, No Answer, or Attendant Transfer.

3.2.2.6 Call Routing by Auto Attendant

The system incorporates Interactive Voice Response (IVR) capabilities called Customer Call Routing (CCR). After or during a VSF Announcement, the caller may dial digits to select a destination or route for the call. The CCR Table defines the destination type and value associated with the digit dialed by the caller in response to the index, a VSF Announcement (01-70). Up to 70 single-level Audio Text menus may be assigned or, multi-level menu structures (maximum 70 levels) can be established using one menu as a destination for the previous level.

Table 3.3.2.6 Customer Call Routing Auto Attendant Destinations

DESTINATION	VALUE RANGE
Station	10~33
Station Group	620~631
Common Speed Dial	200~999
DVU Announce	01~70
DVU Announce and disconnect	01~70
Networking	Network station number)
Paging Zone	01~10
All Call Paging	n/a
Voice Mail	620~631 & 10~33
Company Directory	n/a
Record VM Greeting	n/a

3.2.2.7 Common Speed Dial

Commonly dialed numbers can be stored by the Attendant or the Administrator in Web Admin. for easy access by stations allowed use of Common Speed Dial bins. Up to 800 Common Speed Dial numbers are available; each Speed Dial number can be up to 23 characters in length and may include special instruction codes.

Special instruction codes available are:

‘*’ as 1st digit Activate Display Security.

LIP Phone users may assign a Flex button for One-Touch access to a specific Common Speed Dial bin.

Stored speed dial number should not include CO access code.

3.2.2.8 Common Speed Zone

The system has 10 Common Speed Dial zones. Common Speed Dial bins assigned to a zone are only available to stations allowed access to that zone. Each zone can be assigned to apply the appropriate Station COS for the speed dial number prior to dialing.

3.2.2.9 CO Hold Music

iPECS SBG-1000 supports two type of Hold Music for CO Line calls. One type of MOH is Hold Tone, and the other type is Record Play. A VSF announcement may be recorded and played as MOH to the connected caller.

3.2.3 System Data

3.2.3.1 System Attributes

The default system codec is defined as G.711. The system codec will be used on all internal communications as well as for other remote devices.

More system attributes are described in the following.

Table 3.2.3.1 System Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
DISA Retry Count	A DISA user is allowed to retry erroneous authentication code entries. This entry sets the number of retries before the system disconnects.	1~9	3
Old Auth Code Usage	System Authorization codes are entered by the user as "*" when this entry is set to ON or "*"+ the Auth code index when this entry is set to OFF.	OFF ON	ON
End code(#) usage in System Auth Code	If this option is set to ON, End code(#) must be entered when system Auth code is entered.	OFF ON	OFF
Station VM Feature Usage	If this option is set to ON, Station VM feature can be used.	OFF ON	OFF
WAN Fault Alarm Disable	The WAN Fault Alarm to Attendant can be disabled. It will be useful to the site which does not use WAN port.	OFF ON	OFF
Emergency Call ATD Notify	Provide notification to attendant when user dial emergency number	DISABLE ENABLE	ENABLE
VM Password check	When ON, check password when a user access to the VSF messages.	No password Password only Station number & password	Station number & password
Override First CO Group	When a user dials 'Access CO in First CO Group' code, the system can search all CO Groups for the first available CO line.	OFF ON	ON

3.2.3.2 Call Feature Timer

A number of timers can be assigned to control and affect many features and functions.

Table 3.2.3.2 Call Feature Timers

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Attendant Recall Timer	Determines the amount of time the attendant receives recall after which the system will disconnect the call.	00~60 (minutes)	01
Call Park Recall Timer	Determines the amount of time before a parked call will recall the station that parked the call.	000~600 (seconds)	120
Camp-on Recall Timer	When a call is transferred using Camp-On, this entry determines the amount of time before the station that transferred the call receives recall.	000~600 (seconds)	030
Exclusive Hold Recall Timer	Determines the amount of time before a call placed on exclusive hold will recall the station.	000~600 (seconds)	060
I-Hold Recall Timer	Determines the amount of time a call that is recalling the station will recall before also recalling at the attendant.	000~600 (seconds)	030
System Hold Recall Timer	Determines the amount of time before a call placed on system hold will recall the station.	000~600 (seconds)	060
Transfer Recall Timer	Determines the amount of time a transferred call will ring at the receiving station before recalling the station that transferred the call.	000~600 (seconds)	060

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
ACNR Delay Timer	If the ACNR Pause Timer expires and no CO Line is available for ACNR recall, the delay timer sets the delay before ACNR again attempts to access a CO line. The ACNR retry counter is not affected by this action.	000~300 (seconds)	030
ACNR Pause Timer	This timer establishes the time between ACNR recall attempts (CIS=5-300).	030~300 (seconds)	030
ACNR Retry Counter	This counter sets the number of recall attempts for ACNR before ACNR is abandoned (CIS=1-9).	1~13	03
ACNR Tone Detect Timer	If call progress tones are not available for ACNR, the system will wait this duration after dialing before considering the called party “busy/no answer”.	001~300 (seconds)	30
Automatic CO Release Timer	If a user accesses a CO path and does not take any action, the system will automatically release the CO path when this timer expires.	000~300 (seconds)	030
CCR Inter-digit Timer	Inter-digit timer used with Customer Call Routing function.	000~300 (100 msec)	030
CO Dial Delay Timer	Delay for through connection to prevent illegal dialing when CO/PBX has slow response.	00~99 (100 msec)	05
CO Release Guard Timer	When a CO Line is returned to idle, the system will deny access for this time to assure the PSTN returns the CO circuitry to idle.	010~150 (100 msec)	020
CO Ring Off Timer	This timer sets the maximum ‘Off’ duration of the incoming ring cycle for the Ring Detect circuitry of the system to detect an abandoned call.	001~150 (100 msec)	060
CO Ring ON Timer	This timer sets the ‘On’ time of the incoming ring cycle for the Ring Detect circuitry of the system to recognize an incoming call.	1~9 (100 msec.)	2
Elapsed Call Timer	Users can receive a periodic tone indicating the length of an outgoing call. This timer sets the time before and between the tones.	060~900 (seconds)	180
Call Forward No Answer Timer	When a user activates No-Answer Forward, calls will ring for this duration before being forward. The Station No Answer timer will take precedence.	000~600 (seconds)	015
DID/DISA No Answer Timer	A DID/DISA call to a busy station will forward to the assigned DID/DISA Destination should this timer expire.	000~255 (seconds)	20
VSF User Maximum Record Timer	This timer sets the maximum duration allowed for the User Greeting in the system’s basic Voice Mail.	000~999 (seconds)	60
VSF Valid User Message Timer	This timer sets the minimum duration allowed for a voice mail message in the system’s basic Voice Mail. Messages shorter than this duration are not stored.	0~9 (seconds)	1
ICM Dial Tone Timer	If a user goes off-hook on the Intercom and takes no action for this timer, the user will receive error tone.	01~20 (seconds)	10
Inter Digit Timer	This timer sets the maximum time allowed between each user-dialed digit. At expiration, the user will receive error-tone.	01~20 (seconds)	03
MSG Wait Reminder Tone Timer	An LIP Phone user will receive periodic reminder tones of a message waiting at intervals of this timer.	00~60 (minutes)	00
Paging Timeout Timer	Determines the maximum duration of a page after which the caller and Page Zone are released.	000~255 (seconds)	15

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Pause Timer	A Timed pause of this duration is used in speed dial and during other automatically dialed digits sent to the PSTN.	1~9 (seconds)	3
SLT Hook Switch Bounce Timer	This timer determines the duration the system considers an actual state change in the hook-switch and not a contact bounce.	01~25 (100 msec.)	01
SLT Maximum Hook Switch Flash Timer	This timer sets the maximum time an SLT user can depress the hook-switch for a Flash signal.	01~25 (100 msec.)	02
SLT Minimum Hook Flash Timer	This time sets the minimum time an SLT user must depress the hook-switch for a Flash signal.	000~250 (10 msec.)	008
Station Auto Release Timer	For an internal call, the system will return a station to idle if the call remains unanswered for this duration.	000~300 (seconds)	060
Prime Line Delay Timer	This timer sets the delay (no action duration) for delayed Prime Line operation.	01~20 (seconds)	05
Enblock Inter Digit Timer	When an ISDN Line is assigned to send digits Enblock, the system will send digits if the user dials "#" or this Enblock inter-digit timer expires.	01~20 (seconds)	3
DTMF Duration Timer	This timer establishes the duration of DTMF tones sent on a CO line.	04~99 (10 msec.)	10
Flex DID Timer	The system will receive DID digits for this timer. After the timer expires, the system will use the last 2 to 4 digits received as the DID digits.	01~99 (100 msec.)	30
CO Flash Timer	This entry sets the duration of a Flash on the CO Line.	000~300 (10msec)	50
SIP Station Registration Timer	Shorter time will make more traffic. More than 10 minute recommended. 0 means registration timer is disabled.	0, 30~3600 (seconds)	3600
Record Warning Repeat Timer	If record warning tone is set and this timer is set greater than 1, it works periodically when it's recorded.	00~999 (seconds)	0
DISA Delay Timer	It is only used for Russia. When DISA incoming call, system is answer for DISA call, after DISA Delay Timer, DISA announcement is played.	0-9	0
DISA Answer Timer	It is only use for Russia When DISA incoming call, System is connected after DISA Answer Timer	0-9	0
Unsupervised Conference Timer	This timer determines the duration of an "Unsupervised conference" before the station is recalled or the conference is dropped.	00~99 (minutes)	10
Open Loop	This entry sets the duration of open loop that will be recognized as a "Disconnect Signal".	00~20 (100 msec)	0
CO CUT OFF TIMER	Allowed duration of CO call.	00-99 (minutes) 00 means disable	00

3.2.3.3 Day/Night/Timed Schedule

The system can be programmed to automatically select the Ring and COS mode based on time of day and day of week. Three Ring & COS modes are available: Day, Timed and Night. The Ring assignments are as defined in the Call Routing by Line and Ring Assignments Table. COS assignments are made according to Authorization Code & COS.

The start times for Day, Night and start and end times for Timed modes are programmed for each day of the week. After Timed end time the mode goes to Day if time is less than Night mode. The Attendant can override the Automatic selection and select the desired Mode (Day, Night, or Timed) manually.

3.2.3.4 Toll Exception Table

There are three Toll Restriction Tables arranged in pairs. Each pair consists of an Allow Table and a Deny Table. Each Toll Exception Table permits entry of 50 Allow codes and 50 Deny codes. Each code can contain up to 20 digits including digits 0-9, "#" as a wild card (any digit) and "*" as the end of entry mark.

Based on Table entries, stations or DISA users are allowed or denied dialing specified numbers. The following rules apply to establish restrictions based on the Table entries:

- If entries are only made in the Allow Table, only those numbers entered can be dialed, all other dialed numbers will be restricted.
- If entries are only made in the Deny Table, only those numbers entered will be restricted and all other numbers can be dialed.
- When there are entries in both the Allow and Deny Table pair, if the number is in the Deny Table, the number will be restricted otherwise the number can be dialed without restriction.

3.2.3.5 Emergency Dialing

The Emergency Code Table is used to identify emergency numbers, which when dialed, will override all COS dialing restrictions. An Emergency Code number may be up to fifteen (15) digits in length.

3.2.3.6 SMDR Attributes

Station Message Detail Recording (SMDR), which is output over TCP channel, contains details on both incoming and outgoing calls. Various SMDR attributes can be assigned including; output records for all calls or LD only, call cost per pulse when using call metering, etc. The following Table describes SMDR Attributes:

Table 3.2.3.6 SMDR Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Call Metering	Selects the call-metering signal from the PSTN to indicate call cost	NONE NPR	NONE
Advice of Charge (Only when BRIU or BRIU2 installed)	When assigned, the system will analyze the Advice of Charge information in the Facility Message according to the ETSI specifications with appropriate regional protocol support.	No Service Italy & Spain Finland Australia Belgium ETSI STD	No Service

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Save Enable	The system can output all outgoing call records (ON), or to allow for PSTN call set-up times, only records for calls that exceed the SMDR Timer (OFF, refer to Start Timer Attribute).	ON OFF	OFF
Print Enable	The system can output SMDR records automatically as they occur (real-time) or only when requested. When this attribute is ON, SMDR output is automatic at call completion.	ON OFF	ON
Record Type	The system can record all outgoing calls or only long distance calls. Long distance calls are identified by the LD digit count and LD codes assigned in “Long Distance Call Digit Counter” and “Long Distance Code” below.	Long Distance All call	All Call
Long Distance Call Digit Counter	Dialed numbers, which exceed the assigned LD digit count are considered long distance calls for SMDR and COS purposes.	07-15	08
Print Incoming Call	The system can output records for Incoming calls as well as outgoing calls. If enabled, incoming as well as outgoing calls are recorded.	ON OFF	OFF
Print Lost Call	When incoming call records are enabled, the system can also provide records for unanswered incoming (abandoned) calls.	ON OFF	ON
Records In Detail	The system can output detailed call records (ON) or summary call information (total number of calls, cost and cost for each station).	ON OFF	ON
Hidden Dialed Digit	For security purposes, digits dialed for an outgoing call can be hidden and replaced with “*”. This field defines the number of digits to hide the trailing digits	0~9	0
SMDR Currency Unit	The unit of currency used for call cost can be identified with 3-characters for easy reference.	Max 3 Characters	
SMDR Cost Per Metering Pulse	When call metering is provided by the PSTN, the cost per metering pulse can be assigned.	6-digits	000000
SMDR Decimal Position	This value determines the position of the decimal in the Cost per Pulse entry above, starting from the right most digit.	0~5	2
Record Start Guaranteed Time	To allow for call set-up times through the PSTN, a “Valid call timer” can be set.	000~250 (msec)	000
Long Distance Code	For SMDR and COS purposes, five (5) Long Distance codes of up to two (2) digits each can be assigned. If dialed as the 1st digits, the call is considered an LD call.	5 two digit LD codes, use * as wild card (any digit)	

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
SMDR CLI or Ring Service I	For incoming calls, the system will send the defined data item for “Field I”. The data item may be CLI, CPN or Ring Service Time. Note the User dialed number is always provided for an outgoing call.	RING CLI CPN	RING
SMDR Ring/CLI/CPN Service II	For incoming calls, the system will send the defined data item for “Field II”. The data item may be CLI, CPN or Ring Service Time.	RING CLI CPN NONE	NONE
Print MSN	Print MSN number Information in SMDR Record	ON OFF	OFF
Print Order No	Print record number as part of SMDR output, will reset to 1 when SMDR capacity is reached or SMDR Mail Auto Delete Set above is enabled.	ON OFF	OFF
LCD Display Priority	Caller ID can be overwritten on Duration or Cost LCD column or not.	Caller ID Duration / Cost	Caller ID
SMDR ICM Save	When enabled, intercom call data is stored as part of the SMDR data.	ON OFF	OFF
SMDR ICM Print	When enabled, intercom call data is printed as part of the On-line SMDR.	ON OFF	OFF
SMDR Disconnect Cause	When enabled, the disconnect cause is stored in Off-line SMDR data and printed as parted of the On-line SMDR.	ON OFF	OFF
SMTP Mail Server Address	SMTP Mail server address to send e-mail SMDR reports.	100 Characters	
SMTP Mail Server ID	This field defines the user’s ID for SMTP Mail server. If user’s ID and password is assigned, SMTP Mail server will check the validation of user ID and password.	Max 40 Characters.	
SMTP Mail Server Password	This field defines the user’s password for SMTP Mail server. If user’s ID and password is assigned, SMTP Mail server will check the validation of user ID and password.	Max 20 Characters.	
SMDR Sender Mail Address	Sender e-mail address to send the SMDR e-mail reports.	40 Characters	-
SMDR Receiver Mail Address	Receiver e-mail address to receive the SMDR e-mail reports.	40 Characters	
SMDR Mail Send Weekly Set	Sets day of week to send SMDR data weekly (0 for no weekly data, 1-7 for Monday through Sunday).	N/A day	N/A
SMDR Mail Send Daily Set	Sets time-of-day for SMDR data sent on a daily basis (00 for no daily records, 01-23 for hour of the day).	00~23	00
SMDR Mail Auto Send Set	If the SMDR buffer is full, the system will automatically send a notification e-mail.	ON OFF	OFF
SMDR Mail Auto Delete Set	Delete SMDR records after sending e-mail.	ON OFF	OFF

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
SMTP Security	Choose SMTP Security	No Security SSL TLS	No Security
SMTP Port	Choose SMTP Port	1 ~ 65535	25

3.2.3.7 International Call

International call can be restricted by prefix matching. International call prefix, all international call restriction and CO-CO international call restriction can be programmed in this admin. The following Table describes International Call:

Table 3.2.3.7 International Call

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
International Prefix	International prefix is used for classification of international call.	Max 2 digits	00
All International call	All outgoing international call can be restricted by this attribute.	Enable Disable	Enable
CO-CO International call	Transfer or forward to International call can be restricted by this attribute.	Enable Disable	Disable

3.2.3.8 Alarm Attribute

Alarm can be set by this attribute.

Table 3.2.3.8 Alarm Attribute

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Alarm Enable	Alarm Enable	ON/OFF	ON
Alarm Contact Type		Close/Open	Close
Alarm Mode	You can choice Alarm or Door Bell.	Alarm/Door Bell	Alarm
Alarm Signal Mode	How many times repeat?	Repeat/Once	Repeat

3.2.4 Station Group Data

Stations can be grouped so that incoming calls will search (hunt) for an idle station in the group. The iPECS SBG-1000 System supports 7 different hunt processes: Circular, Terminal, Ring, Pick-Up, VSF-Voice Mail, IPCR, Networking Voice Mail.

The Station Group capacities for the iPECS SBG-1000 systems are shown in the Table:

Table 3.2.4 Station Group Data

ITEM	CAPACITY
Number of Groups	12
Stations in a Group	24

Certain types of groups can incorporate announcements, which are given to the calling party. The system's VSF can store up to 70 announcements for use with Station Groups.

> **NOTE:** *A station can belong to multiple groups if the groups are of the same type. When a Station Group is assigned to a group type, the group attributes revert to the default values.*

3.2.4.1 Station Group Assignment

Under Station Group Assignments the type, members and Pick-Up attributes are assigned.

Table 3.2.4.1 Station Group Assignment

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Group Type	Defines the type of station group.	N/A Circular Terminal Ring Pick-Up VSF-VM IPCR NET-VM	N/A
Pick-up Attribute	Stations can pick-up group calls ringing at other stations in the group. This does not apply to VSF groups.	OFF ON	OFF
Member	Assign stations as members of a station group.		-

3.2.4.2 Station Group Attributes

Each type of group has a different set of available attributes relating to announcements, timers, overflow, etc. The following Tables provide descriptions for the attributes and data entries required.

Table 3.2.4.2-1 Terminal & Circular Group Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
VSF Announce 1 Timer	If all stations in the group are busy when a call is received, the call may continue to wait (queue) for an available station. If the queue period exceeds the VSF Announce 1 timer, the call is sent to a VSF announcement. If the timer is set to 000, the call will receive the first announcement, in full, prior to the hunt process (guaranteed announcement).	000~999 (seconds)	015
Guar-Annc(Timer 0) Wait If Busy	When a call assigned to receive a guaranteed announcement arrives and all channels are busy, the call may wait with Ringback until a channel is available (ON) or bypass the announcement (OFF).	OFF ON	ON
VSF Announce 2 Timer	After the 1st announcement, the 2nd ANNC TMR is activated. At expiration, if the call remains queued to the group, the call is sent to the assigned 2nd VSF announcement.	000~999 (seconds)	000
VSF Announce 1 Location	The Station Group can be assigned an announcement, which is played if the call remains queued beyond the VSF Announce 1 Timer duration. The announcement location is the VSF Announcement number.	00~70	00: none

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
VSF Announce 1 Auto Drop	If this attribute is selected, the call will drop after the 1st VSF announcement	Check box	
VSF Announce 2 Location	The Station Hunt Group can be assigned a 2nd announcement, which is played if the call remains queued beyond the VSF Announce 2 Timer duration. The announcement location is the VSF Announcement number.	00~70	00: none
VSF Announce 2 Auto Drop	If this attribute is selected, the call will drop after the 2nd VSF announcement	Check box	
VSF Announce 2 Repeat Timer	The 2nd announcement can be repeated to callers that remain in queue at intervals of the announcement 2 repeat timer. Note: VSF Announce 2 Repeat (below) must be “ON”.	000~999 (seconds)	000
VSF Announce 2 Repeat	After the 2nd announcement, if the call remains queued to the group, the 2nd VSF announcement can be repeated at the Announce 2 Repeat Timer interval, defined above.	ON OFF	OFF
Overflow Destination	A call to the group will continue to route through the group until answered or all group members have been tried. The call will remain at the last station or routed to the assigned overflow destination. If VSF Announcement is selected, Auto Drop can be checked.	STA or Hunt Number, VSF Announce, System SPD	
Overflow Timer	A call to the group will remain at the last station in the group or can be sent to the assigned Overflow Destination after expiration of the Overflow Timer.	000~600 (seconds)	180
Warp-Up Timer	After terminating any call, a Group member will be maintained in a busy state for the duration of the Wrap-Up timer.	000~999 (seconds)	002
No Answer Timer	Calls to a station in the group are directed to the station, if unavailable or unanswered in the No Answer Timer, the call can be routed based on the assigned hunt process.	00~99 (seconds)	15
Pilot Hunt	A circular/terminal hunt group can be set so that only calls to the pilot number (station group number) will hunt.	ON OFF	ON
REPT No Member	If a call is received and no members are on-duty, an ICM call will return re-order tone, while a CO call will be routed to the Attendant.	ON OFF	OFF
Music Source	A Music source can be assigned so that calls to the group will receive audio from the assigned source in place of ring-back tone.	Ring-Back Tone Record Play	Ring-Back Tone
Allow Forward Member	A member activating Call forward, may be placed in an unavailable state for hunt group calls (ON). When OFF, group calls are sent to the member as normal.	ON OFF	ON
VSF Wait Station	When a call overflows or routes to the VM group, a station number is used to identify the Mailbox for the group messages.	Station Number	
Mail Box Password	The password associated with a group Mailbox is defined here. The password is used in conjunction with the group as with a normal station.	Max 12 digits	
Forced Forward Destination	Calls to a hunt group may forward calls directly to a defined destination.	STA or Hunt grp. VSF Annc SysSpeed	

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Forced Forward Dest Usage	Calls to a hunt group may forward calls directly to a defined destination. Forced Forward must be enabled for the group.	OFF ON	OFF
Group Name	An group name can be designated	12 character	
Max Queued Call Counter	When the number of calls queued to the group match this parameter, new calls will receive error tone and be disconnected after the VSF Announcement 1, if assigned, is played.	00-99	99

Table 3.2.4.2-2 Ring Group Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
VSF Announce 1 Timer	If all stations in the group are busy when a call is received, the call may continue to wait (queue) for an available station. If the queue period exceeds the VSF Announce 1 Timer, the call is sent to a VSF announcement. If the timer is set to 000, the call will receive the first announcement, in full, prior to the hunt process (guaranteed announcement).	000~999 (seconds)	015
Guar-Annc(Timer 0) Wait If Busy	When a call assigned to receive a guaranteed announcement arrives and all channels are busy, the call may wait with Ringback until a channel is available (ON) or bypass the announcement (OFF).	OFF ON	ON
VSF Announce 2 Timer	After the 1st announcement, a 2nd announcement Timer is activated. At expiration, if the call remains queued to the group, the call is sent to the assigned 2nd VSF announcement.	000~999 (seconds)	000
VSF Announce 1 Location	Each Ring Group can be assigned an announcement, which is played if the call remains queued beyond the VSF Announce 1 Timer duration. The announcement location is a VSF Announcement number. An entry of 00 indicates no announcement.	00~20	00: none
VSF Announce 1 Auto Drop	If this attribute is selected, the call will drop after the 1st VSF announcement	Check box	
VSF Announce 2 Location	The Ring Group can be assigned a 2nd announcement, which is played if the call remains queued beyond the VSF Announce 2 Timer duration. The announcement location is a VSF Announcement number. An entry of 00 indicates no announcement.	00~-20	00: none
VSF announce 2 Auto Drop	If this attribute is selected, the call will drop after the 2nd VSF announcement	Check box	
VSF Announce 2 Repeat Timer	The 2nd announcement can be repeated to calls that remain in queue at intervals of the VSF Announce 2 Repeat Timer. Note: VSF Announce 2 Repeat below must be “ON”.	000~999 (seconds)	000
VSF Announce 2 Repeat	After the 2nd announcement, if the call remains queued to the group, the 2nd VSF announcement can be repeated at the VSF Announce 2 Repeat Timer interval, defined above.	ON OFF	OFF
Overflow Destination	A call to the group will continue to route through the group until answered or all group members have been tried. The call will remain at the last station or routes to the assigned Overflow Destination. If VSF Announce is assigned, Auto Drop is available.	Station or Group Number, VSF Announce, System SPD	

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Overflow Timer	A call to a group will remain at the last station in the group or route to the assigned Overflow Destination after expiration of the Overflow Timer.	000~600 (seconds)	180
Warp-Up Timer	After terminating any call, a Hunt Group member will be maintained in a busy state for the duration of the Wrap-Up Timer.	002~999 (seconds)	002
Music Source	A Music source can be assigned so that calls to the group will receive audio from the assigned source in place of ring-back tone.	Ring-Back Tone Record Play	Ring-Back Tone
Maximum Queued Call Counter	When the number of calls queued to the group match this parameter, new calls will receive an error tone and be disconnected after the VSF AA announcement is played (if assigned).	00-99	99
Allow Forward Member	When a member is forwarded to another station, if this option set OFF, the member receives a incoming hunt call.	OFF : no FWD ON : FWD	ON
Group Name	An group name can be designated	12 character	
VSF Wait Station	When an ring group call overflows or routes to the VM group, a station number is used to identify the Mailbox for the ring group messages.	Station	
Mail Box Password	The password associated with an ring group Mailbox is defined here. The password is used in conjunction with the ring group as with a normal station.	12 digits	
Forced Forward Destination	Calls to a hunt group may forward calls directly to a defined destination.	STA or Hunt grp. VSF Annc SysSpeed	
Forced Forward Dest Usage	Calls to a hunt group may forward calls directly to a defined destination. Forced Forward must be enabled for the group.	OFF ON	OFF

Table 3.2.4.2-3 PICK-UP GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Auto Pick Up	If a group member Station is ringing, another member of the Group can Pick-Up the ringing call from their station by simply going Off-hook.	ON OFF	OFF
All Ring	When a call is received to a member of the Pick-Up Group in the Tone Ring mode, all members will ring. Note: Auto Pickup must be set to ON.	ON OFF	OFF

Table 3.2.4.2-4 VSF GROUP ATTRIBUTES

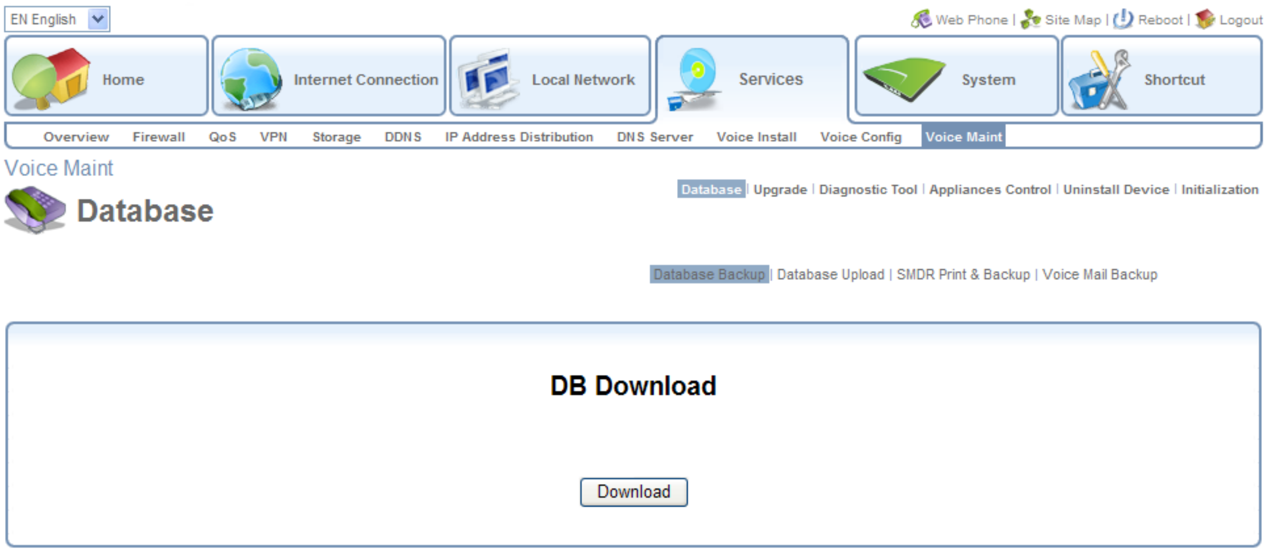
ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Time Set (day) *For future use only*	When voice messages are stored in the VSF, the system will maintain (store) the message for the maximum number of days set in this program (1 to 365 days). (Not used currently)	00-99 (day)	0
Time Out (sec): *For future use only*	This timer determines the inter-digit time for a VSF-AA or a VM session. If this timer expires while the VSF AA or VM is awaiting user input, the system will assume the remote party has disconnected and will return the channel to idle. (Not used currently)	00-99 (seconds)	15
Group Name	An group name can be designated	12 character	

3.2.4.3 IPCR Agent

This table used for matching agent ID to station number. If it's done, the station with agent ID is automatically recorded about internal, external call.

3.3 VOICE MAINTENANCE

Selecting Maintenance from the Main menu will display the Maintenance page (shown).



The Maintenance Main menu item permits download of all or portions of the system's database, and downloading and viewing of SMDR data. The user also can set trace direction or target and system logs. Other functions include uninstall registered station, delete voice mail box for each station, and initialize call attributes.