iPECS

iPECS UCP
(Unified Communications Platform)

Administration & Programming Manual

Please read this manual carefully before operating system.

Retain it for future reference.

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Revision History

SW version	ISSUE	DATE	DESCRIPTION OF CHANGES
	1.0	March, 2014	First Release
	1.1	June, 2014	Updated the content according to upgrading SW
1.2.x	1.2	June, 2015	We updated the following features: - On-line user guide - Terminal attributes (Small popup use, large popup timer, SLT open loop time) - Web Access Authorization - NTP attributes - Digit Conversion Table - System attributes - U-LOOP, PBX code insertion for emergency call - Disconnect with Inband information - Temp License expiration notify - Trace Log via Web - Add System date and time in Installation wizard - Etc.
2.0.x	1.3	Jan. 2016	- Added Speed code number plan in PGM100. - IPCR announce only for incoming call. - Updated Strong password in PGM 160/161. - Updated 'Do not overwrite station name in PGM 211. - Added Attendant ring mode in PGM 257. - Added the more items for Alarm in PGM 163. - Added Flexible button default table (PGM 239) - Added Preset flexible button default (PGM 240) - Added System DST mode in PGM 439. - Added LDAP server setting in PGM 160/161. - Added Arabic language. - Updated Message wait indication LED in PGM 112. - Updated Company directory in Station Name Display. - Added Station User Greeting in Station Data. - Added the search box in Maintenance. - Added MOH Management in Maintenance. - Updated DB management by adding Comment field. - Updated Install wizard - DTMF repeat tone does not have nothing to do with PSTN in PGM160-15 - Added Call wait signal and duration in PGM 113. - Added some admin (Alternate/Secondary signal port, Local route ID, Remote route ID, Sending Name option) in PGM 324. - Added SLT Pulse and SLT pulse-MW type in PGM 110. - Added Lift Handset for Page & Privacy in PGM 111. - Added Short Modem in PGM 112. - Added NFC Authorization Code use in PGM 112. - Added rel 180 after 183 and "Add 'user=phone' param" in PGM 133.

			- Added Short Modem Timer in PGM 182~182, 186 - Added Alarm mode to send email to the address in PGM 163. - Added the special character to DB management. - Added Ring detection register setting in Analog parameter. - Added LCD Dimming for LIP-9000 Series & LDP-9240D in PGM 112. - 'Go to assignment' function added to Personal group overview and can check the master & member list at a glance. - Added Station ICR scenario (1201) to Station Data. - Changed the name from UCS Standard to UCS data. - Added MOH management in Maintenance. - Added Keyset admin password to Install Wizard.

The contents of this document are subject to revision without notice, due to continued progress in methodology, design, and manufacturing. Ericsson-LG Enterprise shall have no liability for any error or damage of any kind resulting from the use of this document.

Table of Contents

1. INTRODUCTION

1.1 Manual Application

This document provides detailed information covering the configuration of the UCP database and maintenance of the UCP. The manual also details the power-up and initialization routines and the Station Web Portal.

The manual is written for the experienced installer with knowledge of telephony terms, and functions of small and mid-sized business telecommunications systems.

1.2 Manual Organization

This manual is organized in ten (10) major sections including:

- **Section 1 Introduction**: This section introduces the content and organization of the manual.

- **Section 2 System & Admin Information**: In this section general information on System capacities, power-up routines and the system initialization process are detailed. In addition, this section discusses the process for registration of IP and softphones with the UCP system.

- **Section 3 Station Administration**: This section provides details on configuring the system employing a station allowed administrative access. Step by step procedures are given along with brief but concise descriptions of the various configuration parameters and available settings.

- **Section 4 Web Services Overview**: Provides general information on access to the Web services, the structure Web pages and navigation features of the Web services.

- **Section 5 Web Setup Wizard**: This section discusses access to and use of the iPECS UCP setup wizard.

- **Section 6 Web Administration:** Similar to the Station Administration section, the Web Administration section provides procedures and descriptions for the configuration parameters and settings available using a Web browser.

- **Section 7 Maintenance**: The Maintenance section provides details on managing the UCP including database upload and download, software upgrade, and user access management using the Web browser interface.

- **Section 8 Station Program**: This section discusses the configuration of the features and functions available in the portal.

- **Section 9 On-line web user guide**: This section explains the frequent use of features to a user. We didn't describe this section because you can easily get the information on the web by clicking [User's guide] of login page.

- **Appendix A**: The Station and Attendant Station User Program codes are listed with the

associated function. These fixed codes are available at the iPECS IP or LDP phones to configure basic functions such as ring tones, activate features and assign features and functions to Flex buttons.

• **Appendix B**: One of the nine Flexible Numbering Plans, which are listed in this Appendix, is selected as the system's Flexible Numbering Plan. Each of these Numbering Plans includes all feature and resource access codes, and any individual code can be changed.

2. SYSTEM & ADMIN INFORMATION

2.1 System capacities

This manual provides detailed information on the database management of the iPECS UCP Series systems. The iPECS UCP Series is available with several versions of the call server configuration. Several other variations exist between systems based on the model employed as shown in Table 2.1-1 System Capacity Chart.

Table 2.1-1 System Capacities

ITEM	UCP100	UCP600	UCP2400	Remark
Main Cabinet	10 Slots			1 slot used by the PSU
System channels, basic	50	100	600	
maximum	199	600	2400	w/License
Stations	199	600	2400	Total stations and Lines cannot exceed the available System channels
CO/IP Lines (external network channels incl. VoIP)	199	600	998	
UCS Standard Clients				
Registrations	100	200	400	
Simultaneous	100	200	400	
UCS Premium Clients				
Registrations	199	600	2400	
Server Redundancy	No	Yes	Yes	Supports local and remote redundancy
Cabinet Power Redundancy	Yes			
Integrated Telephony ports[1]				Two FXS ports are equipped as standard in the UCP100 module; an optional CO/BRI line unit may be equipped or installed.
standard	2 FXS (SLT)	None	None	
optional	4 CO, or 2 BRI or 4 BRI Lines			
WTIM4/8 modules (Max.)	132			
Max DECT phones	100	255	255	
VoIP Switch channels, [2&3]				
Built-in basic	8	8	0	
Built-in maximum	16	24	0	w/License
System Maximum	100	600	998	w/VOIM and VCIM
VoIP DSP channels,				VoIP DSP channels can be assigned to the MCIU, 2 VoIP = 4 Conf channels. For UCP100 see Note 1.
Built-in max	6	6	0	
Multi-party Conference Unit (MCIU) channels	6/10/14/18	6/18	0	
Maximum MCIMs	30			
Max SIP channels	100	600	1200	
VSF[4]	8-Ch./4 Hrs.	8-Ch./6Hrs.	NA	
	14 Hours	16 Hours	NA	w/License

Table 2.1-1 System Capacities

ITEM	UCP100	UCP600	UCP2400	Remark
UVM capacity, basic	8 Channels at 50 hours			
maximum	16 Channels at 200 hours			w/License
UVM per system	30			
Attendants	50			
Serial Port (RS-232C)	1			
USB Host port	1			
Alarm/Doorbell input	1	2		
External Control Relays	1	4		
Music Source Inputs (Ext)	1	2		
Power Fail Circuit	1	4		
External Page zones	1	2		
Internal Page Zones	100			
System Speed Dial	12,000			
System Speed Dial Zones (Groups)	50			
Station Speed Dial, per Station	100			
Total Station Speed Dial	24,000			
Call park	200			
Last Number Redial	10 (23 digits)			
Save Number Redial	1 (23 digits)			
Standard DSS Consoles/Station	9			
Serial DSS - System	500			
Serial DSS - Station (LIP-8000)	4			
Serial DSS – Station (LIP-9000)	1			
SMDR buffer	30,000			
CO Line Groups	200			
Station & Hunt Groups	200			
Station & Hunt Group Members	200			
Pickup Groups	200			
Pickup Group Member	2,4000			
Personal Groups	1200			
Conference Group - System	160			
Conference Group - Station	100			
Executive/Secretary pairs	100			
Authorization Codes	5200 (Station:2400/System:2800)			
Transparent Networking Table	100			
CLI Msg Wait (Missed calls)	4,000			
Redundancy	N/A	Yes		
Flex DID Table	10,000			
MSN table	2,400			
DID Digits Analysis	4			

Table 2.1-1 System Capacities

ITEM	UCP100	UCP600	UCP2400	Remark
Tenancy (ICM) Groups	100			
ICLID table	500			

NOTE

1. The iPECS UCP-100 is equipped with two (2) FXS ports, and one of several built-in CO Line units may be factory or field installed.

    4 analog CO Lines (UCP-COIU4), uses two (2) VoIP DSP channels
    2 BRI Lines, 2B+D each, (UCP-BRIU2) , uses two (2) VoIP DSP channels
    4 BRI Lines, each 2B+D, (UCP-BRIU4) , uses four (4) VoIP DSP channels

    Note the built-in PSTN interfaces require dedicated DSP resources. These DSP resources reduce the maximum VoIP DSP resources available shown in the above chart.

2. The eight (8) built-in VoIP Switching channels implement agent and packet relay functions. Remote device and network interfaces send RTP traffic to the VoIP channel, which forwards traffic to the appropriate local iPECS device. The VoIP Switching channel also forwards multi-cast packets to the remote end-points and local non-iPECS devices. Only the g.711 codec is allowed unless there is an available VoIP DSP channel.

3. The number of VoIP Switching channels can be increased to the maximum with license installation.

4. Approximately 35 minutes (16 Mbytes) of the VSF and UVM memories are used to provide basic system prompts, the remaining memory can be used for announcements and voice message storage. Note the built-in VSF supports the g.711 Codec only; the UVM supports g.711, g.729, g.723, and g.722 Codecs.

2.2 Initialization

When power is applied to the UCP or the UCP Reset button is pressed, the system will initiate the "Power-up" routine. During the Power-Up routine, the system will check the Initialization switch (pole 4th of UCP Mode Dip Switch). If the switch is in the OFF position, the system will perform a simple Power-Up routine; clear all scratch-pad memory, load run-time programs, establish communications with each registered gateway Module and iPECS terminal, send RESTART commands and load appropriate settings to the Modules and terminals. If a Module or terminal does not respond after several attempts, the system places the device in an out-of-service mode but maintains the database settings. Once the Power-up routine is complete, the system will conduct normal operations.

If the Initialization switch is in the ON position, in place of the Power-Up routine, the system will perform the full Initialization procedure. The initialization procedure will set the system database to default values. Further, during the full initialization procedure, the system will establish

communications with each gateway Module and iPECS terminal for registration. This communication will use the default device IP address and the UCP MAC address for system identification. The system will maintain IP addresses and Sequence Numbers for previously registered gateway Modules and iPECS terminals. These values are employed for subsequent communication and logical assignments of numbering plans, respectively. In addition, the system sends commands to modify all settings to the default values, including IP addresses but maintains the existing Sequence Numbers.

After successful initialization, should a device not respond to several attempts by the system to communicate, the system places the device in an out-of-service mode but maintains the database. Once initialization is complete, set the initialization switch to the OFF position to protect the database. The system must be restarted to complete the initialization.

2.3 Registration

2.3.1 Normal Registration Process

Module & Terminal

When power is applied and an Ethernet link is established, an unregistered device will attempt to discover and register with a local (on the same LAN) iPECS system. The Module or terminal will send a registration request to the assigned iPECS system (UCP) IP address. If no response is received, the device will generate a Multi-cast discovery request for registration.

Remote iPECS Phone & Remote Services Module

A remote device, iPECS Phone or gateway Module, registers with the system using the MAC address of the device. The MAC address must be assigned in the system database and the IP address of the system must be assigned in the remote device. Using this address, the remote device will attempt to register with the assigned iPECS system. When the system receives the registration request, the MAC address is compared with the database to authenticate the remote device. With a matching MAC, the system will accept the registration request and provide the remote device with the appropriate settings. Note that the position of the UCP Registration switch does not affect remote registration.

iPECS Softphone or SIP Phone

iPECS softphones (iPECS Communicator and UCS Client) and SIP phones register with the system employing the User Id and Password. When the system receives the registration request, the Id and password are compared to the Station Login parameters. If a match is found in the Station Login Table, the system registers the device and assigns the phone the Station Number requested (Desired number), if available. As with remote registration, the position of the UCP Registration switch will not affect Softphone or SIP Phone registration.

iPECS system

When power is applied, an Ethernet link is established, and the Registration switch (UCP DIP Switch position 3) is in the ON position, the UCP will send a Multi-cast request to unregistered gateway Modules and iPECS terminals for registration.

When the system receives a valid registration or discovery request, and the Registration switch (UCP DIP Switch position 3) is in the ON position, the system will respond to the gateway Module or terminal with a Registration command including the system IP and MAC address. During the registration process, the Module or terminal will receive data from the system including a Sequence Number, IP address, RTP characteristics, etc., as well as default settings appropriate to the type of Module or terminal. Once registered, the Module or terminal will maintain the system IP and MAC address in non-volatile memory and will not attempt further registrations.

If the Registration switch is in the OFF position, the system will not respond to normal registration requests from a local device.

Sequence Number

Sequence numbers are allocated to the different device types and are assigned sequentially to devices of the same type as shown in the Table 2.4.1 below.

Table 2.4.1 Sequence Number Allocation

Device Type	Start Sequence Number	End Sequence Number
Station	1	2400
CO line	2401	3000
VSF	3001	3100
MISC	3101	3200
MCIM	3201	3230
UCS server	3231	3246
IPCR and 3rd party server	3247	3256
WTIM	3257	3388

2.3.2 Replacement Module Registration

Under certain situations, it is necessary to force the registration of gateway modules and terminals specifically when an UCP, gateway Module or iPECS Terminal is replaced. When replacing a UCP module, gateway Modules and iPECS terminals must be forced to register with the new system. With Module or terminal replacement, the system must recognize the "replacement" status to transfer the existing database values.

When replacing a UCP, the local Web interface is used to access the system. The user may update the system database using the database downloaded from the previous UCP memory.

Using the Terminal mode Command Line interface ("maint > reset ip"), the user provides the new UCP with the IP address of the previous UCP, and issues the Register command. The new system will then send a Uni-cast Register command to each gateway Module and iPECS terminal registered to the previous system. This Register command will include the previous system IP

address. These commands are repeated several times only. As communication is established, the new UCP will update the settings of the gateway Modules and iPECS terminals appropriately. When the gateway Modules and terminals respond, they are registered to the new system.

When replacing a gateway Module, use PGM 103 "Device IP Plan" in Web admin to change the service mode to "Out of Service", change the MAC information for the new module, change the service mode to in-service and install the new gateway Module.

When replacing an iPECS terminal, using PGM 103 "Device IP Plan" in Web admin change the service mode to "Out of Service" change the MAC information for the new iPECS terminal, change the service state to in-service, and install the new terminal.

2.4 General Admin and Menu Structure

The iPECS UCP can be configured to meet each customer's individual needs. System configuration may be accomplished by entering the "Program mode" at an assigned Admin Station or by pointing a Web Browser at the IP address of the, UCP. Section 3 provides a description for data entry using the Admin Station. Section 4 discusses configuration employing the Web browser. Note that some parameters are available through Web Admin and not the Keyset Admin.

Configurable items are organized as "Data" groups with a common affect, i.e. station, system, numbering plan, etc. Items are further grouped into "Programs" forming a multi-layered menu structure. Generally, Program groups are assigned a three-digit "Program" (PGM) code used to access the group from an Admin Keyset. The top-level Data groups include:

- System ID & Numbering Plans
- Station Data
- Board Based Data
- CO Line Data
- System Data
- Station Group Data
- ISDN Line Data
- SIP Data
- Tables Data
- Networking Data
- H.323 Routing Table
- T-NET Data
- Zone Data
- Device Login
- UCS Standard
- DECT Data
- Redundancy Data (for UCP600 & UCP2400)
- Initialization

3. STATION ADMIN PROGRAMMING

3.1 General

Using Station Admin, the Attendant Station or Admin enabled station can be used to configure the System Id & Numbering Plan data for the system.   All other programming of the system and maintenance is accomplished using the Web services as covered in section 4.

3.1.1 LCD & Button Functions

While in the Program mode, the Liquid Crystal Display (LCD) and Flex button LEDs of an Admin Station are used to guide and indicate status of the feature. The dial-pad is most often used to enter data after selecting an item using the Flex buttons. In some cases, pressing a Flex button will toggle the entry with the Flex button LED indicating the status (ON/OFF).

For Programs with multiple Flex button selections, the volume controls (**[VOL UP]** and **[VOL DOWN]** buttons) may be used to select the next or previous item. The **[SPEED]** button is generally employed as a delete button to erase existing entries. However, where noted, it may be used to confirm a range input. Pressing the **[CONF]** button, returns to the first step of the data entry procedure for the Program without storing unsaved entries.

The **[SAVE]** button is used to store data after entry. If there are no conflicts in the entered data, confirmation tone is received and the data stored. If a conflict exists, error tone is provided and newly entered data are not saved. Generally, corrected data may be entered and stored without restarting the entry procedure from the 1st step.

3.1.2 Alphanumeric Data Entries

In some cases, an alphanumeric entry is required. Two (2) dial-pad digits represent each character of an alphanumeric entry, as shown in Table 3.1.2–1 below. Use the Table to determine the two digits that must be entered from the dial-pad for each character.

Table 3.1.2-1 ALPHANUMERIC DIAL-PAD ENTRIES

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

3.2 Data Entry Mode

Using Station Admin, data entry is accomplished from an Admin Station or station assigned for data entry (Station Data Common Attributes PGM 111 in Web Admin). After initialization and registration, all iPECS IP or LDP Phones may access the system database. In addition, as default, there is no Station Admin password defined. To enter the Program mode from the Admin Station follow the procedure below. In the left column of the chart are the LCD displays and in the right column are step-by-step instructions to modify database items.

PROCEDURE:	
STATION 100 (T) 04 SEP 01 02:49 PM	1. Press the [PGM] button. Dial '*' and '#'.
ENTER ADMIN PASSWORD	2. Enter the Admin password. Confirmation tone is received. As a default, there is no password and this step is skipped.
ADMIN PROGRAM START	
ENTER PGM NUMBER	3. To select a program, use the instructions in the following sections, starting with "Press the **[PGM]** button" and dial the specified Admin Program code.

3.3 Procedure for Data Entry

The following sections provide specific instructions for entering data from the Admin Station once in the Program mode. Each section provides descriptive information, step-by-step instructions and Tables for determining appropriate entries.

3.3.1 System ID — PGM 100

Under System ID, the country is identified using the international dial codes (Country Code). If the Country Code requires changing, the system must be initialized to restructure memory and create the country specific defaults, gain, frequencies and other system characteristics specific to the country and regional regulatory requirements.

To change the Country Code:

- Set the UCP Module Mode Dip Switch pole 4 to the ON position,

- Follow the procedure below to modify the Country Code,

- Press the reset button on the UCP Module, power the system OFF and ON, or use PGM 450 to initialize the system,

- After initialization, reset the switches as needed; Mode Switch pole 4 should be OFF.

A twenty-four (24) character SITE NAME and the local Area Code(s) are defined in this program. The SITE NAME is primarily useful for the installer/programmer as a reference to the customer.

In addition, under this program the system can be programmed to select one of eight (8) Flexible Number Plans, refer to Appendix B. Individual items from the selected Numbering Plan can be changed under Flexible Numbering Plan part A to D – PGM 106 to 109- in section 3.3.2.5.

PROCEDURE:	
ENTER PROGRAM NUMBER	1. Press the **[PGM]** button and dial 100.
SYSTEM ID PRESS FLEX KEY (1-6)	2. Select the desired Flex button (1~6), refer to Table 3.3.1-1. For COUNTRY CODE, refer to Table 3.3.1-2 for appropriate entries.
	3. Use the dial-pad to enter desired System Id data. For System Reset, button 6, press **[Save]** to reset the System Id to default.
	4. To store the System Id data press the **[Save]** button.

Table 3.3.1-1 SYSTEM ID (PGM 100)

BTN	DISPLAY	REMARK	RANGE	DEFAULT
1	COUNTRY CODE 1	Refer to Table 3.3.1-2 below. Note system must be re-initialized if changed.	4 digits	1
2	CUSTOMER SITE NAME ……………………………	Refer to Table 3.1.2-1 for alphanumeric dial-pad entries.	24 character	

Table 3.3.1-1 SYSTEM ID (PGM 100)

BTN	DISPLAY	REMARK	RANGE	DEFAULT
3	MY MULTI AREA CODE ENTER TABLE NO(00-40)	First, enter the Multi-Area Code table index, 00 ~ 40, then enter the area code. Multiple area codes may be entered.	6 digits	
4	NUMBERING PLAN (1-9) 2	Refer to Appendix B for details of Numbering Plan selection.	1-9	2
5	PREFIX USAGE (1:ON/0:OFF) : OFF	Enable/Disable 8-digit Station Numbering Plan. Assign the Prefix codes in the 8-Digit Numbering Table (PGM238).	0: OFF 1: ON	OFF
6	SYSTEM ID SYSTEM RESET	This selection returns the System Id to default.		

Table 3.3.1-2 COUNTRY CODES

COUNTRY	CODE	COUNTRY	CODE	COUNTRY	CODE
America (U.S.A)	1	Argentina	54	Australia	61
Bahrain	973	Bangladesh	880	Belgium	32
Bolivia	591	Brazil	55	Brunei	673
Burma	95	Cameroon	237	Chile	56
China (Taiwan)	886	CIS	7	Colombia	57
Costa Rica	506	Cyprus	357	Czech	42
Denmark	45	Ecuador	593	Egypt	20
El Salvador	503	Ethiopia	251	Fiji	679
Finland	358	France	33	Gabon	241
Germany	49	Ghana	233	Greece	30
Guam	671	Guatemala	502	Guyana	592
Haiti	509	Honduras	504	Hong Kong	852
India	91	Indonesia	62	Iran	98
Iraq	964	Ireland	353	Israel	972
Italy	39	Japan	81	Jordan	962
Kenya	254	Korea	82	Kuwait	965
Liberia	231	Libya	218	Malta	356
Luxembourg	352	Malaysia	60	Morocco	212
Mexico	52	Monaco	377	Nigeria	234
Netherlands	31	New Zealand	64	Pakistan	92
Norway	47	Oman	968	Paraguay	595
Panama	507	P.N.G	675	Portugal	351
Peru	51	Philippines	63	Senegal	221
Qatar	974	Saudi Arabia	966	Spain	34
Singapore	65	South Africa	27	Sweden	46
Sri Lanka	94	Swaziland	268	Tunisia	216
Switzerland	41	Thailand	66	United Kingdom	44
Turkey	90	U.A.E.	971	Y.A.R.	967
Uruguay	598	Venezuela	58		

3.3.2 NUMBERING PLANS DATA — PGM 102 to 109

3.3.2.1 System IP Address Plan — PGM 102

The System IP Address Plan sets several IP addresses including the UCP Module LAN port IP address (UCP IP Address) required for external VoIP calls, the IP address for the router, and the system's internal private IP address Plan. Note the UCP and Router addresses must be routable IP addresses for access to an external VoIP network, remote access by a gateway/board or terminal and remote Web access. Any VOIM must also have a routable IP address for access to/from an external VoIP network or remote user.

When Automatic IP Assignment, button 7, is enabled, the system will assign IP addresses to each local gateway module terminal using the assigned System IP address range. These addresses are used for communications with the system.

The system may be connected to a LAN that is segmented by two separate private IP address schemes. This segmenting technique is often used to separate voice and data devices. However, with this segmenting technique, the system would normally treat the segmented Terminals such as iPECS Soft Phones, as remote devices, using valuable WAN bandwidth to communicate with the system. Assigning the "Second Sys IP address" with a valid IP address from the second segment permits the system to communicate with the devices directly over the LAN.

iPECS can be installed behind a NAPT server, if the NAPT server provides fixed address translation and port forwarding to the system. In this case, the system will employ the "Firewall IP address", button 10, as the fixed public IP address for communication with remote devices. This address must be assigned as the "UCP" address in the remote device.

PROCEDURE:	
ENTER PROGRAM NUMBER	1.　Press the **[PGM]** button and dial 102.
SYSTEM IP ADDRESS PLAN Press FLEX KEY (01-17)	2.　Select the desired button 1~19, refer to Table 3.3.2.1-1.
	3.　Use the dial-pad to enter desired IP addresses. Use an "*" to enter a dot (".")
	4.　Press the **[Save]** button to store IP address entries.

Table 3.3.2.1-1 SYSTEM IP ADDRESS PLAN (PGM 102)

BTN	DISPLAY	DEFAULT	REMARK
1	UCP IP ADDRESS 10　.10　.10　.2	10.10.10.2	This is the UCP IP address. A Public IP Address is required for remote user and external VoIP network access. IPv4 format
2	UCP SUB NET MASK 255.255.255.0	255.255.255.0	

Table 3.3.2.1-1 SYSTEM IP ADDRESS PLAN (PGM 102)

BTN	DISPLAY	DEFAULT	REMARK
3	ROUTER IP ADDRESS 10 .10 .10 .1	10.10.10.1	IP Address of the default gateway for external network (WAN) access. Required for shared voice and data LAN, external VoIP and remote Web access.
4	SYSTEM START IP ADDRESS 10.10.10.10	10.10.10.10	Start of range for private IP addresses assigned by the system to Modules/Terminals.
5	SYSTEM END IP ADDRESS 10.10.254.254	10.10.254.254	End of range for private IP addresses assigned by the system to Modules/Terminals.
6	SYSTEM SUB NET MASK 255.255.0.0	255.255.0.0	
7	AUTOMATIC IP ASSIGN (1:ON/0:OFF): ON	ON	The system will automatically assign IP addresses to modules and terminals (ON) or, when OFF, IP addresses are assigned manually in PGM 103 Device IP Address Plan or via DHCP.
8	SECOND SYS IP ADDRESS 0 .0 .0 .0	0.0.0.0	When devices are located on a different private address on the same network, enter the UCP IP address for the second LAN.
9	SECOND SYS SUB NET MASK 255.255.0 .0	255.255.0.0	
10	FIREWALL IP ADDRESS 0 .0 .0 .0	0.0.0.0	When the system is installed behind a NAPT server, the fixed IP Address provided by the NAPT server must be assigned in this field. Also, use this IP address for the UCP address in remote devices.
11	FIRST START MAC ADDR 000000000000	00.00.00.0 0.00.00	A range of MAC addresses can be entered to register devices. This entry is the start address of the first range.
12	FIRST END MAC ADDR 000000000000	00.00.00.0 0.00.00	A range of MAC addresses can be entered to register devices. This entry is the end address of the first range.
13	SECOND START MAC ADDR 000000000000	00.00.00.0 0.00.00	A range of MAC addresses can be entered to register devices. This entry is the start address of the second range.
14	SECOND END MAC ADDR 000000000000	00.00.00.0 0.00.00	A range of MAC addresses can be entered to register devices. This entry is the end address of the second range.
15	SYSTEM IP ADDRESS PLAN SYSTEM RESET		Returns the System IP Address Plan to default values.
16	UCP DNS IP ADDR 0 .0 .0 .0	0.0.0.0	IP Address of Domain Name Server that iPECS will use to resolve URL to an IP address. The DNS provides the resolution after receiving the name from iPECS.

Table 3.3.2.1-1 SYSTEM IP ADDRESS PLAN (PGM 102)

BTN	DISPLAY	DEFAULT	REMARK
17	UCP DHCP (1:ON/0:OFF) : OFF	OFF	The iPECS UCP includes a DHCP client and can obtain an IP address from a local DHCP server.

3.3.2.2 Device IP Address Plan -PGM 103

To register an IP device:

As IP gateway Modules and terminals register to the iPECS, a Sequence number is assigned, which indicates the order of registration. In addition, based on the type of device (CO/IP gateway/board, Terminal) the system assigns a logical number (Station Number or CO/IP Line Number). The Sequence Numbers for CO gateway, Terminals, etc. are independently assigned based on the type of gateway. These Sequence Numbers are employed to provide a relationship between the physical MAC address and the logical port numbers of the device.

The system may assign a default private IP address to each Sequence Number. If desired, this program may be used to modify the assigned IP address for each gateway/board and iPECS Phone.

Each local gateway and terminal can be assigned to employ ARP (Address Resolution Protocol). If ARP is disabled, the system will employ the Ethernet MAC address, layer 2 switching to eliminate the need for IP traffic overhead, reducing overall LAN traffic.

The system normally employs IP multi-cast protocol to respond to a registration request from a gateway or terminal. When the device is separated from the system by a router, the system must use the IP uni-cast protocol. This is established by the "Local Device" assignment. When disabled (Off), the system will send an IP uni-cast message to the device in response to a registration request.

PROCEDURE:	
ENTER PROGRAM NUMBER	1. Press the **[PGM]** button and dial 103.
DEVICE IP ADDRESS PLAN PRESS FLEX KEY (1-7)	2. Select the desired Flex button. Button 1: CO & VOIP Gateway/Board Button 2: Stations Button 3: MISU Button 4: UVMU, UVM Button 5: MCIM Button 6: SYSTEM RESET Button 7: WTIM
	3. Use the **[VOL UP]** and **[VOL DOWN]** buttons to see next/ previous IP Address. Refer to Table 3.3.2.2-1 for display information.

PROCEDURE:	

	4. Press Flex 1~6 to select the attribute desired as shown in Table 3.3.2.2-1.
	    Button 1: IP address
	    Button 2: MAC address
	    Button 3: ARP
	    Button 4: REGISTRATION
	    Button 5: CPU Type
	    Button 6: Device (Board) ID

	5. Use the dial-pad to enter desired data. For IP and MAC addresses, an "*" is used to enter a dot (".")
	6. Press the **[Save]** button to store the data entry.

Table 3.3.2.2-1 DEVICE IP ADDRESS PLAN (PGM 103)

BTN	DISPLAY	FEATURE	DEFAULT
1	**CO/IP Line** 001-001 :B40EDCBF5606 VOIP 1 :10 .10 .10 .2	LCD shows: Line 1: Sequence Number, 2 or 3 digits MAC Address, 12 digits Line 2: Module Type, 4 characters First Logical port number IP Address, 7~12 digits	CO & VoIP Gateway IP address set sequentially, from the range in PGM 102.
1–1	SET IP ADDRESS VOIP 1 : 10.10.10.2	Use Flex button 1 to set the device's IP address in IP v4 format.	10.10.10.10~254
1–2	SET MAC ADDRESS 001-001 : B40EDCBF5606	Use Flex button 2 to enter the device's MAC address in the system memory.	None
1–3	ARP (1:ON/0:OFF): OFF	Use Flex button 3 to enable/disable ARP mode, which employs ARP to locate a device. Otherwise, layer 2 switching is employed with local devices.	OFF
1–4	REGISTRATION (0:UCAST/1:MCAST): MCAST	Use Flex button 4 to enable/disable Local Device Mode, which defines the device as on a common LAN with the UCP.	MCAST
1–5	CPU TYPE MSC1K	Flex button 5 displays the type of CPU employed in the device.	
1–6	DEVICE(BOARD) ID PRIB	Flex button 6 displays the Board type designation.	
2	**iPECS IP Phone** 001-001 : B40EDCBF5606 KTU 1000 :10.10.10.10	LCD shows: Line 1: Sequence Number, 3 digits MAC Address, 12 digits Line 2: Station Type, 3 characters Station Number, 2~4 digits IP Address, 7~12 digits	Station IP address set sequentially, from the range in PGM 102.
2–1	SET IP ADDRESS KTU 1000 :10.10.10.10	Use Flex button 1 to set the device's IP address in IP v4 format.	10.10.10.10~254

Table 3.3.2.2-1 DEVICE IP ADDRESS PLAN (PGM 103)

BTN	DISPLAY	FEATURE	DEFAULT
2–2	SET MAC ADDRESS 0001-0001 : B40EDCBF5606	Use Flex 2 button to enter the device's MAC address into system memory.	None
2–3	ARP (1:ON/0:OFF): OFF	Use Flex button 3 to enable/disable ARP mode, which employs ARP to locate a device. Otherwise, layer 2 switching is employed with local devices.	OFF
2–4	REGISTRATION (0:UCAST/1:MCAST): MCAST	Use Flex button 4 to enable/disable Local Device Mode, which defines the device as on a common LAN with the UCP.	MCAST
2–5	CPU TYPE MSC1K	Flex button 5 displays the type of CPU employed in the device.	
2–6	DEVICE(BOARD) ID HYIB	Flex button 6 displays the terminal type designation.	
3	**Miscellaneous Unit** 001-017 : B40EDCBF5606 MISU :10.10.10.2	LCD shows: Line 1: Sequence Number, 2 digits MAC Address, 12 digits Line 2: "MISU" IP Address, 7~12 digits	IP address of built-in MISU set automatically,
3–1	SET IP ADDRESS MISU :10.10.10.2	Use Flex button 1 to set the device's IP address in IP v4 format.	10.10.10.10~254
3–2	SET MAC ADDRESS 001-003 : B40EDCBF5606	Use Flex button 2 to enter the device's MAC address into system memory.	None
3–3	ARP (1:ON/0:OFF): OFF	Use Flex button 3 to enable/disable ARP mode, which employs ARP to locate a device. Otherwise, layer 2 switching is employed with local devices.	OFF
3–4	REGISTRATION (0:UCAST/1:MCAST): MCAST	Use Flex button 4 to enable/disable Local Device Mode, which defines the device as on a common LAN with the UCP.	MCAST
3–5	CPU TYPE MSC1K	Flex button 5 displays the type of CPU employed in the device.	
3–6	DEVICE(BOARD) ID MISU	Flex button 6 displays the MISU type designation.	
4	**Voice Mail Unit**	LCD shows:	IP address set

Table 3.3.2.2-1 DEVICE IP ADDRESS PLAN (PGM 103)

BTN	DISPLAY	FEATURE	DEFAULT
	001-015 : B40EDCBF5606 UVMU :10.10.10.2	Line 1: Sequence Number, 2 digits MAC Address, 12 digits Line 2: device type and IP address.	sequentially, from the range in PGM 102.
4–1	SET IP ADDRESS UVMU :10.10.10.2	Use Flex button 1 to set the device's IP address in IP v4 format.	10.10.10.10~254
4–2	SET MAC ADDRESS 001-015 : B40EDCBF5606	Use Flex button 2 to enter the device's MAC address into system memory.	None
4–3	ARP (1:ON/0:OFF): OFF	Use Flex button 3 to enable/disable ARP mode, which employs ARP to locate a device. Otherwise, layer 2 switching is employed with local devices.	OFF
4–4	REGISTRATION (0:UCAST/1:MCAST): MCAST	Use Flex button 4 to enable/disable Local Device Mode, which defines the device as on a common LAN with the UCP.	MCAST
4–5	CPU TYPE MSC1K	Flex button 5 displays the type of CPU employed in the device.	
4–6	DEVICE(BOARD) ID UVMU	Flex button 6 displays the type designation.	
5	**Conference Module** 001-018 : FFFF0000FFFF MCIM : 10 .10 .10 .2	LCD shows: Line 1: Sequence Number, 2 digits MAC Address, 12 digits Line 2: device type and IP address.	IP address set sequentially, from the range in PGM 102.
5–1	SET IP ADDRESS MCIM :10.10.10.2	Use Flex button 1 to set the device's IP address in IP v4 format.	10.10.10.10~254
5–2	SET MAC ADDRESS 001-018 : FFFF0000FFFF	Use Flex button 2 to enter the device's MAC address into system memory.	None
5–3	ARP (1:ON/0:OFF): OFF	Use Flex button 3 to enable/disable ARP mode, which employs ARP to locate a device. Otherwise, layer 2 switching is employed with local devices.	OFF
5–4	REGISTRATION (0:UCAST/1:MCAST): MCAST	Use Flex button 4 to enable/disable Local Device Mode, which defines the device as on a common LAN with the UCP.	MCAST
5–5	CPU TYPE MS828	Flex button 5 displays the type of CPU employed in the device.	
5–6	DEVICE(BOARD) ID MCIM_V	Flex button 6 displays the type designation, MCIM_V.	

Table 3.3.2.2-1 DEVICE IP ADDRESS PLAN (PGM 103)

BTN	DISPLAY	FEATURE	DEFAULT
6	DEVICE IP ADDRESS PLAN SYSTEM RESET	If the **[Save]** button is pressed, the system will reset and restart.	
7	**Wireless DECT Module** 001-003 : 00405A142C67 WTIM : 10:10:10:14	LCD shows: Line 1: Sequence Number, 2 digits MAC Address, 12 digits Line 2: "WTI4" and IP Address, 7~12 digits	IP address set sequentially, from the range in PGM 102.
7–1	SET IP ADDRESS WTIM :10.10.10.14	Use Flex button 1 to set the device's IP address in IP v4 format.	10.10.10.10~254
7–2	SET MAC ADDRESS 001-003 : 00405A142C67	Use Flex button 2 to enter the device's MAC address into system memory.	None
7–3	ARP (0:ON/1:OFF): OFF	Use Flex button 3 to enable/disable ARP mode, which employs ARP to locate a device. Otherwise, layer 2 switching is employed with local devices.	OFF
7–4	REGISTRATION (0:UCAST/1:MCAST):MCAST	Use Flex button 4 to disable/enable Local Device Mode, which defines the device as on a common LAN with the system. MCAST enables Local Device Mode.	MCAST
7–5	CPU TYPE MS828	Flex button 5 displays the type of CPU employed in the device.	
7–6	DEVICE(BOARD) ID WTIM	Flex button 6 displays the type designation, WTIM.	

3.3.2.3 CO Device Sequence Number -PGM 104

The system configures the CO/IP Line numbers as discussed in section 2. Each CO/IP Line module is assigned a registration "Sequence" number and a starting CO/IP Line number based on the Sequence number. In place of the default numbering, the Sequence number associated with the device can be changed thus altering the starting CO/IP Line numbers for a module.

PROCEDURE:	
ENTER PROGRAM NUMBER	1. Press the **[PGM]** button and dial 104.
0001 0002 0003 0004 0005 2401	2. Press the Flex button (1~6) for the desired Sequence Number, use the **[VOL UP]** and **[VOL DOWN]** buttons for the next/previous set of six Sequence Numbers.

0006 0007 0008 0009 0010	3. Using the dial pad, enter new slot numbers. Note slot numbers cannot be duplicated and duplicates will cause an error. The **[SPEED]** button may be used to erase the slot number associated with the selected Sequence Number.
	4. Press the **[Save]** button to store the new data.

3.3.2.4 Flexible Station Numbering Plan -PGM 105

Each terminal (iPECS IP and LDP Phone, iPECS softphone, SIP phone, and SLT) is assigned a Sequence Number during the registration process. The Sequence Number is a 4-digit number starting at 0001, which is incremented as each terminal device is registered. At registration, station numbers increment sequentially with the Sequence Number and are assigned starting at station 1000 for Sequence Number 0001. The Station Numbering Plan allows the station numbers to be two (2) to four (4) digits and up to eight (8) digits in length when using the 8-digit Table.

Note, if the 8-Digit Table is used, the prefix digits for each station must be entered.

PROCEDURE:	
ENTER PROGRAM NUMBER	1. Press the **[PGM]** button and dial 105.
0001　　0002　　0003　　0004 1000　　1001　　1002　　1003	2. Use either of the two methods below to change the station number associated with a Sequence Number. Note pressing the **[SPEED]** button twice clears all station number assignments. The **[VOL UP] / [VOL DOWN]** buttons are used to view the next/previous two station Sequence Numbers.
0001　　0002　　0003　　0004 1000　　1001　　1004　　1008	3. Range entry: Using the dial-pad, enter a station number range (first & last station number). The range assignment begins with the first station number shown by the LCD and continues to the end of the entered range.
	4. Single entry: 4.1 Press Flex button 1~4 to select the desired Sequence Number from those shown by the LCD 4.2 Dial new station number.
	5. Press the **[Save]** button to store the new station numbers.

3.3.2.5 Flexible Numbering Plan part A to D - PGM 106 to 109

Feature dial codes for the system can be assigned using the system's Flexible Numbering Plan. Feature codes should be one (1) to four (4) digits in length and must not conflict with other codes or the Flexible Station Numbering Plan. For example, Feature dial codes 53 and 536 represent a conflict. The system will generate error tone and will not update the database. Tables 3.3.2.5-1 to 3.3.2.5-4 below show the defaults for Numbering Plan 1. Appendix B provides the default values for each of the eight basic Numbering Plans; select the basic Numbering Plan in PGM 100.

PROCEDURE:	
ENTER PROGRAM NUMBER	1. Press the **[PGM]** button and dial 106 for Part A 107 for Part B 108 for Part C 109 for Part D
FLEX NUMBERING PLAN A PRESS FLEX KEY (01-24)	2. Select the desired button (01~24); refer to Table 3.3.2.5-1 to Table 3.3.2.5-4 for **PGM** 106 to 109 respectively.
	3. Use the dial-pad to enter desired data. Where a range is required, input the first and last numbers in the range.
	4. Press the **[Save]** button to store the new Numbering Plan data.

Table 3.3.2.5-1 FLEXIBLE NUMBERING PLAN PART A (PGM 106)

BTN	DISPLAY	FEATURE	DEFAULT
1	INT PAGE ZONES START& END:*301-*400	Internal Page Zone access codes	*301~*400
2	INT ALL CALL ENTER NEW #:*543	Internal All Call Page access code	*543
3	MEET ME PAGE ENTER NEW #:*544	Meet-Me-Page answer code	*544
4	EXT PAGE ZONE 1 ENTER NEW #:*545	External Page Zone 1 access code	*545
5	EXT PAGE ZONE 2 ENTER NEW #:*546	External Page Zone 2 access code	*546
6	EXT ALL CALL ENTER NEW #:*548	External All Call Page access code	*548
7	ALL CALL PAGE ENTER NEW #:*549	All Call Page access code	*549
8	SMDR ACT CODE ENTER ENTER NEW #:*550	Dial code to signify the start of an SMDR Account Code.	*550
9	FLASH CMD TO CO ENTER NEW #:*551	Dial code to generate a Flash on the active CO Line.	*551
10	SLT LAST SPD DIAL ENTER NEW #:*552	SLT Last Number Redial feature access code.	*552

Table 3.3.2.5-1 FLEXIBLE NUMBERING PLAN PART A (PGM 106)

BTN	DISPLAY	FEATURE	DEFAULT
11	DND ENTER NEW #:*553	Dial code to activate Do-Not-Disturb.	*553
12	CALL FWD ENTER NEW #:*554	Dial code to activate Call Forward.	*554
13	SPD DIAL PGM ENTER NEW #:*555	Speed Dial programming access dial code for SLTs.	*555
14	MSG WAIT ENABLE ENTER NEW #:*556	Dial code to activate a Message Wait/Call Back.	*556
15	MSG WAIT RETURN ENTER NEW #:*557	Dial code to return a Message Wait/Call Back.	*557
16	SPD DIAL ACCESS ENTER NEW #:*558	SLT Speed Dial access code	*558
17	DND/FWD CANCEL ENTER NEW #:*559	Dial code to cancel DND/FWD/MSG Wait.	*559
18	CO SYS HOLD ENTER NEW #:*560	Dial code to place a CO call on System Hold.	*560
19	SLT PGM MODE ENTER ENTER NEW #:*561	User program mode entry dial code for SLTs.	*561
20	ATTD UNAVAILABLE ENTER NEW #:*562	Dial code to place attendant in the "unavailable" mode, attendant only.	*562
21	ALARM RESET ENTER NEW:*565	Dial code to terminate Alarm contact signal.	*565
22	GROUP CALL PICK-UP ENTER NEW #:*566	Group Call Pick-up dial code	*566
23	UNIVERSAL ANSWER ENTER NEW #:*567	Universal Night Answer dial code.	*567
24	ACCNT CODE WITH BIN ENTER NEW #:*568	Dial code for entering an Account code.	*568

Table 3.3.2.5-2 FLEXIBLE NUMBERING PLAN PART B (PGM 107)

BTN	DISPLAY	FEATURE	DEFAULT
1	WALKING COS ENTER NEW #:*569	Dial code to activate Walking Class-of-Service.	*569
2	ACD AGENT ON/OFF DUTY ENTER NEW #:*571	Dial code to toggle ACD Agent or Supervisor ON and OFF duty.	*571
3	ACD SUPERVISOR LOGIN ENTER NEW #:*572	Supervisor login dial code.	*572
4	ACD SUPERVISOR LOGOUT ENTER NEW #:*573	Supervisor logout dial code.	*573
5	ACD HELP CODE ENTER NEW #:*574	Agent code requesting Supervisor help and Supervisor Help request Response code.	*574
6	ACD CALLS IN QUEUE ENTER NEW #:*575	Dial code to display calls in queue.	*575
7	ACD SUPERVISOR STATUS ENTER NEW #:*576	Dial code to display group status.	*576
8	ACD SUPERVISOR MONITOR ENTER NEW #:*577	Dial code to activate Supervisor monitor.	*577
9	ACD REROUTE QCALL ANS ENTER NEW #:*578	Dial code to reroute call after answer.	*578
10	ACD REROUTE QCALL NO AN ENTER NEW #:*579	Dial code to reroute call prior to answer.	*579
11	CAMP-ON ANSWER ENTER NEW #:*621	Dial code to answer a Camped On call.	*621
12	CALL PARK LOCATIONS START#:#601-#800	Dial code to place/retrieve a call in a Park location.	#601~#800
13	STA GRP PILOT NUMBER START #:*401-*500	Station group pilot numbers	*401~*500
14	STA USER VSF FEATURES ENTER NEW #:66	VSF feature access code.	66
15	CALL COVERAGE RING ENTER NEW #:*76	Code for Call Coverage button	*76

Table 3.3.2.5-2 FLEXIBLE NUMBERING PLAN PART B (PGM 107)

BTN	DISPLAY	FEATURE	DEFAULT
16	DIRECT CALL PICK-UP ENTER NEW #:*77	Dial code to activate Directed Call Pick-up.	*77
17	ACCESS CO GROUP FEAT ENTER NEW #:89	Dial code to access a CO Line or IP channel from a CO/IP group.	89
18	ACCESS IND CO/IP FEAT ENTER NEW #:88	Dial code to access a specific CO Line.	88
19	ACCESS HELD CO/IP FEAT ENTER NEW #:8*	Dial code to access last held CO Line or IP channel from Hold.	8*
20	ACCESS HELD IND CO/IP ENTER NEW #:8#	Dial code to access a specific CO Line/IP channel from Hold.	8#
21	ACCESS CO IN 1ST CO GRP ENTER NEW #:9	Dial code to access the 1st available CO Line in any accessible group.	9
22	ATTENDANT CALL ENTER NEW #:0	Dial code to call an Attendant.	0
23	VM MSG WAIT ENABLE ENTER NEW #:*8	Dial code for external Voice mail to activate Message Wait indication.	*8
24	VM MSG WAIT CANCEL ENTER NEW #:*9	Dial code for external Voice Mail to deactivate Message Wait indications.	*9

Table 3.3.2.5-3 FLEXIBLE NUMBERING PLAN PART C (PGM 108)

BTN	DISPLAY	FEATURE	DEFAULT
1	DOOR OPEN 1 ENTER NEW #:#*1	Dial code to activate Door 1 contact (open door 1)	#*1
2	DOOR OPEN 2 ENTER NEW #:#*2	Dial code to activate Door 2 contact (open door 2).	#*2
3	DOOR OPEN 3 ENTER NEW #:#*3	Dial code to activate Door 3 contact (open door 3).	#*3
4	DOOR OPEN 4 ENTER NEW #:#*4	Dial code to activate Door 4 contact (open door 4).	#*4

Table 3.3.2.5-4 FLEXIBLE NUMBERING PLAN PART D (PGM 109)

BTN	DISPLAY	FEATURE	DEFAULT
1	MCID REQUEST ENTER NEW #:*0	Dial code to activate Malicious Call ID Request in ISDN Supplementary service. Not available in USA version	*0
2	AME FEATURE ENTER NEW #: *564	Dial code to assign an Answering Machine Emulation Flex button.	*564
3	US-CONF TMR EXTENSION ENTER NEW #:##	Dial code to extend Unsupervised conference time.	##
4	PTT GROUP LOGIN/OUT ENTER NEW #:#0	Push-To-Talk group login and logout dial code. The station must have a PTT button for proper operation.	#0
5	ACD AGENT P LOGIN ENTER NEW #:*581	ACD Agent Primary Login code	*581
6	ACD AGENT P LOGOUT ENTER NEW #:*582	ACD Agent Primary Logout code	*582
7	ACD AGENT S LOGIN ENTER NEW #:*583	ACD Agent Secondary Login code	*583
8	ACD AGENT S LOGOUT ENTER NEW #:584	ACD Agent Secondary Logout code	*584
9	ACD AGENT WRAPUP END ENTER NEW #:*585	ACD Agent wrap-up end code.	*585
10	TNET CM LOGIN/OUT ENTER NEW #:*586	When Central Control networking (TNET) is employed, a station can be manually logged in or out of the Central system using this code.	*586
11	ENTER INTO CONF-ROOM ENTER NEW #:*59	Code for a station to enter a conference room	*59
12	ENTER INTO CONF-GROUP ENTER NEW #:*68	Code to open a conference group	*68
13	STATION ICR ENTER NEW #:*587	Code to activate Station ICR	*587
14	PICK UP GROUP PICK-UP ENTER NEW #:*588	Pick Up Group Call Pick-up dials code.	*588

Table 3.3.2.5-4 FLEXIBLE NUMBERING PLAN PART D (PGM 109)

BTN	DISPLAY	FEATURE	DEFAULT
15	EMERGENCY PAGE ENTER NEW #:*589	Code for emergency page	*589
16	REMOTE MEX CONTROL ENTER NEW #:*580	Code to control the mobile extension settings remotely	*580
17	ALL GR AGENT ON/OFF DUT ENTER NEW #:*58*	Code to change the state of the Agent ON/Off duty in all Station groups	*58*
18	SLT ACNR CODE ENTER NEW #:*58#	In SLT, user can ACNR feature by using this numbering plan.	*58#
19	ACD SUPERVISOR RING MODE ENTER NEW #:*570	Code to check and change ACD group Ring mode by ACD group supervisor.	*570
20	COMPANY DIRECTORY NAME ENTER NEW #:*563	Code to check and change recording station subscribe name of Company Directory feature. (USA Only)	*563
21	ISDN SUPP HOLD ENTER NEW #:*57*	ISDN Supplementary Service for HOLD	*57*
22	ISDN SUPP CONF ENTER NEW #:*57#	ISND Supplementary Service for Conference (Not supported).	*57#
23	FORCED SEIZE BUSY STN/CO ENTER NEW #:*56*	Busy Station / CO can be connected with entering this Code.	*56*
24	ADDED FLEX NUMBER PLAN PRESS FLEX KEY (1-5)		
24-1	OVERRIDE DND/CFW ENTER NEW #:*56#	[56#] + Station number, then the station's DND or CFW setting will be overridden.	*56#
24-2	CALL BACK CANCEL ENTER NEW #:....	Code is used to cancel call back.	
24-3	XFER TO VSF ANNC NO ENTER NEW #:*55*	When a station is talking over a CO line User, [Transfer] + [55*] + Valid system announcement (01-70), then Outside user can hear system announcement, and system starts DISA service.	*55*

Table 3.3.2.5-4 FLEXIBLE NUMBERING PLAN PART D (PGM 109)

BTN	DISPLAY	FEATURE	DEFAULT
24-4	CCR ENTER NEW #:#2	This code is used with Digit Conversion to direct a call to a specific System Announcement. The conversion modifies the received digits to a four digit string "#2xx" where xx indicates the announcement number	#2
24-5	UCS SYS CONF GRP JOIN ENTER NEW #:5*0	After a UCS client activates a UCS Conference Group, other users may dial this code and the group number to enter the Conference.	5*0

3.3.2.6 8-Digit Table –PGM 238

The iPECS UCP supports a Station Numbering Plan of up to eight digits. By combining a prefix digit string of up to six (6) digits with the Add Digit count (digit count from the Station Numbering assigned in PGM 105), the Station Numbering Plan can support up to eight digits. Note that multiple prefixes (8 Digit Strings) can be assigned with varying Add Digit counts. In addition, in case of a conflict, the Prefix digit string will have priority over the Flexible Numbering Plan thus disabling the feature associated with the digit string.

PROCEDURE:	
ENTER PROGRAM NUMBER	1. Press the **[PGM]** button and dial 238.
DIGIT 8 TBL ENTER TBL NO(01-30)	2. Use the dial-pad to enter the desired table number.
Digit 8 TBL 1 : Empty	3. Press the desired Flex button; refer to Table 3.3.2.6-1.
	4. Use the dial-pad to enter the required data.
	5. Press the **[Save]** button to store the data entry.

Table 3.3.2.6-1 8-Digit Table (PGM 238)

BTN	ATTRIBUTE/DISPLAY	DESCRIPTION	RANGE	DEFAULT
1	TBL 1 : SET 8 digit 8 Digit 1 : ….....	This field defines the prefix digits for the Station Number Plan.	Up to 6 digits	N/A
2	TBL 1: SET ADD DIGIT ADD DIGIT : 0	This field assigns the number of digits from the Station Numbering Plan PGM 105.	0 ~ 4 digit	0

4. WEB SERVICE

4.1 General

The iPECS UCP incorporates a Web Server employed by the system's Web Service:

1) Setup Wizard

2) Administration: PGM Base & Function Base

3) Maintenance: File upload, Database, Multi language, SMDR, VSF, Voice Mail, Trace, other function.

4) Station Portal

The default database includes assignment of a private IP address to the system. This address (10.10.10.2) may be used to access the system from the LAN. However, a routable IP address must be assigned for access from a remote location.

To access the iPECS Web Server requires:

1. Operating iPECS series system

2. IP address assigned in the system and is known

3. TCP/IP port assigned for the UCP LAN port and is known

4. iPECS UCP connected to an accessible LAN

5. iPECS UCP password (Maint, Admin, and User) if any, is known

4.1.1 PC/Browser

- MS Explore 10.0, Chrome 24.0, Firefox 18.0 or higher version is recommended (HTML5 support required)

- Windows PC, at least 32MB RAM (64MB or more RAM is recommended)

- NIC (Network Interface Card)

4.1.2 Environment for LAN connection

- IEEE 802.3, 10/100 Base T

- Static/DHCP addressing

- Firewall, requires Network Administrator to allow access

- Remote access requires a routable IP address for the iPECS UCP Web Server. This must be assigned to the system prior to access.

4.1.3 Web Browser setting

Web browsers may store (cache) a copy of the iPECS Web pages in a cache memory. The Web browser may use these copies to provide a "quick view". If the Web page has been altered by data entered in Station Admin or a file upgrade, the cached copy will be out-of-date and could cause unexpected system operation. To assure proper page views and data entry, the browser should be set to eliminate the use of the cached pages. For Explorer, in "Internet Options", enable refresh on "Every visit to the page", for other browser, the procedure may be different.

4.1.4 Password Encryption

When enabled in PGM 162, iPECS UCP implements decryption of the password employing RC-6 block encryption. iPECS UCP employs a Sun Java Virtual Machine applet to implement AES encryption. The PC entering the Password must have a JAVA Virtual Machine and the JRE (Java Runtime Environment) Explorer option enabled to handle encrypted passwords properly. The Sun JVM is downloaded from the Java home page (www.java.com). Once downloaded, execute the downloaded file. To enable the Explorer JRE option,

1. From the Explorer menu, select Internet Options-Advanced.

2. From the Advanced Internet Options check the **[Use JRE….]** Option.
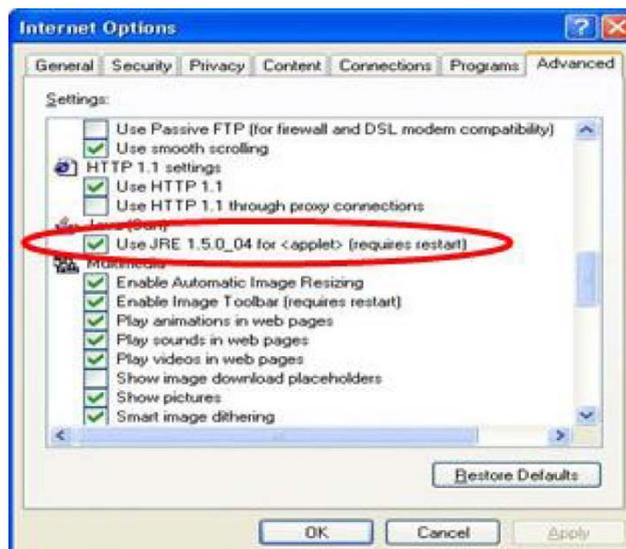
3. After restarting the computer, access the iPECS UCP Web Manager, section 4.2. "Applet iPECSPwd started" will display in the bottom left corner to indicate password encryption is active.

4.2 iPECS UCP Web Access & Login

In the browser 'ADDRESS' field, enter the UCP IP address (default is https://10.10.10.2 or http://10.10.10.2) and select 'GO'. The Web server goes to iPECS UCP Web Services Login page, Figure 4.2-1.

The UCP Web services include the Admin & Maintenance functions and the Station Program User Portal. To access the Admin and Maintenance function, a valid User ID and Password matching the assignments in the System Password PGM 162 or User Management in the Maintenance section must be entered. The default User ID is 'admin' and the password is '1234'.
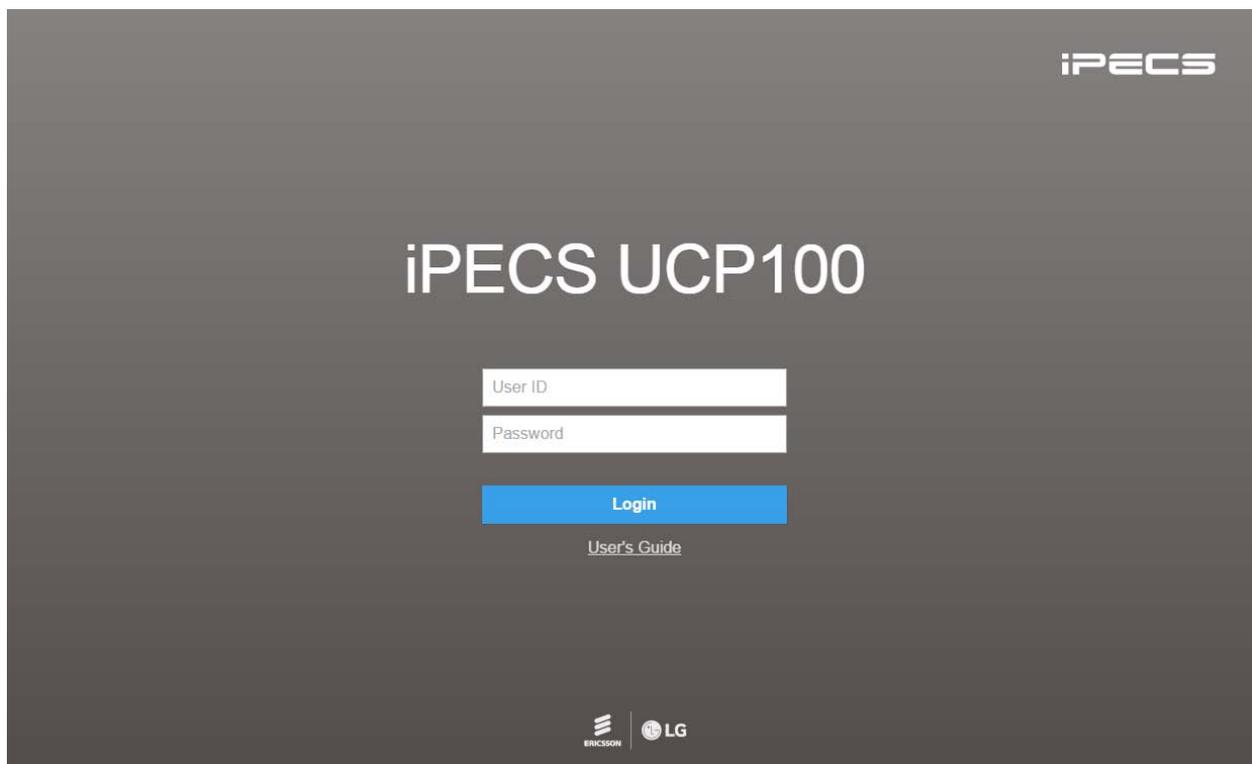
Figure 4.2-1 iPECS UCP Home page

For reference, we explain Admin and Maintenance based on UCP100. The home page is displayed according to UCP type: UCP100, UCP600, UCP2400.

The exact specification according to UCP type is described "*Section 2.1 System capacity*" in this manual.

To access the Station program, the user must enter **Station number and full Authorization code** (station number and Auth code) as defined in *Tables Data – Station Authorization Codes (PGM 227)*. The system will go to the Station Program Main page as below. For detail information, refer to the title "*Station Program (User Portal)*".
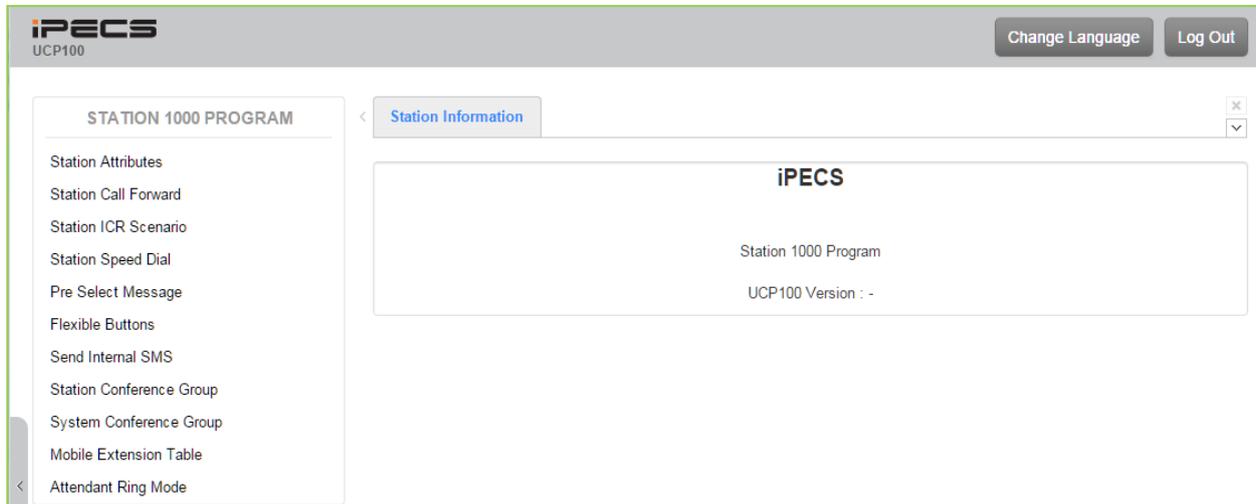
Figure 4.2-2 iPECS UCP Station program

4.3 Web Admin & Maintenance Main Page Overview

In the Web Admin Login screen (see section 4.2), enter the User ID and Password then click the
[Login] button to access the iPECS Admin & Maintenance Main Page as shown in Figure 4.3-1.
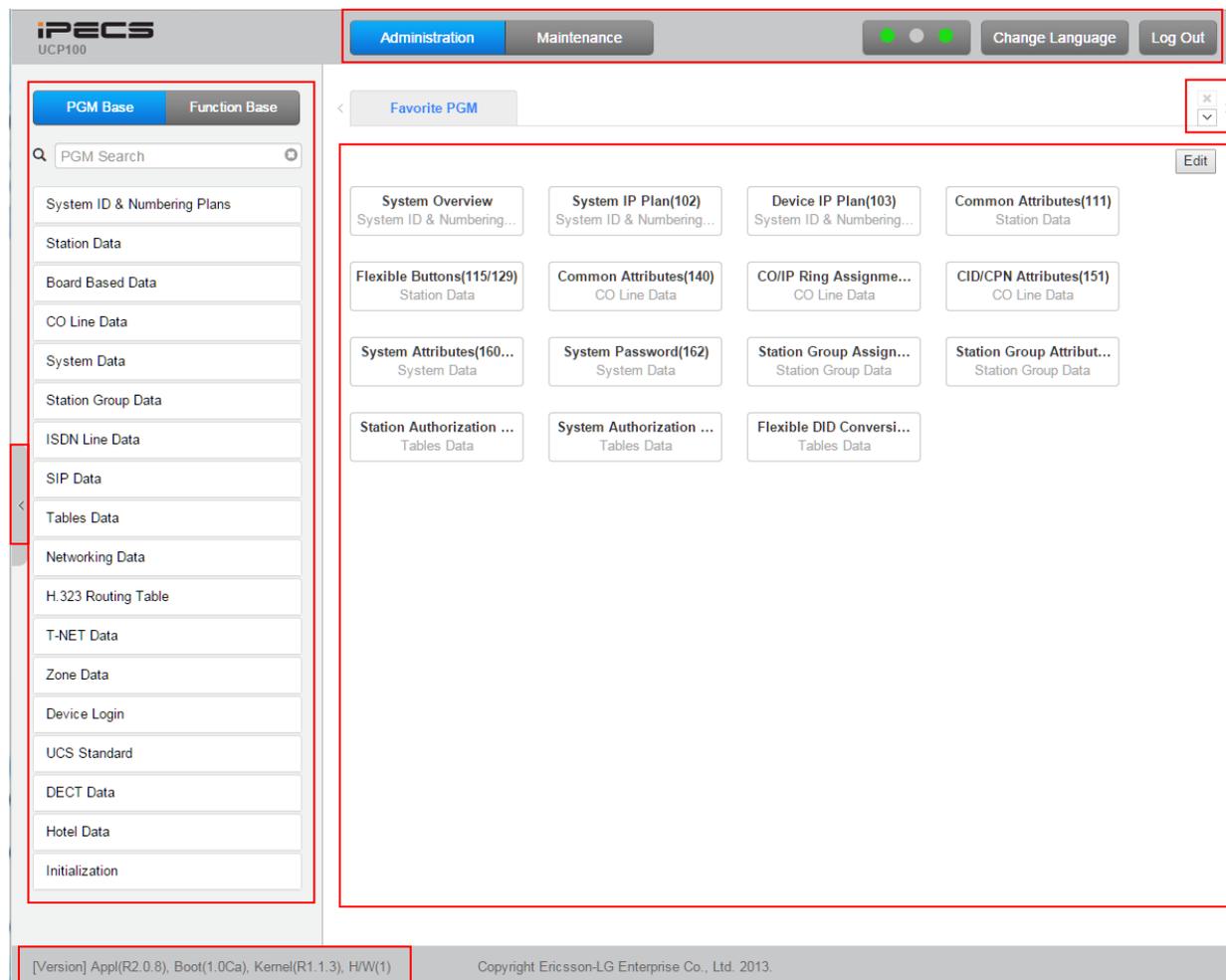
Figure 4.3-1 iPECS UCP Admin & Maintenance Main Page

The Admin & Maintenance Main Page has three sections,
> **Menu bar** – Upper frame
> **Directory & Navigation section** – Left frame
> **Favorite Programs and Entry section** – Central frame

Items in the Menu bar are mouse-clickable for selections of:
> **Administration** – accesses the system database.
> **Maintenance** – SW upgrade, Database, Multi Language, SMDR, VSF, Voice Mail, Trace, etc.
> **Change Language** – change the desired language.
> **Log out** – log out the web server

What is the meaning about Buttons and Text on page?

1)   Hide menu by clicking the left arrow in the middle of window on left side.

2)   X button at the top of the page on the right side functions whether the current tab or all tabs are closed or not.

3)   [⌄]   button will display all PGM by click and check which PGM is using.

4)   The Application, Boot, and Kernel version can find at the left side of bottom.

5)   User can check the status indicator at the bottom on the right side of page by displaying the text such Disconnected or Read Only User.

> ➤ Disconnected: discconected to Web Admin Server without working for a long time. User can connect to Web Admin Server by clicking Refresh button.

> ➤ Read Only User: User who access to Web Admin Sever has no authority to set the function as the maintenance user and just is only for read.

Notification

Connection Lost.
If you want to reconnect, Click button
Refresh.

Refresh   Cancel

Copyright Ericsson-LG Enterprise Co., Ltd. 2013.   ⚠ Disconnected
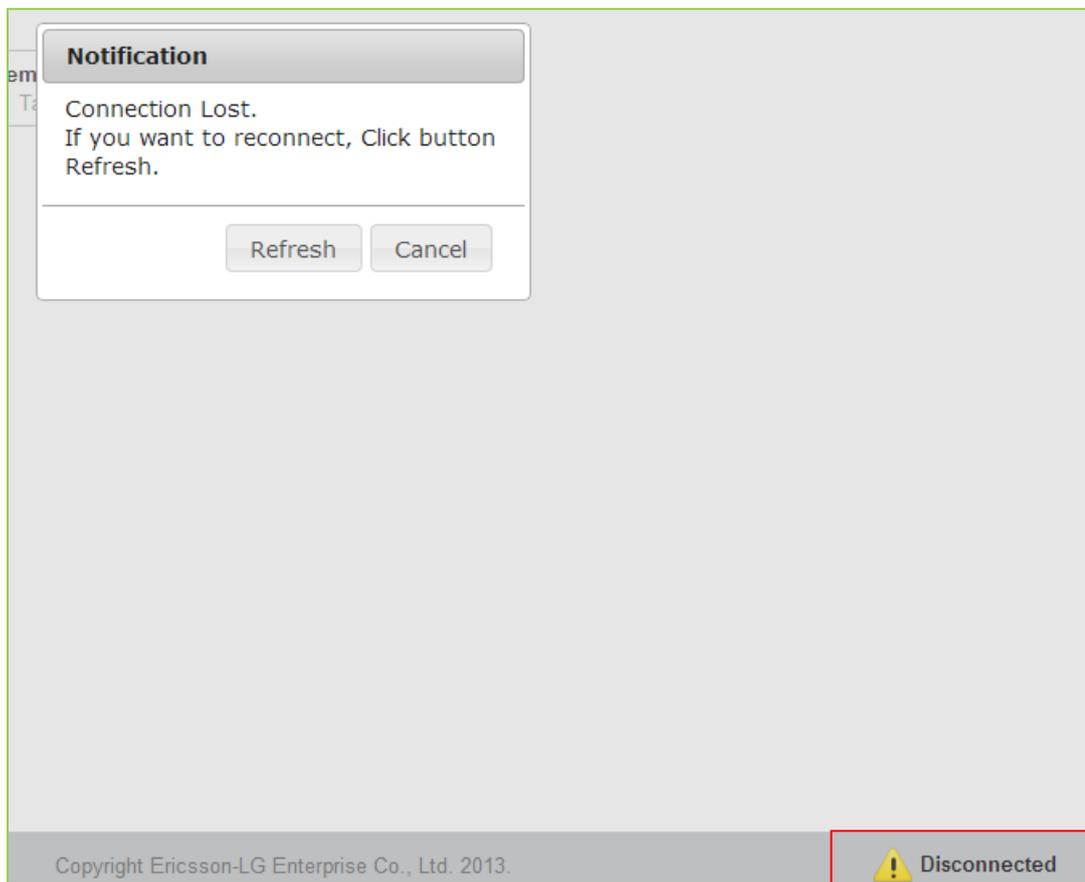
License State Display

In UCP system, a Software Maintenance license is needed in order to upgrade system software. If "No Maint. Period" or the color of SW Maint. is GREEN, system software can be upgraded. Major version upgrade is possible in "No Maint." or "Maintenance" Period but Minor version upgrade is possible in "Warranty" Period. In other states, both upgrade and downgrade are not allowed.

- First LED : SW maintenance
- Second LED: Temporary license
- Third LED: T-Net LCM/Redundancy

SW Maintenance State

The following figure is SW Maintenance stat for SMB. System Alarm is occurred 30 days once before expiration and for 7 days before expiration.
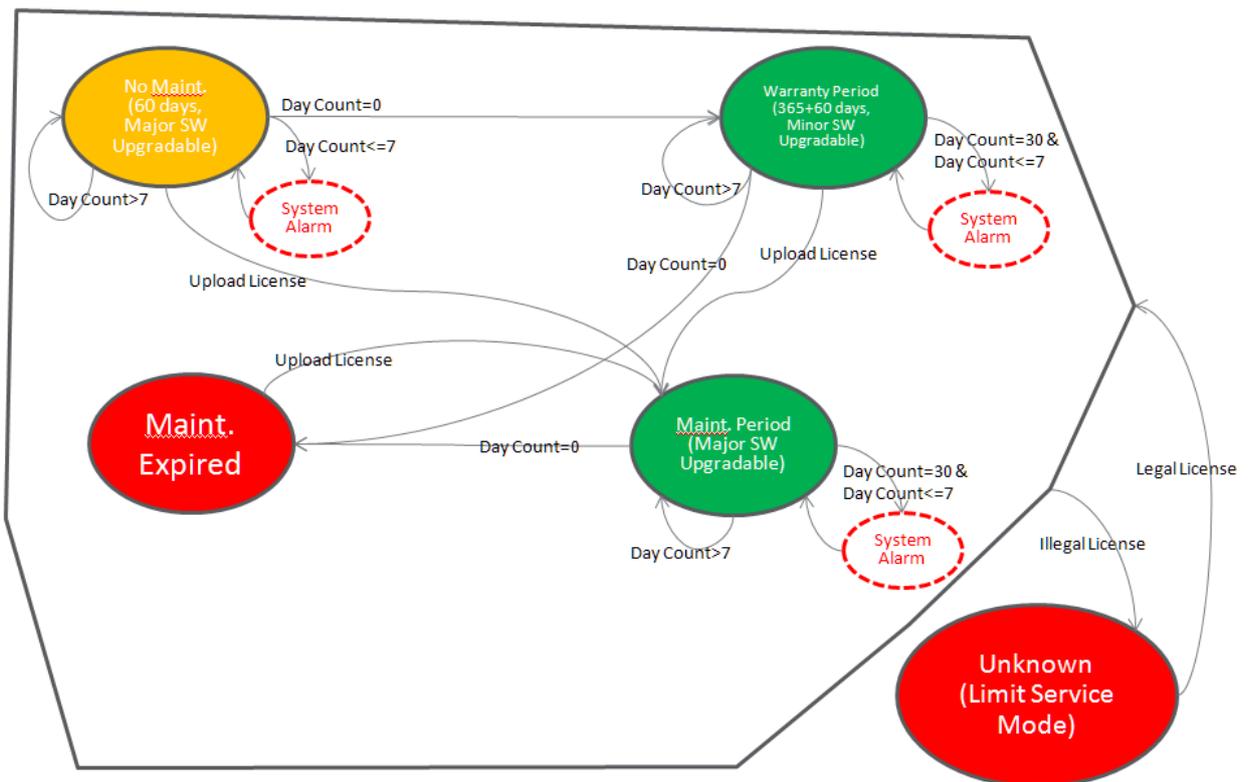
Temp License State

Temp License can be activated totally 6 times. It is valid for 60 days at first activation and is valid 30 days from second activation to the last. System Alarm is occurred from 7 days before expiration.
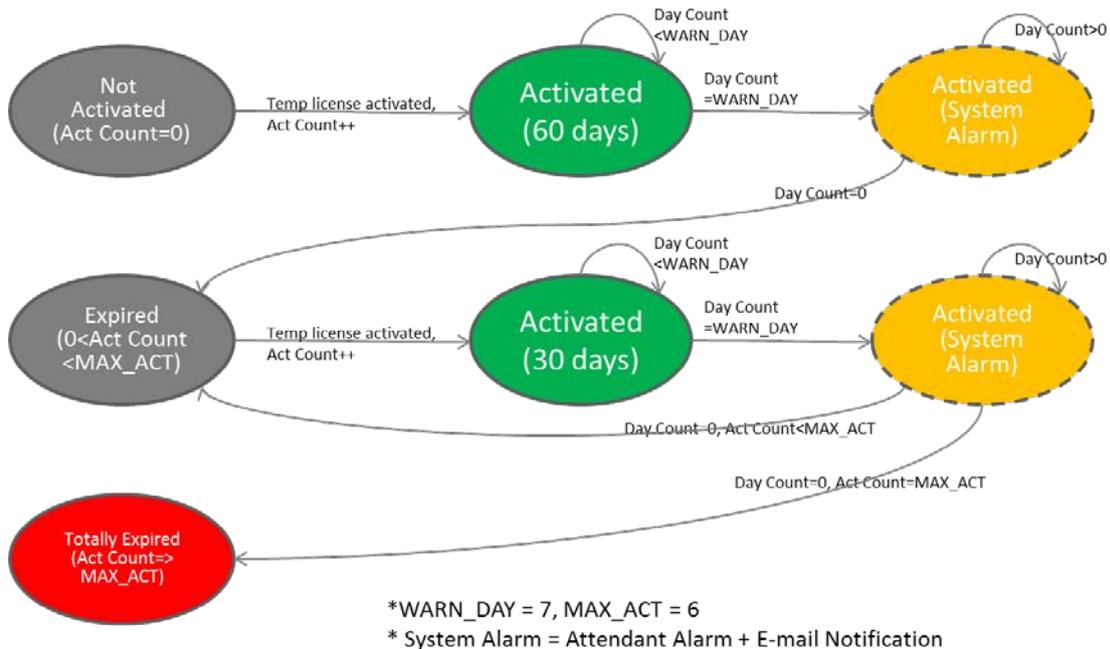
T-Net or Redundancy State

T-Net LCM license is valid for 60 days if the connection is broken. And Redundancy is valid for 60 days if eMG system is slave, active state, and the connection is broken. But the limitation is not applied in Master eMG. System Alarm is occurred from 7 days before entering "Limited Service Mode".

- Below cases are always "Normal".
1. TNLS license
2. Master system when redundancy
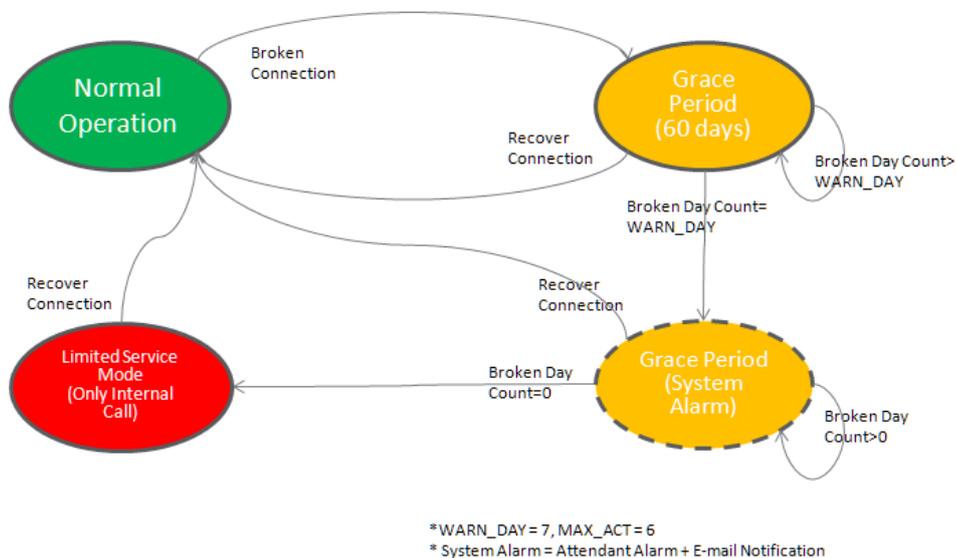
4.3.1 Favorite Program Groups

To ease access to frequently used program, the iPECS Admin Main Page displays a Favorite PGM list. The Favorite PGM list buttons, when selected, return the associated Web page. Up to 20 favorites are configured using the Edit button in the upper right of the page.
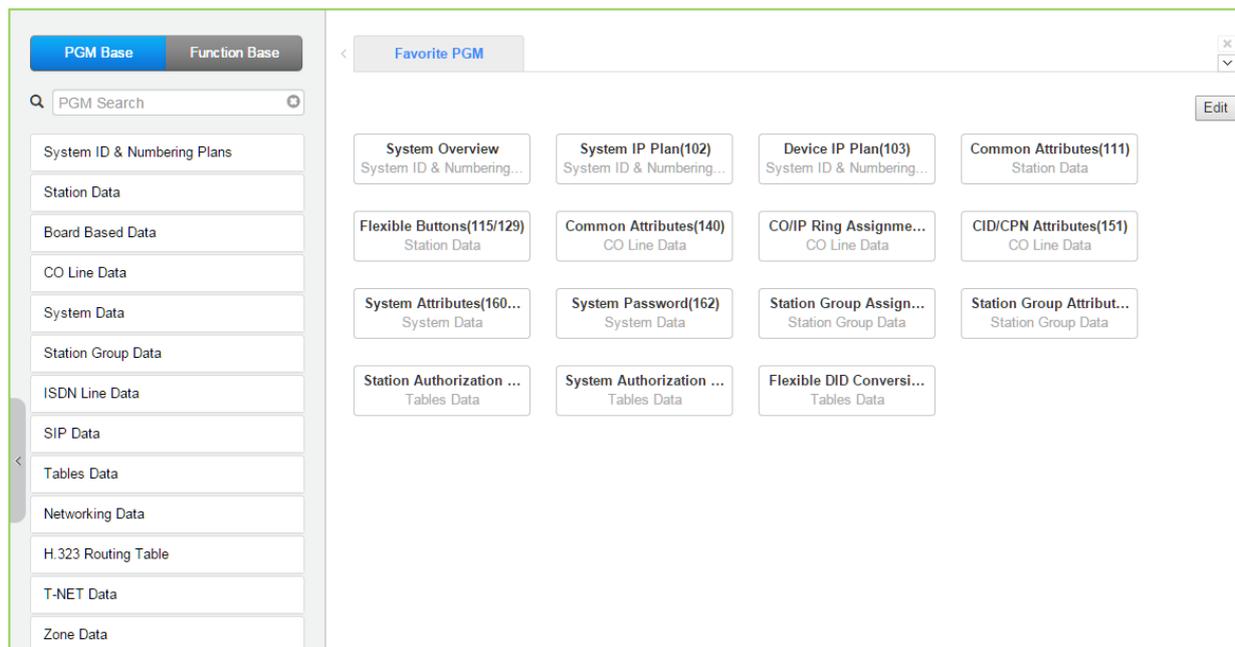
Figure 4.3.1-1 Favorite Program Groups

To register a Favorite program, click Edit button. The following page will be displayed.

- Clear button: Clears the check box for all programs.
- Save button: Saves the Web page, PGMs with checked boxes are stored as Favorites, up to 20.
- Back button: Returns to the previous page.
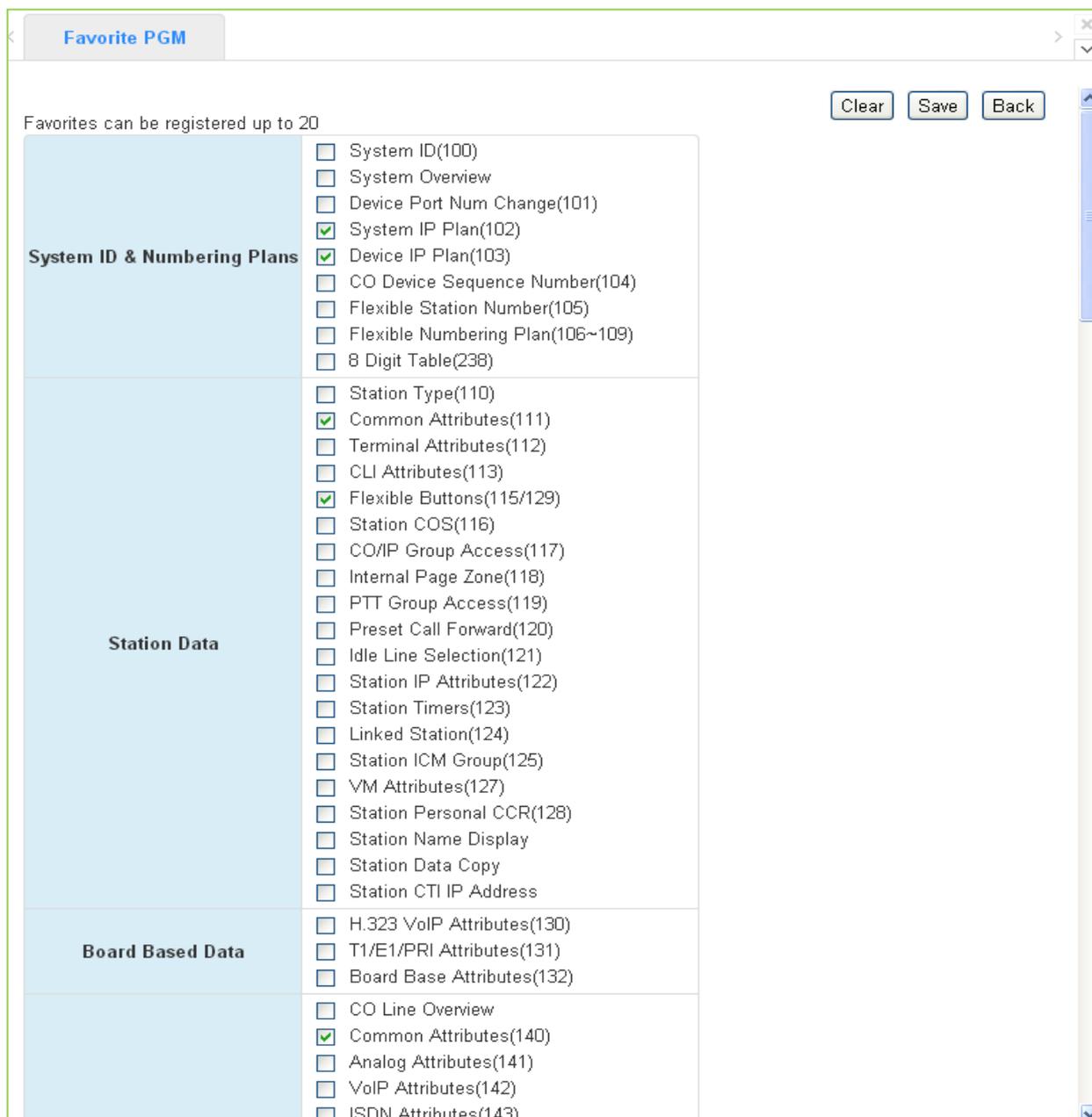
Figure 4.3.1-2 Favorite Program Groups for edit

4.3.2 Using Function Base

Function Base is designed to customize the feature or PGM by configuring the Function base so that customer can be easy to use the desired and frequent feature.

There are two buttons: Common Function List and User Function List. The feature or PGM can register up to 20 as Favorites.

1) Common Function List

Generally, iPECS UCP supports the basic function list as default. The available function is 5 (DID Setting Scenario, Network Scenario, SIP Extension Registration, SIP Trunk Configuration, Station Group Scenario). The 5 functions can't be deleted or editable.
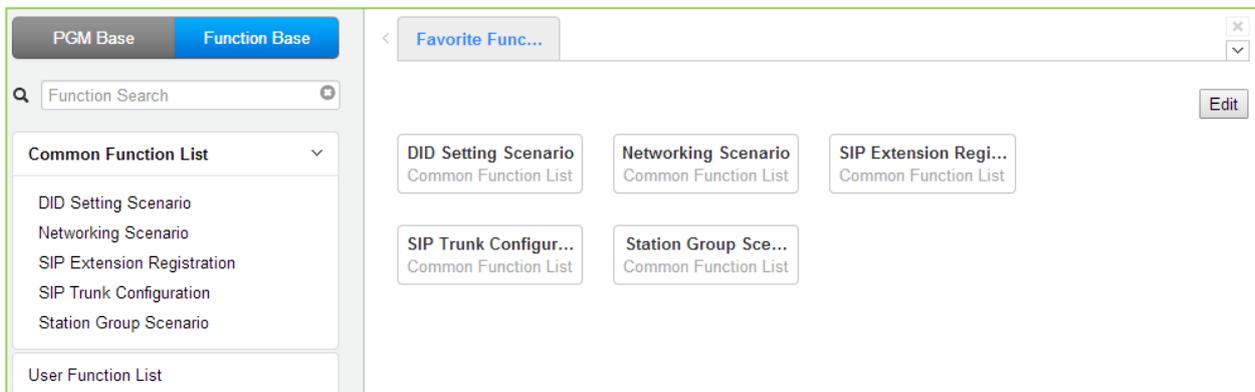
Figure 4.3.2-1 Common Function List

2) User Function List

To configure the user function list, click Maintenance button and then you can see the Function Program in the left frame and click the sub menu '*User Function Management*' as the following figure. On this web page, you can add or delete the function.
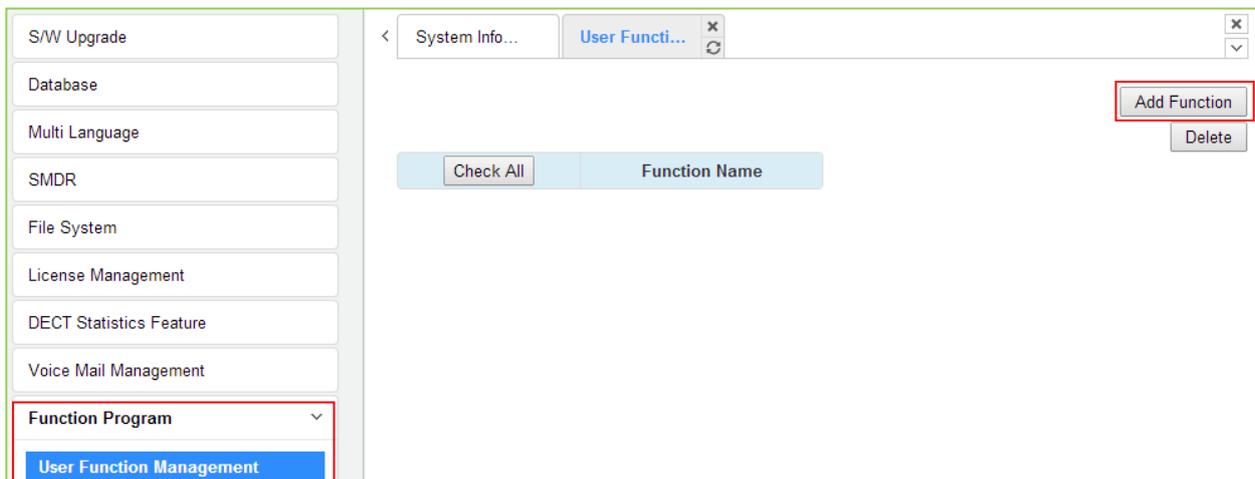
Figure 4.3.2-2 User Function Management

Adding Function

This step is divided into 5 steps and you can make the desired function each step. The following is the guide to make Name and add the function on each step:

1) To enter the function name, English, Numbering, Underscore (_) and Parentheses are available. Function name has to be filled out.

2) To enter the step name, English, Numbering, and Special letters except Double quotation marks are available.

3) The step name doesn't need to be filled out, but the function has to be configured each step.

4) To cancel or close this tab, click the close button (X) and pop up the blow;

1. Click the Add Function button.

2. On the below window, click the desired PGM in the left frame. First fill out the function name and step name. To configure the step 1, click the Make table to check the desired function and then click the Save button.

   ✓ Check All: check all functions

   ✓ Save: Save the checked functions

   ✓ Initialize: Initialize the checked functions

3. The selected functions are displayed and click the Save button after checking each functions. The rest steps are the same as the step 1.

4. Finally, click the Save Function to save and then click the OK button.

5. To check the function, click Function Base button in the left frame and User Function List and you will see the following figure.

   You can enable or disable the function by checking each function and then click the save after setting Value. Also move to each step by clicking the next button or Previous button.

Deleting Function

To delete the user function, click Maintenance at the top of window and then click *the Function Program -> User Function Management.* Check the desired function to delete and click the delete button.

4.3.3 iPECS Web Page Navigation

The Navigation frame appears in the left after login; refer to Figure 4.3-1. Selecting a Program group from either the Navigation pane or the Favorites list will display the selected Web entry page.

Each of the system's data entry Web pages includes a frame for data display and modification. To modify data, click in the data field, either a drop-down menu will appear for entry selection or a cursor will appear in the field and the user may type in the data required. Once all new data for a Web page has been entered, the **[Save]** button must be clicked to send the new page to the system and save the modified data.

In some cases, where mentioned, it may be necessary to reset the system. The system can be reset manually as described in the *iPECS UCP Hardware Description and Installation Manual* by selecting the Reset System button on the Initialization Web page or using the reset button on the UCP Module.

4.3.4 General Web Page Features

4.3.4.1 Web Page Range Entries

On many of the Web pages for Station, CO Line and Tables Data a range of station, lines or table indices must be entered to display the appropriate data entry page. In this case, a range of devices or indices can be selected by entering the lowest and highest device number separated by a dash. In addition, a comma can be used to enter non-sequential numbers. Note no space characters should be entered before or after the dash or comma. Note the data entry page displays the data for the lowest device or table index entered.

4.3.4.2 Table Check Boxes

Data entry pages that require a range entry, only display the data for the lowest device number or Table index entered in the range. To assure that only the appropriate data is changed for all entries in the range, a check box is located in front of each attribute. When the page is saved, only data for attributes with the box checked are saved for the range while data for unchecked attributes are not modified.

4.3.4.3 Sorting Displayed Data

Charts in the data pages typically allow the data to be sorted based on a given column in either ascending or descending order. In the column header, the sorting symbol displays to indicate the sorting function is available for the column.

5. WEB SETUP WIZARD

After the system is initialized, Install Wizard is displayed. The Wizard presents pages in sequence with parameters that should be verified or commonly need modification prior to operation of the system:

1) System Upgrade
2) Nation Code
3) System Time & Date
4) Station Number
5) Flexible Numbering Plan
6) CO Ring Assignment
7) License Upload
8) Maintenance ID & Password
9) IP Information

5.1 System Upgrade

Click the Select files button and then open the pop-up folder. Select the desired file to upload to the system's memory and click the **[Start]** button. The file is sent to the system's memory, saved and automatically loaded upon a system reset or restart.
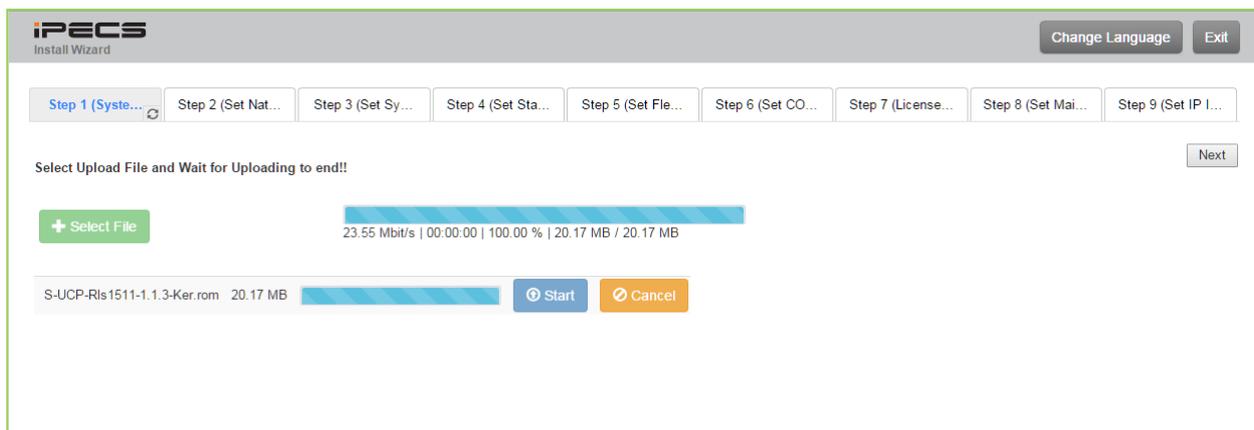
Figure 5.1 System upgrade

5.2 Nation Code

During initialization, the system employs the Nation Code to establish the default gains and tones for the various interfaces (analog CO Lines, ISDN lines, etc.), as well as the numbering plans for the specific country. The gains, in particular must be set to comply with the local regulatory requirements. Normally, the Nation Code will be set at the factory, however, assure the Nation code matches the system location.

Changing Nation Code

1) Dip Switch pole 4 of UCP100 (or UCP600, UCP2400) moves to ON.

2) Change Nation Code by clicking in the combo box.

3) Click Save button and then the system start resetting to apply.

In addition, all other data will be initialized, so the Nation Code should be properly set prior to other programming. You can change the numbering plan for your situation.
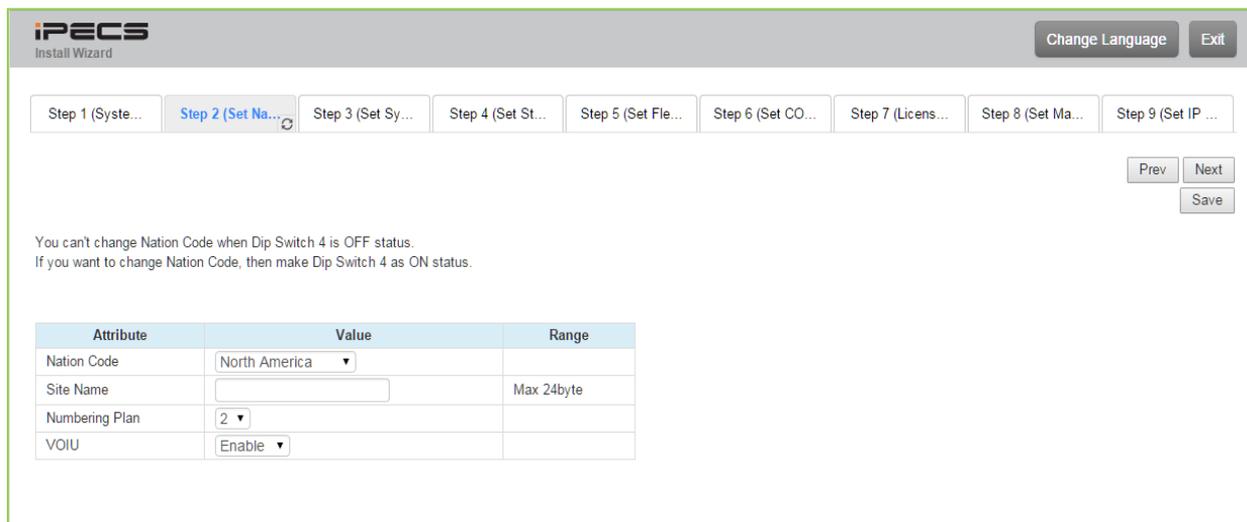
Figure 5.2 Nation code

5.3 Set System Date and Time

You can set System Date and time in this page.
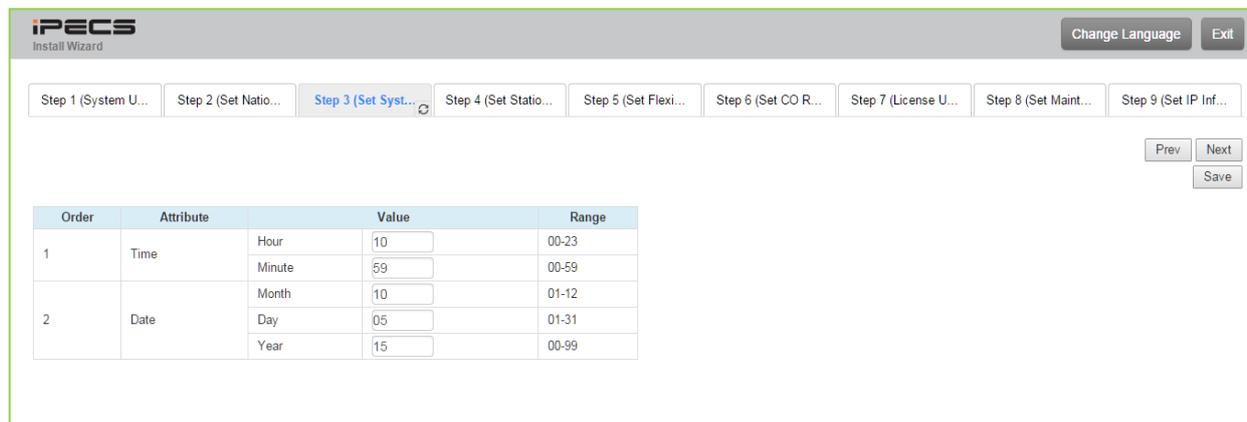
Figure 5.2 System Date and Time

5.4 Set Station Number

You want to change the current station number to New station number. In this page, you can change the station number.

Figure 5.4-1 Set Station Number

5.5 Set Flexible Numbering Plan

The Flexible Numbering Plan defines the various digit strings (codes) users may dial to access system resources (outside lines, page zones, etc.) and features. In the wizard, codes for features that commonly may require modification are displayed. For more detailed information, refer to the title "*Flexible Numbering Plan*".
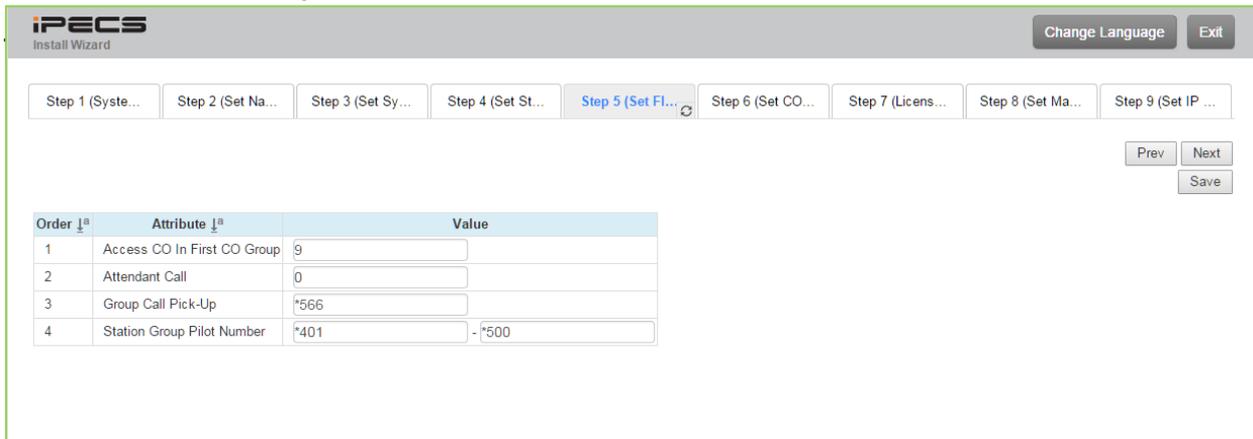
Figure 5.5 Set Flexible Number Plan

5.6 Set CO Ring Assignment

CO Ring Assignment establishes how the system will route incoming calls. The wizard assignments cover all CO/IP Lines in the system and the Ring assignments for Day, Night and Timed Ring modes can be established. For details on Ring assignments, refer to the title "*CO/IP Ring Assignment*".
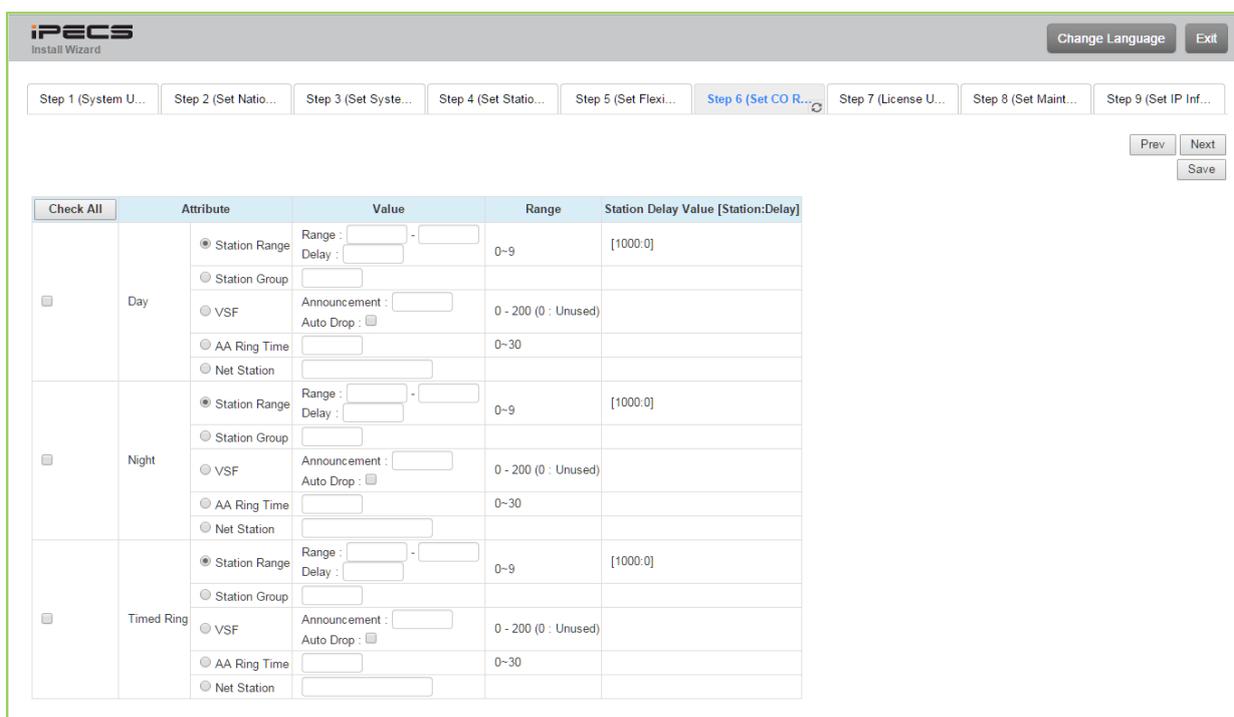
Figure 5.6 Set CO Ring assignment

5.7 Set License Upload

Before License upload, make sure that the date of system is set correctly. Click the Select files button and then open the pop-up folder. Select a valid license file to upload to the UCP system and click the **[Start]** button. If the file which is sent to the system is "System License File", it will be saved and automatically applied without restart. The enabled features by uploading license file can be shown in "System Overview" page.
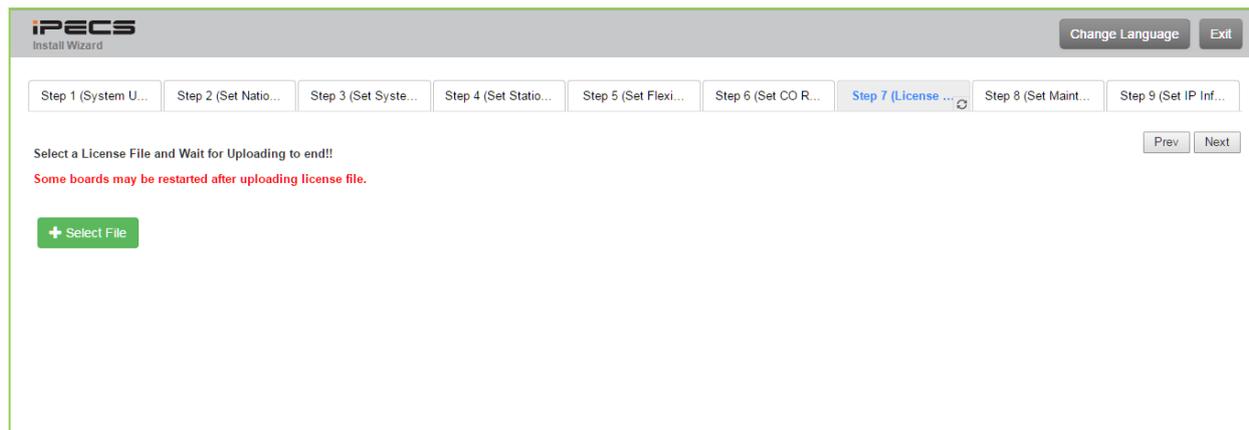
Figure 5.7 Set license upload

5.8 Set Maintenance Password

The Wizard includes a password entry page. **It is strongly recommended that a unique User ID and strong password be entered to minimize the risk of admin and maintenance access by unauthorized personnel.**

In order to finish the final step, you should register one or more than a maintenance ID. If not so, the Wizard can't go on to the next.

Also, Keyset admin password can be registered in this page.
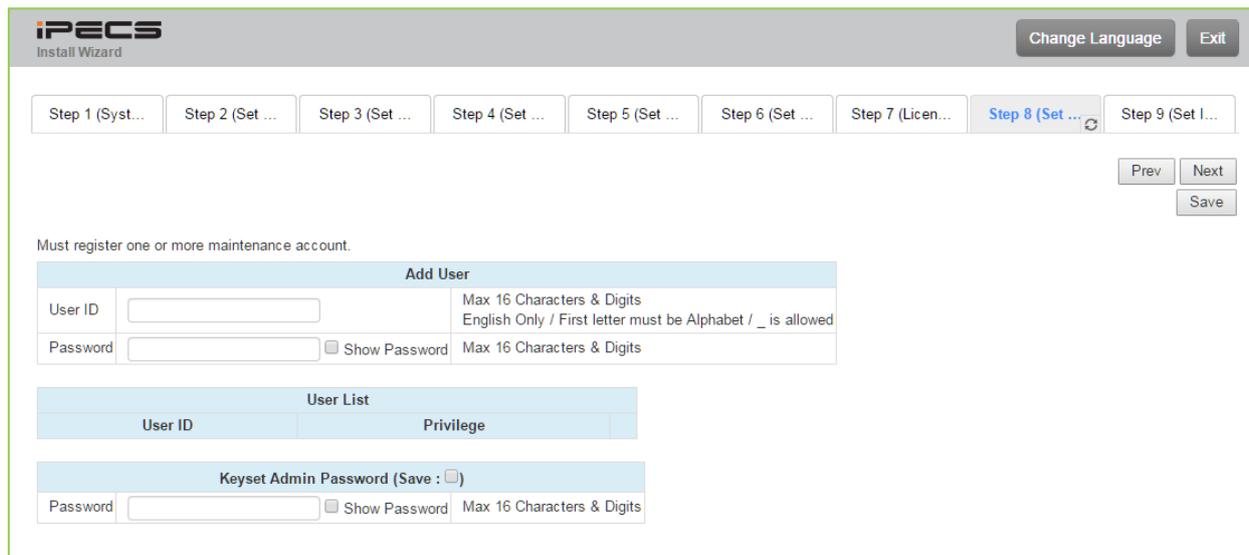
Figure 5.8 Set Maintenance Password
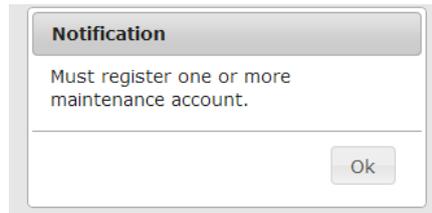
NOTE

The new information will be in effect immediately upon saving the information. When a new Admin User ID and Password are saved, the Web login screen appears. A new Admin session will be required using the new login credentials.

5.9 Set IP Information

The IP Information page establishes the UCP call server (UCP module) IP address scheme. DHCP can be enabled or static addressing can be configured. When the system is behind a firewall, the Firewall address must be entered allowing proper operation with remote users, devise and SIP trunks. A DNS (Directory Name Server) for Domain Name resolution should also be entered on this page. Note that changing any IP address in the system requires a system reset; the reset does not initialize data. For additional details, refer to the title "*System IP Plan*".
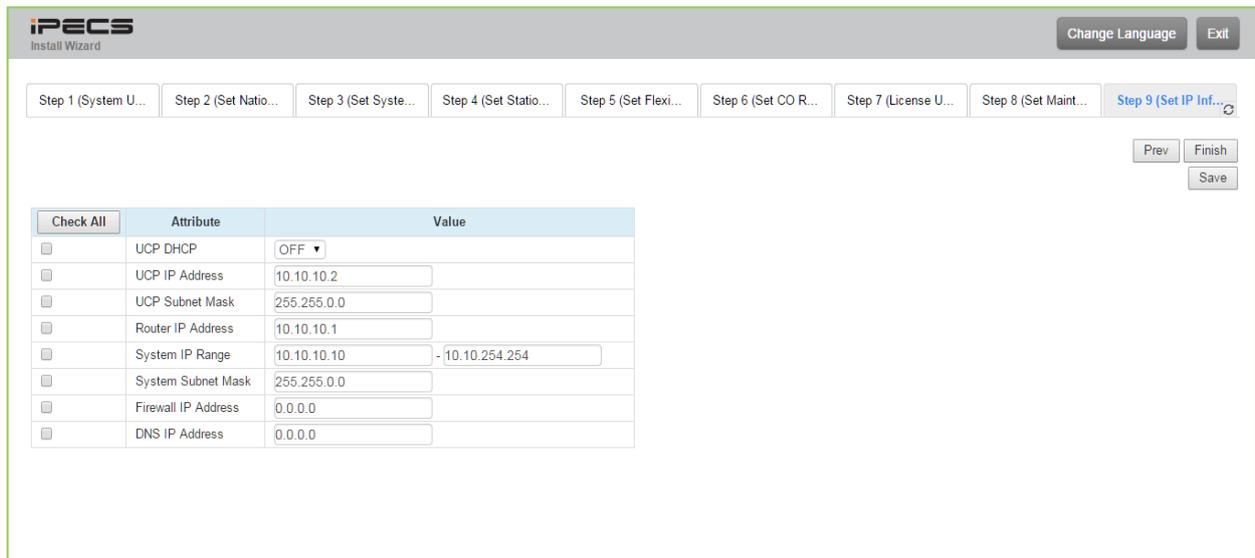
Figure 5.9 Set IP Information

6. WEB ADMINISTRATION

6.1 System ID & Numbering Plans

Selecting the System ID & Numbering Plans expands the Navigation frame to display the available Program groups as shown in the below figure.
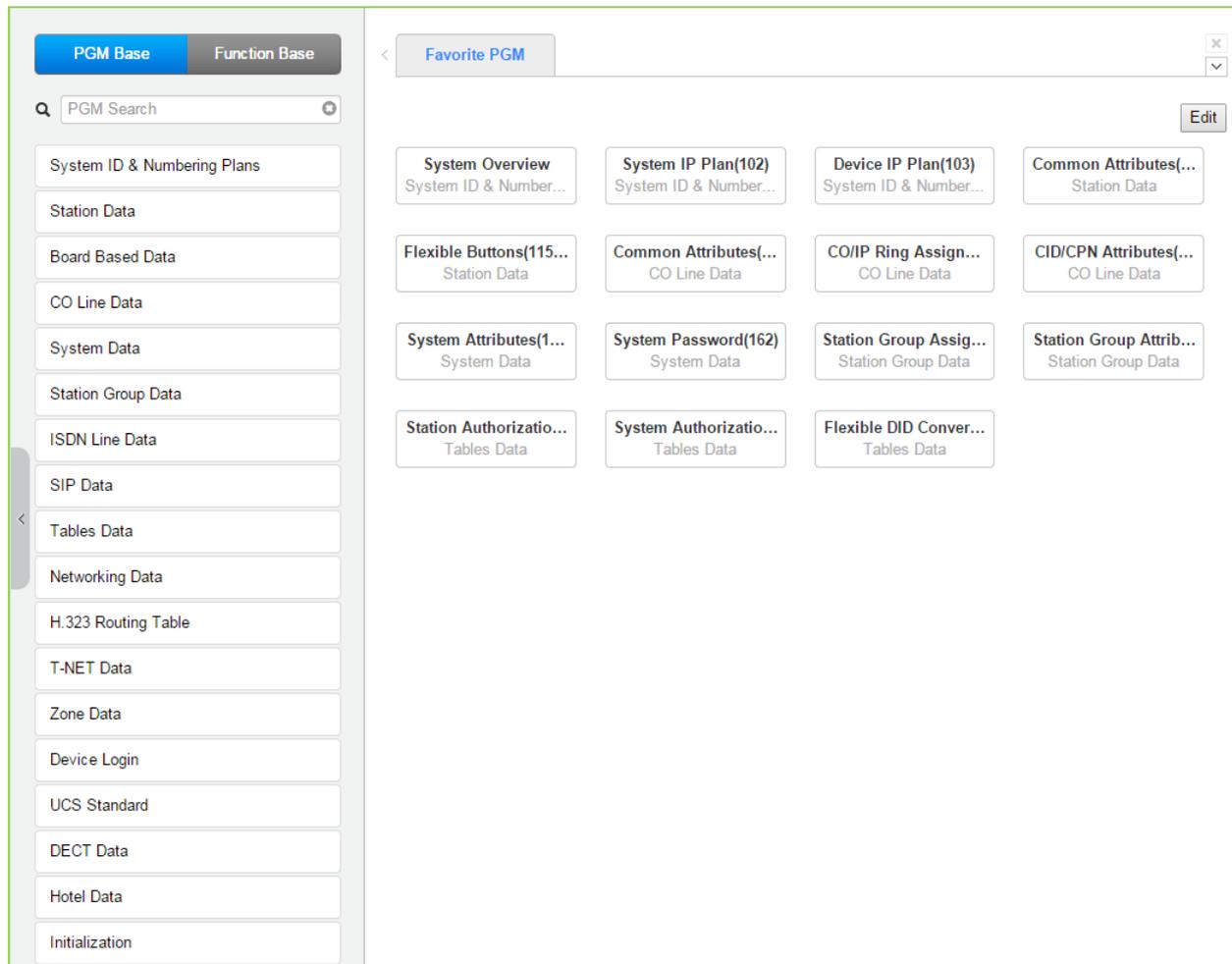
Figure 6.1-1 System ID & Numbering Plans sub-menu

6.1.1 System ID – PGM 100

Selecting System ID will display the following Input Entry page. Click **[Reset System]** to restart the system after changing Nation code, Site name, Site detail, My area code, Multi area code, Numbering Plan, Enable or Disable for VoIU.

Figure 6.1.1-1 System ID

Under System ID, the country is identified using the international dial codes (Nation Code). A twenty-four (24) character Site Name, Site detail and the local My Area Code maybe defined. This information is used to set gain, frequencies and other system characteristics specific to the country and regional regulatory requirements. The Site Name is primarily useful for the installer/programmer as a reference to customer.

In addition, the system employs one of the nine (9) basic Flexible Numbering Plans as detailed in Appendix B. Individual items from the selected Numbering Plan can be changed under "*Flexible Numbering Plan*".

The built-in VoIP DSP channels (VOIU) can be disabled allowing use of the DSPs for the Multi-

party Conference function. Note the capability to disable the VOIU is only available when dip switch 4 of the UCP module is in the ON position.

Note

To change the Nation Code or disable the VOIU, the UCP Module "Mode" dip switch pole 4 must be ON; you cannot change the Nation code while the switch is OFF.

Numbering Plan

The system employs one of the nine (9) basic Flexible Number Plans as detailed in Appendix B. Individual items from the selected Numbering Plan can be changed under Flexible Numbering Plan in section 6.1.8.

Note

The numbering plan on this manual is based on Numbering plan 2. The value (or feature code) may be different according to Numbering plan.

Speed Numbering

You can easily memory the system speed dial code and station dial code by selecting the desired type depending on the situation among the following types:

1) Type (0): System speed (20000~31999), Station speed (000~099)
2) Type (1): System speed (2000~9999), Station speed (000~099)
3) Type (2): System speed (2000~7999), Station speed (000~099)
4) Type (3): System speed (2000~4999), Station speed (000~099)
5) Type (4): System speed (200~999), Station speed (00~19)
6) Type (5): System speed (20~99), Station speed (00~19)
7) Type (6): System speed (10~99), Station speed (00~09)
8) Type (6): System speed (1~9), Station speed (0~0)

The notifications is displayed 'Speed Numbering type is successfully changed. Please refresh this page to reload Speed numbering data." According to selecting Speed numbering type, the range of System speed dial and Station speed dial is changed.

6.1.2 System Overview

Selecting System Overview will display the following System Overview page. This page shows the system capacity, the available applications, and Device ID list based on installed Module, Device, and the available license. Note that data can't be entered on this page.

Note

System overview may show different figure between the below and yours according to installing Module, Device, etc.

Figure 6.1.2-1 System Overview for UCP100

6.1.3 Device Port Number Change – PGM 101

Selecting Device Port Num Change (101) will display the following input entry page.

Figure 6.1.3-1 Device Port Num Change

The system supports changing port count of a device or deleting devices. In this window, the Device ID, Mac address, IP address and port count of the device can be modified. Also checking the **[Device Delete]** check box will delete the device.

Devices may be registered by clicking **[Virtual Registration]** button on "*Device Port Number Change*" page. Devices can be registered with or without a MAC address and the number of ports associated with the device may be limited. This may be useful for example to implement a "partial T1" gateway where only some of the T1 channels are available. Note to virtually register a SIP trunk, use the Device ID Type of CO and device as VOIM, enter the desired number of channels. If you want to use virtual MAC address when you register Device, please check the check box in front of MAC address input field. Click **[Register]** button after finishing the desired value.
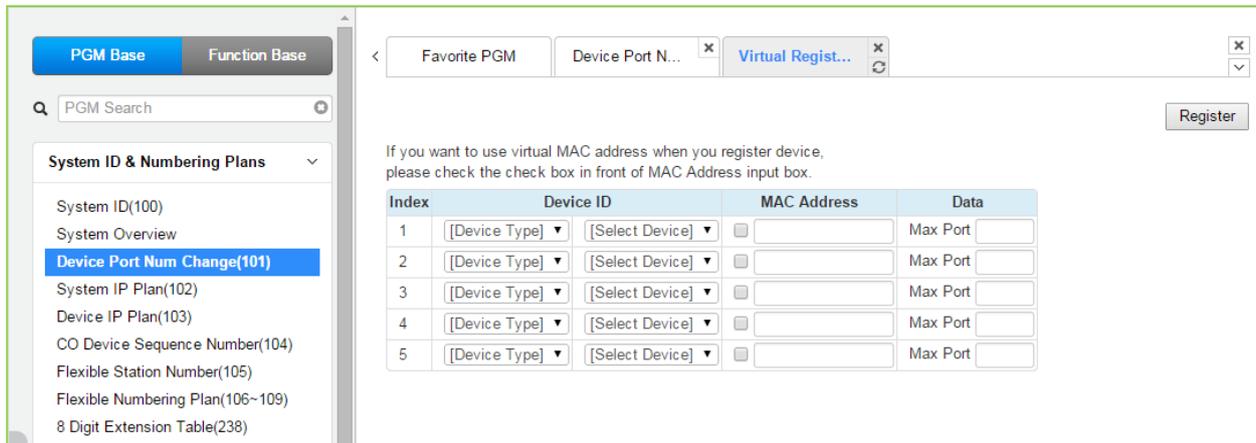
Figure 6.1.3-2 Virtual Registration

6.1.4 System IP Plan - PGM 102

Selecting System IP Plan will display the following input entry page. Use the check boxes to indicate which attributes to modify and the data for checked attributes is stored for the entire range of stations when saved.
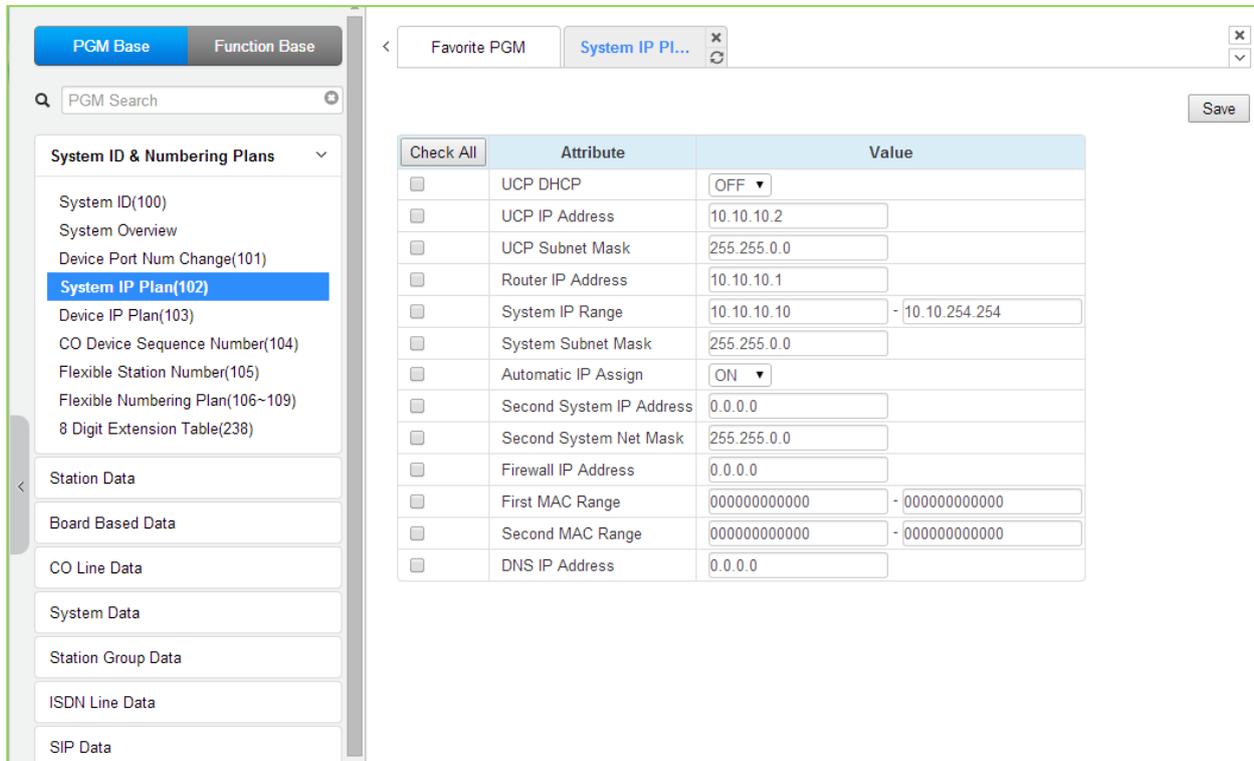
Figure 6.1.4-1 System IP Plan (PGM 102)

The System IP Address Plan sets several IP addresses including the UCP module IP address that is required for external VoIP calls, the IP address for the router, and the system's internal private IP address Plan. Note that the LAN and Router addresses must be routable IP addresses for access to an external VoIP network, remote access by an iPECS Phone or remote Web access. When used, the LAN port of any VOIM (Voice over IP device unit) must also have a routable IP address for access to/from an external VoIP network and a remote iPECS device.

When "Automatic IP Assign" is enabled, the system will assign IP addresses to each iPECS IP terminal and gateway Modules including any VOIM using the System IP address range defined. These addresses are used for communications between the system and other VOIMs and terminals.

The system may be connected to a LAN that is segmented by two separate private IP address schemes. This segmenting technique is often used to separate voice and data devices. However, with this segmenting technique, the system will normally treat the segmented devices such as IP softphones, as remote devices, using valuable WAN bandwidth. Assigning the "Second Sys IP address" with a valid IP address from the second segment permits the system to communicate directly with the devices over the LAN.

iPECS can be installed behind a NAPT server, if the NAPT server provides fixed address translation and port forwarding to the system. In this case, the system will employ the "Firewall IP address" as the fixed IP address for communication with remote devices. This address must be

assigned as the "UCP" address in the remote device.

Table 6.1.4-1 SYSTEM IP ADDRESS PLAN

ATTRIBUTE	DESCRIPTION	DEFAULT
UCP DHCP	Controls the DHCP client function for the UCP.	OFF
UCP IP Address	Public IP Address of the LAN port that is required for remote user and external VoIP network access. IPv4 format	10.10.10.2
UCP Subnet Mask	Defines the system subnet for UCP IP addresses.	255.255.0.0
Router IP Address	IP Address of router for external network (WAN) access. Required for shared voice and data LAN, external VoIP and remote Web access.	10.10.10.1
System IP Range	Range for private IP addresses of Modules/Terminals.	
System Subnet Mask	Define the system subnet for private IP addresses.	255.255.0.0
Automatic IP Assign	The system automatically assigns IP addresses to modules and terminals (ON) or, when OFF, IP addresses are assigned manually in Device IP Address Table or from the DHCP address assigned to the device.	ON
Second System IP Address	When devices have different address schemes on the same LAN, enter an IP address from the second LAN for use by the UCP system.	0.0.0.0
Second System Net Mask	Net mask of the second private IP addresses	255.255.0.0
Firewall IP Address	When the system is installed behind a NAPT server, the fixed IP Address provided by the NAPT server must be assigned here. Also, use this IP address to identify the UCP in remote devices.	0.0.0.0
First MAC Range	MAC Address Range to register a device regardless of the 3rd Dip Switch.	000000000000~ 000000000000
Second MAC Range	MAC Address Range to register a device regardless of the 3rd Dip Switch.	000000000000~ 000000000000
DNS IP Address	IP Address of Domain Name Server, which iPECS UCP will use to resolve a URL to an IP address. The DNS provides the resolution after receiving the URL from iPECS.	0.0.0.0

6.1.5 Device IP Plan - PGM 103

Selecting Device IP Plan will display the following input entry page.

Figure 6.1.5-1 Device IP Plan (PGM 103)

This page displays all of the devices registered to the iPECS UCP by grouping into the type of device (CO, STA, etc.).

If the station is assigned to Master or Slave by linked station, the bracket '[M] or [S]' will be displayed by the station number in PGM 101, 103, 105, 124 of Web page.

Selecting the SVC button for a device will toggle the service mode between in and out-of-service. The device type can be modified as well as the MAC and IP address. Using the SVC check box to place a device out of service, an errant device can be replaced without affecting the database. After placing the device out-of-service, it can be removed, replaced and the MAC address of the new device entered. The SVC check box is used to bring the new device into service employing the database from the replaced device.

The system may assign a default private IP address to each device. If desired, this program may be used to modify the assigned IP address for each Gateway/board and iPECS Phone.

The device mode, connected to a local or remote LAN, is displayed. Each Gateway/board and terminal can be assigned for "*ARP*". With ARP disabled, the system will employ the Ethernet MAC address of the device to send iPECS protocol messages to the device. This reduces the overall LAN traffic by eliminating ARP messages and the need for IP address headers in the messages.

The system normally employs IP multi-cast protocol to respond to a registration request from a Gateway/board or terminal.

When the device is separated from the system by a router, the system must use the IP uni-cast protocol. This is established by the "*Local Device*" assignment. When disabled (OFF), the system will send an IP uni-cast message to the device in response to a registration request.

6.1.6 CO Device Sequence Number - PGM 104

Selecting CO Device Sequence Number will display the following input entry page. Selecting the blue colored text in the Table header will sort the table based on the selected column. Click **[Save]** button after changing Value to apply.

Figure 6.1.6-1 CO Device Sequence Number

The system configures the CO/IP Line numbers. Each Module is assigned the starting CO/IP Line number based on the registration order. In place of the default Order Numbering, the "*Order*" number assigned in this PGM can be used to reorder the CO/IP numbering.

6.1.7 Flexible Station Number - PGM 105

Selecting Flexible Station Number will return the following data entry page. This page permits changes in the Station Numbering Plan using one of three methods:

Station Index Range: use to change the station numbers associated with a range of "Order Numbers" (the indexes). The "Start Station Number" is assigned to the station with the first index then the station number is incremented by one and assigned to the next station in the range. The process is repeated until the station number with the last index is changed.

Station Number: enter the station number to search. The station number is displayed and then you can change the new station number as you wish.

Multiple Station Number to change: there are two ways to search the station number for change; Enter Index Range or Enter Station Range.

You can change station numbers over a range of stations using the "Start Station Number" as the first station number for the range. The station number is incremented by one for each successive station in the range.
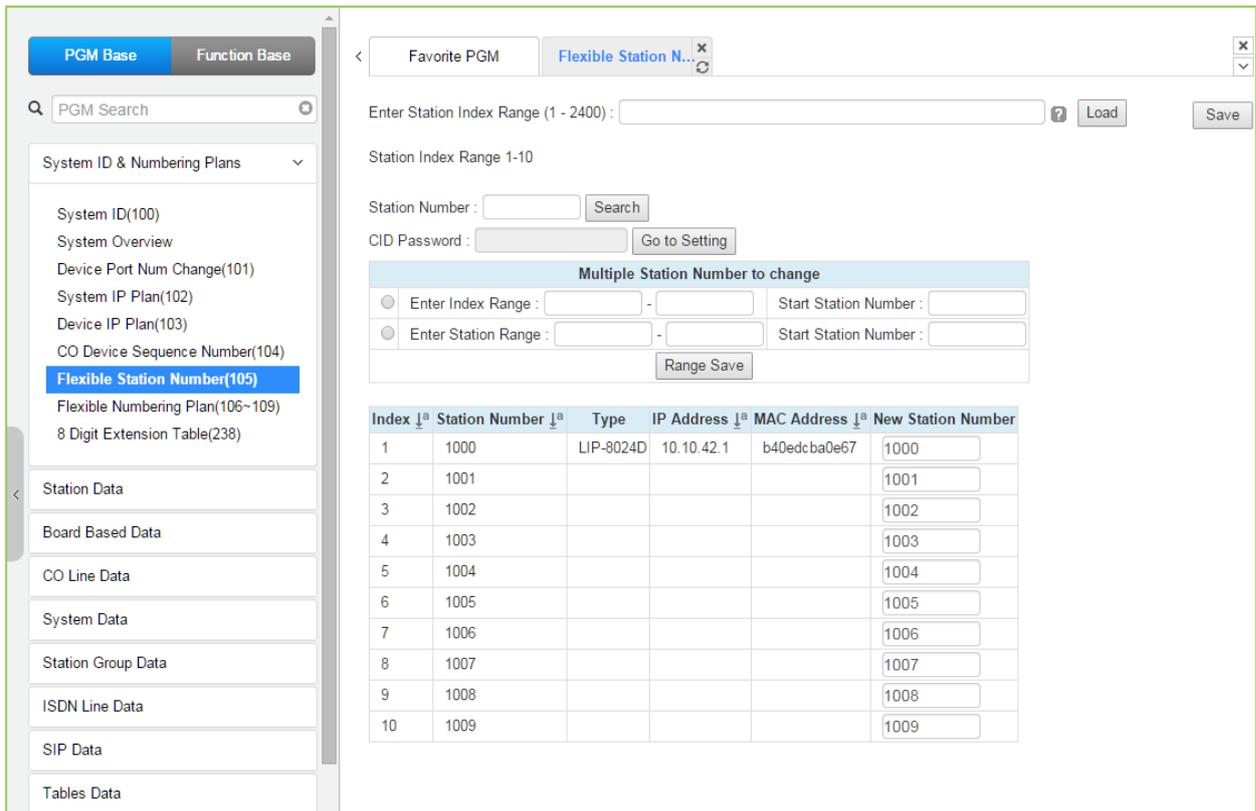
Figure 6.1.7-1 Flexible Station Number

The station Index number is incremented from 1 as each terminal device is registered. At registration, station numbers increment sequentially with the index and are assigned starting at station 1000. The Station Numbering Plan allows the station numbers to be two (2) to eight (8) digits in length. If the 8-Digit Table is employed, the full station number including the Prefix digits must be entered in this table.

Setting CID Password directly

You can set the CID password to click **[Go to Setting]** button. After clicking it, you will move to the following PGM 162 and set the CID password, and then save CID password to mark tick on the save box and click **[Save]** button.

6.1.8 Flexible Numbering Plan - PGM 106 ~ 109

Selecting Flexible Numbering Plan will display the following input entry page. Selecting the blue colored text in the Table header will sort the table based on the selected column. Click **[Save]** button after changing Value to apply.
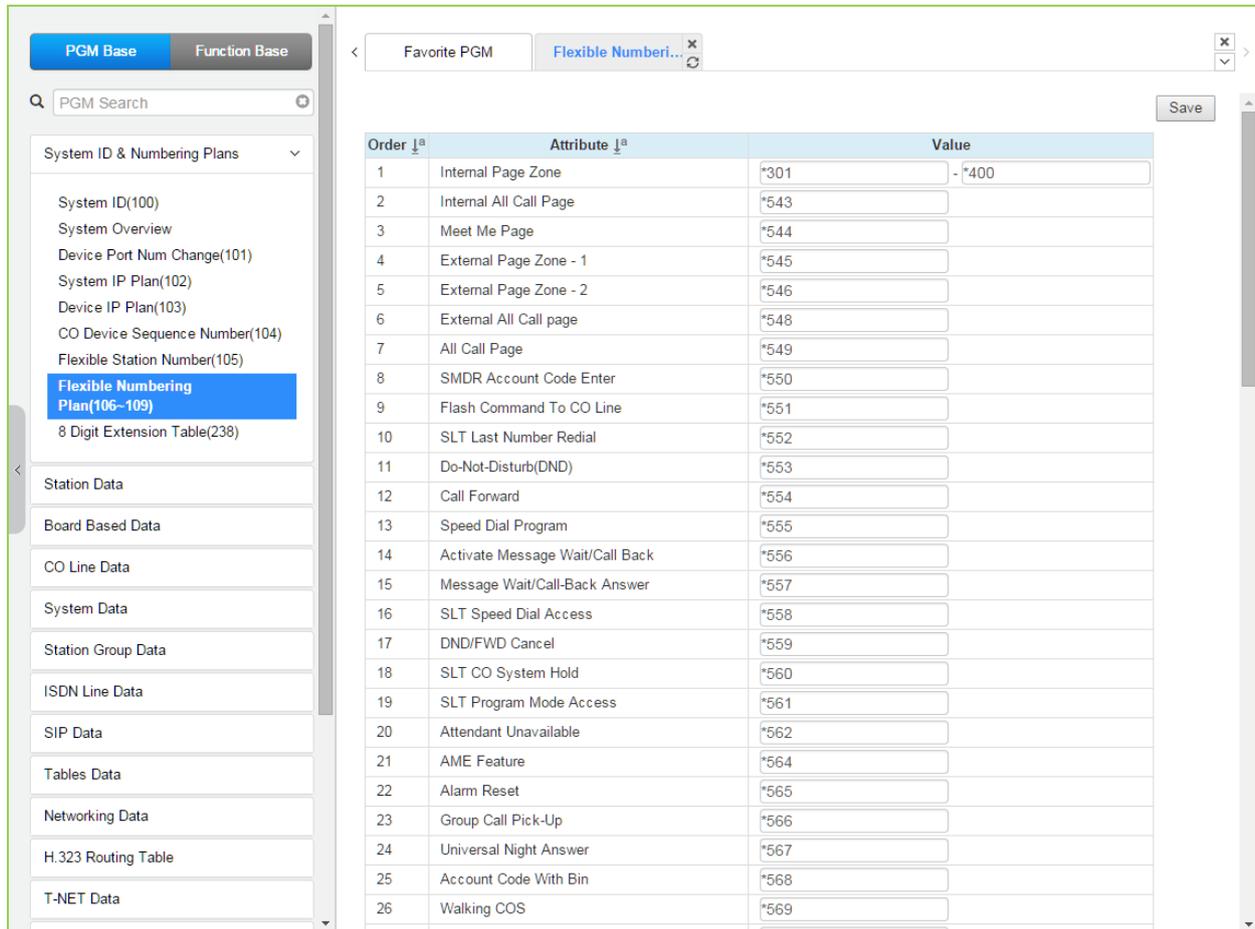
Figure 6.1.8-1 Flexible Number Plan

Feature dial codes for the system can be assigned using the system's Flexible Numbering Plan. Feature codes should be one (1) to four (4) digit in length and must not conflict with other codes or the Station numbering. For example, Feature codes 53 and 536 represent a conflict. The system will not update the database until correct data is entered. If the 8-Digit Table is employed, the information is saved even if a conflict with the Flex Numbering Plan exists. In this case, the Station Number will have priority.

The below table provides a brief description for each feature and the default codes as they appear in **Numbering Plan 2**. The default values for other numbering plans, which can be selected on the title "*System ID (100)*" and the other numbering plan is described in Appendix B.

Table 6.1.8-1 FLEXIBLE NUMBERING PLAN CODES

Order	ATTRIBUTE	DESCRIPTION	DEFAULT
1	Internal Page Zone	Internal Page Zone access codes.	*301~*400
2	Internal All Call Page	Internal All Call Page access code.	*543

Order	ATTRIBUTE	DESCRIPTION	DEFAULT
3	Meet Me Page	Meet-Me-Page answer code.	*544
4	External Page Zone – 1	External Page Zone 1 access code.	*545
5	External Page Zone – 2	External Page Zone 1 access code.	*546
6	External All Call Page	External All Call Page access code.	*548
7	All Call Page	All Call Page access code.	*549
8	SMDR Account Code Enter	Dial code to signify the start of an SMDR Account Code.	*550
9	Flash Command To CO Line	Dial code to generate a Flash on the active CO Line.	*551
10	SLT Last Number Redial	SLT Last number redial feature access code.	*552
11	Do-Not-Disturb (DND)	Dial code to activate Do-Not-Disturb.	*553
12	Call Forward	Code to activate Call Forward.	*554
13	Speed Dial Program	SLT Speed Dial programming access code.	*555
14	Activate Message Wait/Call Back	Code to activate Message Wait/Call Back.	*556
15	Message Wait/Call-Back Answer	Code to return Message Wait/Call Back.	*557
16	SLT Speed Dial Access	SLT Speed Dial access code.	*558
17	DND/FWD Cancel	Code to cancel DND/FWD/MSG Wait.	*559
18	SLT CO System Hold	Code to place a CO Line call on System Hold.	*560
19	SLT Program Mode Access	SLT user program access code.	*561
20	Attendant Unavailable	Code to make attendant "unavailable".	*562
21	AME Feature	Dial code to assign an Answering Machine Emulation Flex button.	*564
22	Alarm Reset	Code to terminate an Alarm signal.	*565
23	Group Call Pick-Up	Group Call Pick-up code.	*566
24	Universal Night Answer	Universal Night Answer code.	*567
25	Account Code With Bin	Dial code for entering an Account Code.	*568
26	Walking COS	Dial code to activate Walking Class-of-Service.	*569
27	ACD Agent ON/OFF Duty	Code to toggle ACD Agent ON and OFF duty.	*571
28	ACD Supervisor Login	Supervisor login code.	*572
29	ACD Supervisor Logout	Supervisor logout code.	*573
30	ACD Help Code	Agent & Supervisor code for Supervisor help.	*574
31	ACD Call In Queue Display	Dial code to display calls in queue.	*575
32	ACD Supervisor Status	Dial code to display group status.	*576
33	ACD Supervisor Monitor	Dial code to activate Supervisor monitor.	*577
34	ACD Reroute Queued Call Answer	Code to reroute call after answer.	*578
35	ACD Reroute Queued Call No Answer	Code to reroute call prior to answer.	*579
36	Camp-On Answer	Dial code to answer a Camped On call.	*621
37	Call Park Locations	Dial code to place/retrieve a call in a system Park Orbit.	#601~#800
38	Station Group Pilot Number	Station group pilot number.	*401~*500
39	Station User VSF Features Access	VSF feature access code.	66
40	Call Coverage Ring	Code for Call Coverage button.	*76
41	Direct Call Pick-Up	Dial code to activate Directed Call Pick-up.	*77
42	Access CO Group	Dial code to access a CO Line from a group.	89
43	Access Individual CO/IP	Dial code to access a specific CO/IP Line.	88
44	Access Held CO/IP	Dial code to access the last held CO/IP Line from Hold.	8*
45	Access Held Individual CO/IP	Dial code to access a specific CO/IP Line from Hold.	8#

Order	ATTRIBUTE	DESCRIPTION	DEFAULT
46	Access CO In First CO Group	Dial code to access the 1st available CO/IP Line in any accessible group.	9
47	Attendant Call	Dial code to call Main Attendant.	0
48	VM MSG Wait Enable	Dial code for external Voice mail to activate Message Wait indication.	*8
49	VM MSG Wait Cancel	Dial code for external Voice Mail to deactivate Message Wait indication.	*9
50	Door Open	Dial code to activate Door 1 contact.	#*1
51	Door Open	Dial code to activate Door 2 contact.	#*2
52	Door Open	Dial code to activate Door 3 contact.	#*3
53	Door Open	Dial code to activate Door 4 contact.	#*4
54	MCID Request	Dial code to activate Malicious Caller Id (Except USA version).	*0
55	Unsupervised Conference Time Extension Code	Dial code to extend unsupervised conference time.	##
56	PTT Group Logon/Logoff	Push-To-Talk group login and logout dial code. The station must have a PTT button for proper operation.	#0
57	ACD Agent Primary Login	ACD Agent Primary Login code.	*581
58	ACD Agent Primary Logout	ACD Agent Primary Logout code.	*582
59	ACD Agent Secondary Login	ACD Agent Secondary Login Code.	*583
60	ACD Agent Secondary Logout	ACD Agent Secondary Logout Code.	*584
61	Wrap-up End	ACD Agent Wrap-up end code.	*585
62	T-NET CM LOGIN/OUT	When Central Control networking (TNET) is employed, a station can be manually logged in or out of the Central system using this code.	*586
63	ENTER INTO CONF ROOM	Code for a station to enter a Conference Room.	*59
64	ENTER INTO CONF-GROUP	Code for a station to initiate a Conference Group.	*68
65	STATION ICR	Code for a station to activate ICR forward.	*587
66	PICK UP GROUP PICK-UP	Pick Up Group Call Pick-up dialing code.	*588
67	EMERGENCY PAGE	Code for emergency page.	*589
68	REMOTE MEX CONTROL	Code to control the mobile extension settings remotely.	*580
69	Agent ON/OFF Duty In ALL GRP	Code to change the state of the Agent ON/Off duty in all station groups.	*58*
70	SLT ACNR	Dial code for an SLT to activate ACNR (Auto Call Number Redial)	*58#
71	ACD Supervisor Ring Mode	Code to check and change ACD group Ring mode by ACD group supervisor.	*570
72	Company Directory Name	Code to check and record a user's Name greeting for the Company Directory feature.	*563
73	ISDN Supplementary HOLD	ISDN Supplementary HOLD Numbering Plan Code.	*57*
74	ISDN Supplementary Conference	ISDN Supplementary Conference Numbering Plan Code (Not supported).	*57#
75	Forced Channel Seize	Code to disconnect an existing call and seize the CO/IP Line or connect to the station.	*56*
76	Override DND/Forward	Dial code to override DND or Call Forward activated by a station.	*56#
77	Cancel call back	Code to cancel a Callback request.	
78	Transfer to VSF Number	While on a CO/IP Line call, this code may be used to	*55*

Order	ATTRIBUTE	DESCRIPTION	DEFAULT
		transfer a call to a valid system announcement, [Transfer] + [*55*] + valid system announcement (01-200). The outside party receives the system announcement and DISA service activates.	
79	CCR	This code is used with Digit Conversion to direct a call to a specific System Announcement. The conversion modifies the received digits to a four digit string "#2xx" where xx indicates the announcement number.	#2
80	Room Type Conf Group Join	After a UCS client activates a UCS Conference Group, other users may dial this code and the group number to enter the Conference.	5*0

6.1.9 8 Digit Extension Table - PGM 238

Selecting 8 Digit extension table will display the following input entry page. Click **[Save]** button after changing Value to apply.
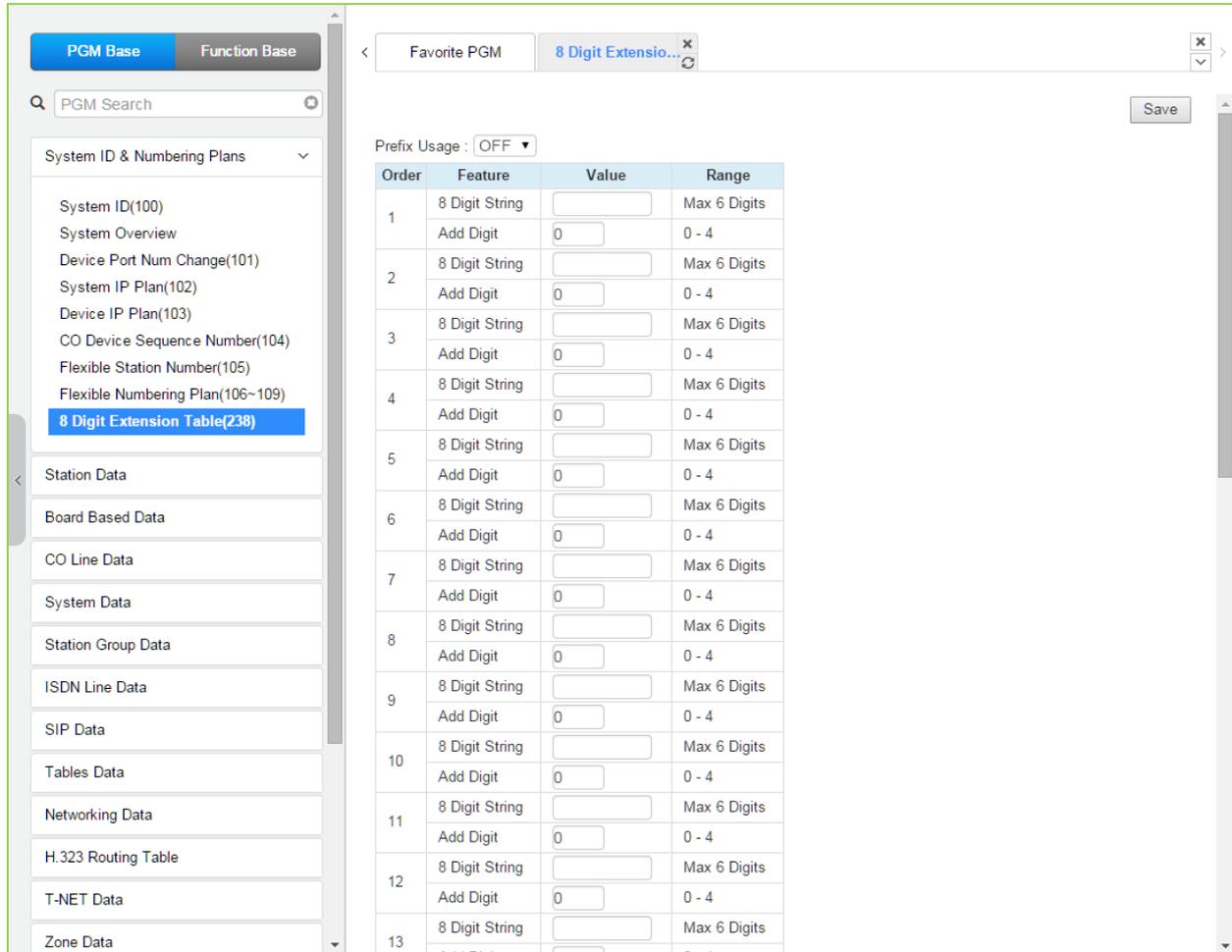
Figure 6.1.9-1 8 Digit Table

The iPECS UCP supports a Station Numbering Plan of up to eight digits. By combining a prefix digit string (8 Digit String) of up to six (6) digits with the Add Digit count (digit count from the Station Numbering assigned in PGM 105), Station Numbering can support up to eight digits. Note that multiple prefixes (8 Digit Strings) with varying ADD Digit counts can be assigned. In addition, in case of a conflict, the Prefix digit string will have priority over the Flexible Numbering Plan thus disabling the feature associated with the digit string but allowing the station to receive calls.

6.2 Station Data

Selecting the Station Data group will display the Station Data sub-menu displayed in the left frame as the below figure.
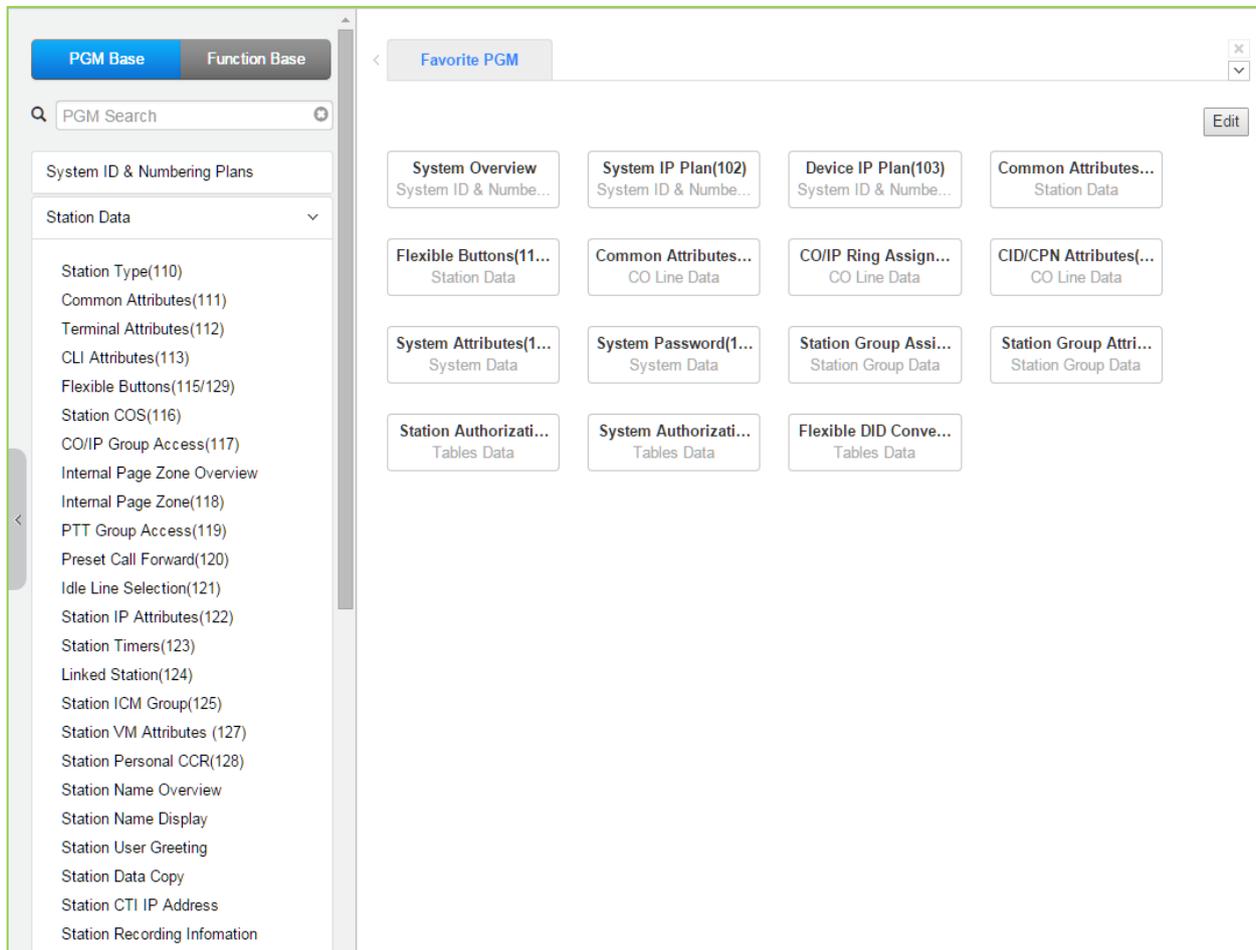
Figure 6.2-1 Station Data Main Page

6.2.1 Station Type - PGM 110

Selecting Station Type will display the Station Type data input entry page. Enter a valid station range and click **[Load]** to modify the Station Type data. Click **[Save]** button after changing Value to apply.
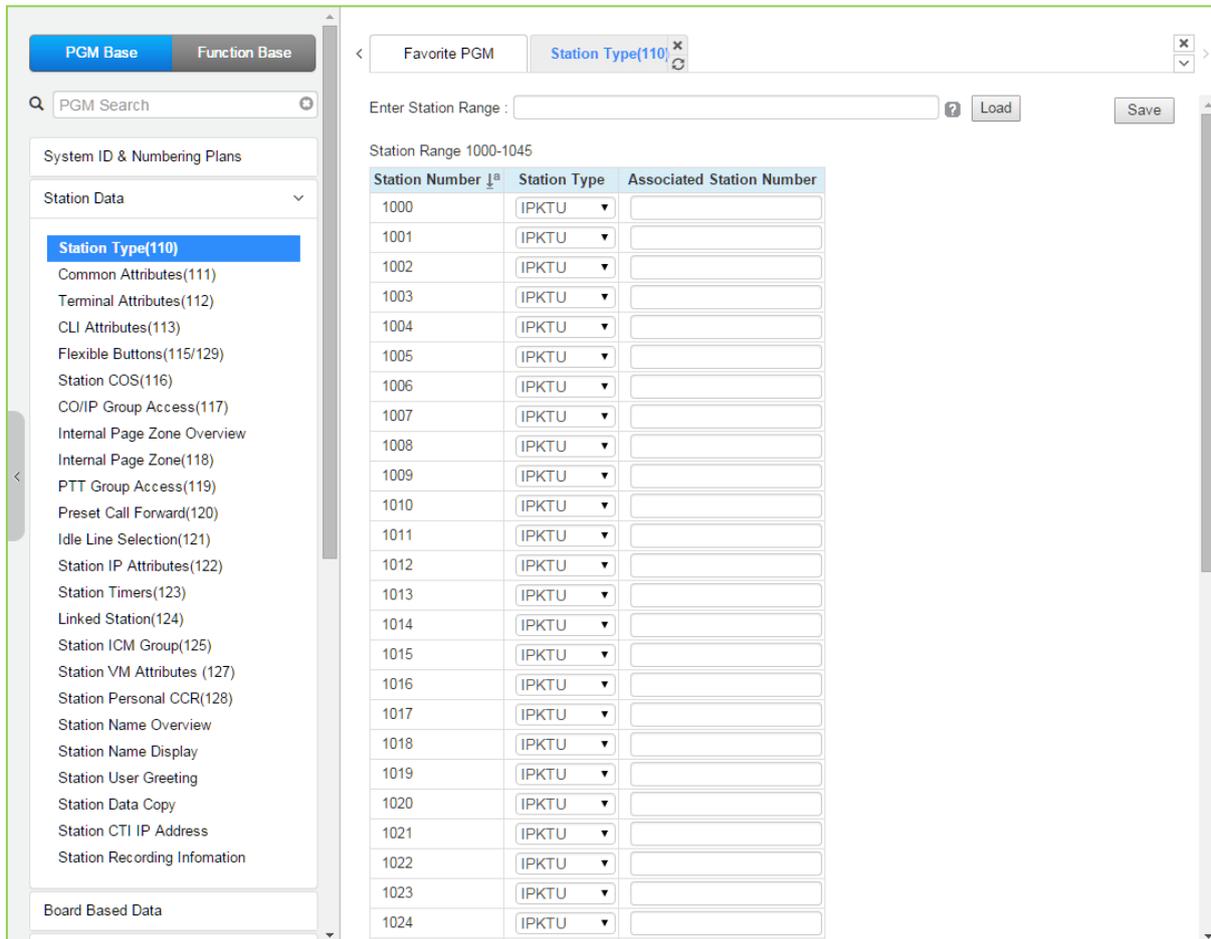
Figure 6.2.1-1 Station Type

Each station is assigned a type selected from the "*Station Type*" drop-down menu. The type is used by the system to recognize the station's capability and set default Flexible button configurations.

In addition, for standard iPECS DSS/DLS consoles the "*Associated Station Number*" is required so the system will recognize the station that is used with the console. Note this is not used with the Serial DSS/BLF Consoles.

In case of seleting SLT type, there are 6 types as below:

- SLT (DTMF)
- SLT (Pulse)
- SLT (DTMF VOL-MW)
- SLT (Pulse-MW)
- SLT (DTMF FSK-MW)
- SLT (DTMF POL-MW)

6.2.2 Common Attributes - PGM 111

Selecting Common Attributes will display the common attributes data input page. Enter a valid station range and click the **[Load]** button to enter Common Attributes data.

Use the check boxes to indicate which attributes to define; data for checked attributes is stored for the entire range of stations by clicking **[Save]** Button after changing Value.
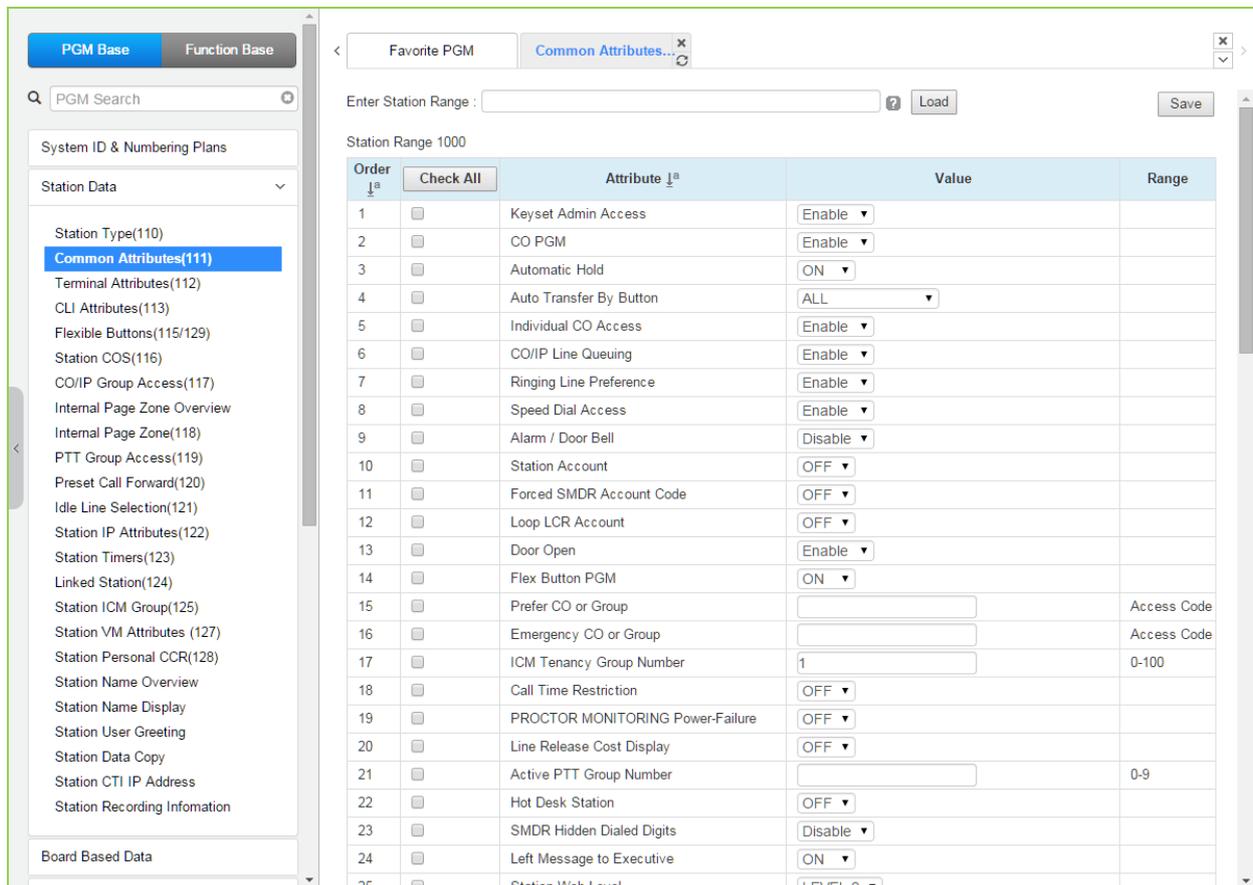
Figure 6.2.2-1 Common Attributes

Common Attributes define features and functions available to the station. Generally, the entry will turn the feature ON (enable) or OFF (disable). Refer to the following table for a description of the features and the input required.

Table 6.2.2-1 Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Keyset Admin Access	When enabled, the station can access the system Database.	Disable Enable	Enable
CO PGM	A station can be permitted to change the CO/IP Line numbers (ports) associated with a Flexible button.	Disable Enable	Disable
Automatic Hold	With Auto Hold enabled, the system will place an active external call on hold if the user presses a CO/IP Line or DSS button.	OFF ON	ATD: ON Others: OFF
Auto Transfer By Button	Transfer a talking call to a new making call by press Station (DSS)/CO (Loop)/U-Loop flexible button.	OFF ALL	ALL

Table 6.2.2-1 Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
		Except CO to CO	
Individual CO Access	If allowed, the stations may use dial codes to access individual CO/IP Lines.	Disable Enable	Enable
CO/IP Line Queuing	When enabled, the station may queue for the next available Line if All Lines Busy signal is received.	Disable Enable	Enable
Ringing Line Preference	When enabled, calls that ring the phone can be answered by going off-hook without selecting the CO/IP Line button.	Disable Enable	Enable
Speed Dial Access	The station is permitted for accessing to System Speed Dial bins when this parameter is enabled.	Disable Enable	Enable
Alarm/Door Bell	The station will receive Alarm/Door Bell signals if enabled.	Disable Enable	Disable
Station Account	When allowed (ON), the station user must enter an authorization code to access CO/IP Lines.	OFF ON	OFF
Forced SMDR Account Code	When allowed (ON), the user must enter an Account code to place an outgoing call.	OFF ON	OFF
Loop LCR Account	The Station may be required to enter a Station Authorization code to access LOOP LCR operation.	OFF ON	OFF
Door Open	This feature enables Station to use Door open feature.	Disable Enable	Enable
Flex Button PGM	The ability to assign Flexible buttons of the iPECS IP and LDP Phones can be controlled. When allowed here, the user may assign features to Flexible buttons but requires special assignments to configure CO/IP Line buttons.	OFF ON	ON
Prefer CO or Group	The System will seize this CO/IP Line or a Line from the CO/IP group number when the station dials "9" (First available CO access code).	CO/IP Line/ CO Group number	Access code
Emergency CO or Group	This feature defines the CO/IP Line or Group employed by the system to place Emergency Assistance calls.	CO # or CO Group #	Access code
ICM Tenancy Group Number	The station is assigned to one of the 100 ICM Tenancy Groups.	0~100	1
Call Time Restriction	All outgoing calls will disconnect at expiration of the Call Restrict Timer. The Cut-Off Timers can be set the time for Station and CO Line. For cut off timer of Station, refer to the title "*Station Timers (123)*". For CO line, refer to the title "*CO Line data: Common Attributes (144)*".	OFF ON	OFF
Proctor Monitoring Power-Failure	This feature enables use of PABX ANI Link device for E-911 support. This feature uses only for SLT port.	OFF ON	OFF
Line Release Cost Display	When a CO/IP line is released, the disconnect cause or call-cost is displayed in the LCD of the iPECS IP or LDP Phone.	OFF ON	OFF
Active PTT Group Number	A station can be assigned to a PTT (Push-to-Talk) group and the available group. So the station can place and receive PTT announcements for the group.	0~9	
Hot Desk Station	If Station can be assigned as a Hot Desk phone, Users and agents can login and use resources of the system through the Hot Desk phone.	OFF ON	OFF
SMDR Hidden Dialed Digits	If this feature enables, you can hide the dialed digits on SMDR output.	Disable Enable	Disable
Left Message to	When a call forwards to the Secretary of an Executive/Secretary	OFF	ON

Table 6.2.2-1 Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Executive	pair, messages can be left for the Executive (ON) or Secretary (OFF).	ON	
Station Web Level	Based on this level, the user will be able to view the specified programs within the Station User Web portal. Level 1 can view all programs, Levels 2 and 3 are configured in the title "*System data: Station Web Authorization*". This feature is configured only on Web PGM.	LEVEL 1 ~ LEVEL 3	LEVEL 2
Headset page mode	When a page announcement is received, it is normally received over the Speaker of the iPECS IP or LDP Phone. For headset users, the page can sent to the headset or both the headset and speaker.	Speaker Headset Both	Speaker
Progress Indication	When employing a non-ISDN terminal, specifically a modem or analog FAX, the ISDN call SETUP message must include this message and "Progress Indication" should be set to "ON".	OFF ON	OFF
3.1 KHz Audio	When an analog device (SLT or FAX) uses an ISDN Line in the system, the Information Element of the ISDN SETUP message must indicate the device only has 3.1 KHz audio capabilities. If an SLT or analog FAX will be allowed access to the ISDN Lines, this feature must be "ON"	OFF ON	OFF
Pick-Up by Flex button	When a station receives a call, the DSS/BLF button at other stations will flash and, if allowed, other stations may use the button to answer (pick-up) the call.	OFF ON	ON
Prepaid Call	It is useful feature for user to pay the money within the budget (Prepaid) for outgoing calls. For more information, refer to the below feature "*Prepaid money*".	OFF ON	OFF
Prepaid Money (0 - 999999) & Used Prepaid Money	An amount can be associated with the station as prepayment for outgoing calls. The assigned "Pre-paid Money" is reduced by the calculated call cost (Call Metering or cost/minute). This parameter displays the remaining funds for outgoing calls. Prepaid Money is often used in small hospitality businesses. The "Used Pre-paid Money" displays the money that has been used.	000000 – 999999 000000 - 999999	0 0
SIP USER TABLE INDEX	These fields reference the index to the SIP User ID Attributes Table. The referenced SIP User Id may be employed for several of the SIP headers for outgoing calls and may be employed for incoming call routing as configured in the SIP CO Attributes PGM 133, "Id Assigned Station". Up to three SIP User Id indices can be assigned to each station to permit the use of up to three SIP Trunk service providers.	0~2400	0
SIP USER TABLE INDEX2		0~2400	0
SIP USER TABLE INDEX3		0~2400	0
Station Web Language	User can select the desired language for Station Web.	English, Local	English
Lift Handset for Page	If an iPECS IP or LDP Phone user attempts to page using the speakerphone, pre-selection will be activated and the display shows 'Lift Handset for Page when Lift Handset for Page is ON". If Lift Handset for Page is OFF, then User can make page on speakerphone without lift handset.	OFF ON	ON
Privacy	If Privacy option is ON, then any person could not barge-in to the station and Attendant can't intrude to Station. Also, if Privacy option is ON, any person can't forcefully disconnect the station.	OFF ON	OFF

Table 6.2.2-1 Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Call Coverage Attributes			
Call Coverage Mode	The Call Coverage feature permits an iPECS IP or LDP Phone user to receive ring and answer calls for other stations.	OFF ON	OFF
Call Coverage Delay Ring	When a covered station rings, the **{CALL COVERAGE}** button LED will flash at the covering station and the station will receive ring (immediate or delayed by 0 to 15 ring cycles).	0~15	0
Call Coverage On Busy	Call Coverage can be configured to provide coverage while the covered station is idle (OFF) or both idle and busy (ON).	OFF ON	ON
Call Coverage Through Mobile Extension	Call Coverage can be configured to deliver coverage ring to the assigned Mobile phone (ON) when call coverage is active.	OFF ON	ON
Call Coverage On Busy Range	When Call Coverage Busy is enabled, the system can send coverage ring for outside calls only or both internal and outside calls.	External Call Only, External and Internal Call	External Call Only
Call Coverage Delay Ring Method	Call Coverage delayed ring can be determined by the covered (originator) or covering (member) settings for the delay.	by Originator, by Member	by Originator
Call Coverage For Wakeup Ring	Call Coverage can deliver coverage ring for Wake-Up signals.	OFF ON	OFF
Call Coverage Ring Type on Member	The station can be assigned to receive normal ring for a covered call or only a flashing of the Call Coverage button. For LIP phones, Muted ring or a muted single burst can be used to notify the station.	Silence, Normal Ring, Muted Ring-80xxE only, One Burst-80xxE only	Normal Ring
Tone/Ring Attributes			
Call Time Tone	A tone can be sent periodically with indicating the elapsed time of an outgoing CO/IP call. The Elapsed Call Timer determines the period between tones. For more information about Elapsed Call timer, refer to the title "*System data: System timer (180~182, 186)*".	OFF ON	OFF
Camp On Tone	When the Camp-on feature is enabled for the station, if the station is busy and receives a Camp-on request, the LCD indicates the camped on call. In addition, if Camp-on tone is enabled, the Camp-On tone is sent to the station as an audible signal for the camp-on.	OFF ON	ON
ICM Dial Tone Source	The dial tone sources can be selected for each station.	Dial Tone, Int/Ext Music1, Ext Music2, VSF MOH, SLT MOH 1~5, VSF MOH2, VSF MOH3	Dial Tone
ICM Ring Back Tone Source	The ring back tones can be selected for each station.	Ring Back Tone, Int/Ext	Ring Back Tone

Table 6.2.2-1 Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
		music1, Ext Music2, VSF MOH, SLT MOH 1~5, VSF MOH2, VSF MOH3	
Off-Hook Ring Type	While the station is Off hook, the type of ring can be adjusted as a single burst, muted normal ring, system ring or no audible ring signal.	Burst, Mute, System Attributes, Silence	Refer to System attributes
SIP Color Ring	With a SIP based AA/VM that supports color ring, color ring may be provided to SIP pones. The group number associated with the external AA/VM should be entered as the SIP Color Ring source.		Station number
Tone table index	Selects one of five Tables to determine the tones sent to the Station. The individual tones are configured in PGM 410 to 417 under the Maintenance.	1-5	1
Gain table index	Selects one of three Tables employ to set the gain from a device to other device types. PGM 410 to 417 under the Maintenance tab set the individual gains.	1-3	1
Digit conversion table	One of the 32 Digit Conversion Tables can be configured for use for this CO/IP Line.	1~32	
Routing Attributes			
Call Forward	When allowed, Call Forward can be activated by the station.	OFF ON	ON
DND	Enables DND to be activated by the station. The station can be limited to activate DND for outside calls (CO call only) or for internal calls (ICM call), if desired.	OFF, ALL, ICM call, CO call only	ALL
Off-Net Forward	A station must be allowed Off-Net Fwd to forward external incoming calls outside the system or otherwise establish a CO-to-CO connection (Unsupervised Conference).	Disable Enable	Enable
ACD Group Service	When unavailable, DID/DISA calls to the station can route to the ACD Group to which the station is a member.	OFF ON	OFF
Ring Group Service	When unavailable, DID/DISA calls to the station can route to the Ring Group to which the station is a member.	OFF ON	OFF
ACD Login Priority	ACD Group members may be assigned as a priority, 0-9. Members with the highest priority receive calls ahead of lower priority members.	0 ~ 9	0
Prime Line	This feature enables Delayed Prime Line (Idle Line) activation; see the title "*Idle Line Selection (121)*" and "*System data: System timer: Prime Line Delay timer (180~182 & 186)*".	HOT WARM	WARM
Auto ACD DND	If an Agent doesn't answer an ACD call in the ACD No Answer timer, the Agent enters an unavailable state with entering the Reason code. The reason code is sent in ACD Event messages.	None, #, *, 0~9	None
FWD if OOS	If a station is Out-of-Service and has previously forwarded calls, the system will forward the calls to the previous destination if this feature is ON.	OFF ON	OFF

Table 6.2.2-1 Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
LDT Table Index	LCR (Least Call Routing) operation for the station will use LDT Table index defined here from 0 to 32.	0~32	1
LDT Zone Number	If the LDT Zone Number of a LDT table (LCR LDT (221)) is equal to this value, the LDT table is available to the station.	1-100	1
Mobile (Web) Client Service	When enabled, the station can activate Call back from the Station Web portal.	Disable Enable	Disable
Click To Call Service	To use click call application, it should be enabled.	Disable Enable	Disable
MS Lync RCC Service	To use MS Lync RCC, it should be enabled.	Disable Enable	Disable
MSN Wait	Calls to an MSN number route to idle stations with a free MSN button for the MSN number. In addition, the call can be routed to busy stations if MSN Wait is ON and the station has a free (idle) corresponding MSN Loop button.	OFF ON	OFF
DID Restriction	A call routed using DID normally routes to the appropriate station. If desired, a Station can be restricted from receiving DID calls.	OFF ON	OFF
DISA Restriction	A call routed using DISA normally routes to the appropriate station. If desired, a Station can be restricted from receiving DISA calls.	OFF ON	OFF
Pre-Selected Msg DND	When the user activates Pre-Selected or Custom Messages, the system can automatically activate DND for the station so that the station will not receive ring and the call is routed by following DND treatment.	OFF ON	OFF
Voice Attributes			
Page Access	Stations must be allowed Page access to send a page over the system's Paging facilities.	OFF ON	ON
Forced Hands-Free Answer	When placing an intercom call, a user can change the ICM signaling mode, Tone Ring to Hands free answer mode or Hands free Answer to Tone Ring.	OFF ON	OFF
Group Listening	Audio is sent to both the handset and speaker with the handset microphone active and speakerphone microphone OFF.	Disable Enable	Enable
Override Privilege	Enables intrusion to gain access to an active CO/IP call.	Disable Enable	Disable
Voice Over	Enables use of Voice Over by the station.	Disable Enable	Enable
Barge In Mode	Barge-in permits an authorized station to intrude into other existing outside/internal calls or to force disconnection of an existing call.	Disable Monitor Only Monitor, Join & Disconnect	Disable
Camp on enable	Station can receive a Camp-on while busy. If 'Camp On Tone' is enabled, the stations receive Camp-on tone, otherwise only the LCD will indicate the camp on call. If the station is not allowed to receive a Camp-On, the calling user receives error tone.	OFF ON	ON
Video Show on Calling (ex. IP Video Door Phone)	When a VoIP video door phone rings to a video-enabled LIP phone, the video streaming commences immediately or after the call is answered. A VoIP channel is required to serve ring-back tone to the Door phone.	OFF ON	OFF

6.2.3 Terminal Attributes - PGM 112

Selecting Terminal Attributes will display the Terminal Attributes data input page. Enter a valid station range and click **[Load]** to enter Terminal Attributes data. Use the check boxes to indicate which attributes to define; data for checked attributes is stored for the entire range of stations by clicking **[Save]** Button after changing Value.
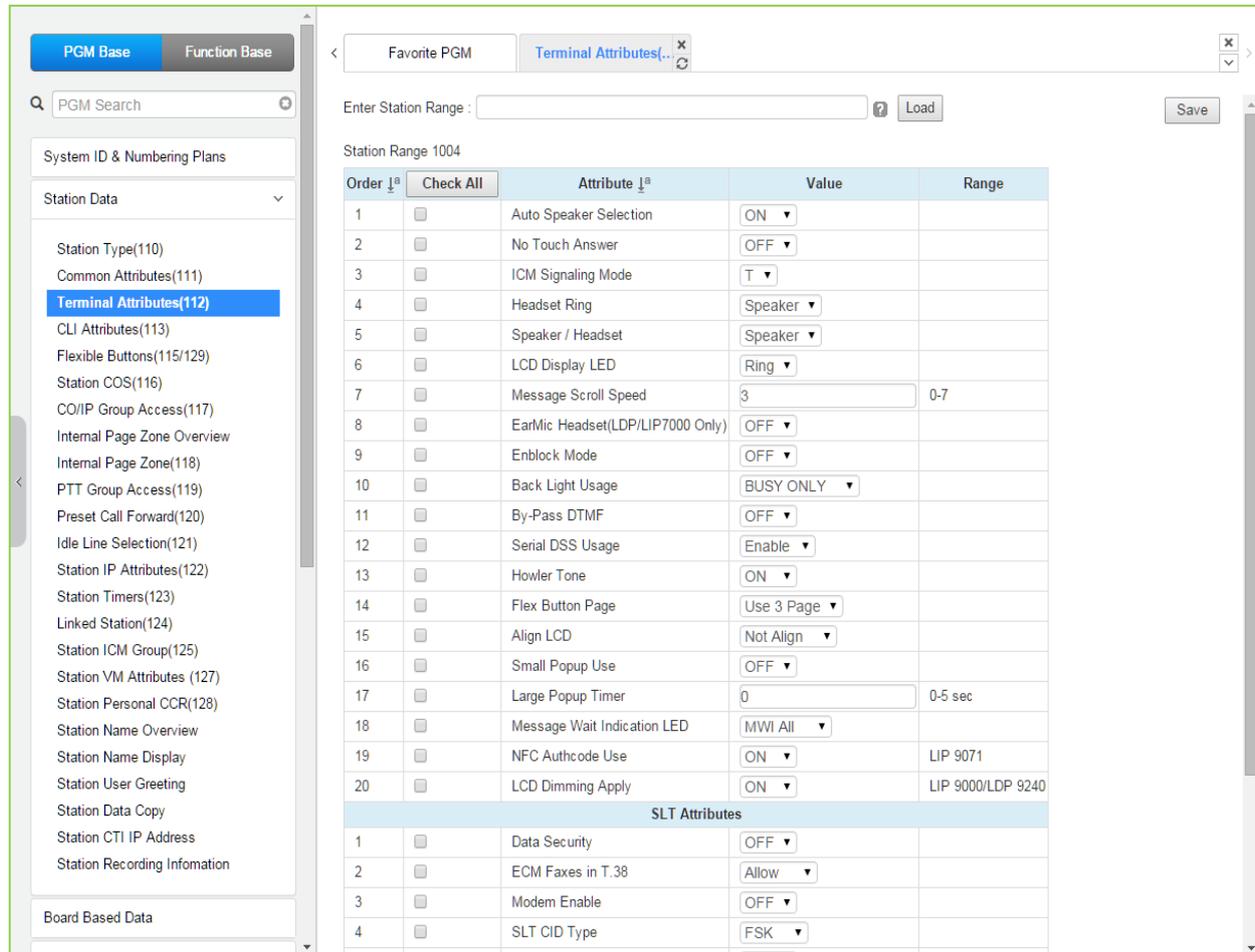
Figure 6.2.3-1 Terminal Attributes (PGM 112)

Terminal Attributes define features and functions available to the terminal itself. Generally, the entry will turn the feature ON (enable) or OFF (disable). Refer to the following table for a description of the features and the input required.

Table 6.2.3-1 Terminal Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Auto Speaker Selection	Enables **[SPEAKER]** activation automatically when a CO/IP Line, DSS or other feature button is pressed; there is no need to lift handset.	OFF ON	ON
No Touch Answer	When On, No-touch answer connects transferred calls to the station's speakerphone automatically after a short tone.	OFF ON	OFF
ICM Signaling Mode	The user can select ICM signaling mode among Hands-free (H), Privacy (P) or Tone Ring (T).	H/ T/ P	T

Table 6.2.3-1 Terminal Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Headset Ring	The user can select how to receive Incoming ring signals among Speaker, Headset or Both.	Speaker Headset Both	Speaker
Speaker/Headset	The user can select between Speaker and Headset for Call.	Headset/ Speaker	Speaker
LCD Display LED	The LED at the top of LCD is an indication for Intercom call ring or Message wait indication.	Ring MWI	MWI
Message Scroll Speed	The user can adjust the scroll speed on message from 0 to 7.	0 ~7	3
EarMic Headset	The user can use Ear Microphone Headset for Internal or external calls.	OFF ON	OFF
Enblock Mode	When ON, digits dialed by the user are stored at the iPECS IP or LDP Phone until explicitly sent by the user. When sent, all dialed digits are sent to the system in a block. En-block mode is only available to iPECS IP or LDP Phones with 3-Soft keys.	OFF ON	OFF
Back Light Usage	The backlight of iPECS IP and LDP Phones is assigned to be OFF, light only when the station is busy, or light constantly. This feature applies to LIP-8000, LIP 9000 and LDP 9000 series phones. In addition, Back Light Usage applies to LDP-7000 phones with a graphic LCD.	Always Off Busy Only Always On	Busy Only
By-Pass DTMF	When detected, DTMF from an SLT may be regenerated by CO/IP Line interface circuitry, the SLT port can bypass detection so DTMF is not detected.	OFF ON	OFF
Serial DSS Usage	Enables an LIP-8000/LIP-9000 phone to have a Serial DSS/BLF Console attached.	Disable Enable	Enable
Howler Tone	Howler tone can be sent to a phone when off-hook.	OFF ON	ON
Flex Button Page	iPECS LIP-9030 and 9040 have 8 and 12 Flexible buttons, respectively. Additional Flex buttons are available using Flex button pages. The phone can have up to 3 pages each with the 8 or 12 Flex buttons. Thus, an LIP-9030 can have 24 Flex buttons and the LIP-9040 can have 36 Flex buttons. The Navigation Up/Down button is used to scroll through the Flex buttons pages assigned.	Use 1 Page Use 2 Page Use 3 Page	Use 3 Page
Align LCD	For the iPECS LIP-9010/20/30/40, character alignment for messages to the phone can be right or left aligned by the system, or alignment controlled by the phone ("Not Align").	Not Align Align Left Align Right	Not Align
Small Popup Use	If this option is set to ON, small popup is displayed on the LCD of LIP-9030/9040. If it is set to OFF, top bar is displayed instead of small popup.	OFF ON	OFF
Large Popup Timer	When Large popup timer is set to any value from 1 to 5 except 0, the display of large popup is disappeared after the timer expired and then the large popup information is displayed at top bar.	0~5 (Sec.)	0
Message Wait Indication LED	User can program MWI (Message Wait Indication) LED according to the following type: 1. MWI All: LED is blinking about all calls and Message 2. VM MWI: blinking in case of getting Voice mail 3. CLI MWI: blinking in case of Calling Line Identification	MWI All, VM MWI, CLI MWI,: SMS MWI, ICM MW	MWI ALL

Table 6.2.3-1 Terminal Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	4. SMS MWI: blinking in case of getting Short message 5. ICM MWI: blinking in case of Incomming call		
NFC Authorization Code Use	If this option is ON, User enters Authorization code to use NFC function for LIP-9071.	OFF ON	ON
LCD Dimming Apply	If this option is ON, LCD dimming is available only for LIP-9000 Series and LDP-9240D. LCD is dark when the phone is paused so some light is needed. So we provide the dimmable lights.	OFF ON	OFF
SLT Attributes			
Data Security	Disables override and camp-on tones to the station. This feature is commonly used for an analog modem or FAX to assure tones do not affect received information.	OFF ON	OFF
ECM Faxes in T.38	ECM stands for Error Correction Mode. If you failed to send something via Faxes, you can send it again until it is successful.	Allow, Prevent	Allow
Modem Enable	When an SLT port is connected to a Modem, the port can be enabled for Modem operation. This will activate Echo Cancellation and disregard any Camp-on/Call Wait to improve modem performance.	OFF ON	OFF
SLT CID Type	Caller ID can be sent to an SLT as FSK or DTMF signals.	FSK, DTMF	FSK
Send SLT CLI Info	When allowed, the system sends CLI (Calling Line Identification) information to SLT.	OFF ON	ON

SLT Flash Mode	When an SLT activates a Hook-flash, the system will perform one of the following operations:
	    Flash Transfer – active call placed on hold and dial tone is returned.
	    Flash-Drop – the active call is dropped.
	    Flash-Ignore – the Hook-flash is ignored, no action is taken.
	    Hold Release – the active call is placed on hold and, if the SLT returns to idle, the call is dropped.
		Flash Transfer
		Flash Drop
		Flash Ignore
		Hold Release
			Flash Transfer

SLT configuration mode	For the South African region, the SLT gain is adjusted based on the SLT Configuration Mode.	Default, Short, Long, Far	Default
Block Back Call	When an SLT attempts to transfer a CO/IP call to another CO/IP Line, the transfer can be blocked and the call released.	OFF ON	OFF
SLT Open Loop Time	Send open loop signal to SLT port when counter party user hangs up SLT.	0~9 (100ms)	0
Short Modem	If this value of a SLT is ON, the SLT is the modem mode in seizing a CO line. When the CO line is LGCM, the {short modem timer} is starting when the SLT seizes the co line. And if the {short modem timer} is expired, the SLT goes to the original mode. When the CO line is ISDN, the {short modem timer} is starting after receiving the ISDN connect message. And if the {short modem timer} is expired, the SLT goes to the original mode.	OFF ON	OFF

6.2.4 CLI Attributes - PGM 113

Selecting CLI Attributes will display the CLI Attributes data input page. Enter a valid station range and click **[Load]** to enter CLI Attributes data. Use the check boxes to indicate which attributes to define; data for checked attributes is stored for the entire range of stations by clicking **[Save]** Button after changing Value.
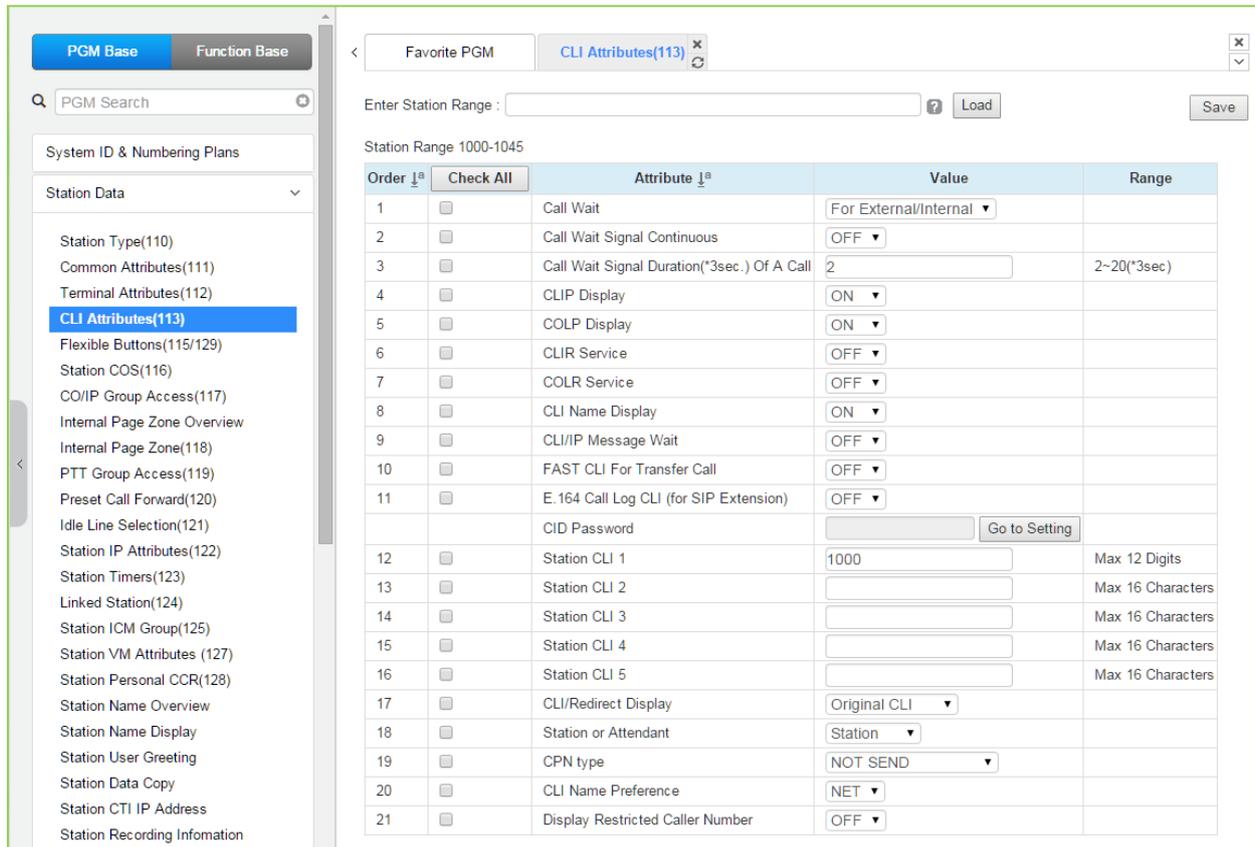
Figure 6.2.4-1 CLI Attributes (PGM 113)

CLI Attributes define Called and Calling Line ID characteristics for the station. Generally, the entry will turn the feature ON (enable) or OFF (disable) or enter is made in the text box. Refer to the following table for a description of the features and the input required.

Setting CID Password directly

You can set the CID password to click **[Go to Setting]** button. After clicking it, you will move to the following PGM 162 and set the CID password, and then save CID password to mark tick on the save box and click **[Save]** button.

Table 6.2.4-1 CLI Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Call Wait	When a busy station receives a call, the call may queue to the station instead of receiving busy treatment. With Call Wait, the caller will hear Ring-back tone and the CO/IP Line LED flashes. Also, the CLI for the new incoming call displays.	OFF, For External/Internal, For External, For Internal	For External/Internal

Table 6.2.4-1 CLI Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Call wait signal continuous	When this filed set On, you will get the indication of Call wait signal continuously without returning to the current conversation. In case of Off, you will return to the current conversation after 1 cycle of Call wait signal indication.	ON, OFF	OFF
Call wait signal duration (*3sec.) of a call	You can set the call wait signal duration of a waiting call: the default is 2 (2*3sec.) and the range is from 2 to 20.	2-20 (*3sec)	2
CLIP Display	CLIP (Calling Line Identification Presentation), a carrier service, sends the number of the calling party to the system in the Call SETUP message. If enabled here, the number will be shown in the iPECS LIP and LDP Phone LCD.	OFF ON	ON
COLP Display	COLP (Connected Line ID Presentation), a carrier service, sends the number of the answering party to the system in the Call CONNECT message. If enabled here, the number will be shown in the iPECS LIP and LDP Phone LCD.	OFF ON	ON
CLIR Service	CLIR (Calling Line Identification Restriction), a carrier service, removes calling party ID sent from the ISDN to the called party with a RESTRICT instruction in the SETUP message. If enabled here, the system will send the RESTRICT instruction when an outgoing ISDN call is placed.	OFF ON	OFF
COLR Service	COLR (Connected Line ID Restriction), a carrier service, removes connected party ID sent from ISDN messages to the calling party with a RESTRICT instruction in the CONNECT message. If enabled here, the system will send the restrict instruction when the station answers an ISDN call.	OFF ON	ON
CLI Name Display	When the CLI data in the call SETUP message from the carrier matches a number in Speed Dial, the system can display the name associated with the Speed Dial bin, if set to ON.	OFF ON	OFF
CLI/IP Message Wait	A log of missed calls with Caller ID can be maintained for the user, permitting the user to call back the identified party. Up to 4000 entries are maintained in the log, system-wide.	OFF ON	OFF
FAST CLI For Transfer Call	If this feature is set to ON, when a transferred call is routed to an SLT or DECT phone, the CLI for the CO/IP call is sent to the SLT or DECT phone instead of transferring station number.	OFF ON	OFF
E.164 Call Log CLI (for SIP Extension)	When enabled, the E.164 format CLI is sent to SIP Extensions for an incoming CO/IP call.	OFF ON	OFF
Station CLI 1	When not restricted by CLIR or COLR, this entry is added to the number in the COLP/CLIP Table defined for the CO/IP Line under the CO Line Data CID/CPN Attributes and used as the caller or answering party id. The specific Station CLI (1-5) is determined as the Station CLI Type assigned in the CID/CPN Attributes for the CO/IP Line.	12 digits	1000
Station CLI 2		16 characters	
Station CLI 3		16 characters	
Station CLI 4		16 characters	
Station CLI 5		16 characters	

Table 6.2.4-1 CLI Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CLI/Redirect Display	When an incoming call is redirected in the carrier's network, the call SETUP message will contain an original and redirected CLI. This selection determines if the iPECS IP and LDP Phone will display the original or redirected number.	Original CLI Redirect CLI	Original CLI
Station or Attendant	The Station CLI added to the CLIP or COLP can be, the CLI associated with the Attendant or the station.	Station Attendant	Station
CPN Type	NOT SEND: CPN is not sent to S0 interface. STATION NUMBER: Station number is sent as CPN to S0 interface. BYPASS FROM NET: Incoming CPN is sent as CPN to S0 interface.	Not send, Station number, By pass from NET	Not send
CLI Name Preference	NET: If CLI name is provided from network, then it is displayed. If CLI name is not provided from network, matched Speed name is displayed. SPD: If CLI is matched with Speed bin and it has name, then matched Speed name is displayed. If CLI is not matched or matched Speed name is not configured, CLI name from network is displayed.	NET, SPD	NET
Display Restricted Caller Number	When {Display Restricted Caller Number} in CLI Attributes(113) of a station is ON, Although the caller number is restricted in ISDN message, the ringing station displays the caller number.	OFF ON	OFF

6.2.5 Flexible Buttons - PGM 115/129

Selecting Flexible Buttons will display the Flexible buttons data input page. Enter a valid Station range and click **[Load]** to enter Flexible button data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of stations when saved.
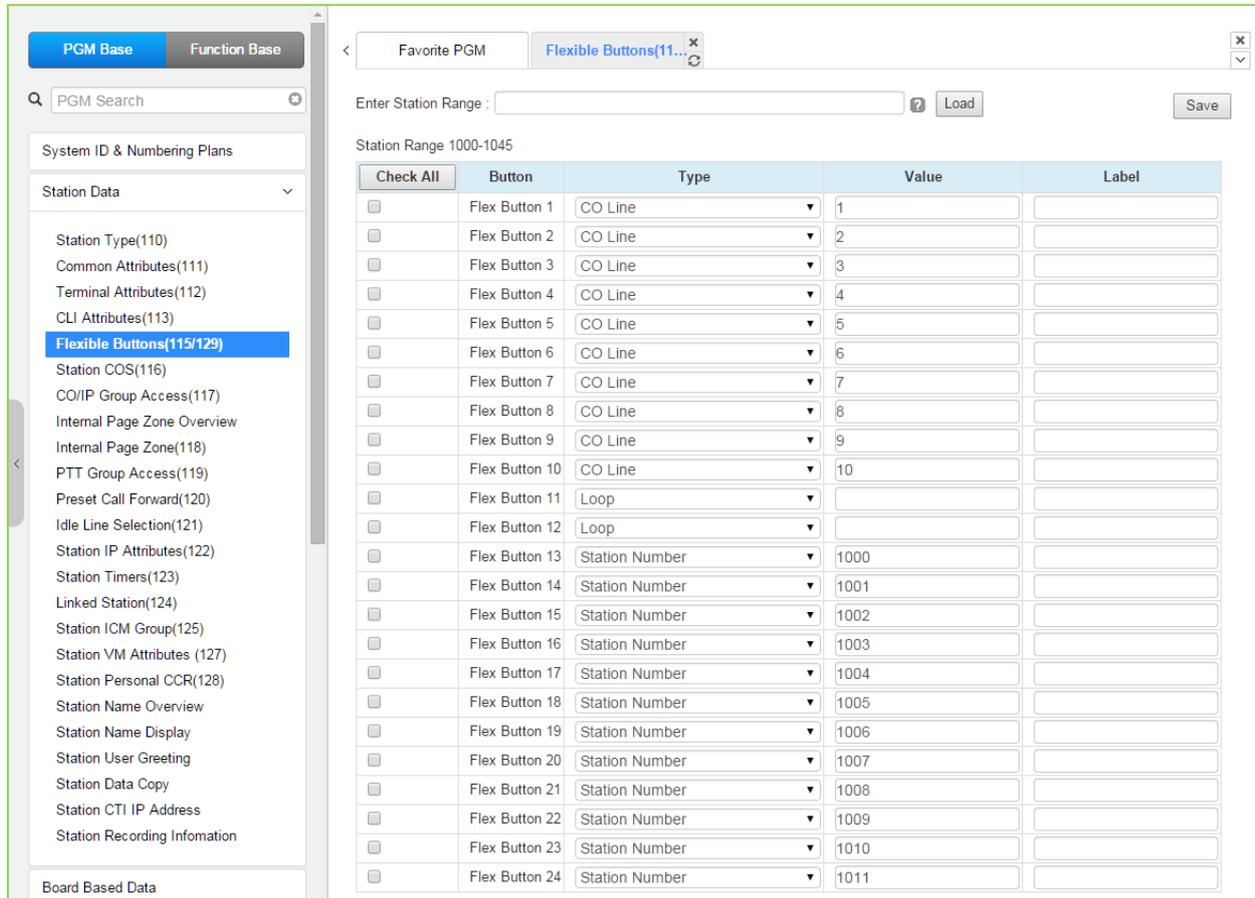
Figure 6.2.5-1 Flexible Buttons Assignment

Each Flex button for each iPECS IP and LDP Phones, and DSS Console can be assigned a function (TYPE) as below. After selecting the Type for a button, enter the value, if required. The types available from the drop-down menu are shown in Table below. In addition, for the LIP Phone models 8040, 8050, and 9000 series and the LSS Console models, a label can be assigned that is used as the designation for the button in the LCD of the phone.

Table 6.2.5-1 FLEXIBLE BUTTON TYPE & VALUE

TYPE	REMARK
N/A	Empty (unassigned), may be defined by the user.
CO Line	Assigns a button to access a defined CO/IP line.
CO Group	Assigns a button to access a free line in the CO/IP Group.
Loop	Assigns a button to access CO/IP Line that does not otherwise appear on the phone.
Station Number	Assigns a button as DSS/BLF for the assigned station number.

Table 6.2.5-1 FLEXIBLE BUTTON TYPE & VALUE

TYPE	REMARK
Programming (Numbering Plan)	Assigns a button to dial a code from the Flexible Numbering Plan as shown in Appendix B.
Programming (PGM)	Assigns a button to perform a User Program function from the Numbering Plan shown in Appendix A.
Station Speed Bin	Assigns a button to a Station Speed Dial bin.
System Speed Bin	Assigns a button to a System Speed Dial bin.
Net Station Number	Refer to Network Numbering Plan Table - PGM 324.
U-Loop	U-Loop button for call wait of internal & external call

6.2.6 Station COS - PGM 116

Selecting Station COS will display the Station COS data input page. Enter a valid station range and click **[Load]** to enter the Station COS data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of stations when saved.
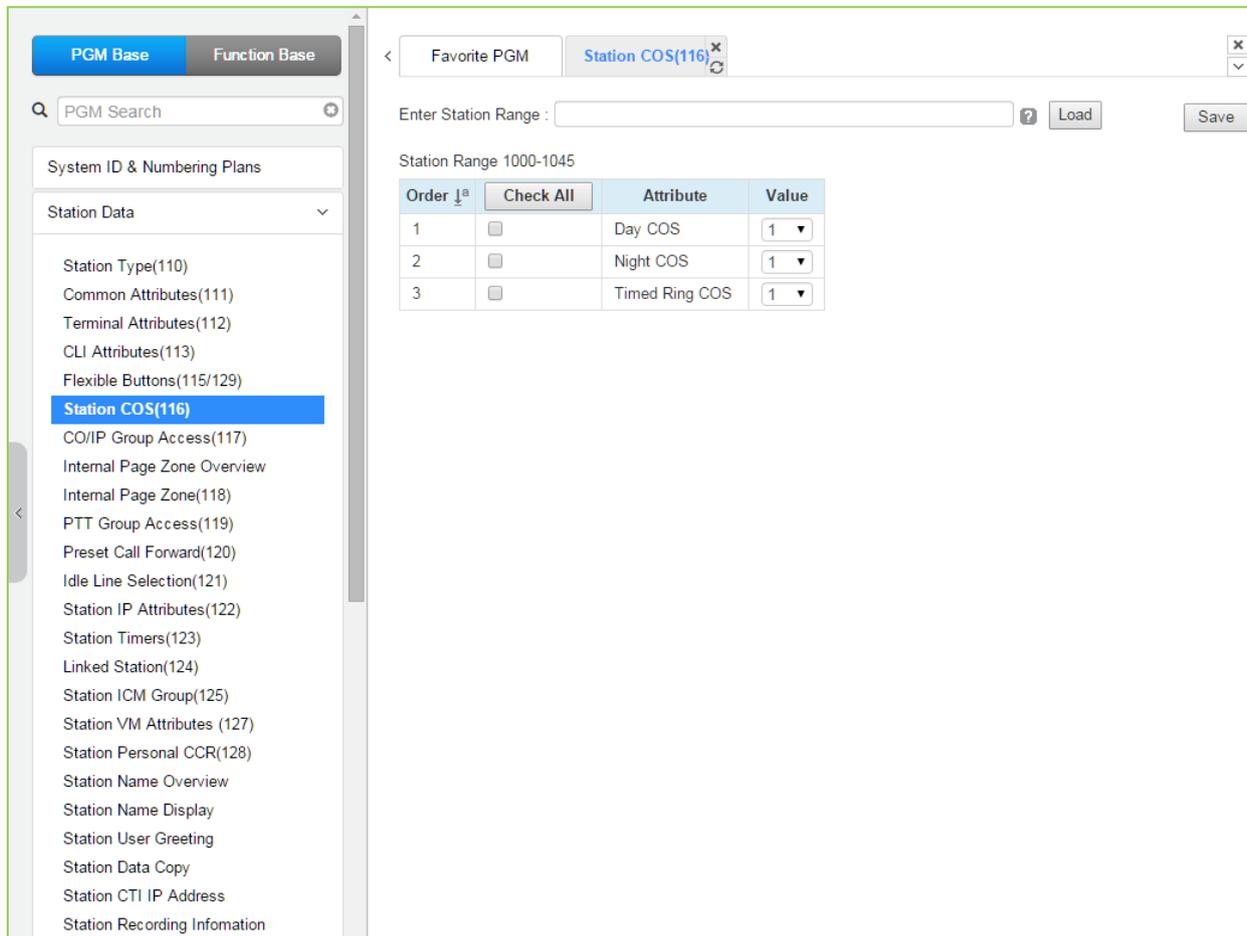
Figure 6.2.6-1 Station COS

In case of PGM 116 is in condition of limited service mode due to License issue, the default value is 7 for all COS type.

All stations are assigned a Class-of-Service (COS), which determines the ability of the user to dial certain types of calls, refer to the below 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. The station COS interacts with the CO Line COS to establish overall dialing or Toll restrictions. This interaction and the resulting restrictions are given in table 'Station/CO COS'.

Long distance calls are determined by the first dialed digit ("0" or "1") and the number of digits dialed. If the first digit dialed is a LD code, default "0" or "1", or, if the number of digits dialed exceeds the assigned LD digit counter in '*System Data SMDR Attributes section*', the call is considered a Long Distance call and appropriate restrictions applied.

Table 6.2.6-1 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 dialed cannot be a Long Distance code, default "0" or "1", and further denied/allowed based on Exception Table C.
6	The leading digits dialed cannot be a Long Distance code & digit count cannot exceed the LD digit counter, default 7 digits, and further denied/allowed based on Exception Table C.
7	Intercom and paging calls are allowed. No outgoing dialing except for emergency calls is allowed on CO Lines.
8	The assignments in the Exception Table D are monitored for allow and deny numbers.
9	The assignments in the Exception Table E are monitored for allow and deny Numbers
10	The assignments in the Exception Table D & E are monitored for allow and deny numbers.
11	The assignments in the Exception Table A & B and D & E are monitored for allow and deny numbers.

Table 6.2.6-2 STATION/CO COS

	CO COS 1	CO COS 2	CO COS 3	CO COS 4	CO COS 5
STA COS 1	No Restriction	No Restriction	No Restriction	Only Local Call (LD code/counter) and Table C	No Restriction
STA COS 2	Exception Table A governs the dialing	Exception Table A governs the dialing	No Restriction	Only Local Call (LD code/counter) and Table C	No Restriction
STA COS 3	Exception Table B governs the dialing	No Restriction	Exception Table B governs the dialing	Only Local Call (LD code/counter) and Table C	No Restriction
STA COS 4	Exception Table A & B governs the dialing	Exception Table A governs the dialing	Exception Table B governs the dialing	Only Local Call (LD code/counter) and Table C	No Restriction
STA COS 5	Local Call only (LD Code, "1" or "0") and Table C	Local Call only (LD Code, "1" or "0") and Table C	Local Call only (LD Code, "1" or "0") and Table C	Only Local Call (LD code/counter) and Table C	No Restriction
STA COS 6	Only Local Call (LD code/counter) and Table C	Only Local Call (LD code/counter) and Table C	Only Local Call (LD code/counter) and Table C	Only Local Call (LD code/counter) and Table C	No Restriction
STA COS 7	In-house dialing only	In-house dialing only	In-house dialing only	In-house dialing only	In-house dialing only
STA COS 8	Exception Table D governs the dialing	Exception Table D governs the dialing	No Restriction	Only Local Call (LD code/counter) and Table C	No Restriction
STA COS 9	Exception Table D governs the dialing	Exception Table D governs the dialing	No Restriction	Only Local Call (LD code/counter) and Table C	No Restriction
STA COS 10	Exception Table D & E governs the dialing	Exception Table D & E governs the dialing	No Restriction	Only Local Call (LD code/counter) and Table C	No Restriction
STA COS 11	Exception Table A & B and D & E governs the dialing	Exception Table A & B and D & E governs the dialing	No Restriction	Only Local Call (LD code/counter) and Table C	No Restriction

6.2.7 CO/IP Group Access - PGM 117

Selecting CO/IP Group Access will display the CO/IP Group Access data input page. Enter a valid station range and click **[Load]** to enter CO/IP Group Access data. Check the appropriate boxes to allow or delete access to each CO/IP Group. Click **[Save]** button after changing Value to apply.
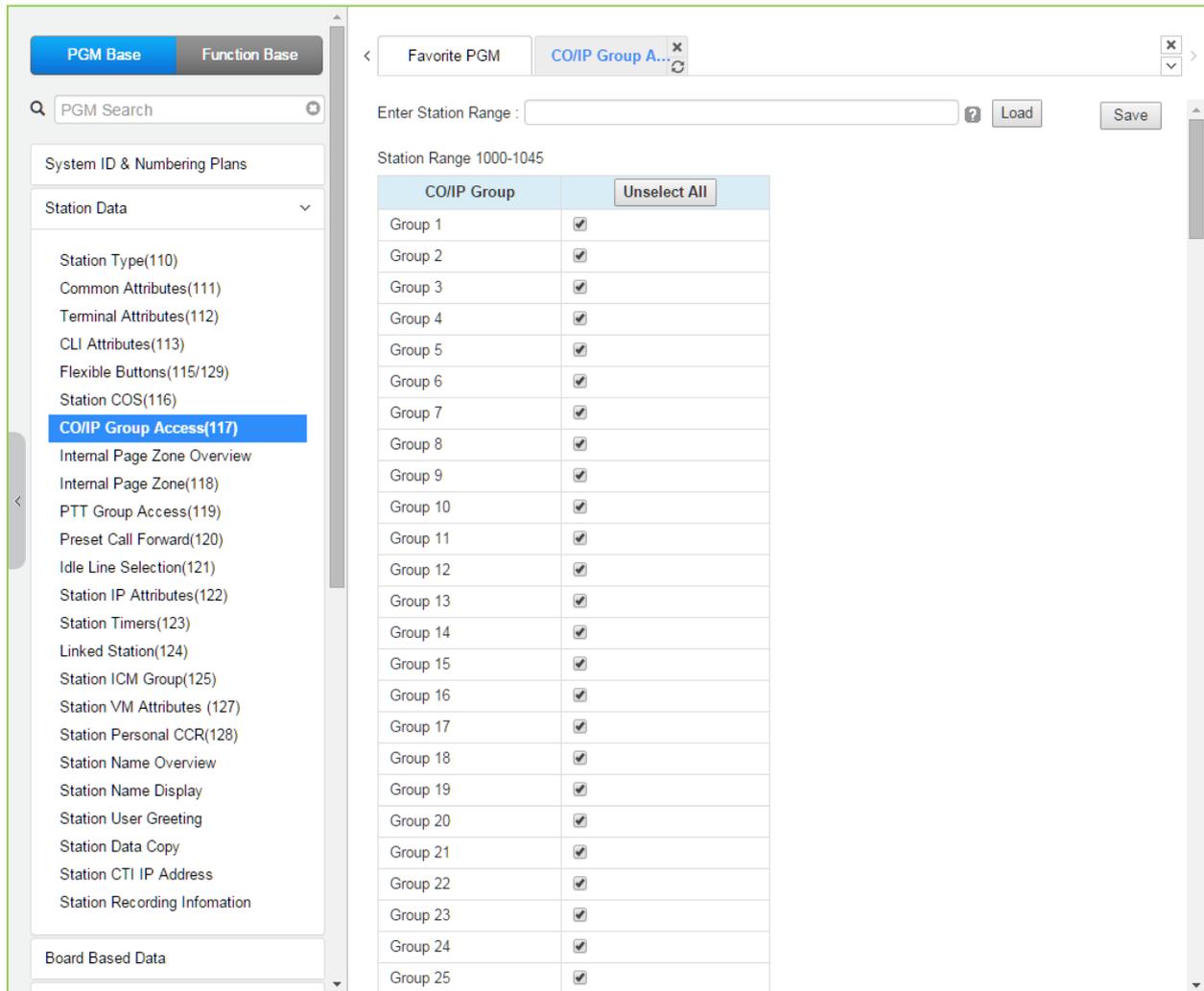
Figure 6.2.7-1 CO/IP Group Access

Stations can be allowed or denied access to CO Lines and IP Channels by group. As a default, all stations are allowed access to all groups except Private Lines (group 201) and unused CO Lines (group 0). The CO Line of an RSGM is assigned as a Private Line by default.

6.2.8 Internal Page Zone - PGM 118

Selecting Internal Page Zone overview will display the current status of the Internal Page Zone as below.

Figure 6.2.8-1 Internal Page Zone overview

Selecting Internal Page Zone will display the Internal Page Zone data input page. Enter a valid station range and click **[Load]** to enter the Internal Page Zone Access data. Check the appropriate boxes to allow the station to receive pages from each Internal Page Zone. Click **[Save]** button after changing Value to apply.
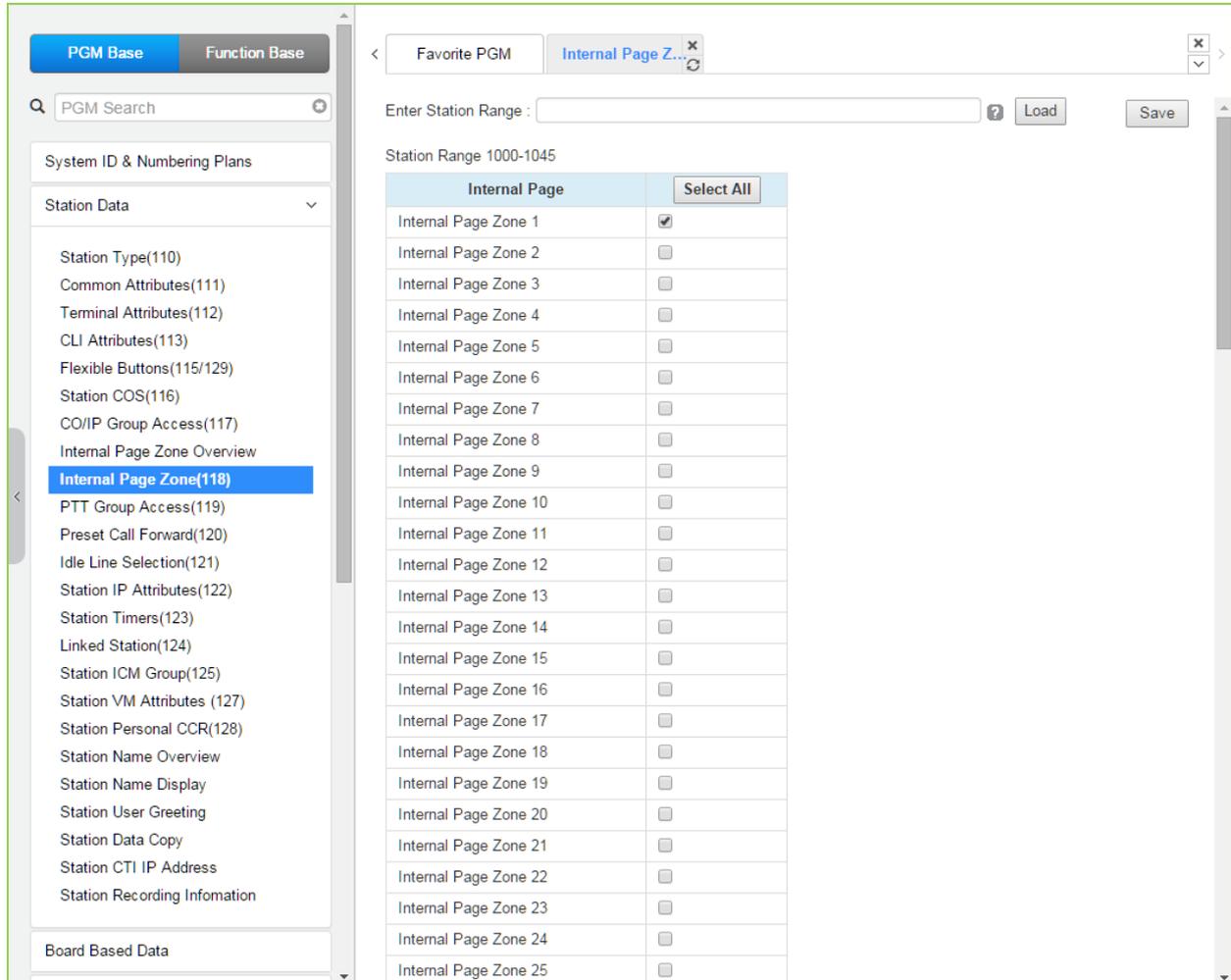
Figure 6.2.8-2 Internal Page Zone

Each iPECS IP and LDP Phone is assigned to receive announcements from each Internal Page Zone. A station can be assigned to any, all or no zones. Note a station not assigned to any Internal Zone will not receive any page announcements including Internal All Call. As a default, all stations except remote stations are assigned to receive pages from zone 1.

6.2.9 PTT Group Access - PGM 119

Selecting PTT (Push-to-Talk) Group Access will display the PTT Group Access data input page. Enter a valid Station range and click **[Load]** to enter the PTT Group Access data. Check the appropriate boxes to allow access to each PTT Group. Click **[Save]** button after changing Value to apply.
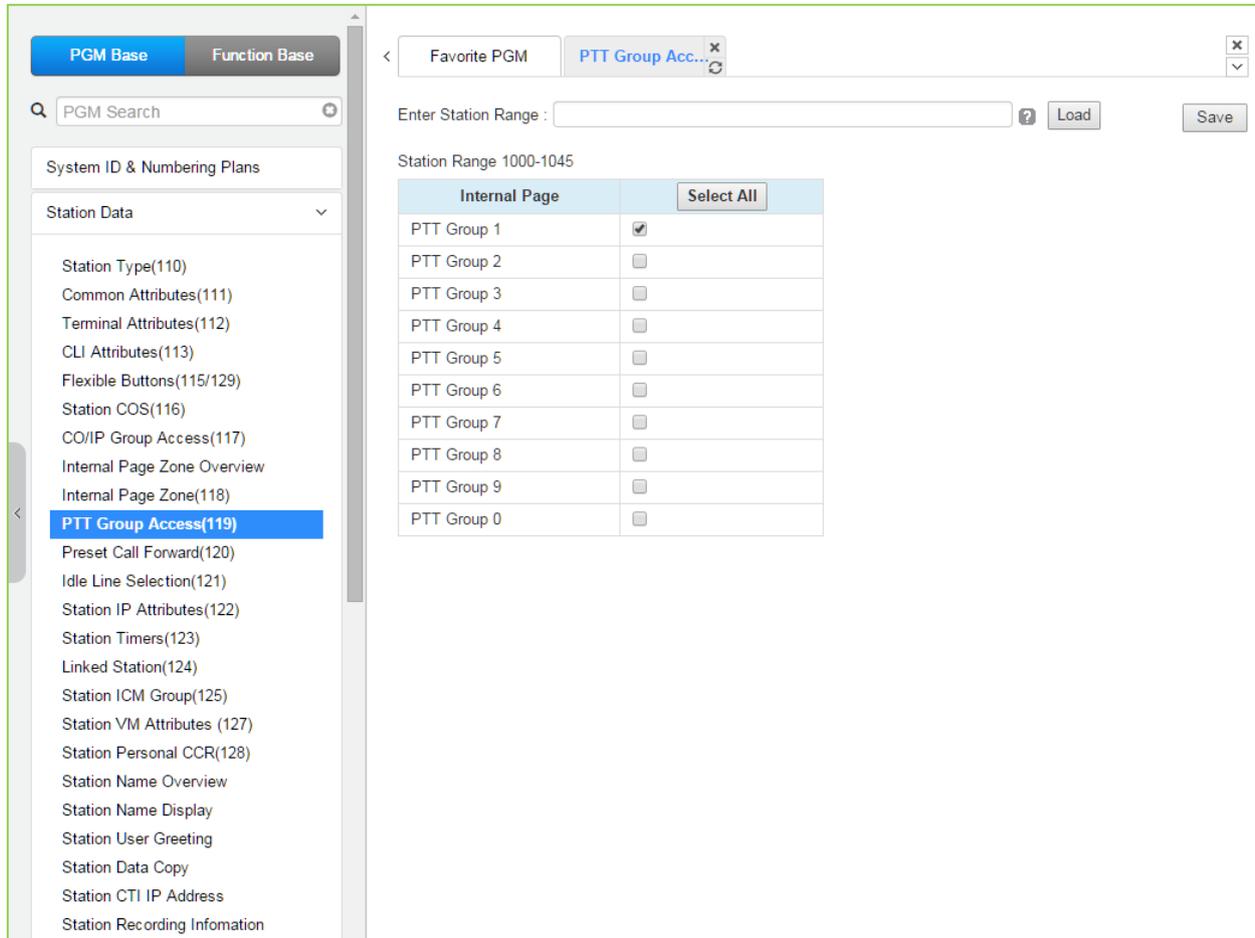
Figure 6.2.9-1 PTT Group Access

Each iPECS IP or LDP Phone is assigned to receive PTT announcements from any combination of the ten (10) PTT groups. Note a station not assigned to any group will not receive PTT page announcements including Internal All PTT group page. As a default, all stations except remote stations are assigned to group 1.

6.2.10 Preset Call Forward - PGM 120

Selecting Preset Call Forward will display the Preset Call Forward data input page. Enter a valid station range and click **[Load]** to enter the Station Preset Call Forward data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of stations when saved.
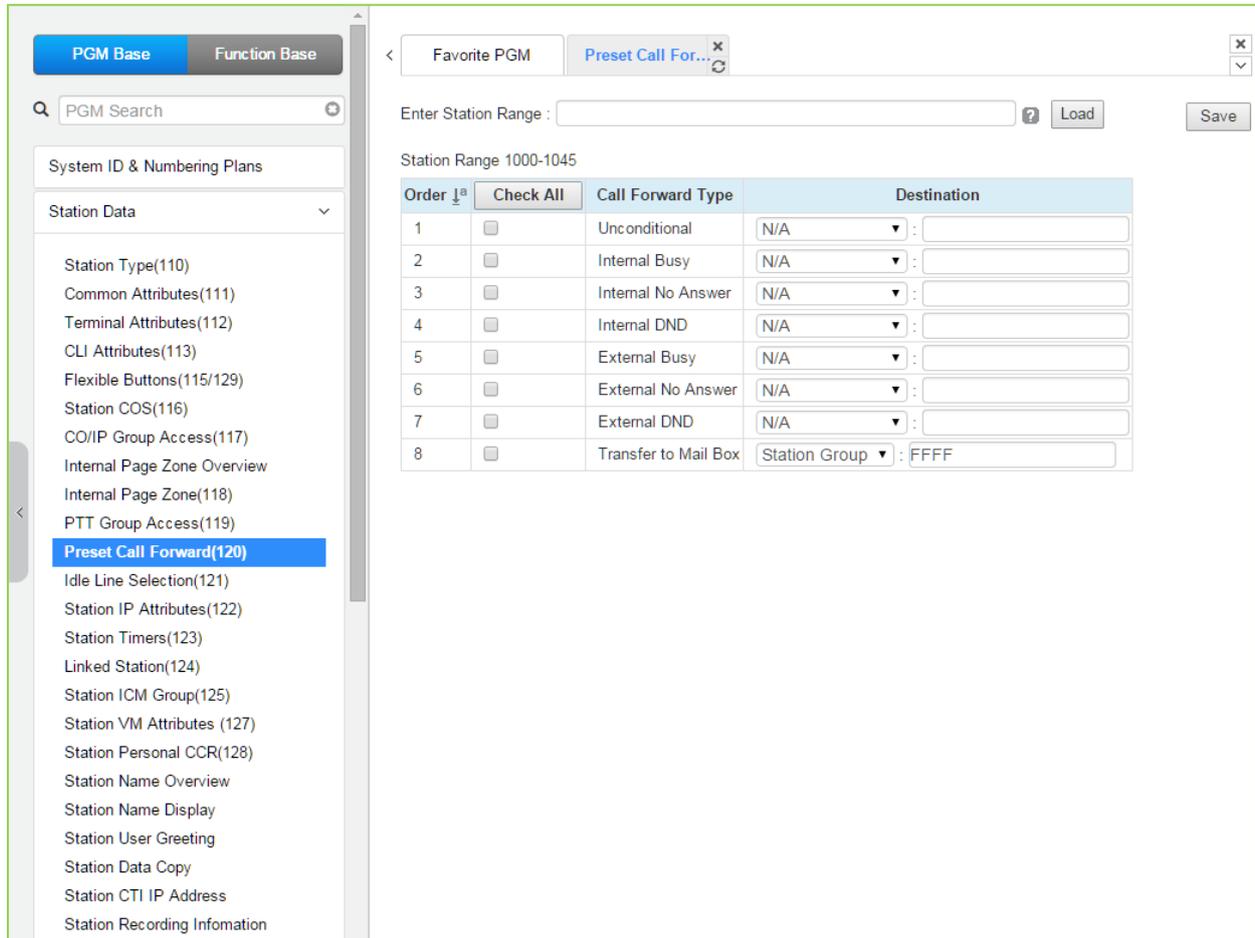
Figure 6.2.10-1 Preset Call Forward

Stations can be programmed so that incoming CO/IP and Intercom calls forward to a preset station or station group. This allows an external or internal call to initially ring at a station and forward to a pre-determined destination. Preset Forward can be separately assigned Unconditional, Internal Busy, Internal No Answer, Internal DND, External Busy, External No Answer, External DND preset forwarding to any station, Station group, System Speed dial bin (off-net) or Station ICR. As a default, no Preset Call Forward is assigned.

For "Transfer to Mail Box" enter the Station Group number of the Voice Mail group. This will permit other users to transfer calls directly to the desired user's Voice Mailbox.

6.2.11 Station ICR Scenario - PGM 1201

Selecting Station ICR Scenario displays the input entry page.

Figure 6.2.11-1 Station ICR Scenario

Station ICR is an extension of call forward where the user enters scenarios to define the call forward feature. Each station has ten (10) routing scenarios that define conditions for routing a user's incoming calls. Each scenario may define time of day, day of week, date, caller ID and destination for incoming calls. In addition, the scenarios may be prioritized; calls are routed to the destination with the highest priority-matching scenario.

6.2.12 Idle Line Selection - PGM 121

Selecting Idle Line Selection will display the Idle Line Selection data input page. Enter a valid Station range and click **[Load]** to enter the Idle Line Selection data. Check the appropriate radial button and enter the value for the Idle Line Selection. Click **[Save]** button after changing Value to apply.
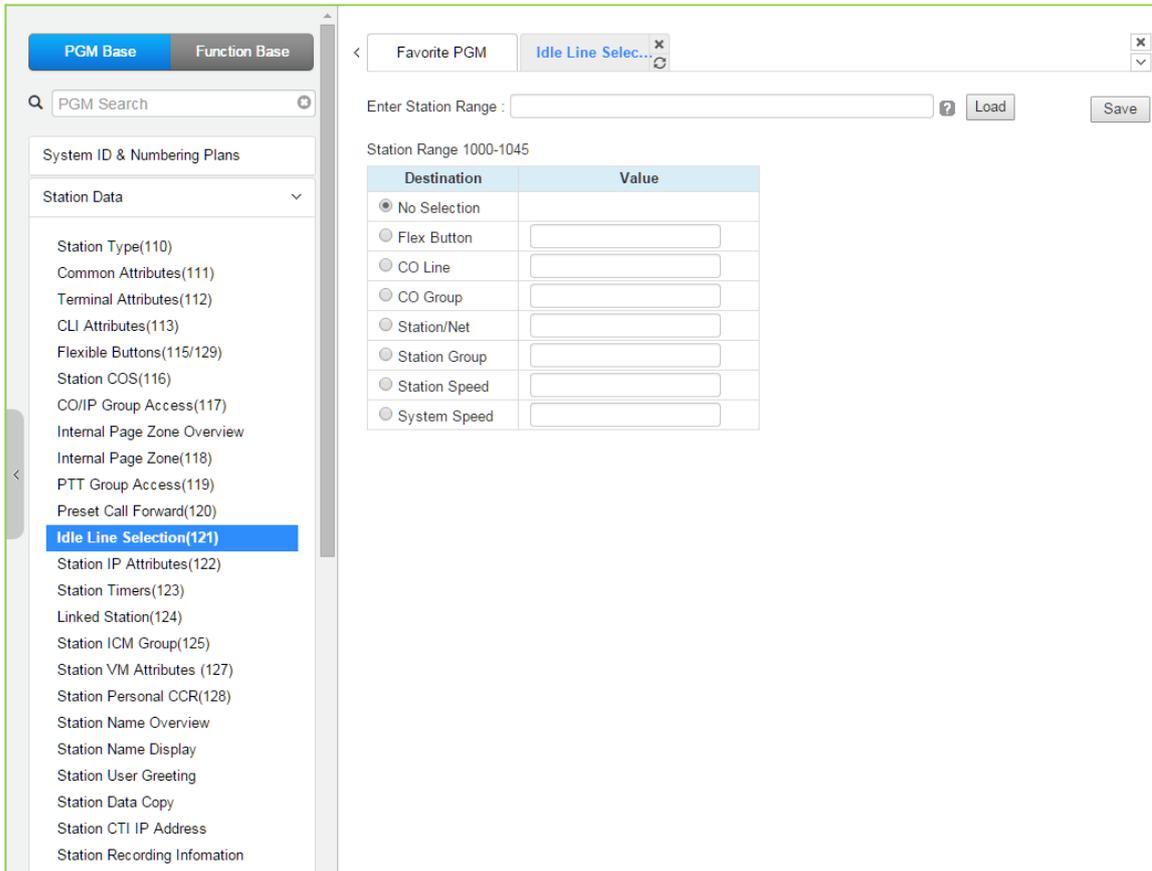
Figure 6.2.12-1 Idle Line Selection

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 access a CO Line, CO/IP Group or call a Station or Station Group as described in the below table when the station goes off-hook. The Idle Line Selection (Prime Line) can be either immediate (Hot) or delayed after going off-hook (Warm). The immediate/delay selection is based on the Hot/Warm assignment in '*Common Attributes - Prime Line*'.

Table 6.2.12-1 IDLE LINE SELECTION TYPE & VALUE

TYPE	DESCRIPTION
No Selection	Returns Intercom dial tone.
Flex Button	Flex button, activates Flex button as if pressed.
CO Line	CO path seizes the CO line.
CO Group	CO Group seizes CO line from the CO/IP Group.
Station/Net	Station, calls the assigned station.
Station group	Station group, calls the assigned Station group.
Station speed	Station speed, calls the number assigned in a station speed bin.
System speed	System speed, calls the number assigned in a system speed bin.

6.2.13 Station IP Attributes - PGM 122

Selecting Station IP Attributes will display the Station IP Attributes data input page. Enter a valid Station range and click **[Load]** to enter the Station IP Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of stations when saved.

Figure 6.2.13-1 Station IP Attributes

Stations are allowed access to the systems VoIP resources based on the Station IP Attributes.

6.2.14 Station Timers - PGM 123

Selecting Station Timers will display the Station Timers input page. Enter a valid Station range and click **[Load]** to enter the Station Timers data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of stations when saved.

Figure 6.2.14-1 Station Timers

Certain timers can be assigned on a station basis. Available timers, description and valid inputs are given in the below table.

Table 6.2.14-1 STATION TIMERS (PGM 123)

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Station Forward No Answer Timer	This timer set the duration that the station will ring prior to Ring-No-Answer Forward. This setting affects both manual and '*Preset Call Forward*' and overrides '*Call forward No Answer timer in System timer of System data*'.	000-600 (seconds)	0
Cut Off Timer	Allowed length of CO/IP calls when station is assigned '*Call Time restriction in Common Attributes of Station data*'.	00-99 (minutes)	0

6.2.15 Linked Station - PGM 124

Selecting Linked Station will display the Linked Station input page. Enter a valid Station range and click **[Load]** to enter the Linked Station data. Click **[Save]** button after changing Value to apply.
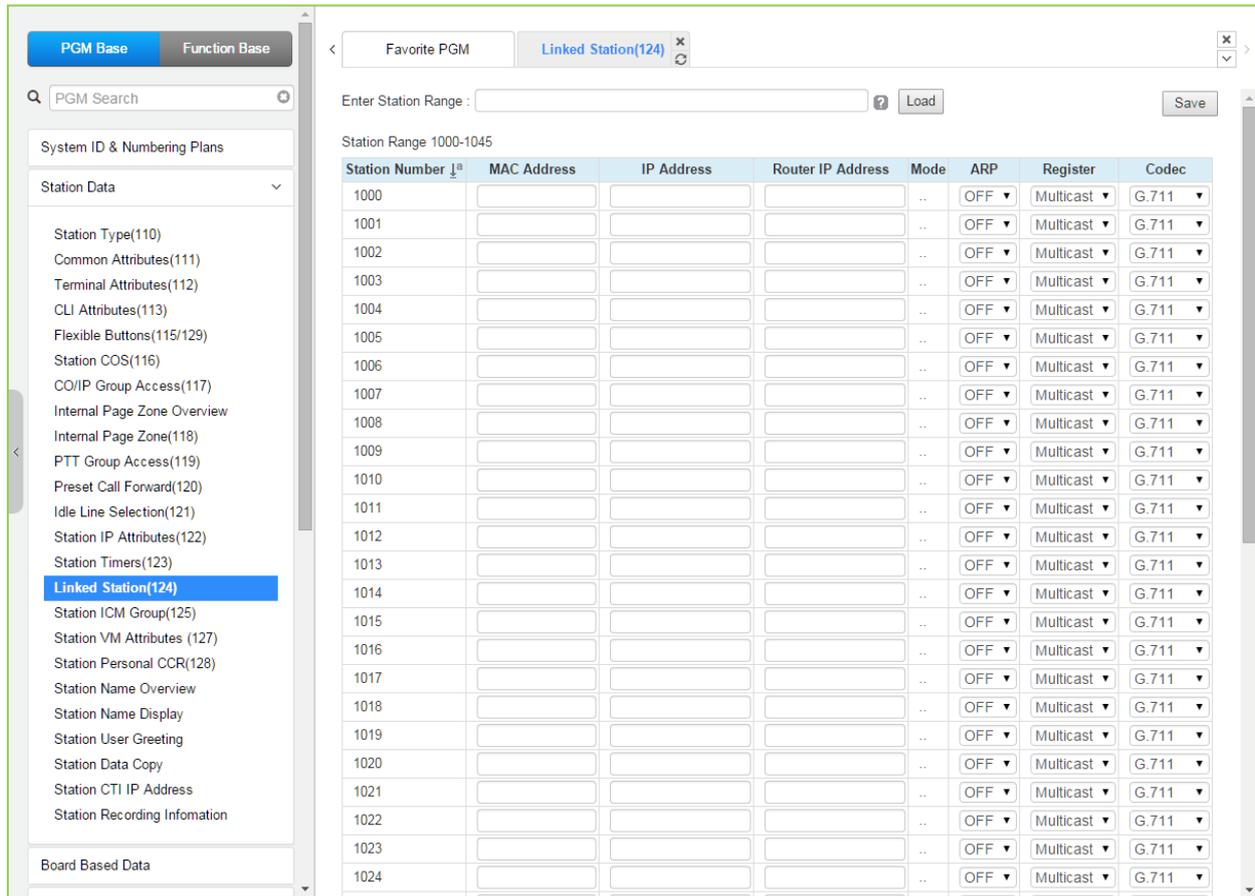
Figure 6.2.15-1 Linked Station Pair

For UCS Client, it is recommended that PGM 443 be employed for an unregistered (MAC Linked pair) station. When unregistered station linking is used, the linked station does not reduce the system's capacity. However, in this case, the linked station must be an iPECS IP Phone. Unregistered linking is allowed only to an iPECS IP Phone. Once UCS Client is linked, it will display in this Web page as well as in PGM 103 and 443.

If the station is assigned to Master or Slave by linked station, the bracket '[M] or [S]' will be displayed by the station number in PGM 101, 103, 105, 124 of Web page.

To link registered stations, the Personal Group feature is used and configured in PGM 260 and 261.

Table 6.2.15-1 LINKED STATION TABLE

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
MAC Address	Set MAC address of linked un-registered station, required data. Note the secondary station must not be registered in the system prior to linking. If needed, delete the device from the system prior to linking.		

Table 6.2.15-1 LINKED STATION TABLE

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
IP Address	The IP Address of the linked station.	IPv4 address	
Router IP Address	Set the Router IP address for a linked station.	IPv4 address	
Mode	This field displays the operating mode of the station, remote or local and NAT or NAPT.	Remote: R/NAPT R/NAT R/NAP R Local/Remote: LO/NAPT LO/NAT LO/NAP LO Local: L/NAPT L/NAT L/NAP L	L (Local)
ARP	If OFF, the system will employ layer 2 switching over the LAN to communicate with the linked station. If ON, the system will employ ARP (Address Resolution Protocol) to determine the IP address of the device, if required, and communicate using IP.	OFF ON	OFF
Register	Normally, iPECS IP devices register using multi-cast packets. When separated from the system, that is when the device is remote, unicast is used.	Multicast Unicast	Multicast
Codec Type	A specific Codec type can be configured for the station.	G.711 G.723.1 G.729 System, use the codec defined for the system	G.711

6.2.16 Station ICM Tenancy Group - PGM 125

Selecting Station ICM Tenancy Group displays the Station ICM Tenancy Group input page. Select ICM Tenancy Group and click **[Load]**, the system will display the ICM Tenancy Group Characteristics. Enter the Station number of the group Attendant and check the appropriate box to allow calling to the group. Click **[Save]** button after changing Value to apply.
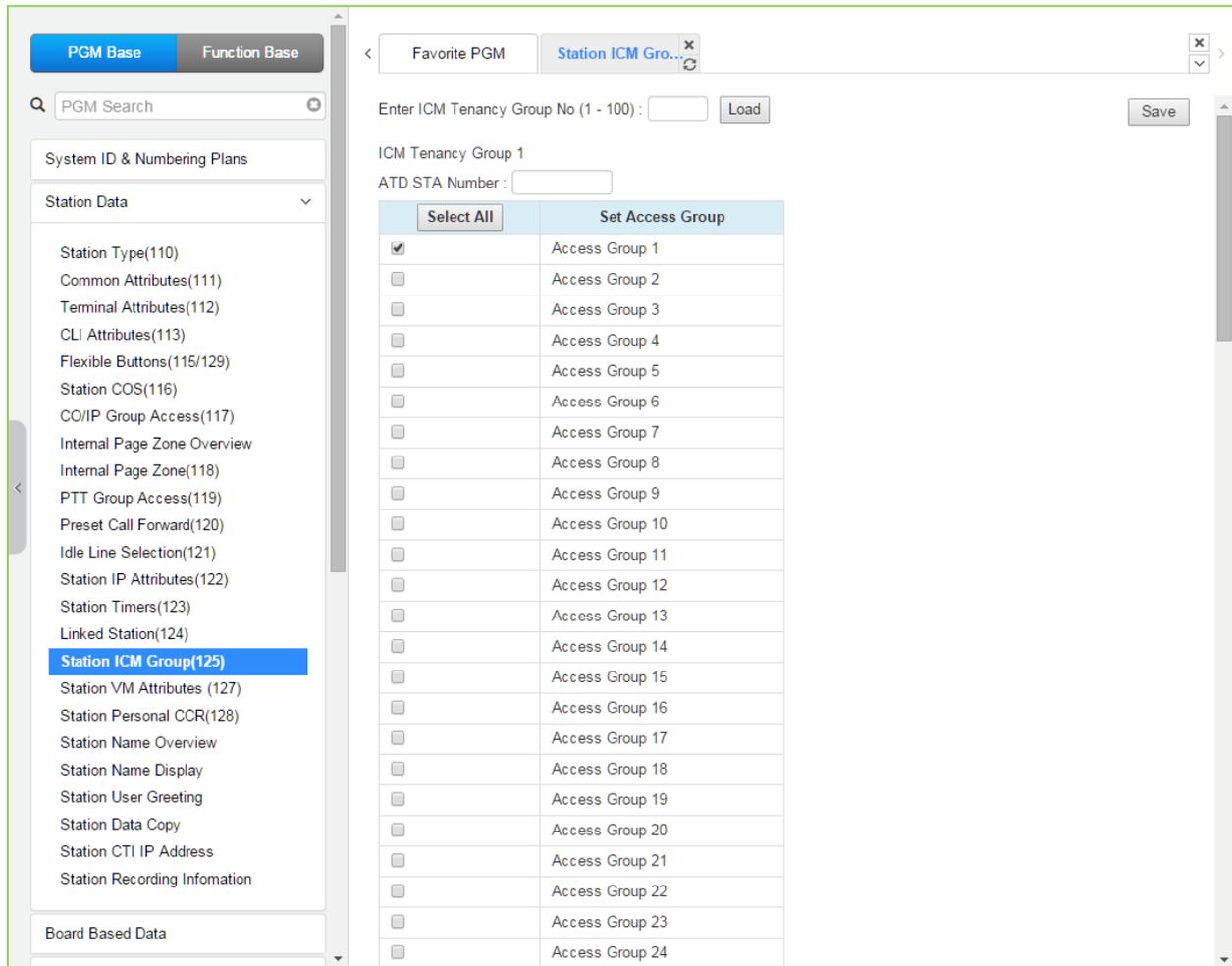
Figure 6.2.16-1 Station ICM Tenancy Group

Stations can be assigned to an ICM Tenancy group under '*Common Attributes section*'. Up to 100 Tenant groups can be defined. Each group is configured to allow or deny placing intercom calls, including transfers, to stations in other groups and an Attendant station can be defined for each group. The Attendant will receive "dial 0" calls and controls Day/Night mode for the group.

Table 6.2.16-1 STATION ICM TENANCY GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
ATD STA Number	Assign the Attendant station for the ICM Tenant group. The ATD receives dial '0' calls and controls Day/Night Service. Any station can be assigned as the Attendant for the Tenancy group.	Station No.	
Group Access	ICM tenancy groups allowed access by stations of the selected group.	1~100	1

6.2.17 Station Voice Mail Attributes – PGM 127

Selecting VM Attributes displays the Station Voice Mail Attributes input page. Enter a valid Station range and click **[Load]** to enter the Station Voice Mail Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of stations when saved.
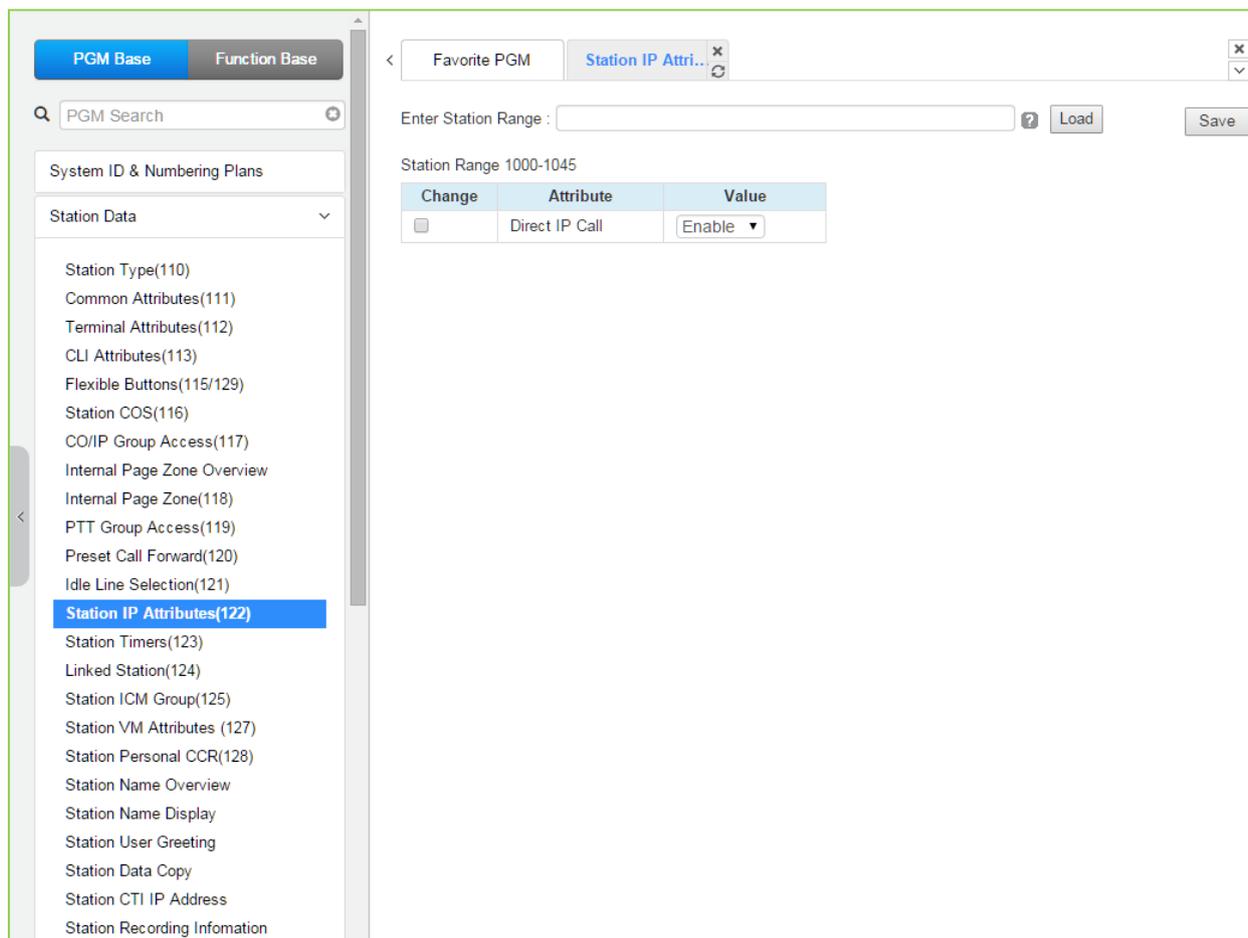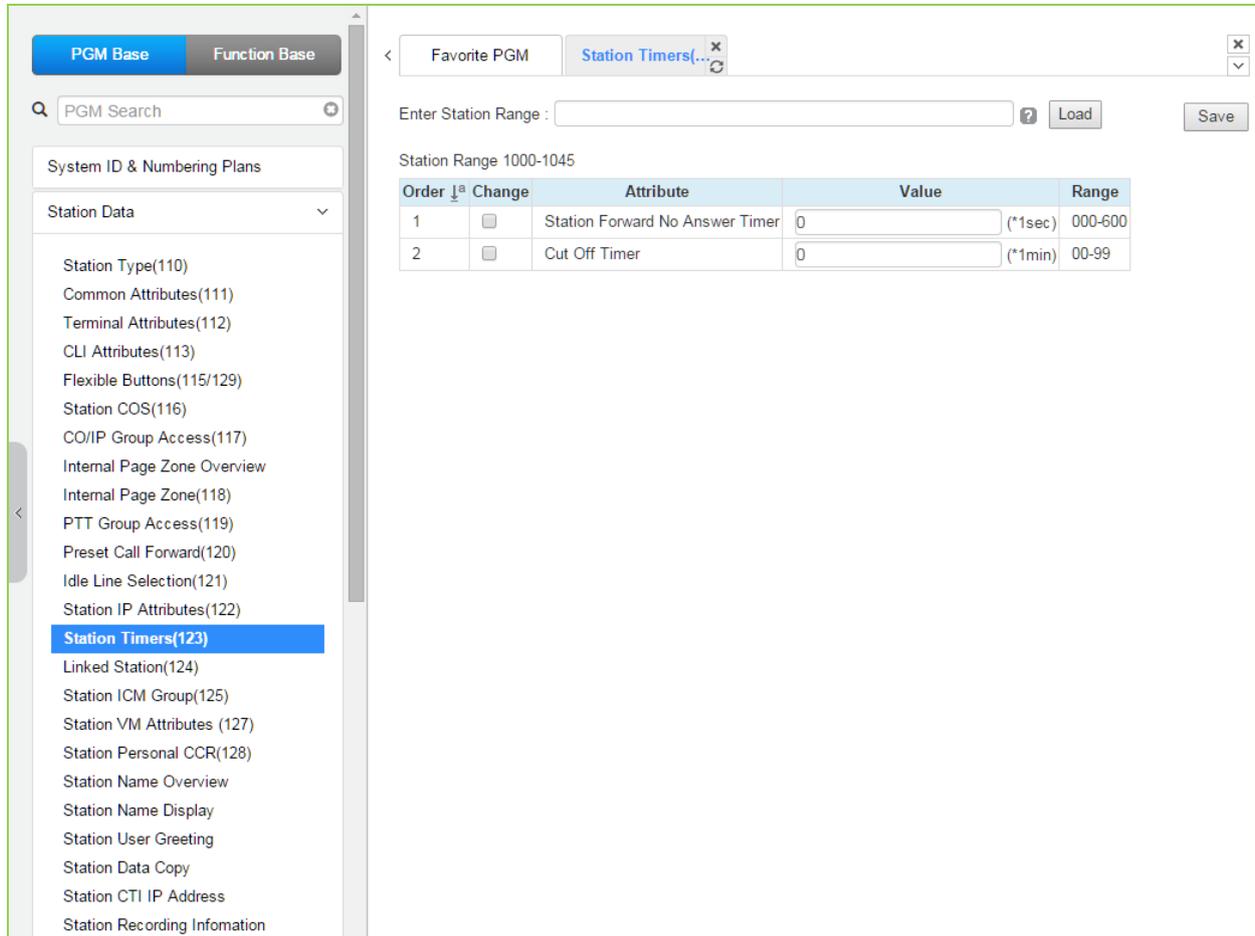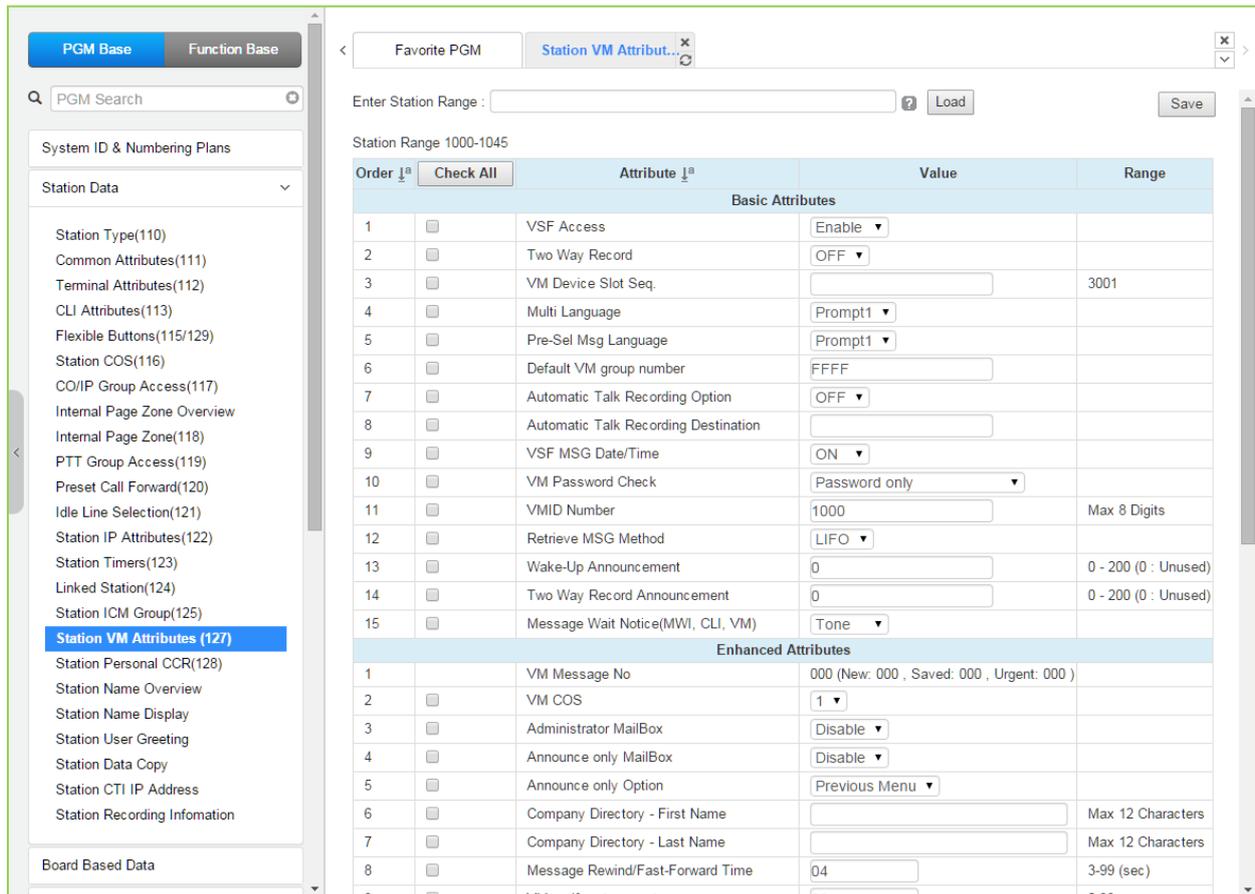

Figure 6.2.17-1 Station VM Attributes

The Station VM Attributes are divided into three sub-sections. The Basic Attributes assign general characteristics of the Station's Voice Mail and announcements. En100hanced Attributes assign characteristics including VM COS, notification, cascading and Company Directory Names. The E-Mail Notification section defines various SMTP server data such as the Station and System E-mail account data.

Table 6.2.17-1 STATION VM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Basic Attributes			
VSF Access	The station must be permitted VSF access to access the built-in Voice Mailbox.	Disable Enable	Enable
Two way Record	When allowed, the station can activate the Two-way record feature to record a conversation.	OFF ON	OFF
VM Device Slot Seq.	The sequence number defines the VSF gateway where messages for the station are stored.	Seq. No.	3001

Table 6.2.17-1 STATION VM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Multi Language	The selected language is employed for prompts played to the user.	Prompt 1~6	Prompt 1
Pre-selected Message Language	The selected language is employed for prompts played to an external CO caller if the called station has activated Pre-selected Message.	Prompt 1~6	Prompt 1
Default VM group number	When the user has no Voice messages and selects the **[Call Back/Msg]** button, the "Default VM group number" is called.		
Automatic Talk Recording Option	This field enables unconditional recording of all calls placed or received by the station. Recordings, in .wav format, are stored at the Call Recording destination defined below.	OFF ON	OFF
Automatic Talk Recording Destination	When Auto Call Recording is defined for a station, the destination for the recording is defined here. The UCS Client Station number, or an IPCR or 3rd party SIP recording server.	Station or Group	
VSF MSG Date/Time	When ON, the date and time the message was received is played to the user prior to the message.	OFF ON	ON
VM Password check	The user may be required to enter a password to access their VSF Voice Mailbox. The password can be defined as none, the Authorization code only or the normal Station Number and Authorization code.	No password Password only Station number & password	Password only
VMID Number	When using an adjunct VM, the system can translate the Mailbox number from the user's station number to the assigned VMID. The system sends the station number or VMID, if assigned, to the VM (in-band, SMDI) in order to identify the appropriate Voice Mailbox.	Max. 8 digit	Station Number
Retrieve MSG Method	Messages stored in the built-in VM may be retrieved in either a FIFO (first-in-first-out) or LIFO (last-in-first-out) order based on the default setting.	FIFO/ LIFO	LIFO
Wake-Up Announcement.	A system announcement can be recorded to play to the station as a Wakeup Announcement.	0-200	0
Two way record announcement	The two way record announcement can be recorded to play to the station as Announcement.	0-200	0
Message Wait Notice(MWI, CLI, VM)	When a user has voice mail, system can provides this as voice prompt instead of dial tone according to option based on station. Disable: System provides normal dial tone when a user goes to off hook status. Tone: System provides warning tone instead of dial tone to give indication when a user goes to off hook status. Prompt: System provides message indication as voice prompt when a user goes to off hook status. DECT phone do not support this feature.	Disable, Tone, Prompt	Tone
Enhanced Attributes			
VM Message No	Number of Voice Messages in the user's Mailbox.		
VM COS	Each Voice Mailbox is assigned one of five VM Classes of Service (COS) that determines basic characteristics of the Mailbox such as message retention. The Voice Mail COS	1-5	1

Table 6.2.17-1 STATION VM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	attributes are set in PGM 253.		
Administrator Mailbox	A Mailbox can be assigned as an Administrator Mailbox permitting broadcast messaging and control other user mailboxes.	Disable Enable	Disable
Announce Only Mailbox	A mailbox can be configured so that the connected party will hear the greeting but cannot leave a message, Announce Only. After the greeting, the call is routed based on the "Announce Only Option below.	Disable Enable	Disable
Announce Only Option	This feature determines if the caller will be disconnected or returned to the previous menu after hearing an Announce Only Mailbox greeting.	Previous Menu/ Hang Up	Previous menu
Company Directory – First Name	This feature is first name of a station for Company Directory feature.	Max. 12 characters	N/A
Company Directory – Last Name	This feature is last name of a station for Company Directory feature.	Max. 12 characters	N/A
Message Rewind/Fast-Forward Time	While listening to a message, the user may rewind or fast forward. The Rewind and Fast Forward time can be set according to user's needs.	3 – 99 (Seconds)	04
VM Notify retry count	If the notification to the user's mobile fails repeatedly, after the retry count, the system will send the call to the Attendant to complete the notification.	00 – 99	03
VM Notify dial time	When the Mobile notification call is placed, the system will terminate the call, considering the retry count of failure, if no digits are received from the mobile phone user before the timer expires.	00 – 99 (Seconds)	15
VM Forward Option	When a left message is forwarded, this feature is applied.	Move / Copy	Move
Cascade Mailbox 1	A left message is copied or moved to 1'st mailbox destination station.	Station No	N/A
Cascade Mailbox 2	A left message is copied or moved to 2'nd mailbox destination station.	Station No	N/A
Cascade Mailbox 3	A left message is copied or moved to 3'rd mailbox destination station.	Station No	N/A
Cascade Mailbox 4	A left message is copied or moved to 4'th mailbox destination station.	Station No	N/A
Cascade Mailbox 5	A left message is copied or moved to 5'th mailbox destination station.	Station No	N/A
Cascade Method	Cascade method can be set.	Disable / Copy / Move	Disable
Cascade Message Type	Cascade Message Type can be selected.	Normal Only / Urgent Only / All	Normal Only
Cascade Apply Timer	Cascade Apply Timer is set.	001 ~250 (min)	000
E-Mail notification [Send Mail]			
[Send Mail] button is used for sending the mail after filling out the below menu.			
SMTP Server Address (IP or Domain name)	This field defines the IP address or URL of the SMTP mail server the system will use to send the notification E-mail.	IP v4 address or Mail server name	
SMTP Port	This field defines the TCP/IP port the system will employ	00001 - 65535	25

Table 6.2.17-1 STATION VM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	when communicating with the SMTP E-mail server.		
SMTP Security connection	The system can support basic security policies when communicating with the SMTP E-mail server. Note the server must also be configured for the selected security protocol.	No Security SSL TLS	No Security
E-Mail Account ID	This field defines the account Identification for the system's E-mail account in the SMTP server.		
E-Mail Account Password	This field defines the password for the system's E-mail account in the SMTP server.		
Sender E-Mail Address (From)	This field defines the E-mail address used to send the new Voice Message notification via E-mail.	E-mail address	
Receiver E-Mail Address (To)	This field defines the E-mail address to notify when a new VSF message is received for the station.	E-mail address	
Attach Message Option	When E-mail notification is enabled in the VM COS (PGM 253), the E-mail may include the Voice message as a wav file attachment. The proper information must be assigned for the receiver mail and system mail account.	OFF ON	ON
Delete Message Option	After sending an E-mail notification, the system will automatically delete the Voice message from the user's Voice Mailbox if this parameter is ON.	OFF ON	OFF

6.2.18 Station Personal CCR Table – PGM 128

Selecting Station Personal CCR displays the Station Personal CCR Table input page. Enter a valid Station range and click **[Load]** to enter the Station Personal CCR Table data. Click **[Save]** button after changing the type or value.
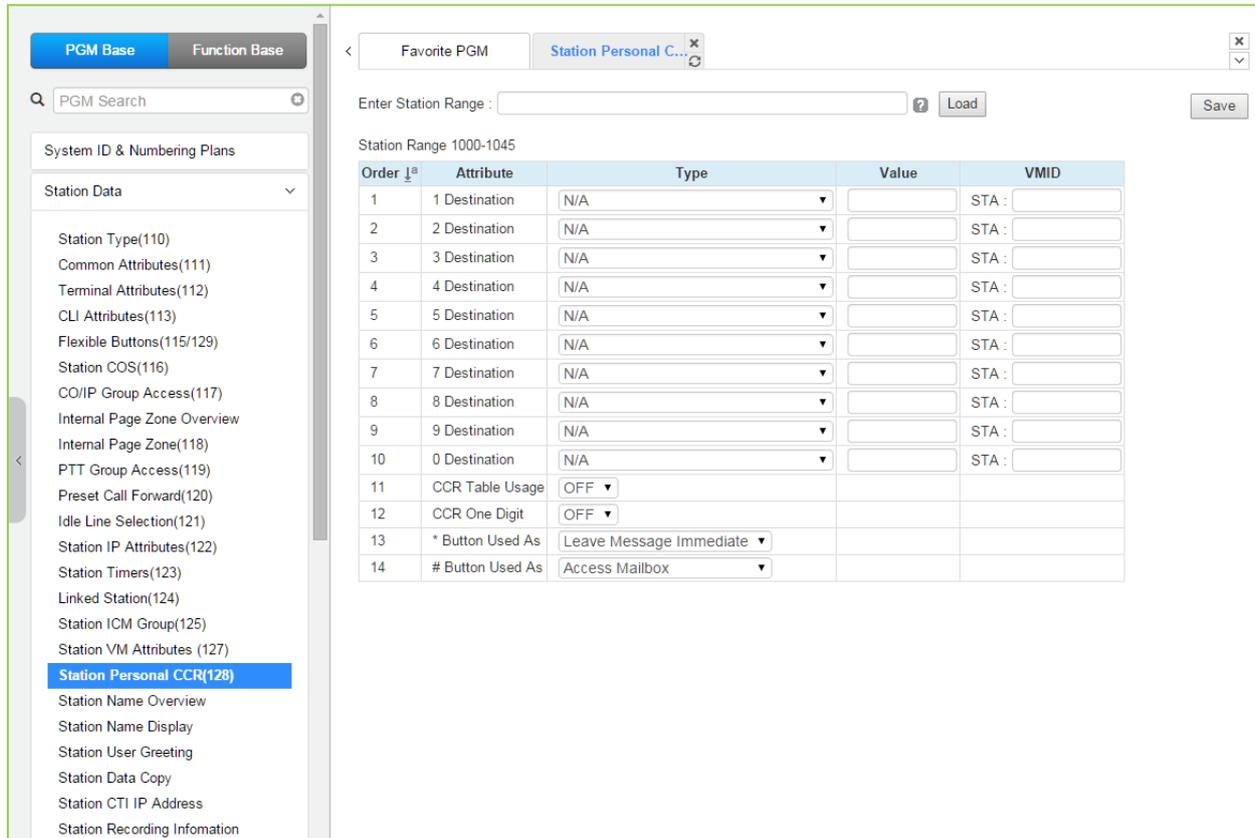
Figure 6.2.18-1 Station Personal CCR Table

A CCR Table can be associated with the station's Voice Mail greeting recorded for the built-in Voice Mail. The greeting should direct the caller to enter a digit when the called user is busy or does not answer.

Table 6.2.18-1 STATION CCR Table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
1-0 Destination	Entries 1 ~ 9 and 0 represent the digit dialed by the outside caller. A destination type and value is defined for each digit. Refer to Table 6.2.17-2 for the destinations available.	…	
CCR Table Usage	This option enables use of the CCR Table. If not enabled, DISA is employed to analyze the caller-dialed digits and determine routing.	OFF ON	OFF
CCR One Digit	When ON, the system will analyze the first digit received only, additional digits dialed by the caller are ignored.	OFF ON	OFF
* Button Used As	The asterisk (*) can be used for the following functions only: • Flex Numbering Plan • Replay Greeting • Access Mailbox • Leave Message Immediate	Refer to Description	Leave Message Immediate

Table 6.2.18-1 STATION CCR Table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
# Button Used As	The pound (#) can be used for the following functions only: • Flex Numbering Plan • Replay Greeting • Access Mailbox • Leave Message Immediate	Refer to Description	Access Mailbox

Table 6.2.18-2 STATION CCR DESTINATIONS

DESTINATION TYPES
Route to a Station
Route to a Station Group
Route with System Speed Dial
Route as PBX Transfer with System Speed Dial (Flash then dial speed dial digits)
Route to VSF Announcement
Route to VSF Announcement and disconnect
Route to Networked Station.
Conference Room
Internal Page
External page
All Call Page
Route to voice mail(station group/station number)
Company Directory
Record VM Greeting
Room type conference group join

6.2.19 Station Name Overview & Display

Selecting Station Name Overview shows the current station name.

Figure 6.2.19-1 Station Name Overview

Selecting Station Name Display will display the Station Name input page. Enter a valid Station range and click **[Load]** to enter the Station Name data. Click **[Save]** button after changing the type or value.

Figure 6.2.19-2 Station Name Display

You can fill out the Station name, Input name (up to 20 digits), company directory-First name & Last name for customer's convenience.

6.2.20 Station User Greeting

Selecting Station User Greeting will display the Station User Greeting data input page. Enter a valid Station range and click **[Load]** to enter the Station Voice Mail Attributes data. Use the check mark to indicate which attributes to modify; data for checked attributes is stored by clicking **[Save]** button.
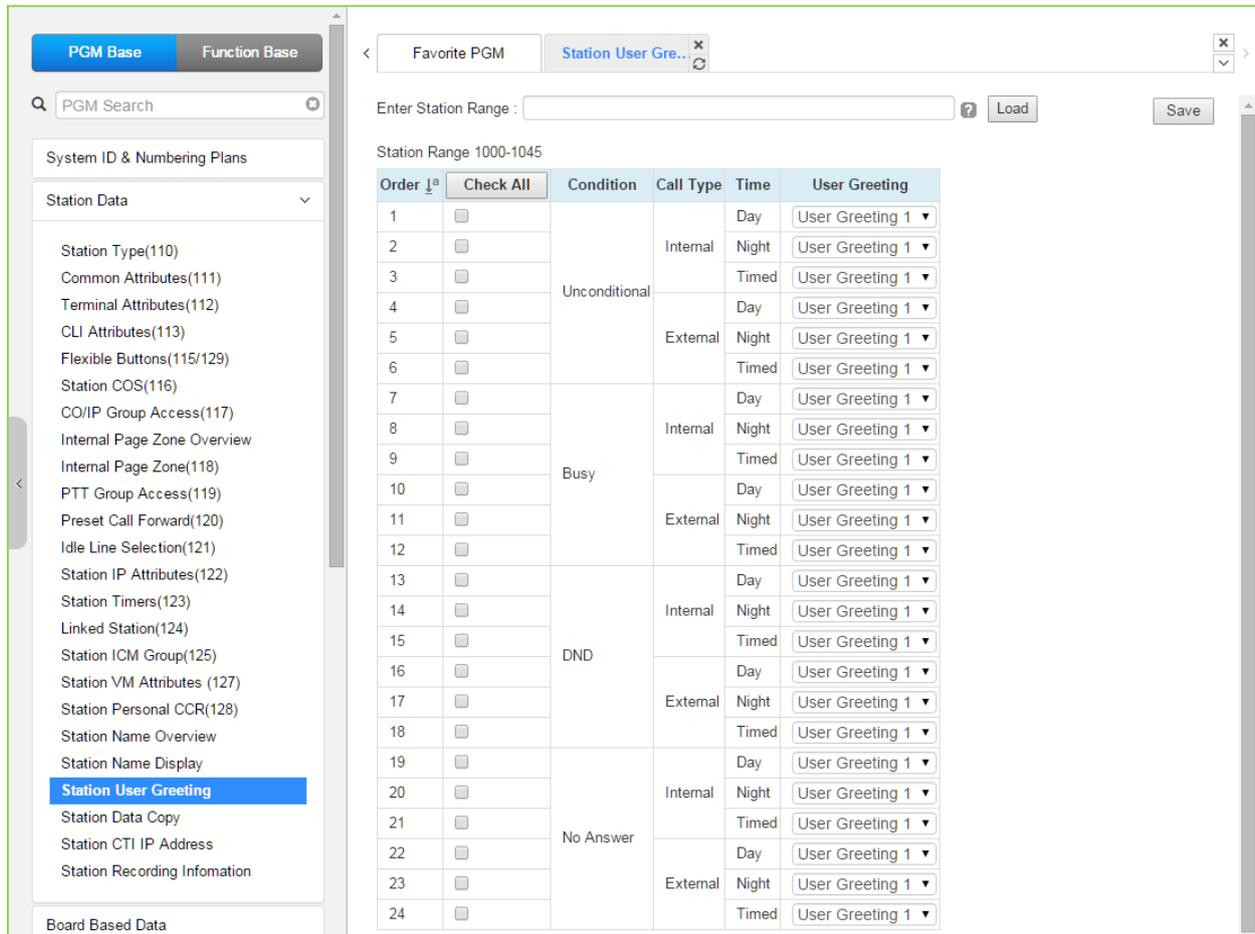
Figure 6.2.20-1 Station User Greeting

There are 4 User greeting for each station.

User can select multi user greeting depending on the condition, call type, and time.

    -. Condition: unconditional, busy, DND, No answer

    -. Call Type: Internal, External

    -. Call Time: Day, Night, Timed

6.2.21 Station Data Copy

Selecting Station Data Copy will display the Station Data Copy data input page. Enter a valid Source Station, Destination Station Range and click **[Copy]** to copy the station data.

Note that this function is not available for an Attendant station.
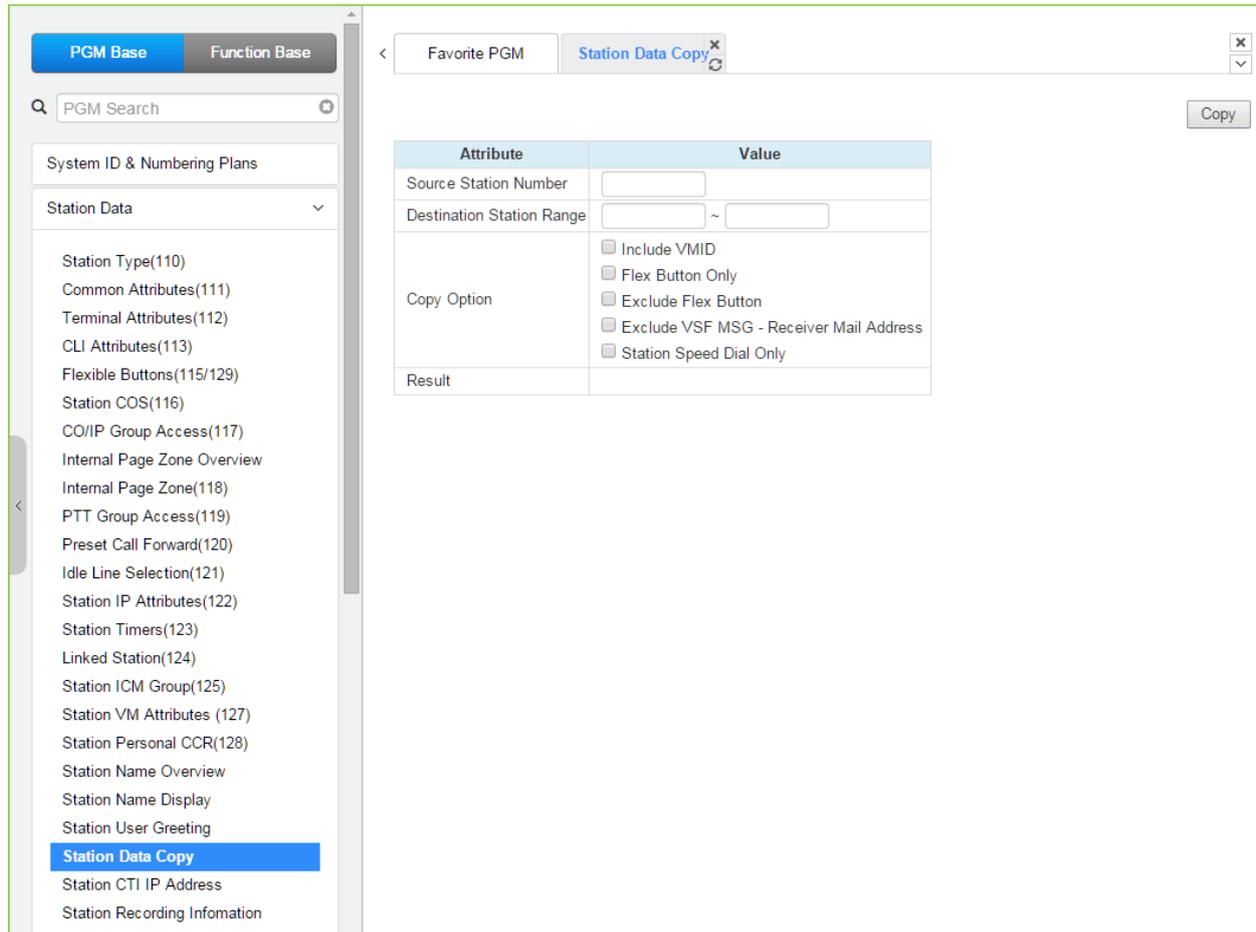
Figure 6.2.21-1 Station Data Copy

6.2.22 Station CTI IP Address (1ˢᵗ Party TAPI)

Selecting Station CTI IP Address displays the following figure. Enter a Station Range and click **[Load]** to display and modify the CTI IP address. Click **[Save]** button after changing the type or value.

Figure 6.2.22-1 Station CTI IP Address

The system supports 1st party TAPI (Telephony Application Programming Interface) for CTI (Computer Telephony Integration). The system will send specific TAPI information to the IP address for activity by the associated station.

6.2.23 Station Recording Information

Selecting Station recording information displays the following figure. Enter a Station Range and click **[Load]** to display and check the current recording information on each station.

Figure 6.2.23-1 Station Recording Information

The {Station Recording information} displays the recording state of voice mail greeting and company directory name with O and X. (O: recorded, X: not recorded)

6.3 Board Based Data

Selecting the Board Based Data program group returns the sub-menu displayed in the left frame as shown in the following figure.
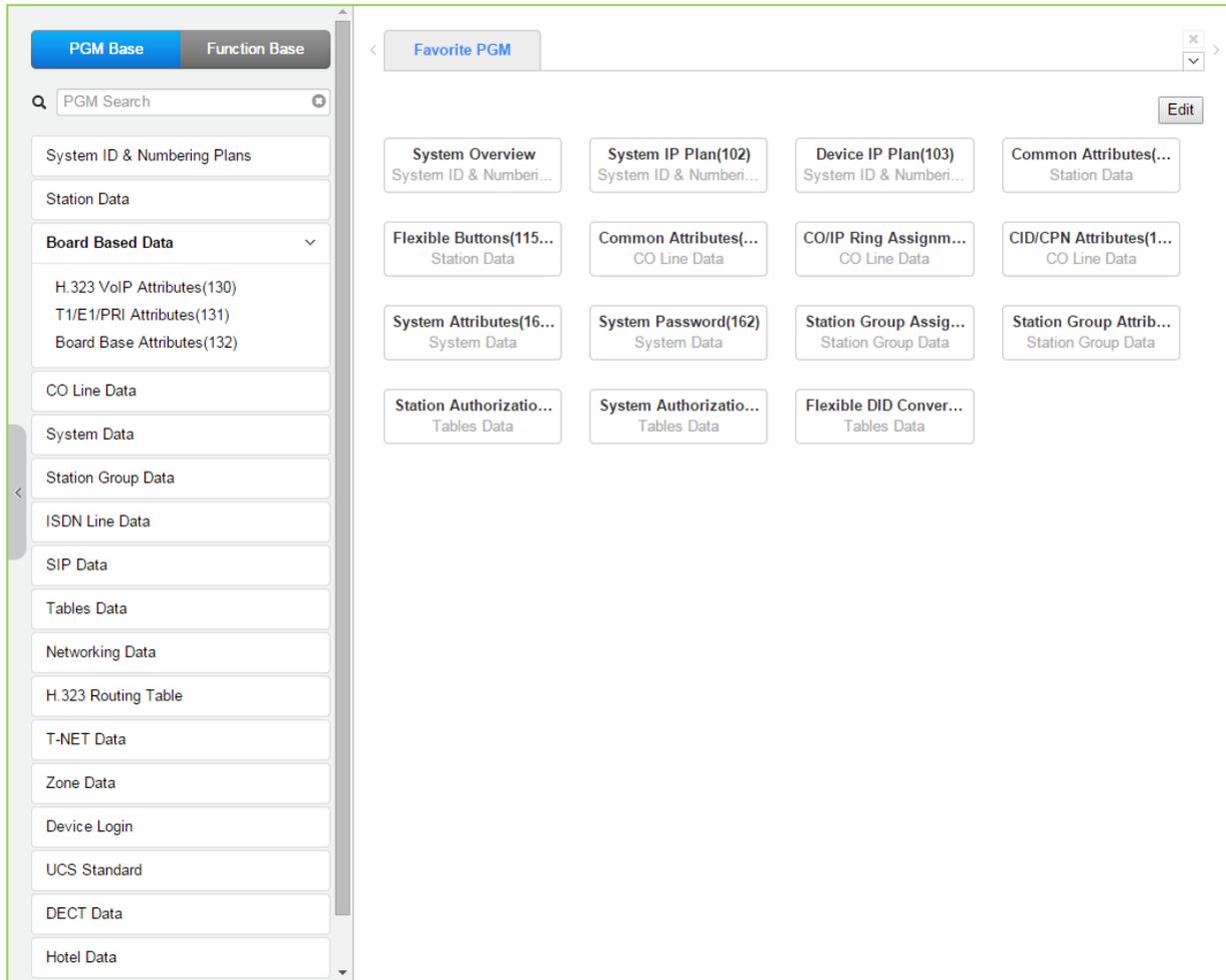
Figure 6.3-1 Board Based Data Main Page

6.3.1 H.323 VoIP Attributes - PGM 130

Selecting H.323 VoIP Attributes will display the H.323 VoIP Attributes data input page. Enter the Vo IP gateway Sequence number (refer to '*Device IP Plan*' if you want to find the sequence number) and click **[Load]** to enter VoIP data. Click **[Save]** button after changing Value.

Figure 6.3.1-1 H.323 VoIP Attributes

The VOIP channels may be used for Distributed Networking, access to SIP or H.323 networks, and for remote iPECS devices. When the standard H.323 VoIP protocol is employed for an external VoIP call, several attributes of these channels can be assigned. The H.323 call set-up mode and tunneling (H.245 Encapsulation) can be established.

Also for H.323 support, a RAS (Registration, Admissions and Status) channel can be defined. The RAS channel IP addresses (uni-cast and multi-cast) as well as the IP port numbers and other H.323 set-up characteristics are defined.

This page also allows setting the IP TOS bit for Diffserv, a commonly recognized packet prioritization protocol. Higher priority packets are given priority in the Default gateway or Layer 3 Switch queue. However, they are the first to be discarded in the event of long queue delays, which may cause excess packet loss and poor voice quality.

Refer to the following table for a description of the features and the input required.

Table 6.3.1-1 H.323 VOIP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
H323 Setup Mode	IP calls can be set-up using the H.323 normal or Fast Start mode.	Normal Fast	Fast
H323 Tunneling Mode	IP calls can be set-up using the H.245 encapsulation (Tunneling).	OFF ON	ON
H.323 Early Media (early H245)	This feature is the ability of two user endpoints to communicate before call is actually established in normal call mode. This feature is not used when 'H323 Setup Mode' is 'Fast'. * Setup: Caller party tries to open early media on receiving the Setup message. * Proceeding: Calling party tries to open early media on receiving the Proceeding message. * Alerting: Calling party tries to open early media on receiving the Alerting message.	Setup, Proceeding, alerting	
H323 DTMF Path	During a call, DTMF can be sent in-band, out-of-band (H.245) or employ RFC2833.	Out-band In-band 2833	IN
DIFFSERV	This feature sets DiffServ pre-tagging for Voice packet. Note high values may cause high packet discard levels.	0~63	4
TCP Keep-Alive	The system will send a polling message every 75 seconds to assure the status of the TCP connection.	OFF ON	ON
Gatekeeper Attributes			
RAS Usage	When this feature is ON, the VoIP channel is used as the Gatekeeper.	OFF ON	OFF
RAS Multicast IP Port	This field defines the Multicast TCP/IP Port for RAS Information of Gatekeeper.	TCP/IP Port	1718
RAS Multicast IP Address	This field defines the Multicast IP address for RAS Information of Gatekeeper.	IP Address	224.0.1.41
RAS Unicast IP Port	This parameter defines the Unicast IP Port for RAS Information of Gatekeeper.	TCP/IP Port	1719
RAS Unicast IP Address	This parameter defines the Unicast IP address for RAS Information of Gatekeeper.	IP Address	82.134.80.2
RAS Keep Alive Time	To maintain a connection, the system and GateKeeper must exchange RAS Information prior to expiration of this timer.	001-999 (seconds)	120
RAS Number Plan Prefix	The number of digits for an outgoing call used in RAS Setup messages.	Max. 23 digit	9
RAS Light RRQ Usage	The system can be assigned to use the simple RRQ (Registration Request) message (ON) or the full RRQ message (OFF).	OFF ON	OFF
RAS Gateway ID	This field is the Gatekeeper ID.	128 characters	
Fail Over Usage	If an H.323 call fails to set up in the 'Call Setup No Response time' below, the system will attempt to place the call on the Failover CO/IP group also defined below.	OFF ON	OFF

Table 6.3.1-1 H.323 VOIP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Call Setup No Response Time	When Failover is enabled, the system places the fail-over call if this timer expires before the system receives a response to setup messages.	0, 3-10 Sec.	5
Failover CO Group Number	The system will place the Fail-over call employing a CO/IP line from the assigned CO/IP Line Group.	1-201 CO group	
VOIP GW Port Usage			
Q.931 Port Range	IP-Binding H.323 signaling option: Q.931 TCP Port Range.	TCP port	2048-2559
H.245 Port	IP-Binding H.323 signaling option: H.245 TCP Port Range.	TCP port	2560-3071
RAS Port Range	IP-Binding H.323 signaling option: RAS UDP Port Range.	UDP port	2048-3071
Media Port Range	IP-Binding media option: Media UDP Port Range.	UDP port	6000-19972
Data Sharing Port Range	IP-Binding option: Data Sharing TCP Port Range.	TCP port	8500-8548
External CODEC Priority configuration			
External CODEC Priority Configuration (1ST ~ 5th priority)	The system supports five Codecs and, for negotiating purposes, the priority of each can be defined. Codecs not assigned a consecutive priority are not available during negotiations with the host.	None g.711-u g.711-a g.723.1 g.729 g.729-a	none

6.3.2 T1/PRI Attributes - PGM 131

Selecting T1/PRI Attributes will display the T1/PRI Attributes data input page. Enter the Device Sequence number (refer to '*Device IP Plan*' if you want to find the sequence number) and click **[Load]** to enter T1/PRI data. Click **[Save]** button after changing Value.
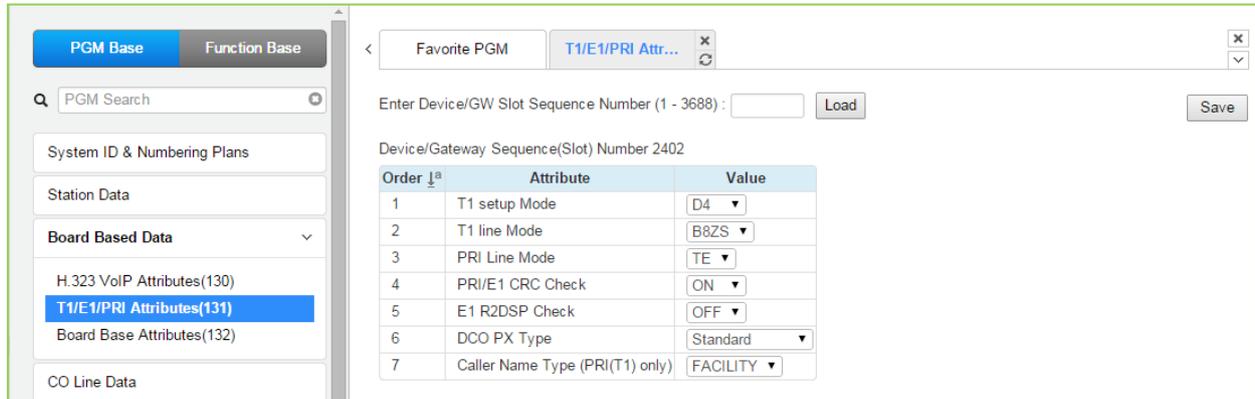
Figure 6.3.2-1 T1/PRI Attributes

Each T1/PRI module can be assigned for various attributes of the interface. The T1 interface framing and line coding can be selected and, for the PRI, TE or NT operation can be selected. Refer to the following table for a description of the features and the input required.

Table 6.3.2-1 T1/PRI ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
T1 Setup Mode	Select T1 Setup mode, D4 frame: Use In-Band Control Protocol ESF: Use Data link Message.	ESF D4	D4
T1 Line Mode	Select T1 line coding (AMI/B8ZS).	AMI B8ZS	B8ZS
PRI Line Mode	Select TE/NT Mode for PRI.	TE/NT	TE
PRI /E1 CRC Check	Enable CRC (Cyclical Redundancy Check).	OFF ON	ON
E1 R2DSP check	Enables DSP check for R2-EI or E1 interface.	OFF ON	OFF
DCO PX Type	Reserved for future usage for R2 E1 Device.	S1240 TDX1B STANDARD CONGES_DIS	STANDARD
Caller Name Type(PRI(T1) only)	If the caller has a name, the Caller Name is sent to the network according to the option (FACILITY, DISPLAY). (USA only)	FACILITY, DISPLAY	FACILITY

6.3.3 Board Base Attributes - PGM 132

Selecting Board Base Attributes will display the Board Base Attributes data input pag. Enter the Sequence number (refer to '*Device IP Plan*' if you want to find the sequence number) and click **[Load]** to enter attribute values. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of devices when saved. Click **[Save]** button after changing Value.
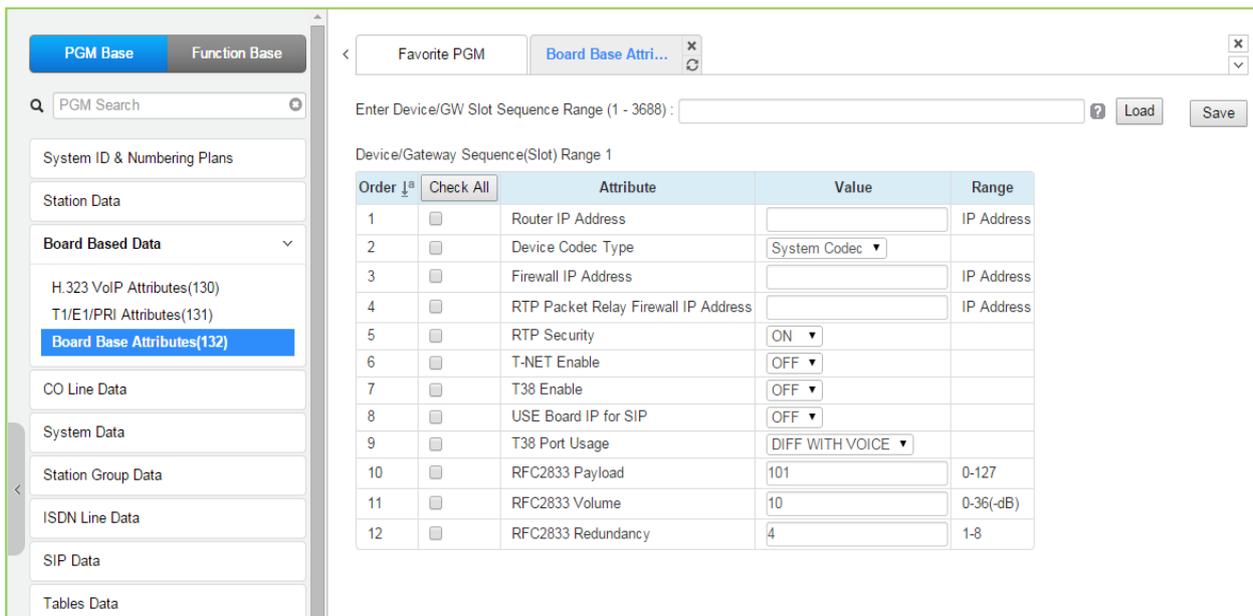
Figure 6.3.3-1 Board Base Attributes

Appliances (Devices and IP Phones) can be connected to the iPECS over a managed WAN without the need to employ a VoIP channel. In this case, the system does not implement security (IPSec) or QoS treatment over the link. To implement the managed WAN connectivity, the iPECS must be assigned with the IP address of the router for all appliances that may attempt a point-to-point connection over the managed WAN, including devices on the iPECS LAN. Note that if the device's Router IP address is not defined, the system will use the Router IP address defined in System IP Plan.

The default codec employed by each device can be specifically defined as G.711, G.729, G.723, G.722, or the system default codec assigned.

Appliances include support for SRTP (Secure Real-Time Protocol), which employs Advanced Encryption Standard (AES) to secure RTP packets. If RTP security is enabled then IPSEC or SRTP is implemented for RTP packet.

Table 6.3.3-1 BOARD BASE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Router IP Address	Enter the Default gateway (Router) IP address associated with the selected devices.	IP Address	
Device Codec Type	Select the Codec type for each device.	G.711 G.723.1 G.729 G.722	System Codec

Table 6.3.3-1 BOARD BASE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
		System Codec	
Firewall IP Address	Enter the Firewall IP address of the selected devices.		
RTP Packet Relay Firewall IP Address	When a VoIP channel of a VOIM is used for both RTP Relay for remote devices and SIP signaling, enter the Firewall IP address associated with the VOIM.		
RTP Security	SRTP implements AES (Advanced Encryption Standard) for packets between other devices with RTP Security enabled. To reduce bandwidth use or to use an external VPN, this parameter must be OFF.	OFF ON	ON
TNET Enable	When a module or station is to be connected in a Centralized Control network (TNET), the device must be enabled for TNET operation.	OFF ON	OFF
T38 Enable	FAX over IP Lines is supported when T38 mode is ON.	OFF ON	OFF
USE Board IP for SIP	When the VoIP channels of a VOIM are used for SIP, such as a SIP Trunk, the IP address of the VOIM must be used for the SIP messages (ON).	OFF ON	OFF
T38 Port Usage	The TCP/IP port employed for T38 packets is negotiated and the system can allow the port to be the same as or different from the preceding Voice packets. The system can also employ NAT to determine the port.	Different from Voice Same as Voice NAT Triggered	DIFF WITH VOICE
RFC2833 Payload	Payload value for RFC2833	0~127	101
RFC2833 Volume	Volume(Gain) of RFC2833 payload	0~36(-dB)	10
RFC2833 Redundancy	Sending time of end packet of RFC2833	1~8	4

6.4 CO Line Data

Selecting the CO Line Data program group returns the sub-menu displayed in the left frame as shown in the following figure.
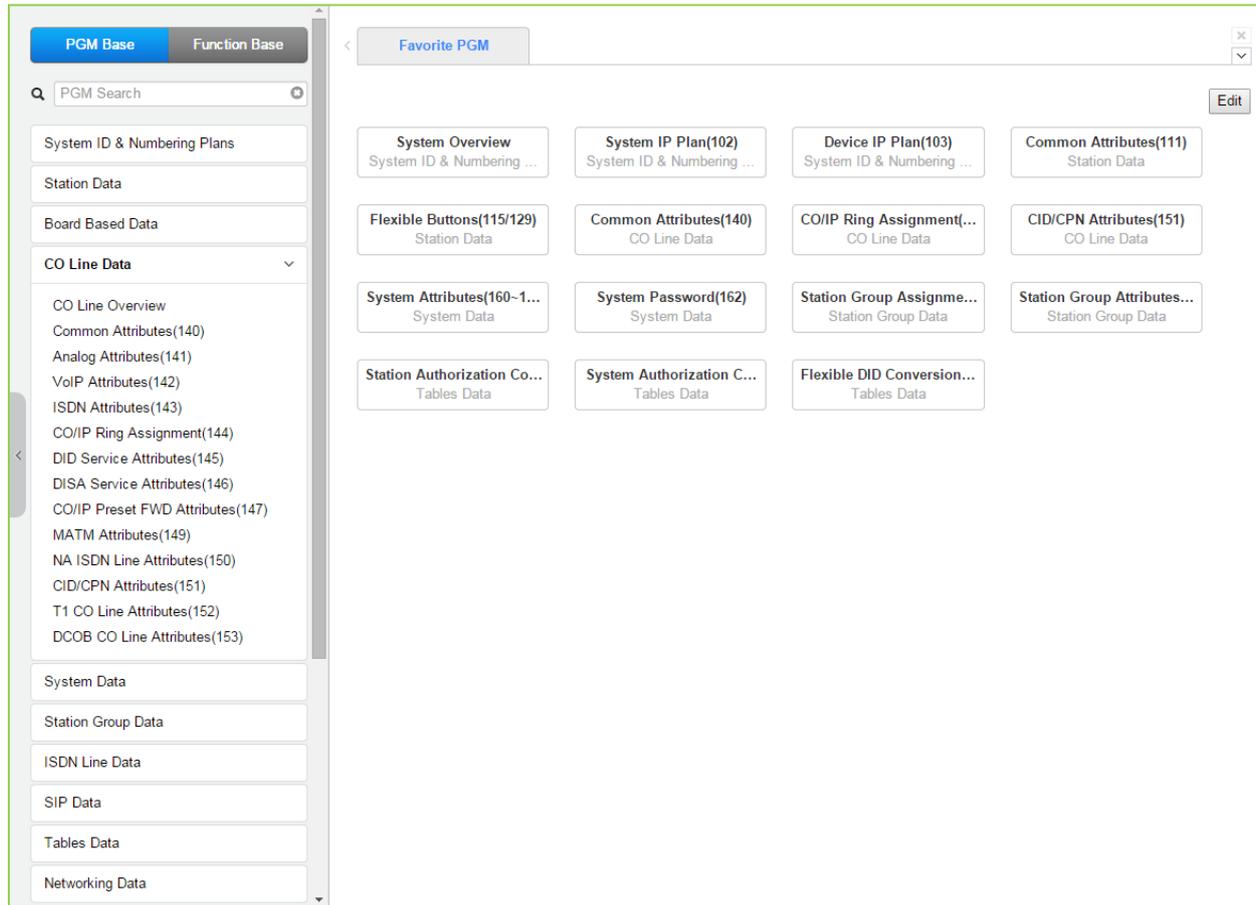
Figure 6.4-1 CO Line Data Main Page

Selecting CO Line Overview will show the current overview of CO line. The information may vary depending on installed modules.

Figure 6.4-2 CO Line Overview

6.4.1 Common Attributes - PGMS 140

Selecting Common Attributes will display the Common Attributes data input page. Enter a valid CO range and click **[Load]** to enter the Common Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.
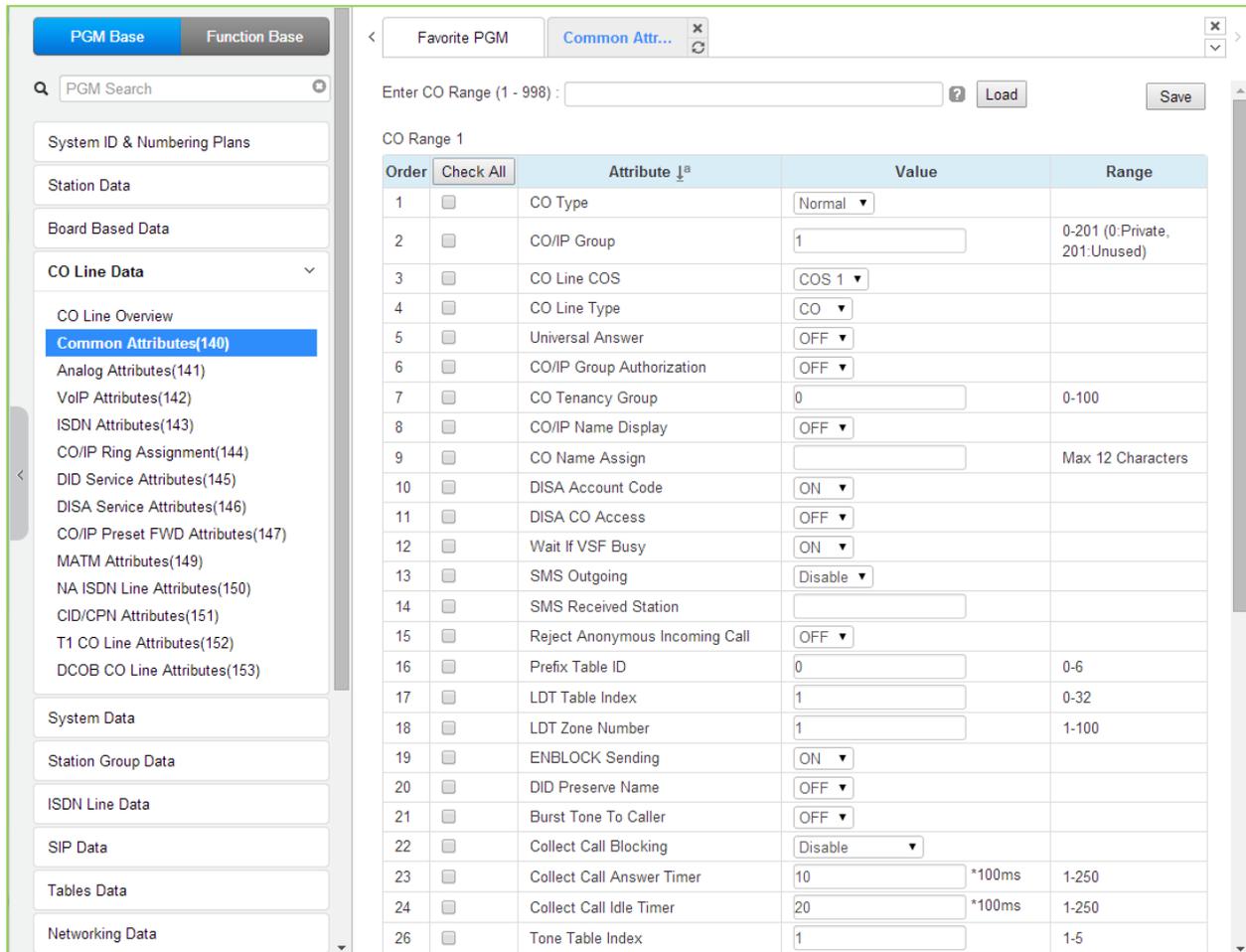
Figure 6.4.1-1 Common Attributes

Common Attributes define various characteristics of CO lines facilities under control of the system. Most characteristics require an ON/Off setting; refer to the following table. Specific descriptions for Class-of-Service are provided in Table 6.4.1-2.

Table 6.4.1-1 Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CO Type	Each CO Line is assigned a type: Normal, DID, TIE Line, or Unused.	Normal DID TIE Unused	DID
CO/IP Group	Each CO/IP Line is assigned to a group; grouping should be based on the Line type, technology and COS.	0-201 (0: Private, 201: Unused)	1

Table 6.4.1-1 Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CO Line COS	Each CO/IP Line is assigned a Class-of-Service that interacts with the Station COS, refer to Table 6.4.1-2 CO COS 1: Station COS applies CO COS 2: Exception Table A governs CO COS 3: Exception Table B governs CO COS 4: Restricts LD calls & Exception Table C CO COS 5: Overrides Station COS 2~6 with no restrictions	COS 1~5	COS 1
CO Line Type	Each CO/IP Line can be assigned a type that indicates connection to a CO Line or a PBX/CTX Line.	PBX CO	CO
Universal Night Answer	Universal Night Answer (UNA) allows any station user to answer a call on the CO/IP line by dialing the UNA code.	OFF ON	OFF
CO/IP Group Authorization	Each CO/IP Group can be assigned to require the user enter an Authorization Code.	OFF ON	OFF
CO Tenancy Group	Only stations in the assigned ICM Tenancy group are permitted access to the defined CO Line.	0-100	0
CO/IP Name Display	The IP Phone display can indicate the CO line/IP channel number or the twelve (12) character name, if assigned below.	OFF ON	OFF
CO Name Assign	Each CO Line and IP group can be assigned a twelve (12) character name for display purposes.	Max. 12 characters	
DISA Account Code	With DISA Account Code "ON", users are required to enter a DISA Authorization code. Codes are entered in '*Authorization Code Table section*'.	OFF ON	ON
DISA CO Access	When enabled, DISA users may access the CO/IP Lines in the system.	OFF ON	OFF
Wait If VSF Busy	When a DID/DISA call assigned to receive a System announcement arrives and all channels are busy, the call may wait with Ring back until a channel is available (ON) or route to the DID/DISA Destination -PGM 167-.	OFF ON	ON
SMS Outgoing	Each CO line can be assigned to support PSTN SMS.	Disable Enable	Disable
SMS Received Station	When a PSTN SMS is received, the system delivers the message to the assigned station.	Station	….
Reject Anonymous Incoming Call	When REJECT ANONYMOUS is enabled, incoming calls without Caller ID are rejected.	OFF ON	OFF
Prefix Table ID	When the Prefix Dialing Table (PGM 206) feature is used, this Prefix Table ID is employed. If this value is set to 0, the Prefix Table feature is disabled.	0 – 6 0: disable	0
LDT Table Index	When LCR is configured, this LDT Table index is referenced for outgoing calls on this CO/IP Line.	0-32 (LDT Table index) 0: Unused	1
LDT Zone Number	If the LDT Zone Number of a LDT table (LCR LDT(221)) is equal to this value, the LDT table is available to this CO Line.	1-100	1
ENBLOCK Sending	This entry determines if the system sends dialed digits to the ISDN line as they are received (OFF), or collects all digits and forwards them in a block. (ON).	OFF ON	ON

Table 6.4.1-1 Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
DID preserve Name	For DID lines, the CLI (Calling Line Identification) is normally displayed only during ringing. If enabled here, the CLI will be displayed for the entire call duration.	OFF ON	OFF
Burst Tone to Caller	When DID Call Wait is assigned for the station, the system can send audible off-hook signals as a short burst of tone for DID calls from this CO/IP Line.	OFF ON	OFF
Collect Call Blocking	This parameter defines the type of Collect Call Blocking supported for E1 CO lines with R2 signaling. (Intended for Brazil only)	Disable Double Answer Double Answer w/Indication	Disable
Collect Call Answer Timer	These parameters determine treatment of an incoming call when Collect Call Blocking Double Answer is	1-250 (100ms)	10
Collect Call Idle Timer	assigned. In this case, the system answers the call and maintains the connection for the Call Answer time then the system will disconnect for the Call Idle time before finally reconnecting the call.	1-250 (100ms)	20
Gain table index	One of three Gain tables can be configured for TDM connections.	1-3	1
Tone table index	One of the five Tone table can be configured for use with the CO/IP Line.	1-5	1
Digit Conversion Table index	One of the 100 Digit Conversion Tables can be configured for use with this CO/IP Line for incoming DID call digit conversion.	1-32	1
Prepaid Call	The Pre-paid Call feature must be enabled for use by a station. User can set the budget for the station call charge in advance.	OFF ON	OFF
Prepaid Money (0-999999) & Used Prepaid Money	An amount can be associated with the station as prepayment for outgoing calls. The assigned "Pre-paid Money" is reduced by the calculated call cost (Call Metering or cost/minute). This parameter displays the remaining budget for outgoing calls. Prepaid Money is often used in small hospitality businesses. The "Used Pre-paid Money" displays the money that has been used.	0-999999 0-999999	0
Automatic call recording destination	When desired, the IPCR server can be used to record calls on the CO/IP Line. This value defines the Station Group of the IPCR server. The IPCR Agent object, see PGM 237, must be assigned as CO Line for proper operation.		
ICLID Usage	An incoming call can be routed to the destination based on Calling Name. Disable - ICLID feature is disabled CLI - ICLID feature is operated with CLI Name - ICLID feature is operated with Name	Disable, CLI, Name	CLI
PBX Code Insertion for Emergency call	System can insert PBX code automatically if pre-configured when a user make an emergency call.	Disable, PBX code1~4	Disable
Timer Attributes			
CO Cut Off Timer	When Call Time Restriction is assignment to the station, this timer defines the allowed call duration.	00-99 minutes 00 = disable	0

Table 6.4.1-1 Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
DISA Delay Timer	Once answered, the system attaches a DTMF receiver to the DISA call to receive caller-dialed digits. User can set the delay time before attaching the DTMF receiver to the CO line. (Intended for Russia only)	0-9 seconds	0
DISA Answer Timer	When a call is received on a CO line with DISA service, the system will answer the call when the DISA Answer Timer expires. (Intended for Russia only)	0-9 seconds	0
Tone Attributes			
CO Line MOH	A held call can be connected to one (1) of ten (10) possible audio sources while on Hold as Music-on-Hold (MOH).	Refer to Sys Hold, Internal/External Music1, External Music 2, VSF MOH, SLT MOH1~5, VSF MOH2~3	Refer to Sys Hold
CO Dial Tone	ISDN Lines may provide a digital signal rather than actual tones. In this case, the iPECS can provide the tones. If the ISDN provides the tone, the Tone is "ON", for an iPECS UCP-generated tone, the tone is set to "OFF".	OFF ON	ON
CO Ring Back Tone		OFF ON	OFF
CO Error Tone		OFF ON	OFF
CO Busy Tone		OFF ON	OFF
CO Line Dial Tone Source	One of eleven sources can be defined as dial tone for use by the CO line.	Dial Tone, Internal/External Music1, External Music 2, VSF MOH, SLT MOH1~5, VSF MOH2~3	Dial Tone
CO Line Ring Back Tone Source	One of eleven sources can be set as ring back tone for use by the CO line.	Ring Back Tone, Internal/External Music1, External Music 2, VSF MOH, SLT MOH1~5, VSF MOH2~3	Ring Back Tone
COL Ring Tone	One of sixteen Ring Tones can be set for use by this CO Line.	00-16 0: NA	0

Table 6.4.1-2 STATION/CO LINE TOLL RESTRICTIONS

	CO COS 1	CO COS 2	CO COS 3	CO COS 4	CO COS 5
STA COS 1	No Restriction	No Restriction	No Restriction	Only Local Call (LD code/counter) & Table C	No Restriction
STA COS 2	Exception Table A governs the dialing	Exception Table A governs the dialing	No Restriction	Only Local Call (LD code/counter) & Table C	No Restriction
STA COS 3	Exception Table B governs the dialing	No Restriction	Exception Table B governs the dialing	Only Local Call (LD code/counter) & Table C	No Restriction
STA COS 4	Exception Table A & B governs the dialing	Exception Table A governs the dialing	Exception Table B governs the dialing	Only Local Call (LD code/counter) & Table C	No Restriction
STA COS 5	Local Call only (LD Code, "1" or "0") and Table C	Local Call only (LD Code, "1" or "0") and Table C	Local Call only (LD Code, "1" or "0") & Table C	Only Local Call (LD code/counter) & Table C	No Restriction
STA COS 6	Only Local Call (LD code/counter) & Table C	Only Local Call (LD code/counter) and Table C	Only Local Call (LD code/counter) & Table C	Only Local Call (LD code/counter) & Table C	No Restriction
STA COS 7	In-house dialing only	In-house dialing only	In-house dialing only	In-house dialing only	In-house dialing only
STA COS 8	Exception Table D governs the dialing	Exception Table D governs the dialing	No Restriction	Only Local Call (LD code/counter) & Table C	No Restriction
STA COS 9	Exception Table D governs the dialing	Exception Table D governs the dialing	No Restriction	Only Local Call (LD code/counter) & Table C	No Restriction
STA COS 10	Exception Table D & E governs the dialing	Exception Table D & E governs the dialing	No Restriction	Only Local Call (LD code/counter) & Table C	No Restriction
STA COS 11	Exception Table A & B and D & E governs the dialing	Exception Table A & B and D & E governs the dialing	No Restriction	Only Local Call (LD code/counter) & Table C	No Restriction

6.4.2 Analog Attributes - PGMS 141

Selecting Analog Attributes will display the Analog Attributes data input page. Enter a valid CO range and click **[Load]** to enter the Analog (PSTN) Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.
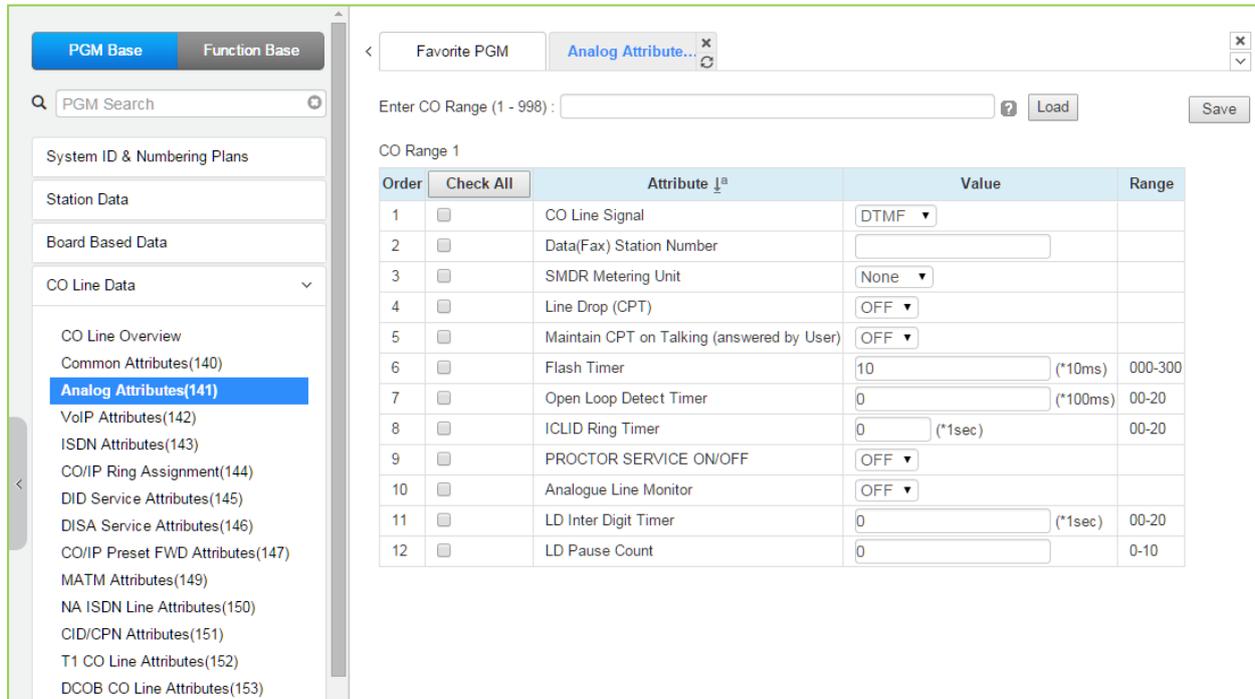
Figure 6.4.2-1 Analog Attributes

Analog Attributes define various characteristics of Analog CO Lines under control of the system. Most characteristics require an On/Off setting; refer to the following table. Specific descriptions for CO Line (SMDR) Call Metering tones are provided in Table 6.4.2-2.

Table 6.4.2-1 Analog ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CO Line Signal	Each analog CO Line can be assigned to send either DTMF or Pulses for dialed digits to the PSTN.	DTMF, Pulse	DTMF
Data(Fax) Station Number	Each CO line can be assigned to recognize a FAX call when a specified station answers.	Station Number	
SMDR Metering Unit	This field selects the call-metering signal from the PSTN to indicate call cost, refer to Table 6.4.2-2.	See Table 6.4.2-2	None
Line Drop (CPT)	Each CO Line can be programmed to disconnect if error tone is detected.	OFF ON	OFF
Maintain CPT on talking (Answer by User)	The system can continuously monitor the CO Line during a call and, if error tone is detected, drop the call.	OFF ON	OFF
Flash Timer	User can set the flash time.	000-300 (*10 msec)	50

Table 6.4.2-1 Analog ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Open Loop Detect Timer	This filed sets the duration of open loop that will be recognized as a "Disconnect Signal".	00-20 (*100 msec)	4
ICLID Ring Timer	When a call is received, the system may use ICLID (Incoming Caller ID) to route the call. The system will delay routing a call for this duration while awaiting ICLID. Set 00 for disabling ICLID routing.	00-20 (*Sec)	4
Proctor Service ON/OFF	Each analog CO line can be assigned to send the station number as DTMF digits for Proctor service.	OFF ON	OFF
Analog Line monitor	The system can monitor and report faults on an analog CO Line.	OFF ON	ON
LD Inter Digit Timer	This time is for setting the duration between digits for LD (Long Distance) call.	00-20 (*1sec)	0
LD Pause count	This count sets the number of Pause for Long distance call.	0-10	0

Table 6.4.2-2 CALL METERING FUNCTION

ENTRY	SMDR METERING UNIT TYPE
00	- None
01	- 50 Hz
02	- 12 KHz
03	- 16 KHz
04	- Singular Polarity Reverse (SPR)
05	- Plural Polarity Reverse (PPR)
06	- No Polarity Reverse (NPR)

6.4.3 VoIP Attributes - PGMS 142

Selecting VoIP Attributes will display the VoIP mode data input page. Enter a valid CO range and click **[Load]** to enter the VoIP Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.
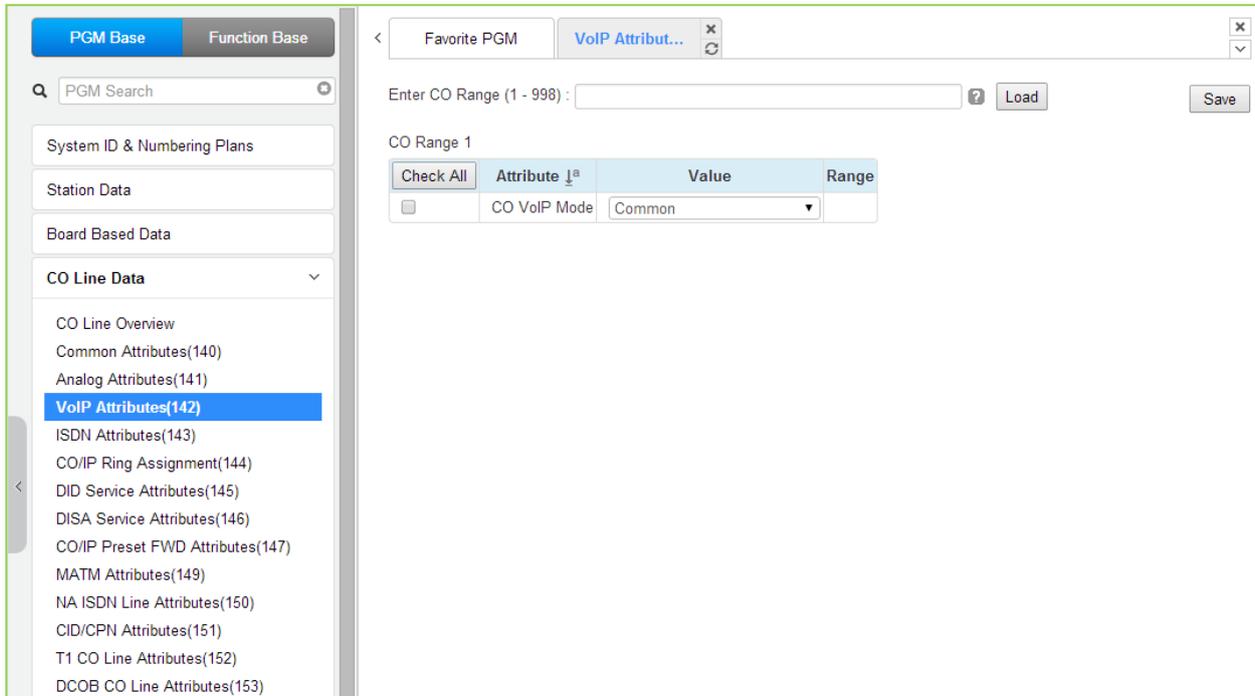
Figure 6.4.3-1 VoIP Attributes

Table 6.4.3-1 VoIP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CO VoIP Mode	The VoIP channels can support iPECS, H.323 or SIP protocols. This field selects the protocol for the VoIP channel(s) according to the user's situation.	COMMON H323 only SIP only RTP-Packet-Relay only H323 & RTP-Packet-Relay SIP & RTP-Packet-Relay	COMMON

6.4.4 ISDN Attributes - PGMS 143

Selecting ISDN Attributes will display the ISDN Attributes data input page. Enter a valid CO range and click **[Load]** button to enter the ISDN Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.
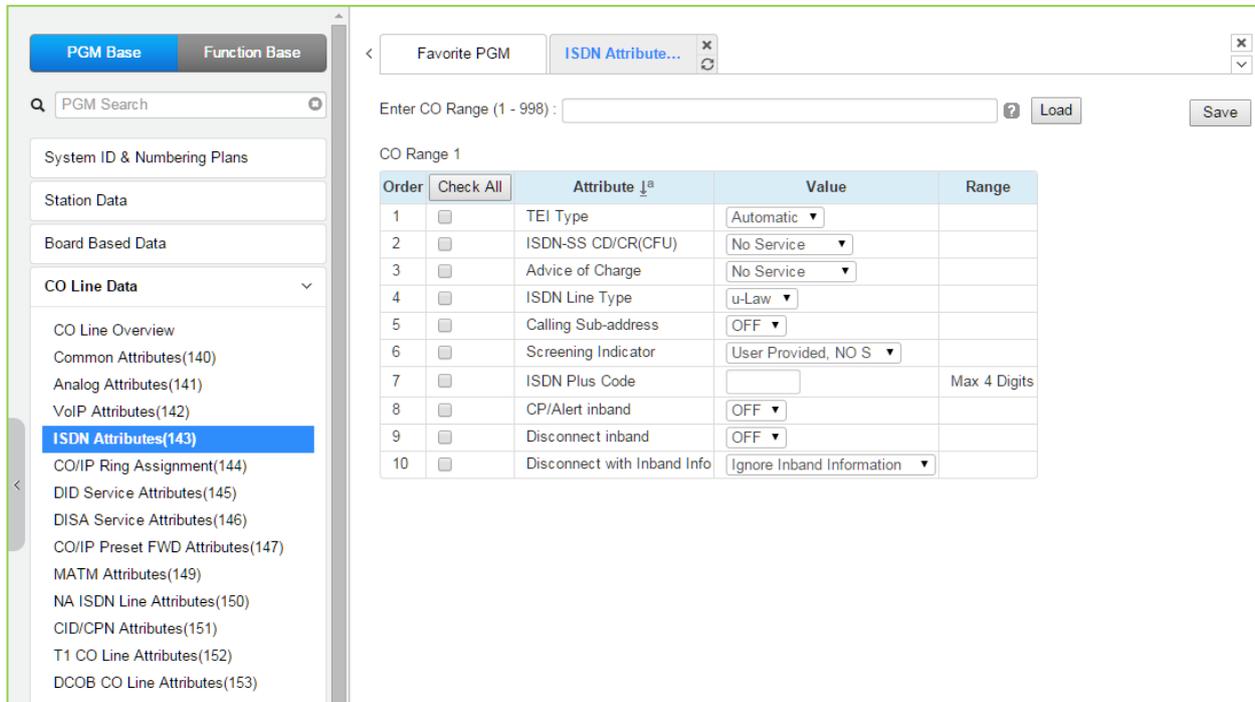
Figure 6.4.4-1 ISDN Attributes

This programming assigns attributes as shown in the following table for the ISDN lines in the entered range.

Table 6.4.4-1 ISDN ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
TEI Type	The TEI (Terminal Endpoint Identifier) is a unique identifier for each device connected to the ISDN line. When the system shares an ISDN connection with other devices, the TEI should be automatic to avoid conflicting with the other connected devices. Otherwise, the Fixed identifier option should be employed.	Fixed, Automatic	Automatic
ISDN – SS CD/CR(CFU)	Permits a user to access to ISDN Supplementary Call Deflection Service. (Except USA version).	No Service, Call Deflection, Call Rerouting	No Service
Advice of Charge	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 AOC, Italy & Spain, Finland, Australia, Belgium, ETSI STD	No Service

Table 6.4.4-1 ISDN ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
ISDN Line Type	The system will encode voice using the A-law or u-law PCM format to match the installed ISDN backbone.	μ-Law, A-Law	μ-Law
Calling Sub-address	For outgoing calls, the user's station number may be included in the ISDN call SETUP message Sub-address field.	OFF ON	OFF
Screening indicator	The ISDN Screening Indicator can be configured.	User Provided-No Service, User Provided-Pass, User Provided-Fail, Network Provided.	User Provided, No Service
ISDN PLUS Code	When the incoming CLI starts with "+", the value assigned for this attribute is inserted in place of the "+" sign.	Max. 4 digits	
CP/Alert inband	When enabled, the system will send progress indication with in-band information in Call proceeding and Alerting messages.	OFF ON	OFF
Disconnect inband	When enabled, the system will send progress indication with in-band information in Disconnect messages.	OFF ON	OFF
Disconnect with Inband Info	When system receives DISCONNECT message with Inband information from PSTN, the following option can be applied: -. Ignore Inband information: Line is disconnected -. Bypass Inband information: Progress Information is transferred -. Wait Release: System wait to send Release message till the other party is disconnected	Ignore Inband information, Bypass Inband information, Wait Release	Ignore Inband information

6.4.5 CO/IP Ring Assignment - PGM 144

Selecting CO/IP Ring Assignment will display the CO/IP Ring Assignment data input page. Enter a valid CO range and click **[Load]** to enter the CO/IP Ring Assignment data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.


Figure 6.4.2-1 CO/IP Ring Assignment

Each "Normal" CO/IP line is assigned to signal (Ring) a station, station group or VSF Announcement for an incoming call. Separate ring assignments can be set for Day, Night, and Timed Ring mode.

A delay from 1 to 9 Ring cycles can also be assigned, based on this assignment, the station/Station group will receive audible ring after a delay of the number of Ring cycles entered. In addition, when assigned to ring a VSF Announcement, the system can be programmed to disconnect after the announcement, 'Auto Drop'.

When CO/IP Lines are programmed to ring the built-in Auto Attendant, a delay of 0 to 30 seconds can be assigned as the AA Ring Time. The delay allows stations to be assigned Ring and to answer prior to signaling the AA. At expiration of the AA Ring Time, the call is sent to the assigned VSF announcement or announcement 1 when no VSF announcement is assigned.

6.4.6 DID Service Attributes - PGM 145

Selecting DID Service Attributes will display the DID Service Attributes data input page. Enter a valid CO range and click **[Load]** button to enter the DID Service Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.
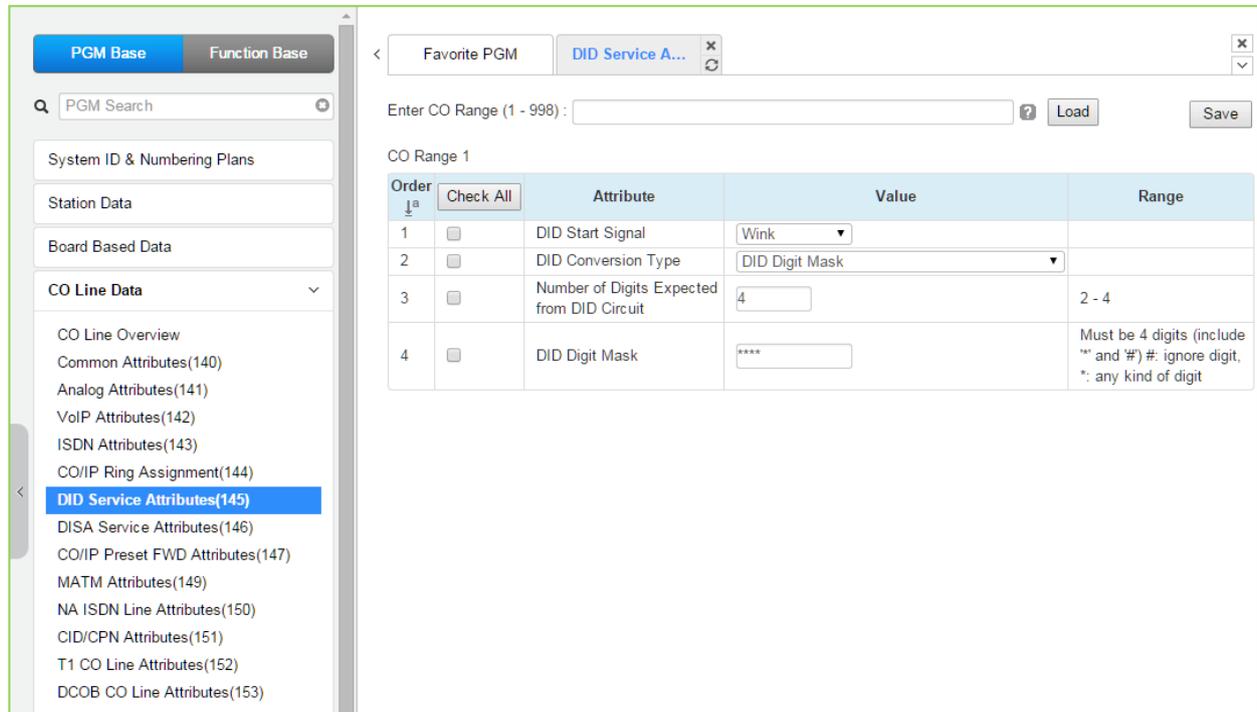
Figure 6.4.6-1 DID Service Attributes

DID lines can be assigned as the type of "Start" signaling and treatment of received digit. Digits can be used "as is" to route the call within the system, digits can be converted and used to route the call, or digits can be converted to a Table index to determine the call routing from a Table look-up. Refer to the following table for additional description of attributes and values.

Table 6.4.6-1 DID SERVICE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
DID Start Signal	Assigns the type of DID start signaling: Immediate, Wink or Delayed dial for the CO/IP Line.	Immediate, Wink, Delayed dial	Wink
DID Conversion Type	The received DID digits can be treated to determine call routing, simple conversion (Convert), use "as is" (no treatment), or modify using Flexible DID Conversion Table (Look-up).	DID Digit Mask, Use 'as is', Modify using Flexible DID conversion table	DID Digit Mask
Number of Digits Expected from DID Circuit	The number of digits expected from the PSTN DID circuit.	2~4	4
DID Digit Mask	DID digit modification sequence: "#" deletes the digit, "*" accepts the digit "as is", a digit (0~9) replaces the digit. The modification is based on the position of the digit (1~4) in the received number.	(0~9, *, #)	****

6.4.7 DISA Service Attributes - PGM 146

Selecting DISA Service Attributes will display the DISA Service Attributes data input page. Enter a valid CO range and click **[Load]** to enter the DISA Service Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.
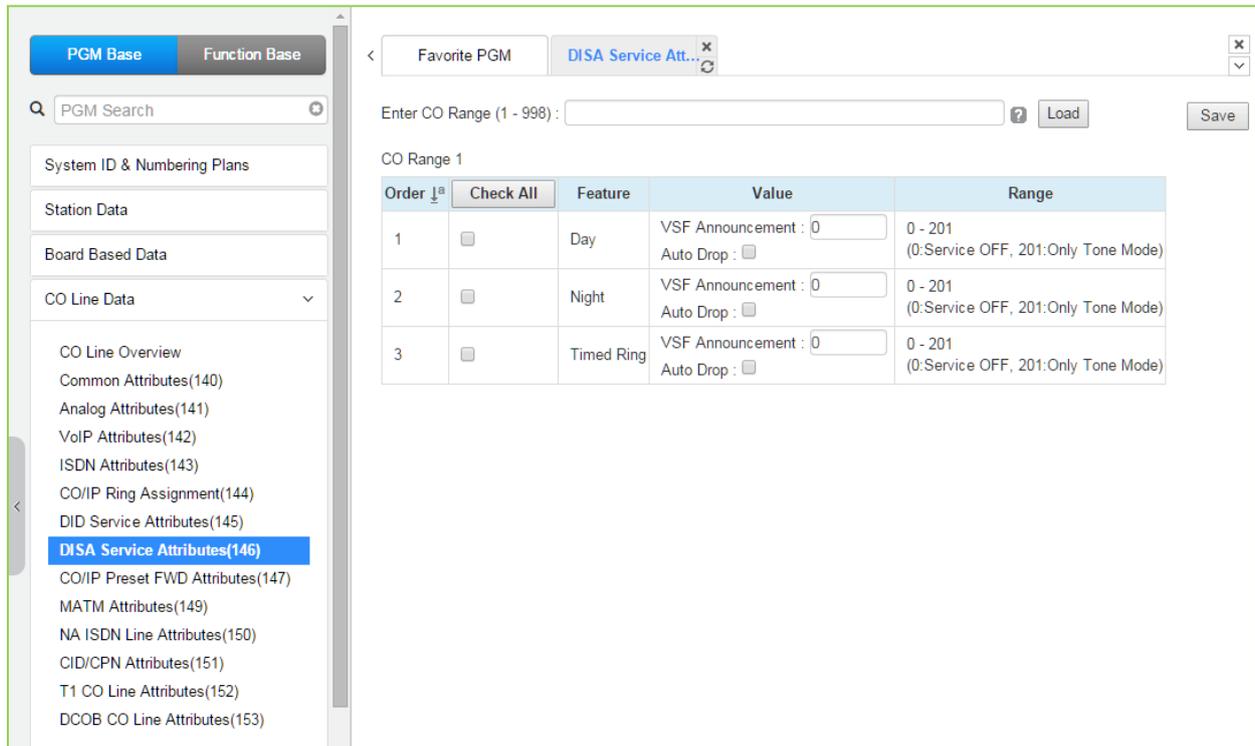
Figure 6.4.7-1 DISA Service Attributes

DISA Service can be enabled on CO lines based on the system operation mode (Day, Night, and Timed). DISA calls may be routed to dial tone and await user dialing (simple routing) or through a multi-layered Audio Text menu assigning a System announcement and a Customer Call Route (CCR) Table Index. The system can be programmed to disconnect after the announcement, 'Auto Drop', or follow the CCR Table routing with a user-recorded System announcement requesting specific inputs from the user.

6.4.8 CO/IP Preset Forward Attributes - PGM 147

Selecting CO/IP Preset Forward Attributes will display the CO/IP Preset Forward Attributes data input page. Enter a valid CO range and click **[Load]** to enter the CO/IP Preset Forward Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.
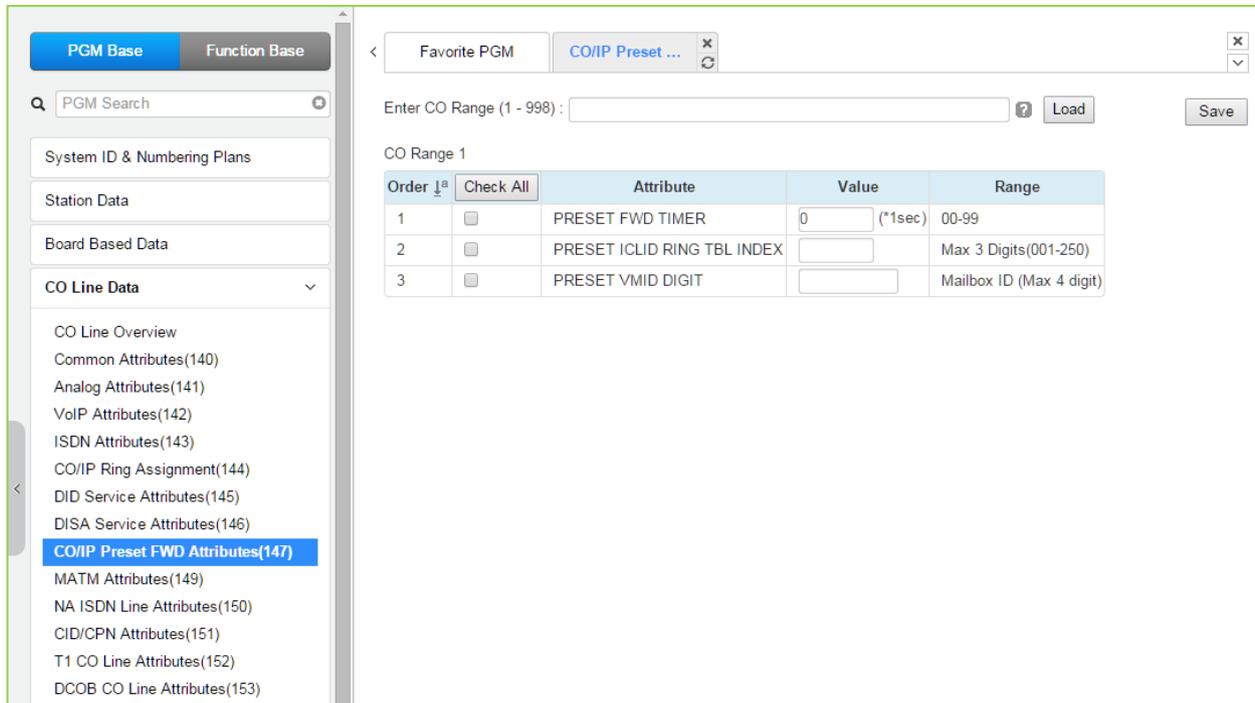
Figure 6.4.8-1 CO/IP Preset Forward Attributes

The CO/IP Preset Call Forward feature enables a CO/IP call to initially ring at multiple stations and forward to a pre-determined destination. A separate timer can be defined for each CO/IP line for no-answer. The destination can be any index to the '*ICLID Ring Table section*' or a Voice Mailbox. The VMID (Voice Mail ID) field selects a specific mailbox when the call forwards to an external VM group.

Table 6.4.8-1 CO/IP PRESET FORWARD ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Preset FWD Timer	An incoming call, which remains unanswered for this timer, is routed as defined in the '*ICLID Ring Table Index section*'.	00-99 (Sec.)	0
Preset ICLID Ring Table Index	If an incoming call remains unanswered after the Preset Fwd time above, the call is routed as defined in the '*ICLID Ring Table index*' defined here.	001-250	
Preset VMID Digit	Each CO/IP line can be assigned a VMID (Voice Mail Id) that is sent to the external VM group to identify the desired Mailbox for the call.	Mailbox ID (Max. 4 digits)	

6.4.9 MATM Attributes - PGM 149

Selecting MATM Attributes will display the MATM Attributes data input page. These attributes are only valid when the Nation Code is Korea. Enter a valid CO range and click **[Load]** to enter the MATM Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.

These attributes are available for Country code 82.
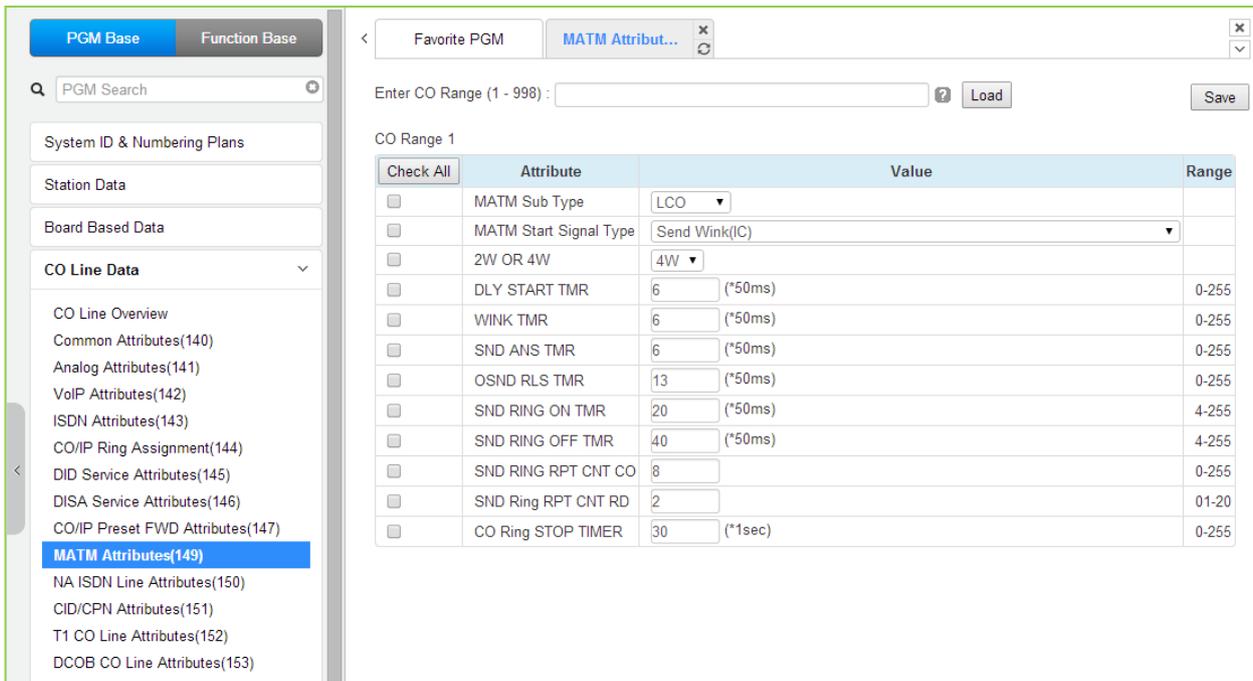

Figure 6.4.9-1 MATM Attributes

Table 6.4.9-1 MATM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
MATM Sub Type	This is the analog trunk type for the MATM option board. COIU : LCO RDIU : RD (ring down, only for national army) LDIU : LD (loop dial, only for national army) EMC = E&M continuous, EMD = E&M discontinuous)	LCO, RD, LD, EMC, EMD, Unused	LCO
MATM Start Signal Type	This is the signal type for the analog trunk. This should be set with alternate trunk type. These can be set by testing with available values.	Send Wink (IC), Wait Seize Ack(OG), Send Wink (IC) and Wait Seize Ack(OG), Send Sub Answer (IC) and Wait Sub Answer(OG)/ Send Wink and	Send Wink (IC)

Table 6.4.9-1 MATM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
		Send Sub Answer(IC)/ Wait Ack and Wait Sub Answer(OG)/ Send Wink and Send Sub Answer(IC) & Wait Wink and wait sub answer(OG)/ Unused Type	
2W or 4W	For E&M Lines, this value should be set as 2 wire or 4wire to match the line.	2W/ 4W	4W
DLY Start Timer	This is timer value for analog signaling.	0~255	6
Wink Timer	This is timer value for analog signaling.	0~255	6
SND ANS Timer	This is timer value for analog signaling.	0~255	6
OSND RLS Timer	This is timer value for analog signaling.	0~255	13
SND RING ON Timer	This is timer value for analog signaling.	4~255	20
SND RING OFF Timer	This is timer value for analog signaling.	4~255	40
SND RING RPT CNT CO	This is timer value for analog signaling.	0~255	8
SND RING RPT CNT RD	This is timer value for analog signaling.	01~20	2
CO Ring STOP Timer	This is timer value for analog signaling.	0~255	30

6.4.10 NA ISDN Line Attributes - PGM 150

Selecting NA (North America) ISDN Line Attributes will display the NA ISDN Line Attributes data input page. Enter a valid CO range and click **[Load]** to enter the NA ISDN Line Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.
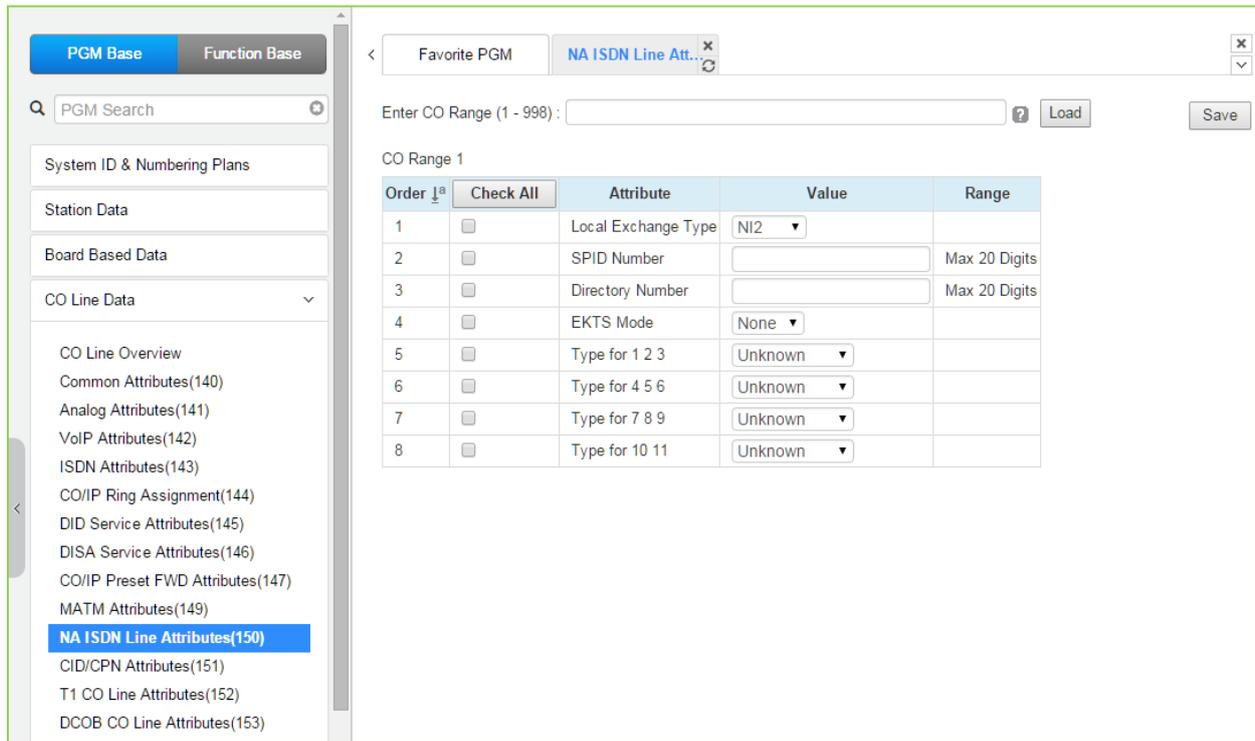

Figure 6.4.10-1 NA ISDN Line Attributes

To comply with the North American ISDN standards, certain attributes must be defined for the system. These include Directory (telephone) Number and Service Profile Id (SPID) for the device. Note that this programming is required only for "Country Code" 1, USA installations. Refer to the following table for information on individual attributes.

Table 6.4.10-1 NA ISDN LINE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Local Exchange Type	The type of ISDN determines several specifics of the protocol and is required for proper operation.	NI 1, NI 2, 5 ESS, Nortel	NI 2
SPID Number	The Service Profile Identifier (SPID) is a number assigned to a fully initializing ISDN terminal and enables the Stored Program Control switching System (SPCS) to identify the ISDN terminal at layer 3 of the D-channel signaling protocol. The SPID is a free-formatted numeric string composed of 9 to 20 numeric {0-9} and International Alphabet (IA5) characters. The SPID uniquely identifies a	Max. 20 digits	

Table 6.4.10-1 NA ISDN LINE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	particular set of subscription parameters assigned to a TSP.		
Directory Number	Initializing terminals are required to store a 7-digit DN in order to perform the compatibility checking procedures that are part of call termination.	Max. 20 digits	
EKTS Mode	The EKTS (Electronic Key Telephone Service) terminal permits a user to operate those features that are specific to EKTS, as well as voice features that may function distinctly in the EKTS environment. EKTS allows a DN to be shared by more than one terminal on the same or on different interfaces.	None, EKTS	None
Type for 1/2/3	ISDN CALLED NO uses the International format, National format, Network format, Subscriber format, or Abbreviated format when the user dials 1~3digits.	Unknown, International, National, Network, Subscriber, Abbreviated	Unknown
Type for 4/5/6	ISDN CALLED NO is constructed with International format, National format, Network format, Subscriber format, or Abbreviated format when the user dials 4~6 digits.	Unknown, International, National, Network, Subscriber, Abbreviated	Unknown
Type for 7/8/9	ISDN CALLED NO is constructed with International format, National format, Network format, Subscriber format, or Abbreviated format when the user dials 7~9 digits.	Unknown, International, National, Network, Subscriber, Abbreviated	Unknown
Type for 10/11	ISDN CALLED NO is constructed with International format, National format, Network format, Subscriber format, or Abbreviated format when the user dials more than 10 digits.	Unknown, International, National, Network, Subscriber, Abbreviated	Unknown

6.4.11 CID/CPN Attributes - PGMS 151

Selecting CID/CPN Attributes will display the CID/CPN Attributes data input page. Enter a valid CO range and click **[Load]** to enter the CID/CPN Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.
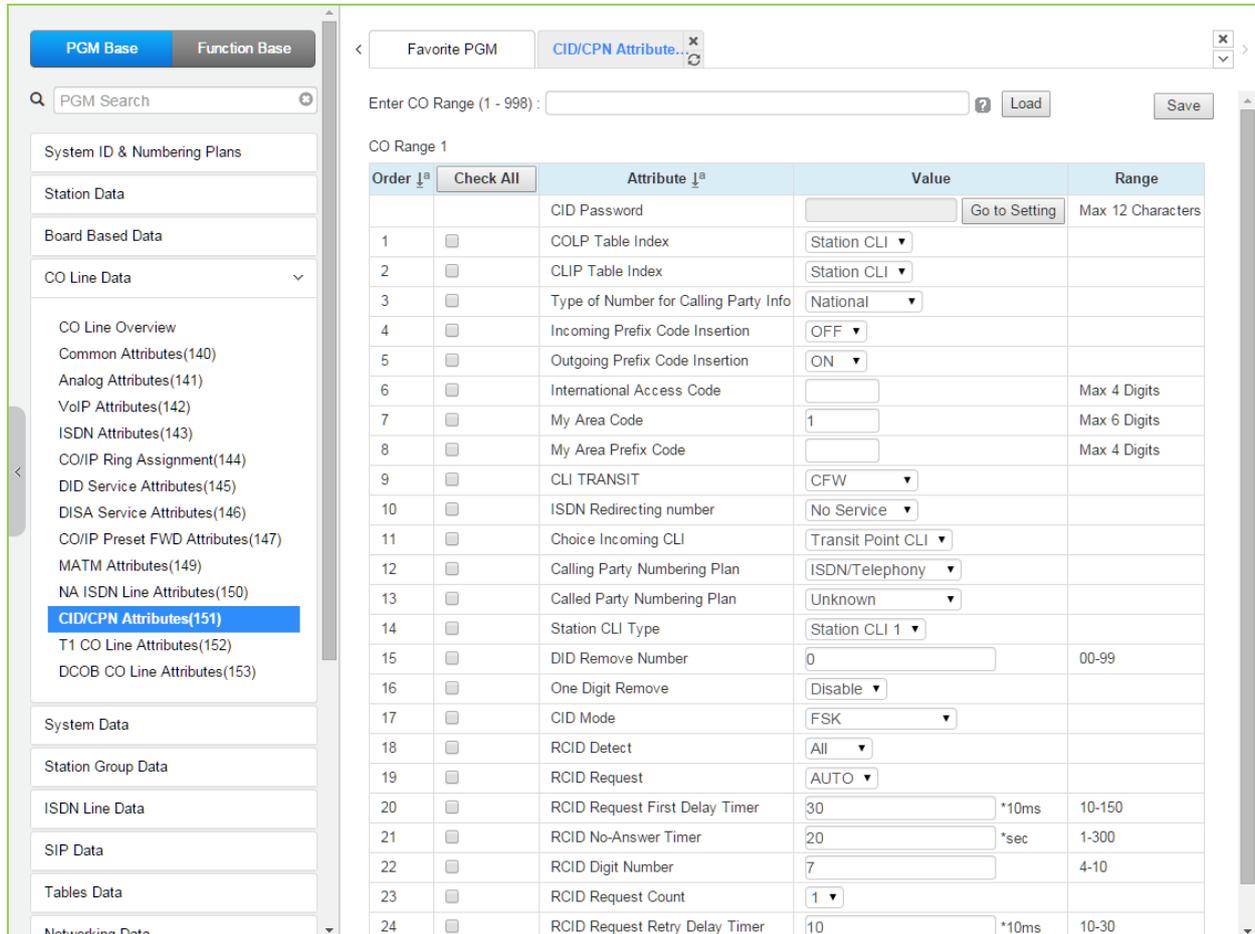
Figure 6.4.11-1 CID/CPN Attributes

Refer to the following table for information on individual attributes.

COLP: Connected Line Presentation / CLIP: Calling Line Identification Presentation

Table 6.4.11-1 CID/CPN ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
COLP Table Index	When an incoming call on a CO/IP Line is answered, the system will send connected party ID using the number from the CLIP/COLP Table entry defined by this parameter with the "Station CLI" as a suffix. The "Station CLI" is the Station CLI Type (1~5) entered below. If Station CLI is selected in the table, only the Station CLI is sent.	N/A, 0-49, Station CLI	Station CLI
CLIP Table Index	When a call is placed on a CO/IP Line, the system will send caller ID using the number from the CLIP/COLP Table entry defined by this parameter with the "Station	N/A, 0-49, Station CLI	Station CLI

Table 6.4.11-1 CID/CPN ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	CLI" as a suffix. The "Station CLI" is the Station CLI Type (1~5) entered below. If Station CLI is selected in the table, only the Station CLI is sent.		
Type of Number for Calling Party Info	For outgoing calls on an ISDN Line, this parameter defines the "Type of Number Plan" provided in Calling Party Information Element of the ISDN call SETUP message.	Unknown, International, National, Unused, Subscriber	National
Incoming Prefix Code Insertion	Regional ISDN providers may use the Local Area Prefix code for special services. In cases where the code is not provided in the incoming call SETUP message, the system can insert the My Local Prefix and My Area Code below in SMDR, LNR, displays etc.	OFF ON	OFF
Outgoing Prefix Code Insertion	Regional ISDN providers may use the Local Area Prefix code for special services. The system can insert the "My Area Code" and "My Local Prefix Code" in the Connect message as defined in those items below.	OFF ON	ON
International Access Code	When an incoming call includes the international Country code in the ISDN call SETUP message, the Country code can be included in the station display. To include the Country code, Incoming Prefix insertion, and CLI display in Station Data must be ON.	Max. 4 digits	-
My Area Code	Regional ISDN providers may use the Local Prefix and Area codes for special services. The system will insert this Local Area Code in the call SETUP messages defined under the Incoming/Outgoing Prefix Code Insertion entries above.	Max. 6 digits	-
My Area Prefix Code	Regional ISDN providers may use the Local Prefix and Area codes for special services. The system will use this code for insertion of the Local Prefix Code in the call SETUP messages if Local Prefix Insertion is enabled above.	Max. 4 digits	-
CLI Transit	When the system must send CLI to the ISDN for an off-net call, the CLI can be either the original caller's CLI or the CLI of the Off-net forwarding/transferring station.	ORI, CFW	CFW
ISDN Redirecting Number	When the system needs to send a Redirecting number to the ISDN for an off-net call, the Redirecting number can be either the original caller's CLI or the CLI of the Off-net forwarding/transferring station. If set as "No Service", the system will not send the CLI information. If configured for OGR CLI (original CLI), the system will send original CLI that is received from incoming CO line. If set to CFW CLI, the system will send the redirecting CLI that is the CLI of the off-net call forwarding station.	NO SERVICE, ORG CLI, CFW CLI	NO SERVICE
Choice incoming CLI	Incoming CLI Choice – When ISDN setup message has two CLIs (Transit Point CLI & Original CLI), this option determines the CLI the system will recognize.	Original CLI, Transit Point CLI	Transit point CLI

Table 6.4.11-1 CID/CPN ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Calling party numbering Plan	ISDN Calling Party Numbering Plan can be programmed.	Unknown, ISDN/Telephony, Data/Numbering, Telex, National Std, Private, Reserved	ISDN/ Telephony
Called party numbering Plan	ISDN Calling Party Numbering Plan can be programmed.	Unknown, ISDN/Telephony, Data/Numbering, Telex, National Std, Private, Reserved	Unknown
Station CLI Type	The Station CLI used with the CLIP or COLP Table index is one of the five Station CLIs defined in '*CLI Attributes-PGM 113*' section as selected here.	Station CLI 1 – Station CLI 5	Station CLI 1
DID Remove Number	When a DID call is received on an ISDN Line, this field determines the number of digits that will be removed starting at the first received digit.	00~99	0
One Digit Remove	Selects One-digit remove mode in ISDN Called Digits for Italy DID.	Enable Disable	Disable
CID mode	The type of CID signal can be assigned according to the type of CID from the carrier.	Disabled, FSK, DTAS FSK, DTMF, RCID	FSK
RCID detect	Russia CID Detect Mode.	LOCAL, ALL	ALL
RCID request	Russia CID Request Mode.	USER, AUTO	AUTO
RCID Request First Delay Timer	Russia CID First Delay Timer.	010–150 (*10 msec)	30
RCID No-Answer Timer	Russia CID NO–Answer Timer.	001–300 (*seconds)	20
RCID Digit Number	Russia CID Digit Number.	4–10	7
RCID Request Count	Russia CID Request Count.	1–3	1
RCID Request Retry Delay Timer	Russia CID Retry Delay Timer.	10–30 (*10 msec)	10

6.4.12 T1 CO Line Attributes - PGM 152

Selecting T1 CO Line Attributes will display the T1 Line Attributes data input page. Enter a valid CO range and click **[Load]** to enter the T1 Line Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.

Figure 6.4.12-1 T1 CO Line Attributes

North American T1 standards require that the T1 terminating device, in this case the iPECS UCP, include various "adjustable" timers and counters as described below.

Table 6.4.12-1 T1 CO LINE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Pause Timer	A timed pause may be included in a Speed Dial number, in which case, the pause time is defined by this entry.	1~9 (seconds)	2
RELEASE GUARD	The release guard timer defines the length of time the system will maintain a Line as busy after the call has been terminated to assure the PSTN has sufficient time to 'clear down' the circuit.	01~60 (100 ms)	20
DT DELAY	The DT (Dial tone) Delay timer defines the duration dial tone must be received for DT recognition.	02~50 (100 ms)	10
INTER DIGIT	The Inter Digit timer defines the duration between digit transmissions.	15~30 (20 ms)	15
WINK Timer	For TIE or DID Lines the Wink timer defines the length of time the 'wink' (T1 TIE line circuit reversal) will last.	7~15 (20 ms)	10
OP RATE	For Pulse signaling, defines the duration and make/break ratio of each pulse.	60-40(10pps), 66-33(10pps),	60-40 (10pps)

Table 6.4.12-1 T1 CO LINE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
		60-40(20pps), 66-33(20pps)	
SEZ DTC	This timer defines the length of a valid 'line seizure' signal.	0~127 (20 ms)	3
RELEASE	For Ground Start, this timer defines the minimum length of time ground will not be applied to the TIP side from the PSTN.	0~127 (20 ms)	7
IASG TY	Incoming Address Signaling Type defines the type of signaling (DTMF or Pulse) expected.	PULSE DTMF	DTMF
RING DTC	The Ring DTC (Detect) timer defines the minimum acceptable length of the Ring-on time during a ring cycle.	2~9 (100 ms)	2
RING STOP	The Ring Stop timer defines the maximum Ring-off time during a ring cycle.	10~60 (100 ms)	60
COLLECT DGT	Collect DGT (digits) defines the number of digits expected on a DID line.	1~6	3
STORE TIME	For DID lines, this timer defines the maximum delay between incoming DID digits.	1~15 (second)	15

6.4.13 DCOB CO Line Attributes - PGM 153

Selecting DCOB CO Line Attributes will display the DCOB Line Attributes data input page. Enter a valid CO range and click **[Load]** button to enter the DCOB Line Attributes data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of CO/IP Lines when saved.
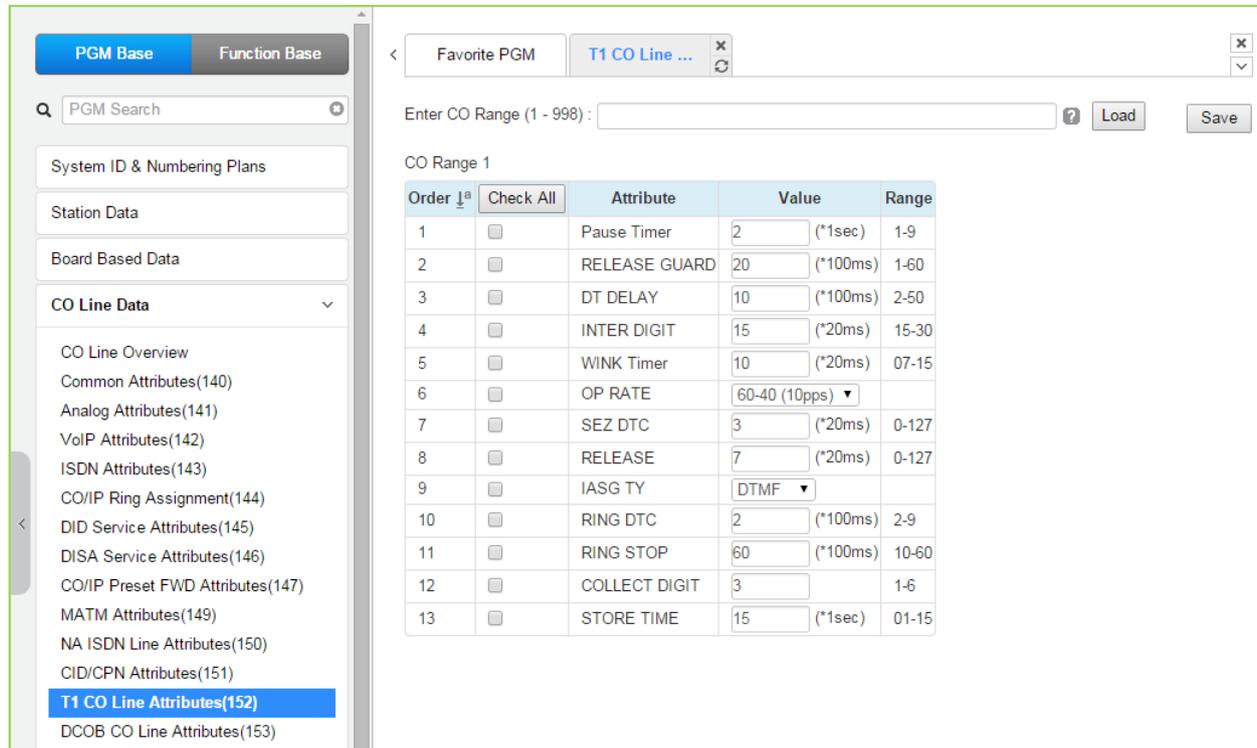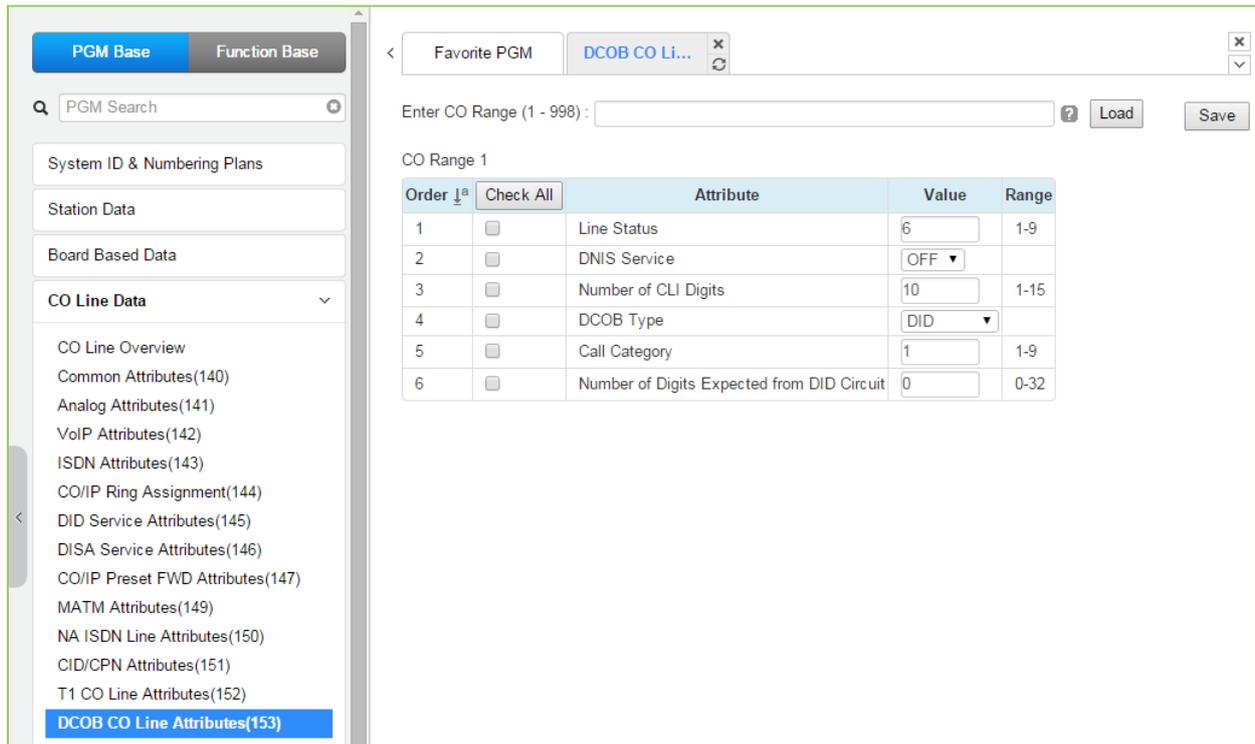

Figure 6.4.13-1 DCOB CO Line Attributes

Table 6.4.13-1 DCOB LINE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Line Status	This parameter defines the code that the system will send to indicate idle line status in accordance with E1R2 specifications.	1~9	6
DNIS Service	In R2, this field determines whether system will send caller information to PX or not.	OFF ON	OFF
Number of CLI Digits	This parameter defines the number of digits expected as the Calling Line Identification from digital CO lines.	01~15	10
DCOB Type	According to this type, the line can be restricted to the type of service, incoming DID and outgoing calls (DOD).	DID/DOD, DOD, DID	DID
Call Category	This parameter defines the code sent in response to a call category request from the network in accordance with E1R2 specifications.	1~9	1
Number of Digits Expected from DID circuit	This parameter defines the number of DID digits expected from digital CO lines. If set to "0", the number of digits defined in PGM 145 is used.	0~32	0

6.5 System Data

Selecting the System Data program group returns the sub-menu displayed in the left frame as shown in the following figure.
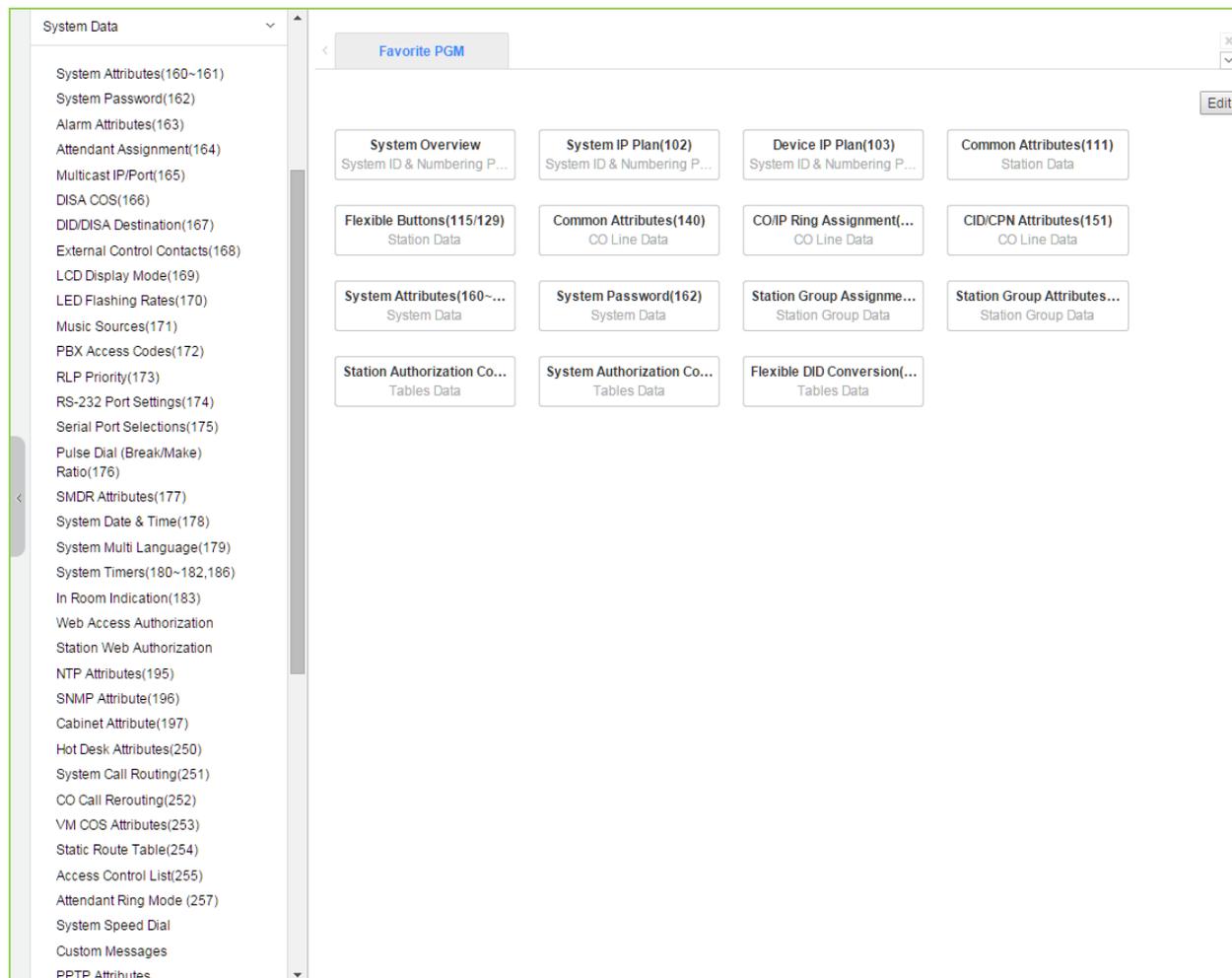
Figure 6.5-1 System Data Main Page

6.5.1 System Attributes - PGM 160 & 161

Selecting System Attributes will display the System Attributes data entry page.
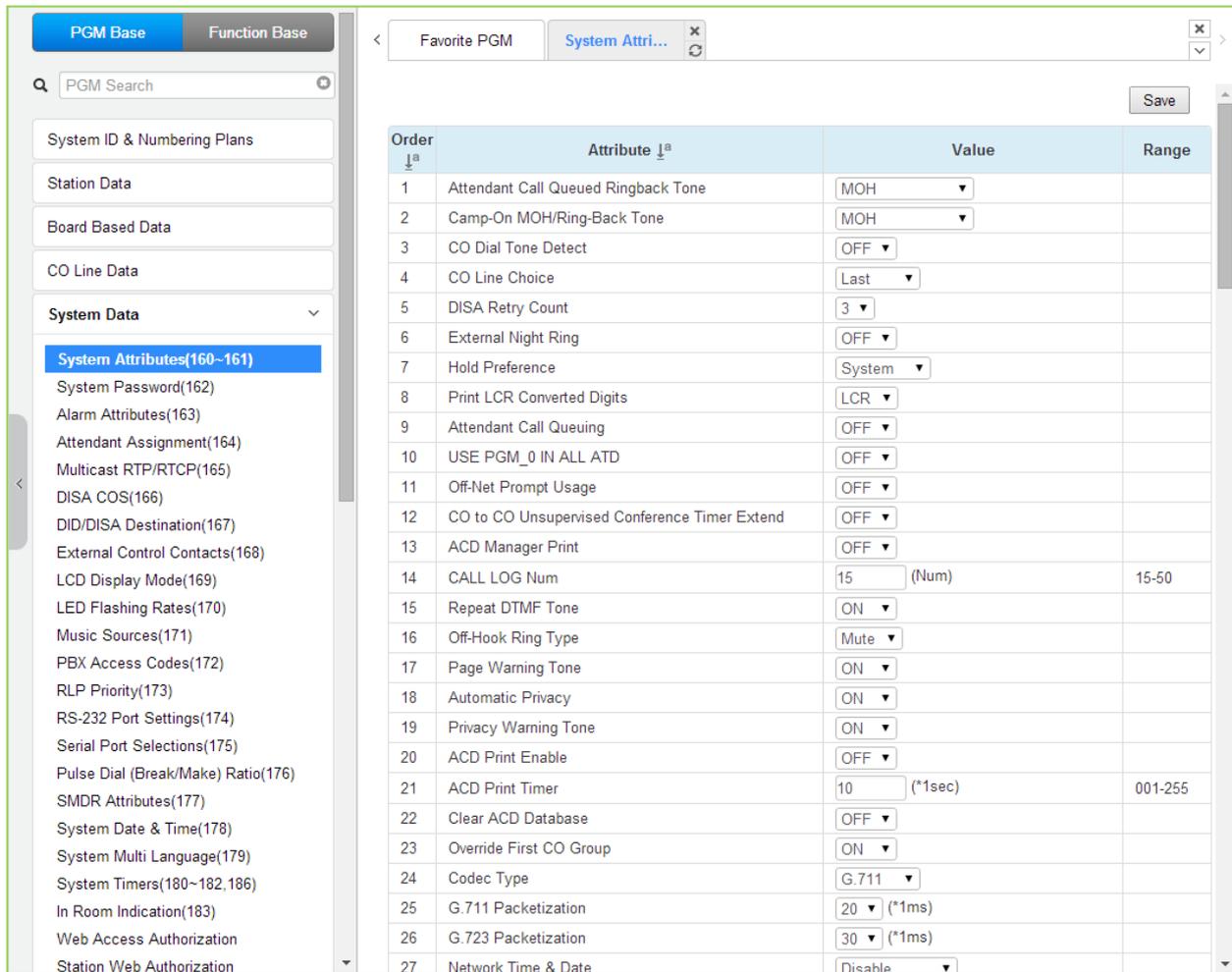
Figure 6.5.1-1 System Attributes

System Attributes define settings that affect system-wide features and functions. Generally, the entry will turn the feature ON (enable) or OFF (disable). Refer to the following table for a description of the Attributes and the data entries required.

Table 6.5.1-1 SYSTEM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Attendant Call Queued Ring Back Tone	When Attendant is busy with receiving call, the system will provide either Ring-back tone or Background music (MOH). If MOH is selected, the Music Source must be defined in '*Music source – PGM171*'.	MOH, Ring-Back Tone	MOH
Camp-On MOH/Ring-Back Tone	When Camp-On is used, the calling station will receive either ring-back tone or Background music (MOH). If MOH is selected, a source must be defined in '*Music source – PGM171*'.	MOH, Ring-Back Tone	MOH
CO Dial Tone Detect	The system can use dial-tone detection or a timed pause for Speed Dial numbers that contain a Pause.	OFF ON	OFF

Table 6.5.1-1 SYSTEM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CO Line Choice	CO Lines can be selected among Round robin, Last, or First according to groups assigned by System.	ROUND, Last, First	Last
DISA Retry Count	A DISA user is allowed to retry erroneous authentication code entries. This entry sets the number of retry count before the system disconnects.	1~9	3
External Night Ring	CO/IP calls, which are assigned UNA (Universal Night answer), can activate the Loud Bell Contact. While in the Night mode, an incoming call will activate the contact.	OFF ON	OFF
Hold Preference	A single depression of the **[Hold]** button places the call on the preferred hold, System or Exclusive.	System, Exclusive	System
Print LCR Converted digits	SMDR will print out the number dialed by either the system's LCR or the user.	LCR, User	LCR
Attendant Call Queuing	The system can be configured to queue incoming calls until Attendant is available.	OFF ON	OFF
USE PGM_0 IN ALL ATD	This field allows Main Attendants to activate Day/Night mode and other System Attendant menus except PGM 06 – Record system announcement.	OFF ON	OFF
Off-net Prompt Usage	When a call is routed to a destination external to the iPECS, the Off Net routing prompt can be played.	OFF ON	ON
CO to CO Unsupervised Conference Timer Extend	When an Unsupervised Conference is established with DISA, Off-Net Fwd etc., the Unsupervised Conference timer determines the duration of the call. If enabled here, the user may extend the duration.	OFF ON	OFF
ACD Manager Print	When the optional ACD Event messages are required, the system must be enabled here to send the events.	OFF ON	OFF
CALL LOG Num	The Call Log saves the Outgoing call, Received call, or Lost call information and can be displayed by pressing Call Log Display Button. The maximum number of Call Log per station can be set.	15~50	15
Repeat DTMF tone	If enabled, the system will repeat DTMF tones to the caller's station when the call is routed to an off-net location.	OFF ON	OFF
Off-Hook Ring Type	Off-hook ring can be a single tone burst or muted normal ring.	Burst Mute	Mute
Page Warning Tone	A warning tone can be sent prior to a page announcement.	OFF ON	ON
Automatic Privacy	Automatic Privacy can be disabled, allowing stations to join an active CO/IP call. A warning tone can be provided, see Privacy Warning Tone below.	OFF ON	ON
Privacy Warning Tone	If desired, a warning tone can be provided when privacy is overridden.	OFF ON	ON
ACD Print Enable	ACD statistics can be periodically sent to the assigned serial port. To provide periodic reports, this feature must be ON.	OFF ON	OFF
ACD Print Timer	This field defines the time, in 10-second increments,	001~255	10

Table 6.5.1-1 SYSTEM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	between the periodic ACD reports assigned above.	(1 sec)	
Clear ACD Database	When a periodic report is sent, the ACD database can be cleared automatically, if "ON".	OFF ON	OFF
Override First CO Group	When a user dials '9', the system can search all CO/IP Groups for the first available CO/IP line.	OFF ON	ON
Codec Type	The default codec can be defined as G.711, G.729, G.722 or G.723.1 for decreased bandwidth needs. The selected codec will be used on all internal communications as well as for remote iPECS devices.	G.711, G.723.1, G.729, G.722	G.711
G711 Packetization	The G.711 voice frame packetization time determines the interval at which voice samples are packetized and sent when the G.711 codec is used.	10, 20, 30 (1 msec)	20
G723 Packetization	The G.723.1 voice frame packetization time determines the interval at which voice samples are packetized and sent when the G.723.1 codec is used.	30, 60 (1 msec)	30
Network Time & Date	The system can use ISDN Network time or NTP to synchronize time with the ISDN or data network. ISDN sync is not available in USA version.	Disable, ISDN CLOCK, NTP	Disable
Incoming Toll Check	The system can invoke COS dialing restrictions when a user dials while connected to incoming call.	OFF ON	ON
Web Server Port/TLS for Web	This field determines the TCP port employed to access the system WEB server. This field also enables Transport Layer Security (TLS) for access to Web Admin.	00001-65535/ Enable/Disable	80/Disable
Auth Retry Count	When an Authorization code is required, the user may attempt to enter a Valid code up to the maximum available count.	1-9	3
Simple Auth Code Usage	System Authorization codes are entered by the user as "*" and the code (ON) or "*"+ the Auth code index and the code (OFF).	OFF ON	ON
COS 7 when Auth Fail	If a user fails to enter a valid Authorization code in the number of attempts assigned in Auth Retry Count above, the station is disconnected or the Station COS is changed to COS 7. In the latter case, the user must employ COS Restore in Station User PGM to return the station to the normal COS.	OFF ON	OFF
Unified Message Format	System Integration Messages are sent out the defined serial or TCP channel.	OFF ON	OFF
Conference Room CO Tel Number	The telephone number associated with a Conference Room can be assigned allowing an external party to access the room Conference room,	Max. 15 digits	
Record warning tone	When call recording is active, a tone can be sent to all connected parties to indicate the conversation is being recorded.	OFF ON	ON
UCP DIFF SERVE	Diff-Serv Code Point applied to packets from LAN port of the UCP.	00-63	46
Device Upgrade Mode	Transfer mode for upgrades from UCP to an iPECS device.	FTP TFTP	FTP

Table 6.5.1-1 SYSTEM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CO Transfer Tone	When a CO call is transferred to a busy extension, Ring Back Tone or Music On Hold will be played to the CO Line. For MOH, define the 'Music source'.	MOH Ring Back Tone	Ring Back Tone
Conference Warning Tone	When a new member joins a conference room, the system provides warning tone to conference members who already joined the conference.	OFF ON	ON
Dummy Dial Tone	When a CO line does not provide dial tone, the system can provide dummy dial tone.	Unused Use	Unused
SIP Station Mode	SIP phones may set up a point-to-point RTP connection (PTP) or to assure a controlled connection, RTP can be routed via a VoIP channel (Routed).	Routed PTP	Routed
SMS Center Number	When the PSTN will be used to send SMS, the phone number of the Short Message Service Center must be entered.	Max. 23 digits	
SMS Center CLI	When the CO/IP Line will be used to receive SMS, the Caller ID expected from the Short Message Service Center must be defined.	Max. 23 digits	
SMS Protocol	The Short Message Service Protocol must be selected to support SMS.	None, ETSI-P1, ETSI-P2, KT-LivingNet, SIP-Text, SIP-XML, KT IP-PBX, SKN IP-PBX, KT XML	None
G.722 Packetization	The G.722 voice frame packetization time determines the interval at which voice samples are packetized and sent when the G.722 codec is used.	10, 20, 30 (1 msec)	20
Transit-out security	The system will check IP address for transit-out calls in the master system. If not valid, the transit-out call is denied.	Unused Use	Use
Emergency call Attendant Notify	The Attendant can be notified when another user in the system dials an Emergency number	Unused Use	Use
3-Way Conference Preference	When 3-way conference calling, the system uses the selected device to establish the conference mixing. When "Local" is assigned, the device, such as a SIP phone, must provide the mixing.	Local MCIM	Local
First digit * in SPD	Normally, when "*" is the first digit in a Speed Dial number, the Display Security feature is activated so that the number is not shown in the LCD of iPECS IP or LDP Phones. Otherwise, the "*" is sent as the first digit to the carrier.	Display Security Digit *	Display Security
Use Strong Password	When enabled, passwords in PGM 162 must be specified as long and random as you make it with more than 6 digits including both numbers and characters.	OFF ON	ON
VSF SMTP port	SMTP port used for the VSF and UVM gateway.	00001~65535	25

Table 6.5.1-1 SYSTEM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CTI IP	When assigned, the system will accept 3rd party TAPI messages only from assigned IP address.		0.0.0.0
Intercom Busy Service	When busy, an intercom call may use Off-hook Voice-Over or Intrusion to connect to the called intercom party.	Voice-over Intrusion	Voice over
Auto save new message	After listening to a voice message, if the user takes no action, the system can automatically save the message or leave the message in the new message category.	Unused Use	Unused
IGMP query usage	The system employs multi-cast packets for registration and certain general functions such as MOH. With some multicast snoop enabled Layer 2 switches, multicast packets will not forward unless an IGMP query device exists in the network. This field enables the IGMP query option and the system sends periodic IGMP query messages to avoid multicast blocking.	OFF ON	OFF
IGMP query interval timer	This timer defines the interval for each IGMP query message.	0~3600 (Sec.)	180
IGMP query all host	IGMP queries are sent to all IP hosts (ON) in the network or to iPECS devices only. For All Hosts, IP address 224.0.0.1 is used, otherwise the iPECS specific 239.20.19.50 IP address is used. If problems occur MOH, the "All Hosts" may correct the problem.	OFF ON	ON
IGMP query generic	This attribute specifies a group address being queried. If ON is selected, all multicast groups are queried. If OFF is selected, the iPECS registering device group (239.20.19.50) is queried only. This should be ON when there is a MOH problem.	OFF ON	OFF
Restrict star and pound	If enabled, calls using "*" or "#" as the first digit are prohibited by the system.	OFF ON	OFF
Restricted Dialing Display After Answer	If a restricted number is called, the system will output an SMDR record for the call after the call is answered.	OFF ON	ON
IP BIND USAGE	If ON, the VoIP channel will apply IP-Binding with the Media port assigned in PGM 130 and 133.	OFF ON	OFF
New 5 Wake Up Usage	With the "New Wake-Up" option, the user can assign five alarm notification times, otherwise only one Wake-Up time can be set.	OFF ON	OFF
Easy 5 Wake Up Usage	If this field is set to ON and New 5 Wake Up Usage is also set to ON, each station user can enter a wake up time without entering wake up type or wake up index.	OFF ON	OFF
Station VM Feature Usage	Enhanced features of the VSF/UVM can be disabled. Features controlled by this setting include as below: - Company Directory - Remote Announcement Recording - Administrator Mailbox - Distribution List - Mark a Message as Private, Urgent, or Delivery Confirmation - Record Greeting through CCR - Voice Mail Class of Service	OFF ON	ON

Table 6.5.1-1 SYSTEM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
End code(#) usage in System Auth Code	If this filed is set to ON, the End code (#) must be entered when a system Auth code is entered.	OFF ON	OFF
Remote VM Access	If this value is OFF, the user cannot access the built-in Voice Mail via an external CO/IP Line.	OFF ON	ON
Transfer Tone Usage	If this value is ON, a Warning tone is served to a station receiving a screened call transfer.	OFF ON	OFF
CID Password Usage CID Password	If this value is ON, a user must enter the CID password to modify a CID setting in admin.	OFF ON	OFF
LCR Dial Tone Detect	If this value is set to ON, the system first checks dial tone on analog CO Lines then LCR dialing is completed. If no dial tone is detected, the call is rerouted to Alternate DMT Index. If LCR type is set to M13, LCR dial tone detect option is not applied.	OFF ON	OFF
ICM Call Log	If this value is set to ON, the system provides an ICM Call log for iPECS IP and LDP Phones.	OFF ON	OFF
Mobile Phone Presence Service port	This is for an interface between UCP and external server that sends http/xml data. Currently the first usage of this port is 'mobile phone presence' service. If set to a valid port, the presence service will be implemented and the system will await messages for presence from the external server.	00001~65535	0
Mobile Phone Presence State Sync	When the system receives presence information for a Mobile extension from an external server, the system synchronizes the station's call state with this presence information. A busy presence will set the station to busy.	OFF DSS LED only Station Status	Station Status
Mobile Phone Presence Force Idle Timer	The system will return a station's state to idle if there is no updated presence information from the external server for the duration of this timer.	0~3600 (Sec.)	0
Attendant Password Usage	If this value is set to ON, system requests a password when a user enters Attendant program ([PGM] + 0, an Attendant enters Speed program, or a user assigns Attendant program code ([PGM] + 0) to Flexible button,	OFF ON	OFF
Pick-up station name usage	When picking up a call for a station, the Name of the station is displayed.	OFF ON	ON
Display LCR mode	The LCD of iPECS IP and LDP Phones can display "LCR MODE" when LCR is activated.	OFF ON	ON
VM Notify to Mobile Extension over CO	If this value is set to ON, an analog Loop Start CO Line can be used for the VM notification call to the Mobile Extension.	Disable Enable	Disable
Display {MEET ME} Soft button	If this value is set to OFF, **{MEET ME}** soft button is not displayed during a paging.	OFF ON	ON
Number of CLI Wait List	The number of records in the missed call log (CLI Msg Wait List) is determined by this value. If the number of missed calls reaches this value, the oldest record is deleted and the new record is added to the list.	0-255	0
Emergency Mailbox	When an Emergency Mailbox Station is assigned, an emergency call placed by a station is recorded		

Table 6.5.1-1 SYSTEM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Station	automatically to the assigned mailbox.		
MSVC XML Port	The XML port used to support Web callback, call-through and iPECS ClickCall application for MSVC (Mobile Service) must be defined.	00001~65535	7878
IPCR Announce for recording warning	When ON, the IPCR server Call Recording notification announcement is played to the caller in place of the warning tone.	OFF ON	OFF
IPCR Announce for only incoming	User can choice IPCR Announce when it's incoming by setting ON.	OFF ON	OFF
Mobile EXT CO Access Password Check	When a mobile extension places an external call using an iPECS CO/IP Line, the user may be required to enter a valid Authorization code to place the call.	OFF ON	OFF
Hold key usage on paging	When ON, the Hold button is used for paging.	OFF ON	ON
Device Info Request Interval	The system checks the status of each device with a polling message at intervals of this timer.	15 ~ 255 seconds	15
Dial By Name List	The Dial by Name feature can allow stations in any ICM tenancy groups to call a station any in ICM Tenancy group. When desired, Dial by Name can be limited to function within allowed Tenancy calling groups, PGM 125.	All Accessible ICM Group only	ALL
Dial By System Speed Name List	If this value is set to "Accessible System Speed Zone Only", the system speed dial number of Accessible system speed zone (System Speed Zone (232)) is displayed in {Dial by Name}.	All, Accessible System Speed Zone Only	ALL
Default Web Language	The user can select the display language used in the Web pages. Either English or the "Local" language can be selected. Note the "Local Language" must be entered in the Multi-Language file in Maintenance section.	English, Local Language	English
No Range Load Limit on Web Admin	OFF: Limited index range will be accepted (eg, 1-100) ON: No limitation of table index (eg, 221, 222, 270, etc)	OFF/ON	ON
Forward limitation by caller COS	Caller (internal station) call to a station offnet forwarded. It can be forwarded by caller COS if it's ON.	OFF/ON	OFF
No CLI Call Log	Leave system Call log even though there is no CLI information if it is ON. Otherwise Call log is not leaved.	OFF ON	OFF
Remote VM Forward Access	If the value is ON, Call Forwarding setting from remote access is enabled. If the value is OFF, Call Forwarding setting from remote access is disabled.	OFF ON	ON
DB Backup to USB monthly	System downloads the database to USB periodically. The user can access, copy and delete files in USB drive via web admin. So you can back up DB by choosing the specific date for Monthly or Weekly.	N/A, 1, 15	
DB Backup to USB weekly		N/A, Mon ~ Sun	
SMDR/ACD/Alarm Mail Attribute			
Common SMTP Attributes			
SMTP Server IP	When the system must send E-Mail SMDR, ACD	12-digits	0.0.0.0

Table 6.5.1-1 SYSTEM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Address	reports or Alarms, the SMTP Mail server IPv4 address for the systems E-mail account is required.		
SMTP Server Domain Address	The URI associated with the SMTP sever domain can be assigned in place of the SMTP Mail IP address above.	Max. 100 characters	
SMTP Port	This field defines the TCP/IP port that system will employ when communicating with the SMTP E-mail server.	1-65535	25
SMTP Security Connection	The system can support basic security policies when communicating with the SMTP E-mail server. Note the server must also be configured for the selected security protocol.	No security SSL TLS	No security
E-Mail Account ID	This field defines the ID for the system's account in the SMTP Mail server. If ID and password are assigned, the SMTP Mail server can check the validity of the user ID and password.	Max. 40 characters	
E-Mail Account Password	This field defines the password for the system's account in SMTP Mail server. If ID and password are assigned, the SMTP Mail server can check the validity of user ID and password.	Max. 20 characters	
Sender E-Mail Address (From)	E-Mail address used by the system as the "From" address when sending Database, SMDR, ACD, and Alarm reports	Max. 40 characters	
Sender System Domain Name	This field is the Domain name of the system used to send the various system SMDR, ACD, and Alarm reports.	Max. 18 characters	
Database Attributes [Send Mail]			
[Send Mail] button is used for sending the mail after filling out the below menu			
Database Mail Send Monthly Set	Sets day of the month for Database to be sent on a monthly basis (NA for no monthly reports, select the day for 1 or 15).	NA/ 1/ 15	NA
Database Mail Send Weekly Set	Sets day of the week to send Database weekly.	Day-of week	N/A
Database Type	Select the desired database type: All database, System speed, Station speed, LCR etc.		All Database
Database Receiver E-Mail Address (To)	E-mail address to receive the Database E-mail reports.	Max. 40 characters	
SMDR Attributes [Send Mail]			
[Send Mail] button is used for sending the mail after filling out the below menu			
SMDR Mail Send Weekly Set	Sets day of the week to send SMDR data weekly.	Day-of-Week	N/A
SMDR Mail Send Daily Set	Sets time-of-day for SMDR data to be sent on a daily basis (00 for no daily reports, 01-23 for hour of the day).	00-23	
SMDR Mail Auto Send Set	If the SMDR buffer is full, the system can send a notification by E-mail automatically.	OFF ON	OFF

Table 6.5.1-1 SYSTEM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
SMDR Mail Auto Delete Set	When set, the system will delete SMDR records after sending the E-mail.	OFF ON	OFF
SMDR Receiver E-Mail Address (To)	E-mail address to receive the SMDR E-mail reports.	Max. 40 characters	
ACD Attributes [Send Mail]			
[Send Mail] button is used for sending the mail after filling out the below menu			
ACD Mail Send Weekly Set	Sets day of the week to send ACD statistic data weekly.	Day-of-Week	N/A
ACD Mail Send Daily Set	Sets time-of-day for ACD statistic data to be sent on a daily basis (00 for no daily reports, 01-23 for hour of the day).	00-23	
ACD Database Delete After Mail Send	When set, the system will delete ACD statistic data after sending E-mail.	OFF ON	OFF
ACD Receiver E-Mail Address (To)	E-mail address to receive the ACD E-mail reports.	Max. 40 characters	
Alarm Attributes [Send Mail]			
Alarm Receiver E-Mail Address(To) & Notification Receiver E-Mail Address(To)	The system will send email to saved E-Mail Address when system alarm is occurred or system restarts.	Max. 40 characters	
LDAP Server Settings			
Server Display Name	When the UCS Client requires access to an LDAP server, the system will provide the LDAP server information to the client. The server name is defined in this field.	15 characters	LDAP Server
Server IP	When an LDAP server is employed, the IP address of the server must be defined.		0.0.0.0
Server Port	When an LDAP server is employed, the TCP/IP port of the server must be defined.	0001-65535	
Require Login	The UCS Client ID and Password may be required for log in to the LDAP server.	OFF ON	ON
Use SSL	When supported by the server, the client can employ SSL (Secure Sockets Layer) for added security.	OFF ON	OFF
Search Base	Server base means Search option. You can get the search option from LDAP Server manager. For example, if OU (Organization Unit) is OC and DC (Directory Company) are ucapp and com, you can give the option "OU=OC, DC=ucapp, DC=com" in this field. You will get the desired directory.		
LDAP User ID	The ID and Password must be required to connect to LDAP Server.		
LDAP User Password			

6.5.2 System Password - PGM 162

Selecting System Password will display the System Password data entry page. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the Keyset Admin, Remote Access Password, or CID password when saved.

It is important to use strong password for lowering overall risks of a security breach. The best way to choose good password are designed to make passwords less easily discovered by intelligent guessing. In addition, we recommend that you change the password frequently.
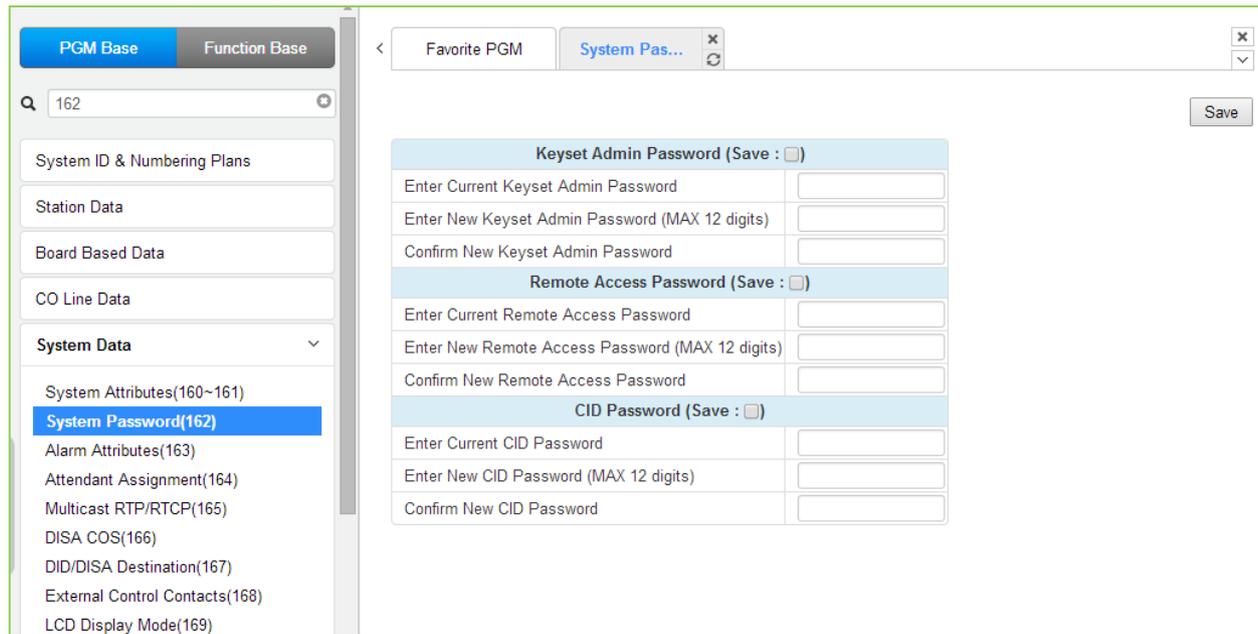
Figure 6.5.2-1 System Password

Access to the system database and maintenance functions can be protected by passwords up to twelve (12) digits. Three passwords can be defined, Keyset Admin, Remote Access and CID.

Check the save box and enter the password and click the save button.

The following success message is displayed if the password is correct and saved successfully.
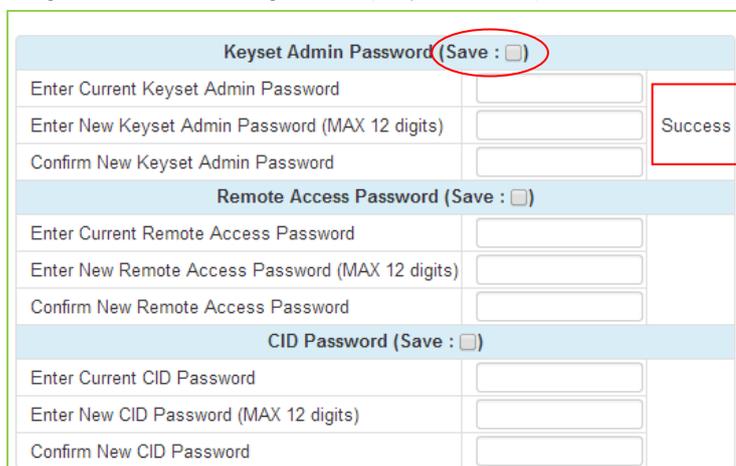
6.5.3 Alarm Attributes - PGM 163

Selecting Alarm Attributes will display the Alarm Attributes data entry page. Click **[Save]** button after changing Value.
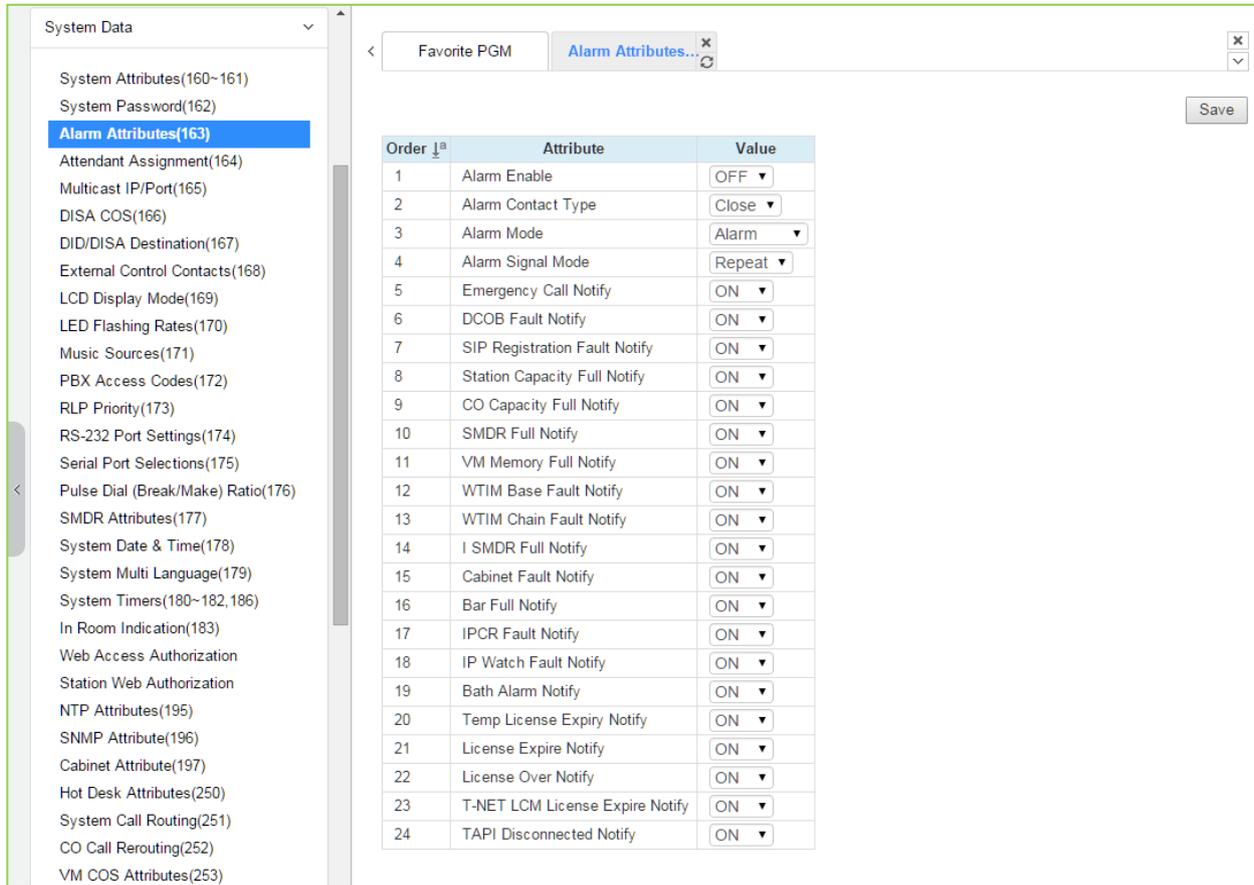
Figure 6.5.3-1 Alarm Attributes

The system can monitor an external contact. This contact is frequently employed as an Alarm indicator or Doorbell. The Alarm attributes define the operation of the external contact. For the Alarm, the signal to assigned stations can be repeating or a single burst, the former is often desired. For the Doorbell, a single tone is sent each time the contact activates. Refer the following table for a description of the features and the data entries required for each attribute.

Additional alarms events can be provided to the Attendant station for various conditions such as an Emergency call or SIP CO registration failure.

When 'Alarm Mode' attributes set 'Alarm', the system sends an email to the address that set in PGM 160 Alarm Attributes 'Alarm Receiver E-Mail Address (To) & Notification Receiver E-Mail Address (To)'.

Table 6.5.3-1 ALARM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Alarm Enable	This field enables/disables the contact monitoring circuitry.	OFF ON	OFF

Table 6.5.3-1 ALARM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Alarm Contact Type	This field establishes the contact state that will activate the Alarm, close or open.	Close/ Open	Close
Alarm Mode	The contact can be configured to function as a doorbell or an alarm.	Alarm/ Door-Bell	Alarm
Alarm Signal Mode	The assigned stations will receive a repeating signal or single burst (Once) of alarm tone.	Repeat/ Once	Repeat
Emergency Call Notify	This field enables/disables the Emergency call notification.	OFF ON	ON
DCOB Fault Notify	This field enables/disables DCOB Fault notification.	OFF ON	ON
SIP Registration Fault Notify	This field enables/disables SIP Registration Fault notification.	OFF ON	ON
Station Capacity Full Notify	This entry enables user to get the notification as alarm when Station Capacity is over the capacity.	OFF ON	ON
CO Capacity Full Notify	This entry enables user to get the notification as alarm when CO Capacity is over the capacity.	OFF ON	ON
SMDR Full Notify	This entry enables user to get the notification as alarm when SMDR is over the capacity.	OFF ON	ON
VM Memory Full Notify	This entry enables user to get the notification as alarm when the memory of Voice Mail is full.	OFF ON	ON
WTIB Base Fault Notify	This entry enables user to get the notification as alarm when WTIB base gets the fault.	OFF ON	ON
WTIB Chain Fault Notify	This entry enables user to get the notification as alarm when WTIB chain gets the fault.	OFF ON	ON
I SMDR Full Notify	This entry enables user to get the notification as alarm when I SMDR is over the capacity.	OFF ON	ON
Cabinet Fault Notify	This entry enales user to get the notification as alarm when the cabinet has fault.	OFF ON	ON
Bar Full Notify	This entry enales user to get the notification as alarm when Bar is full.	OFF ON	ON
IPCR Fault Notify	This entry enables user to get the notification as alarm when IPCR gets the fault.	OFF ON	ON
IP Watch Fault Notify	This entry enales user to get the notification as alarm when IP Wathch has fault.	OFF ON	ON
Bath Alarm Notify	This entry enables user to get the notification as alarm about Bath Alarm.	OFF ON	ON
Temp License Expire Notify	This entry enables user to get the notification before expiring Temp license.	OFF ON	ON
License Expire Notify	This entry enables user to get the notification as alarm before License expires.	OFF ON	ON
License Over Notify	This entry enables user to get the notification as alarm before License is over.	OFF ON	ON
T-NET LCM License Expire Notify	This entry enables user to get the notification as alarm before T-NET LCM license expires.	OFF ON	ON
TAPI Disconnected Notify	This entry enables user to get the notification as alarm when TAPI is disconnected.	OFF ON	ON

6.5.4 Attendant Assignment - PGM 164

Selecting Attendant Assignment will display the Attendant Assignment data entry page. Click **[Save]** button after changing Value.

Figure 6.5.4-1 Attendant Assignment

A maximum of fifty (50) Attendants can be assigned with the iPECS UCP. One is the System Attendant and the rest (49) are Main Attendants. The System Attendant has higher priority in call handling and system management functions. As a default, the System Attendant is assigned as the first Station (1000). Main Attendants are not assigned by default.

6.5.5 Multi-cast RTP/RTCP - PGM 165

Selecting Multi-cast RTP/RTCP will display the Multi-cast RTP/RTCP data entry page.
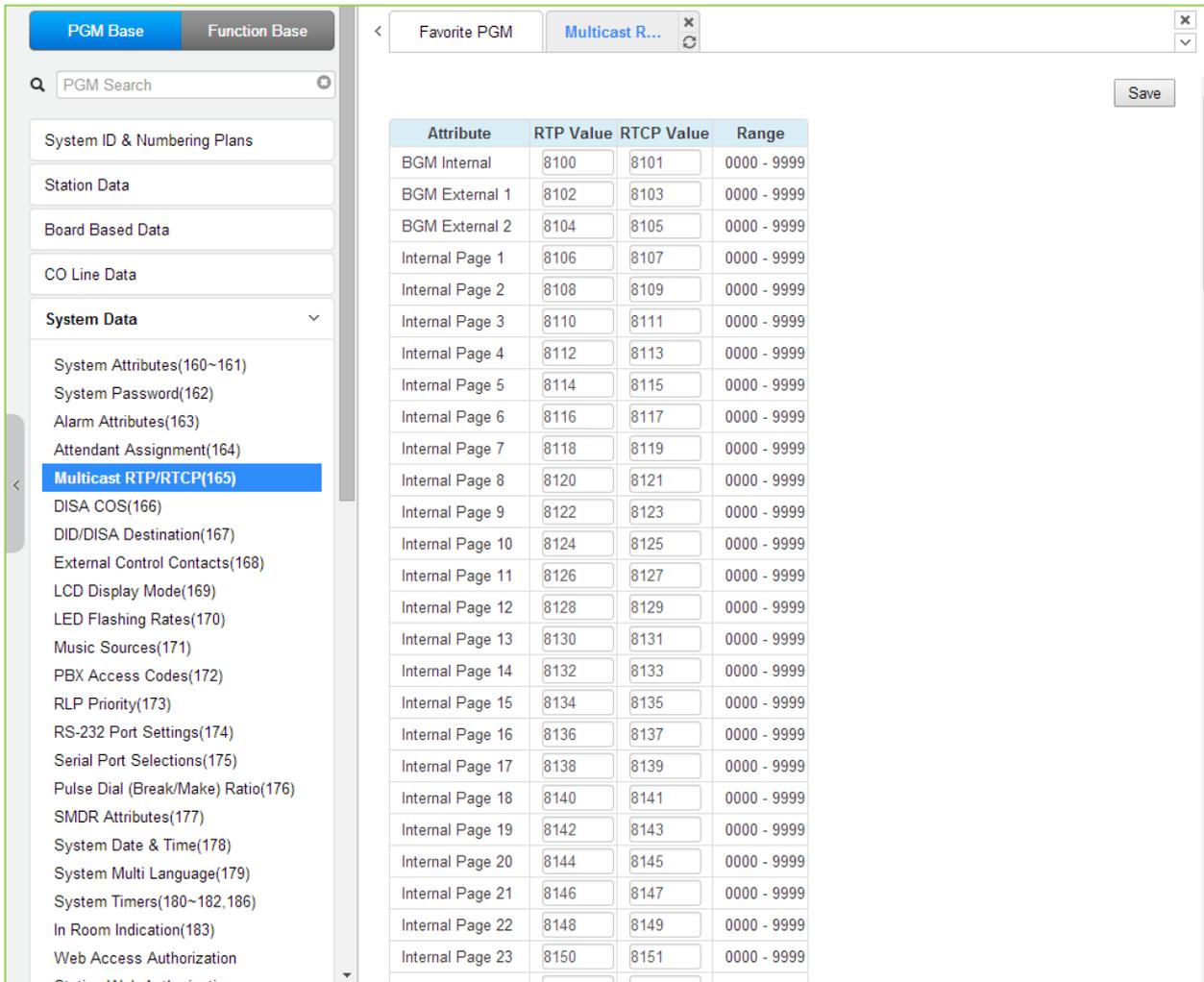
Figure 6.5.5-1 Multi-cast RTP/RTCP

Multi-cast is employed by the system to send BGM, MOH, paging and Push-To-Talk packets. Employing a single multi-cast packet reduces the overall LAN traffic. In some cases, specifically when multiple systems are connected to the same default gateway (router) it may be advantageous to define different IP ports for each system.

Table 6.5.5-3 MULTI-CAST RTP/RTCP

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
BGM Internal	RTP and RTCP ports for internal BGM.	0000-9999	8100 (8101)
BGM External 1	RTP and RTCP ports for external BGM 1.	0000-9999	8102 (8103)
BGM External 2	RTP and RTCP ports for external BGM 1.	0000-9999	8104 (8105)

Table 6.5.5-3 MULTI-CAST RTP/RTCP

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Internal Page 1 ~ Internal Page 100	RTP and RTCP ports for Internal Page 1.	0000-9999	8106~8304 (8107~8305)
Internal Page All	RTP and RTCP ports for Internal All Call Page	0000-9999	8306 (8307)
External Page 1	RTP and RTCP ports for External Page 1.	0000-9999	8308 (8309)
External Page 2	RTP and RTCP ports for External Page 2.	0000-9999	8310 (8311)
External Page All	RTP and RTCP ports for External All Call Page.	0000-9999	8312 (8313)
Page All	RTP and RTCP ports for All Call Page.	0000-9999	8314 (8315)
PTT 1	RTP and RTCP ports for PTT group 1.	0000-9999	8316 (8317)
PTT 2	RTP and RTCP ports for PTT group 2.	0000-9999	8318 (8319)
PTT 3	RTP and RTCP ports for PTT group 3.	0000-9999	8320 (8321)
PTT 4	RTP and RTCP ports for PTT group 4.	0000-9999	8322 (8323)
PTT 5	RTP and RTCP ports for PTT group 5.	0000-9999	8324 (8325)
PTT 6	RTP and RTCP ports for PTT group 6.	0000-9999	8326 (8327)
PTT 7	RTP and RTCP ports for PTT group 7.	0000-9999	8328 (8329)
PTT 8	RTP and RTCP ports for PTT group 8.	0000-9999	8330 (8331)
PTT 9	RTP and RTCP ports for PTT group 9.	0000-9999	8332 (8333)
PTT All	RTP and RTCP ports for PTT group ALL	0000-9999	8334 (8335)
BGM Internal VSF	RTP and RTCP ports for VSF BGM (VSF MOH)	0000-9999	8336 (8337)
SLT MOH 1	RTP and RTCP ports for SLT MOH1	0000-9999	8338 (8339)
SLT MOH 2	RTP and RTCP ports for SLT MOH2	0000-9999	8340 (8341)
SLT MOH 3	RTP and RTCP ports for SLT MOH3	0000-9999	8342 (8343)
SLT MOH 4	RTP and RTCP ports for SLT MOH4	0000-9999	8344 (8345)
SLT MOH 5	RTP and RTCP ports for SLT MOH5	0000-9999	8346 (8347)
VSF MOH2	RTP and RTCP ports for VSF MOH2	0000-9999	8348 (8349)
VSF MOH3	RTP and RTCP ports for VSF MOH3	0000-9999	8350 (8351)

6.5.6 DISA COS - PGM 166

Selecting DISA COS will display the DISA COS data entry page. Click **[Save]** button after changing Value.
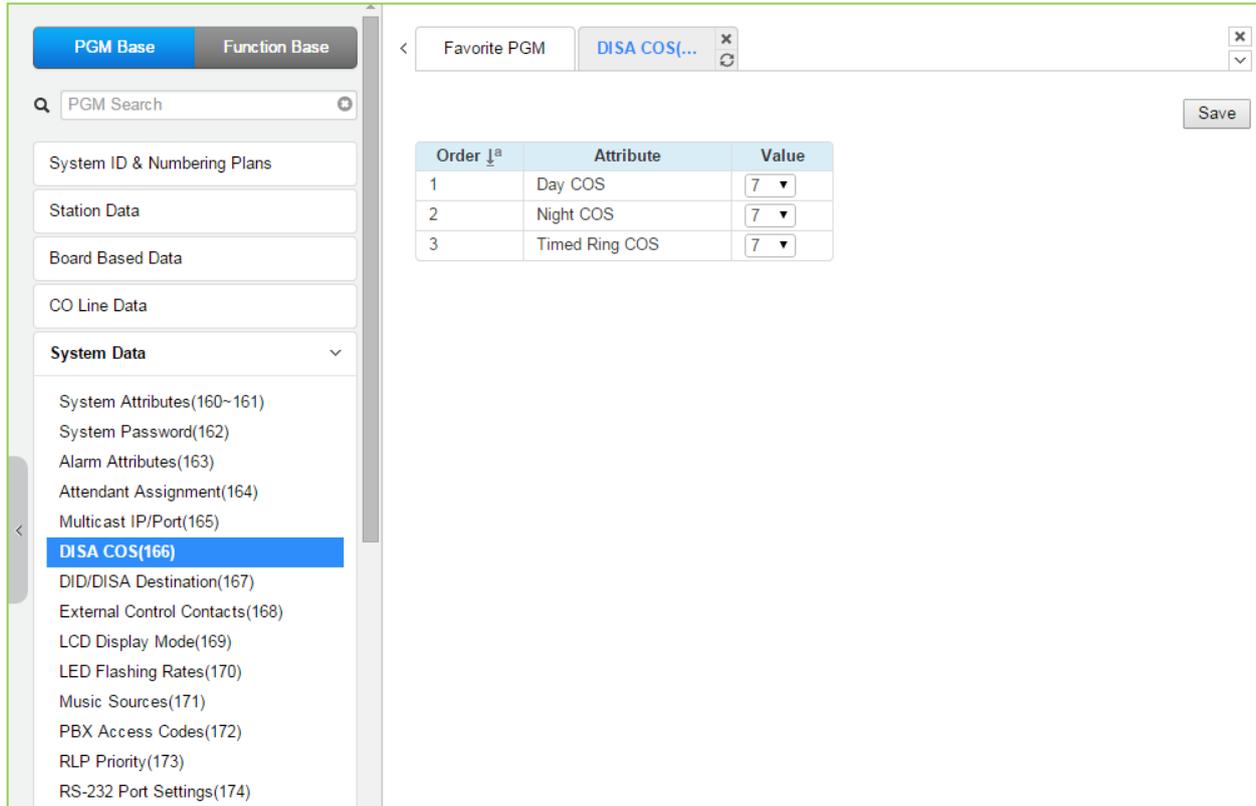
Figure 6.5.6-1 DISA COS

A DISA user is subject to the dialing restrictions assigned in the DISA Class-of-Service (COS). The restrictions applied are the same as with the corresponding Station COS levels 1~11 and interact with the CO/IP COS in the same manner. An assignment can be selected among Day, Night, and Timed Ring mode of system operation. The default for all three modes (Day, Night, Timed ring) of DISA COS is 7, allowing internal calls only.

6.5.7 DID/DISA Destination - PGM 167

Selecting DID/DISA Destination displays the DID/DISA Route table. Enter the desired ICM Tenant Group and the system will display the DID/DISA Destination Attributes. Click **[Save]** button after changing Value.
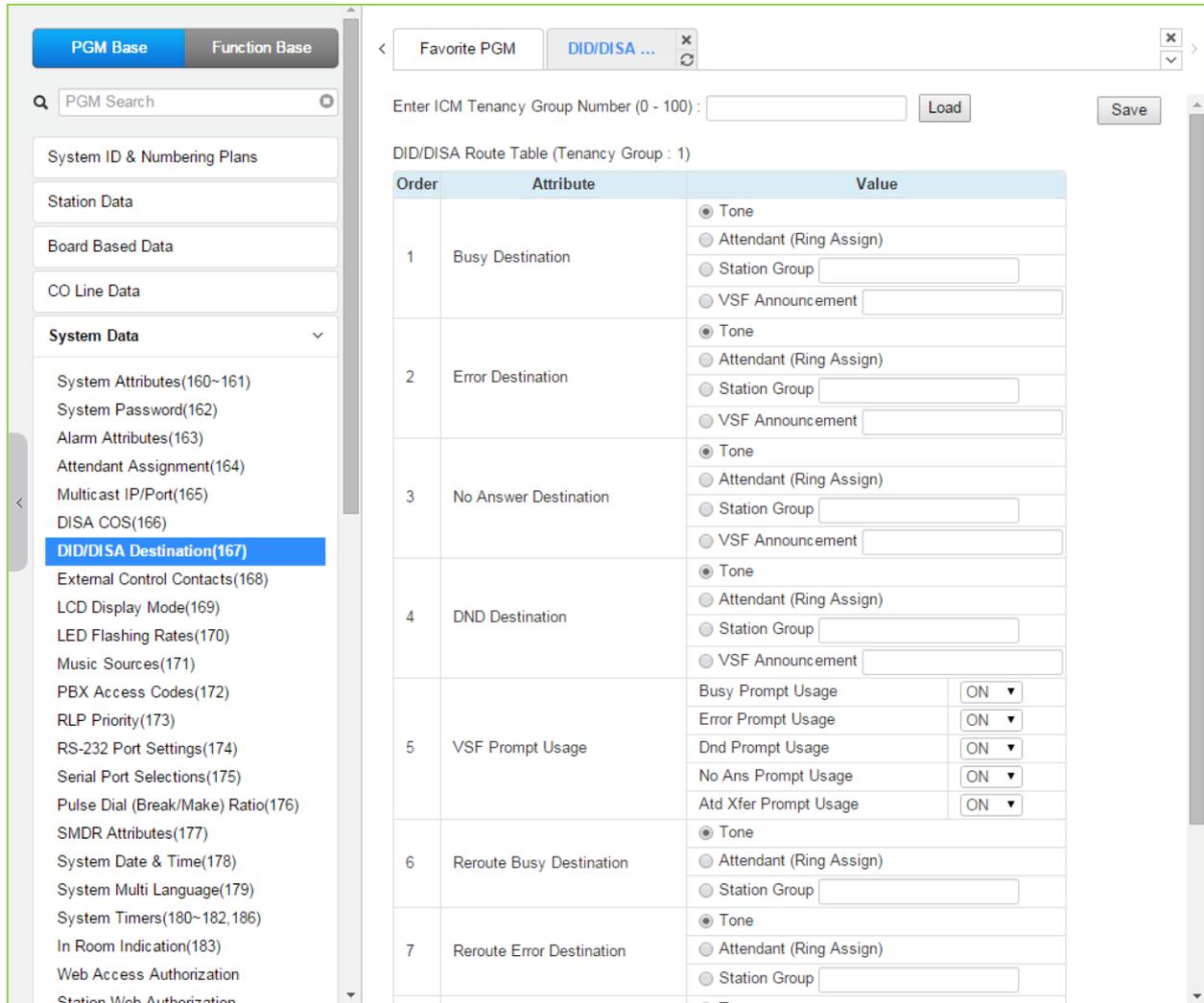
Figure 6.5.7-1 DID/DISA Destination

When a DID line or DISA user dials to the destination (Busy, Error, No Answer, DND, VSF Prompt, Reroute Busy, Reroute Error, Reroute No Answer, Reroute Net CO Busy), the caller will be sent to the assigned destination that is selected according to the ICM Tenancy group of the DID/DISA line. The available destination is separately defined for the current condition and can be routed as Tone, Attendant, Station Group or a VSF (System) Announcement.

For calls on a DID line to a busy station, DID Call Wait can be assigned, refer to '*Common Attributes section*', and the call will queue for the station for the No-Answer time. After the No-answer time, the call routes to the DID/DISA Destination unless forwarded. Also, for DID calls only, announcements (prompts) can be sent from the VSF or UVM gateway to the caller for various conditions, busy, error, DND, No Answer, or Attendant Transfer.

6.5.8 External Control Contacts - PGM 168

Selecting External Control Contacts will display the External Control Contact data entry page. Click **[Save]** button after changing Value.
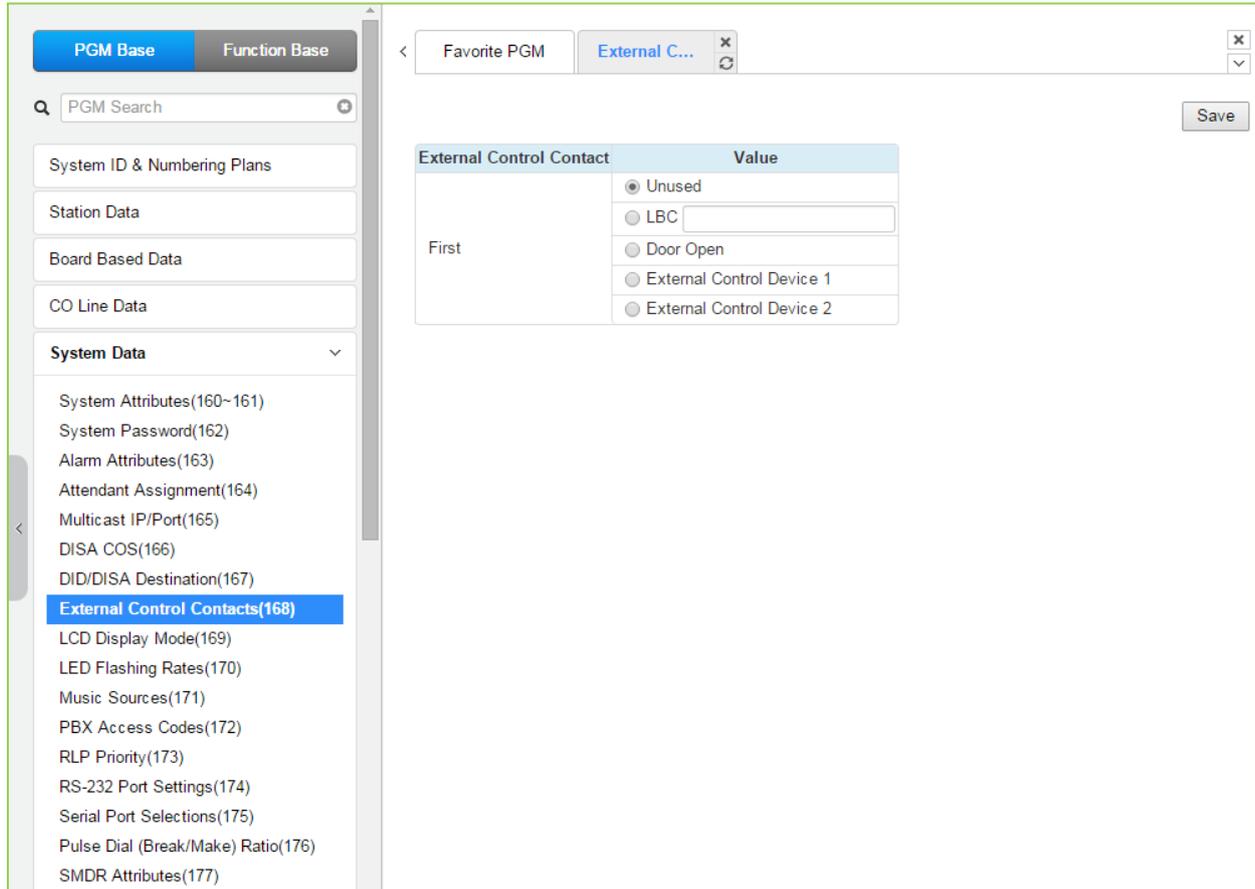
Figure 6.5.8-1 External Control Contact

The system includes a programmable contact, which can be used to control external devices. Refer to the table '*System capacity*' for number of available contacts. Each contact is assigned to activate under one of several conditions. As a Loud Bell Contact (LBC), the contact will activate when the assigned station or group receives an external call. For LBC, when the system is in the Night or Timed Ring mode, the contact will activate for incoming UNA (Universal Night Answer) calls and will ignore any station assignment. The contact may alternatively activate as a Door Lock Release contact, when External Page Zone 1 is accessed or when External Page Zone 2 is accessed.

6.5.9 LCD Display Mode - PGM 169

Selecting LCD Display Mode will display the data entry page. Click **[Save]** button after changing Value.
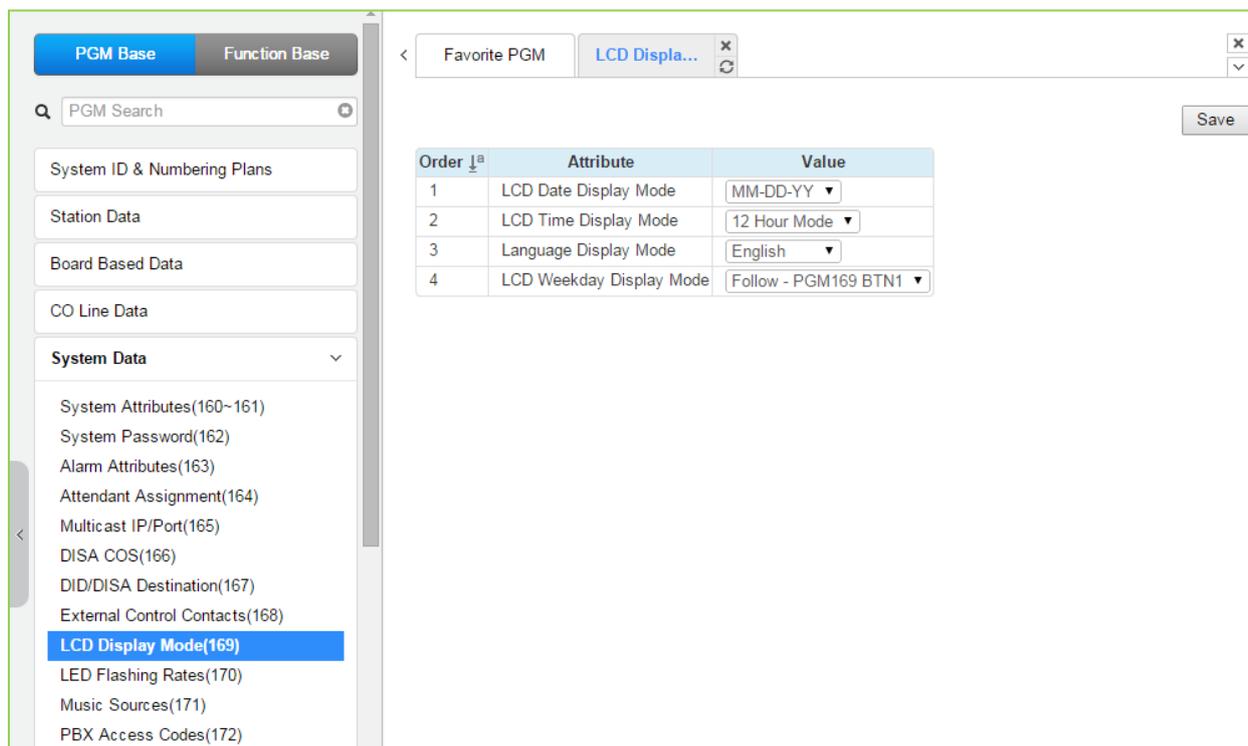
Figure 6.5.9-1 LCD Display Mode

The LCD display mode sets the time (12/24 hr.), date (day/month order) and language. Refer to the following table for a description of the modes and the data entries required.

Table 6.5.9-1 LCD DISPLAY MODES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
LCD Date Display Mode	Sets the Date display as month/day/year or day/month/year.	MM-DD-YY DD-MM-YY	MM-DD-YY
LCD Time Display Mode	Sets the Time display mode as 12 hour or 24-hour (military) time.	12 Hour Mode 24 Hour Mode	12 Hour Mode
Language Display Mode	Sets the Language used in the LIP or LDP Phone display.	Refer to the below table	English
LCD Weekday Display Mode	Sets the Day-of-Week (DoW) display mode: no DoW display mm/dd/DoW display alpha month as mm/dd/DOW, overrides setting of button 1 above display mm/dd/DoW, numeric month display, overrides setting of button 1 above.	No Display, Follow – button 1 above, Type1-(mm/dd/DOW), Type 2-(mm dd/DOW)	Follow - PGM169 BTN1

Table 6.5.9-2 LCD LANGUAGE SELECTION

LANGUAGE
English
Italian
Finnish
Dutch
Swedish
Danish
Norwegian
Hebrew
German
French
Portuguese
Spanish
Korean
Estonian
Russian
Turkish
Polish
Greek
Arabic

6.5.10 LED Flashing Rate - PGM 170

Selecting LED Flashing Rate will display the data entry page. Click **[Save]** button after changing Value.
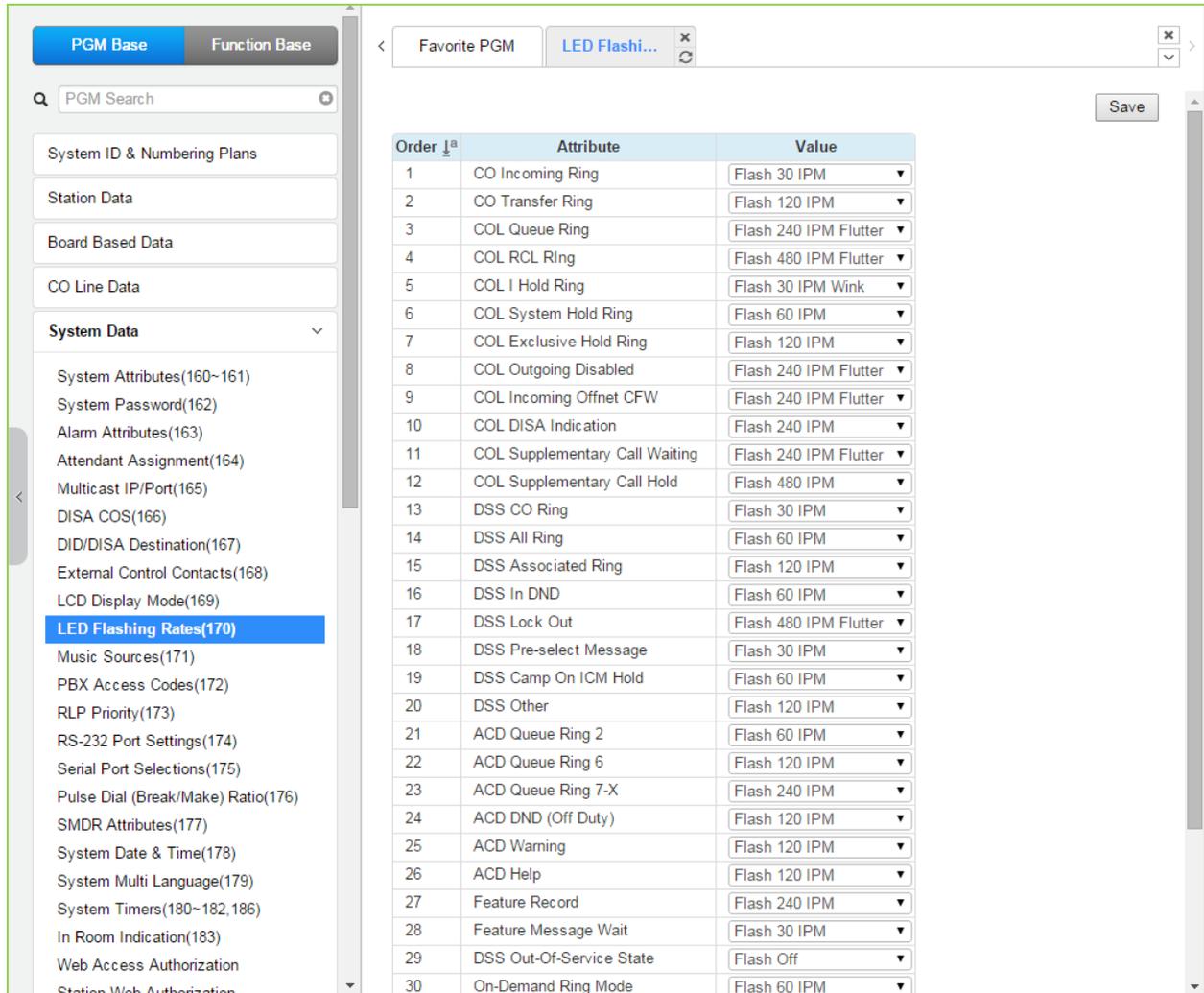
Figure 6.5.10-1 LED Flashing Rate

The LED flash rate for various functions and states can be assigned any one of the system's 14 signals. The various functions and states are shown in the following table. The 14 flash signals available in the system are shown in the table '*LED flash rate*'.

Table 6.5.10-1 LED INDICATION

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CO Incoming Ring	CO button Incoming ring flashing rate	00-14	FLASH 30 IPM (2)
CO Transfer Ring	CO button transfer ring flashing rate	00-14	FLASH 120 IPM (10)
COL Queue Ring	CO button queue call back ring flashing rate	00-14	FLASH 240 IPM FLUTTER (6)

Table 6.5.10-1 LED INDICATION

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
COL RCL Ring	CO button recall ring flashing rate	00-14	FLASH 480 IPM FLUTTER (8)
COL I Hold Ring	CO button I hold flashing rate	00-14	FLASH 30 IPM WINK (12)
COL System Hold Ring	CO button system hold flashing rate	00-14	FLASH 60 IPM (3)
COL Exclusive Hold Ring	CO button exclusives hold flashing rate	00-14	FLASH 120 IPM (10)
COL Outgoing Disabled	CO button outgoing disabled flashing rate	00-14	FLASH 240 IPM FLUTTER (6)
COL Incoming Off-net CFW	CO button incoming off-net call forward flashing rate	00-14	FLASH 240 IPM FLUTTER (6)
COL DISA Indication	CO button DISA indication flashing rate	00-14	FLASH 240 IPM (5)
COL Supplementary Call Waiting	CO button supplementary call waiting flashing rate	00-14	FLASH 240 IPM FLUTTER (6)
COL Supplementary Call Hold	CO button supplementary hold flashing rate	00-14	FLASH 480 IPM (8)
DSS CO Ring	DSS button CO ring flashing rate	00-14	FLASH 30 IPM (2)
DSS All Ring	DSS button ICM ALL ring flashing rate	00-14	FLASH 60 IPM (3)
DSS Associated Ring	DSS button ICM ring associate device flashing rate	00-14	FLASH 120 IPM (10)
DSS In DND	DSS button station is in DND	00-14	FLASH 60 IPM (3)
DSS Lock Out	DSS button station is in lock out	00-14	FLASH 480 IPM FLUTTER (8)
DSS Pre-select Message	DSS button station is in pre-selected message	00-14	FLASH 30 IPM (2)
DSS Camp On ICM Hold	DSS button station is in Camp On ICM hold	00-14	FLASH 60 IPM (3)
DSS Other	DSS button station is in other state	00-14	FLASH 120 IPM (10)
ACD Queue Ring 2	CIQ #1 Threshold	00-14	FLASH 60 IPM (3)
ACD Queue Ring 6	CIQ #2 Threshold	00-14	FLASH 120 IPM (10)
ACD Queue Ring 7-X	CIQ #3 Threshold	00-14	FLASH 240 IPM (5)
ACD DND (Off Duty)	ACD an agent is off duty (ACD DND)	00-14	FLASH 120 IPM (10)
ACD Warning	ACD warning tone	00-14	FLASH 120 IPM (10)
ACD Help	ACD help request/response	00-14	FLASH 120 IPM (10)
Feature Record	FEATURE voice record button	00-14	FLASH 240 IPM (5)

Table 6.5.10-1 LED INDICATION

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Feature Message Wait	FEATURE message wait	00-14	FLASH 30 IPM (2)
DSS Out-of-service state	DSS button a station is in out-of-service state	00-14	FLASH OFF (00)
On-demand Ring mode	DND led of attendant station for ring mode	00-14	FLASH 60 IPM (3)
Night Ring mode	DND led of attendant station for ring mode	00-14	FLASH Steady
Timed Ring mode	DND led of attendant station for ring mode	00-14	FLASH 240 IPM (5)
Auto Ring mode	DND led of attendant station for ring mode	00-14	Flash Off
Page Hold Button	HOLD LED for paging	00-14	FLASH 60 IPM (3)
DSS Station DND(off duty)	DSS button station in Station DND	00-14	FLASH 120 IPM (10)

Table 6.5.10-2 LED FLASH RATE TABLE

Flash Rate	DESCRIPTION
1	Steady On
2	30 ipm flash (30% On)
3	60 ipm flash (30% On)
4	60 ipm double wink (30% On-Off-On-Off & 70% On)
5	240 ipm flash (30% On)
6	240 ipm flutter (30% On-Off-On-Off-On & 70% Off)
7	480 ipm flash (30% On)
8	480 ipm flutter (30% On-Off-On-Off-On & 70% Off)
9	15 ipm flash (30% On)
10	120 ipm flash (30% On)
11	120 ipm flutter (30% On-Off-On-Off-On & 70% Off)
12	30 ipm double flash (30% On-Off-On & 70% Off)
13	480 ipm double wink (30% On-Off-On-Off & 70% On)
14	480 ipm double flash (30% On-Off-On & 70% Off)

6.5.11 Music Sources - PGM 171

Selecting Music Sources will display the Music Sources data entry page. Click **[Save]** button after changing Value.

Figure 6.5.11-1 Music Sources

Music inputs are provided for use as the Background Music and/or Music-On-Hold source inputs. Up to three VSF announcements may be recorded and played as MOH to the connected caller. The "VSF MOH" selection employs System announcement number 201, and for the VSF MOH 2 and VSF MOH 3 selections, any unused announcement can be employed. In addition, up to five SLT ports may be used as MOH to the held caller.

6.5.12 PBX Access Codes - PGM 172

Selecting PBX Access Codes will display the PBX Access Codes data entry page. Click **[Save]** button after changing Value.
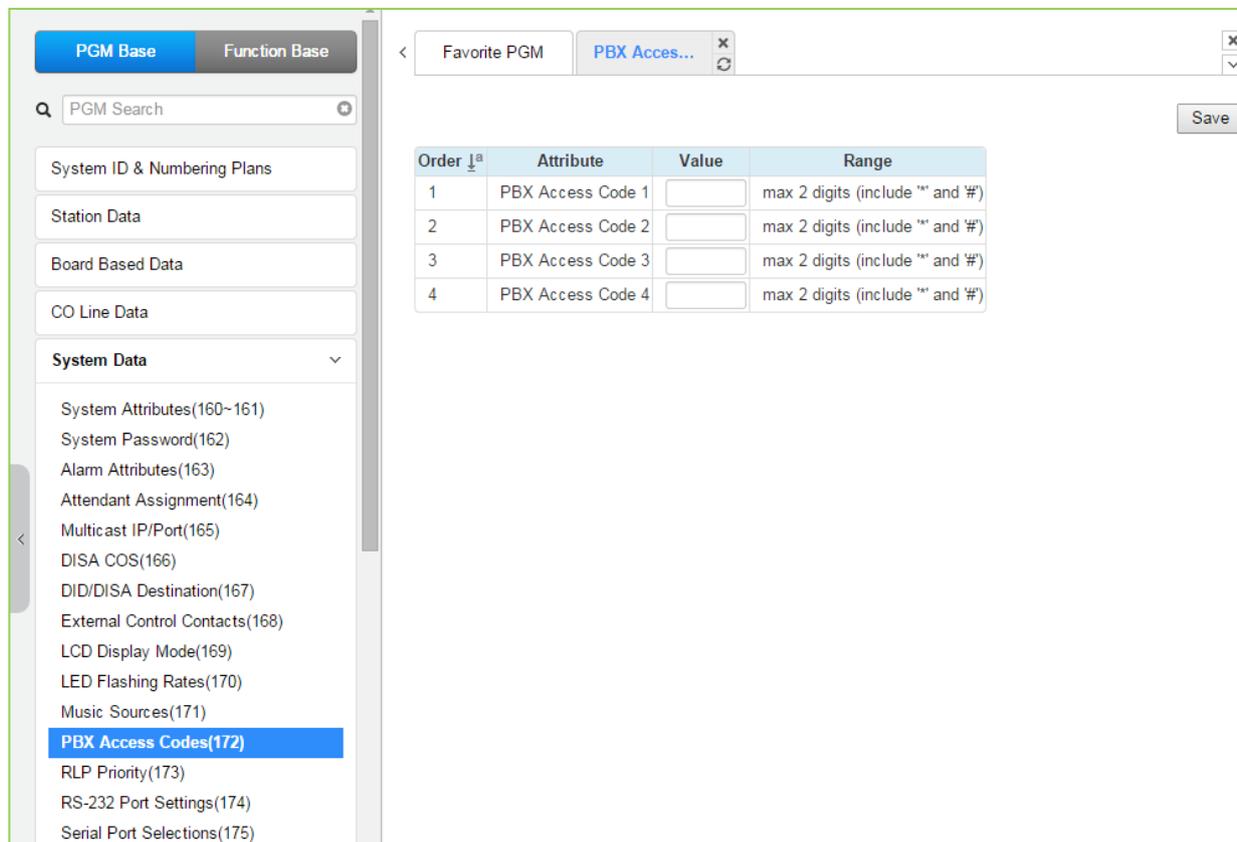
Figure 6.5.12-1 PBX Access Codes

When the system is used "behind" a PBX/CTX, the system needs to recognize the PBX/CTX Trunk access codes to implement proper dialing restriction, tone detection sequences and Flash timing. A maximum of four (4) Trunk Access Codes of one (1) or two (2) digits can be entered.

6.5.13 Ringing Line Preference Priority - PGM 173

Selecting Ring Line Preference Priority will display the Ringing Line Preference Priority data entry page. Click **[Save]** button after changing Value.
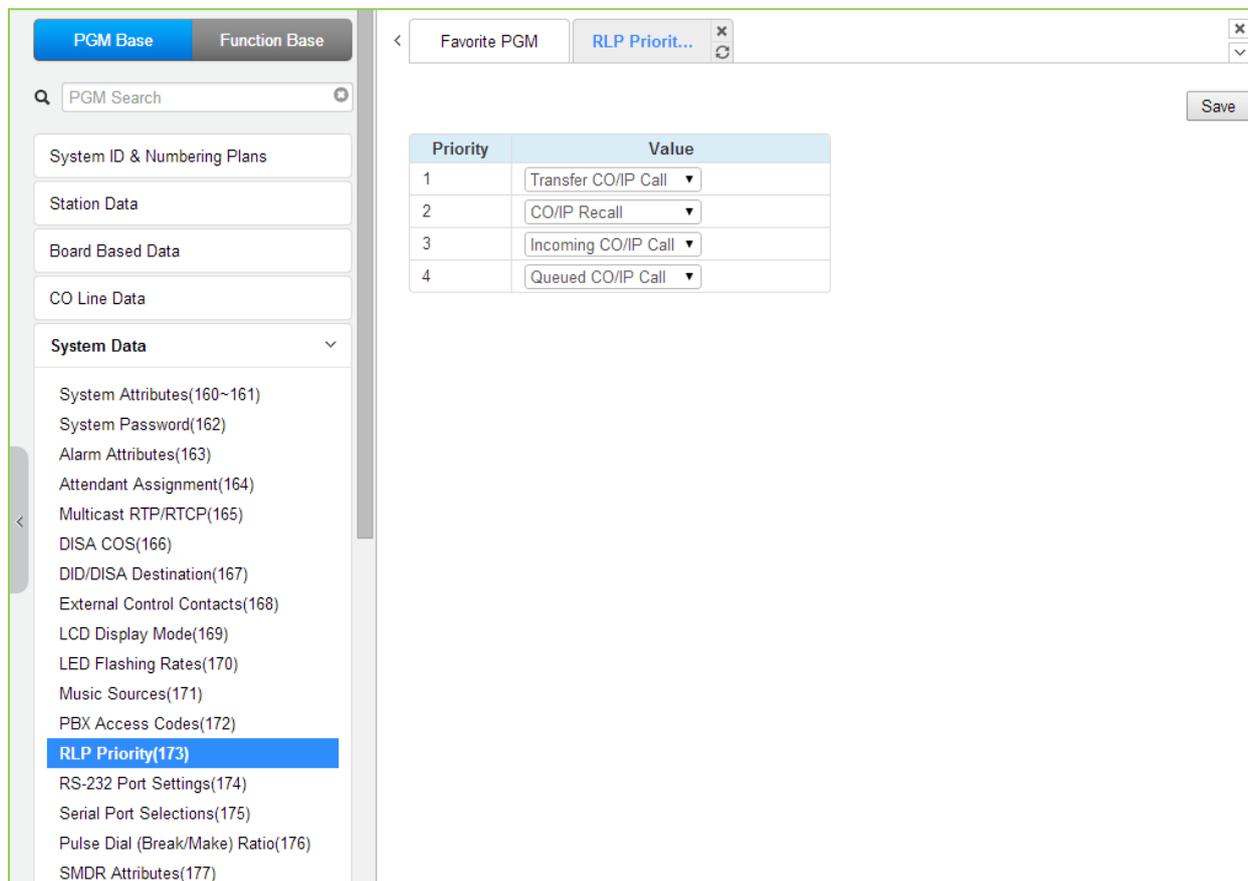
Figure 6.5.13-1 Ringing Line Preference Priority

When multiple calls are ringing at the same time to a station assigned Ringing Line Preference, the order of preference is based on the type of call: CO/IP Transfer, CO/IP Recall, Incoming CO/IP call, Queue CO/IP call. A queued CO/IP call are always assigned the lowest priority.

6.5.14 RS-232 Port Settings - PGM 174

Selecting RS-232 Port and USB 1&2 port Settings will display the RS-232 Port and USB 1&2 port Settings data entry page. Click **[Save]** button after changing Value.
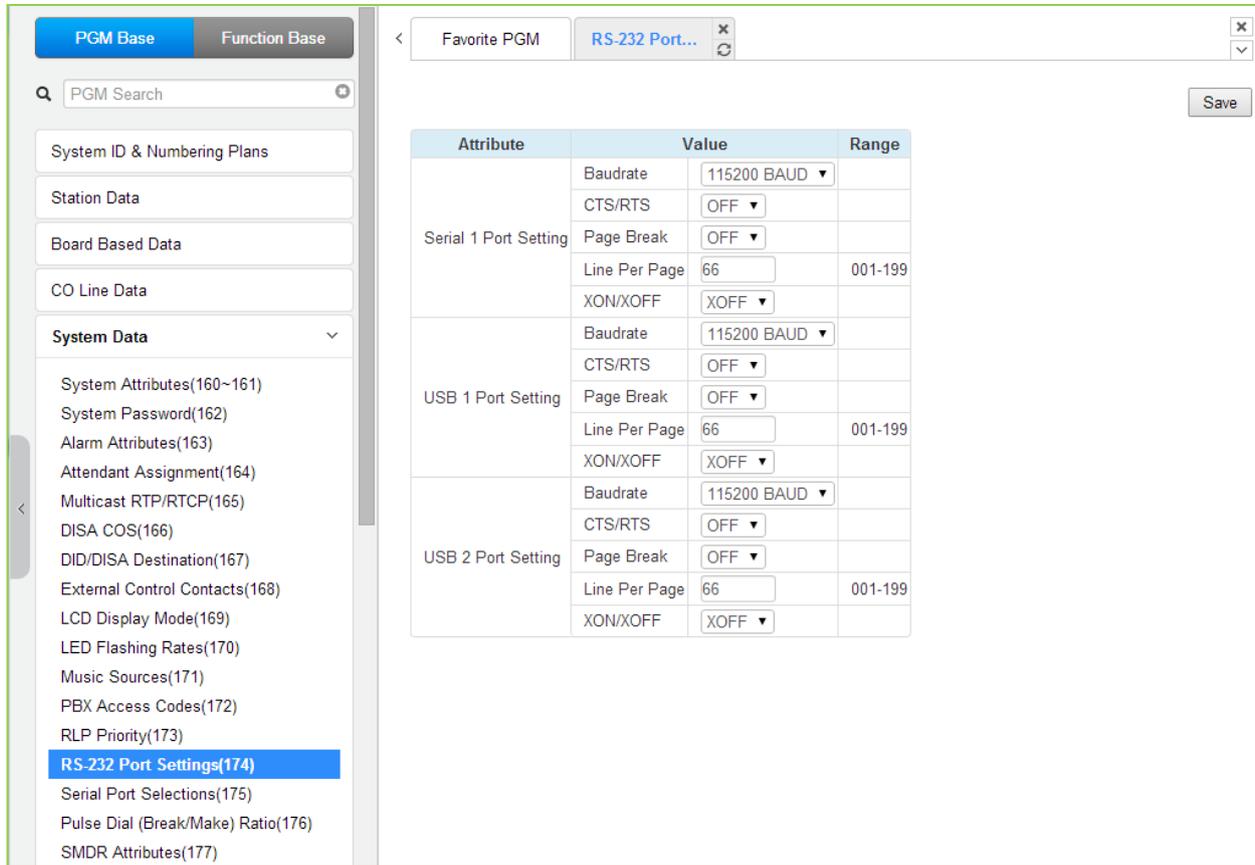
Figure 6.5.14-1 RS232 Port and USB 1&2 port Settings

Certain characteristics of each port are programmable including baud rate, RS 232 control, and page settings. Refer to the following table for a description of the settings and the data entries available.

Table 6.5.14-1 RS232 PORT and USB 1&2 port Settings

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Baud Rate	This field set the BAUD rate for the RS-232 serial port.	Unknown, 9600, 19200, 38400, 57600, 115200	115200
CTS/RTS	The system's RS232 port can support Clear-to-Send (CTS) and Ready-to-Send (RTS), control leads.	OFF ON	OFF
Page Break	The system can send a page break command over the serial port at the end of each page using the "Lines Per Page" below.	OFF ON	OFF
Line Per Page	This field sets the page length, the number of lines that the system will send before sending the page break.	001~199	66
XON/XOFF	This field enables/disables XON/XOFF protocol.	XON XOFF	XOFF

6.5.15 Serial Port Selections - PGM 175

Selecting Serial Port Selections will display the Serial Port Selections data entry page. For each function select the desired output from the drop-down menu and, if a TCP channel is assigned, enter the TCP port. Click **[Save]** button after changing Value.

Figure 6.5.15-1 Serial Port Selections

The system has six (6) serial ports (1 RS232, 3 TCP Channels and 2 USB serial). A serial port is assigned to each function that requires a serial output.

6.5.16 Break/Make Ratio - PGM 176

Selecting Break/Make Ratio will display the Break/Make Ratio data entry page. Click **[Save]** button after changing Value.
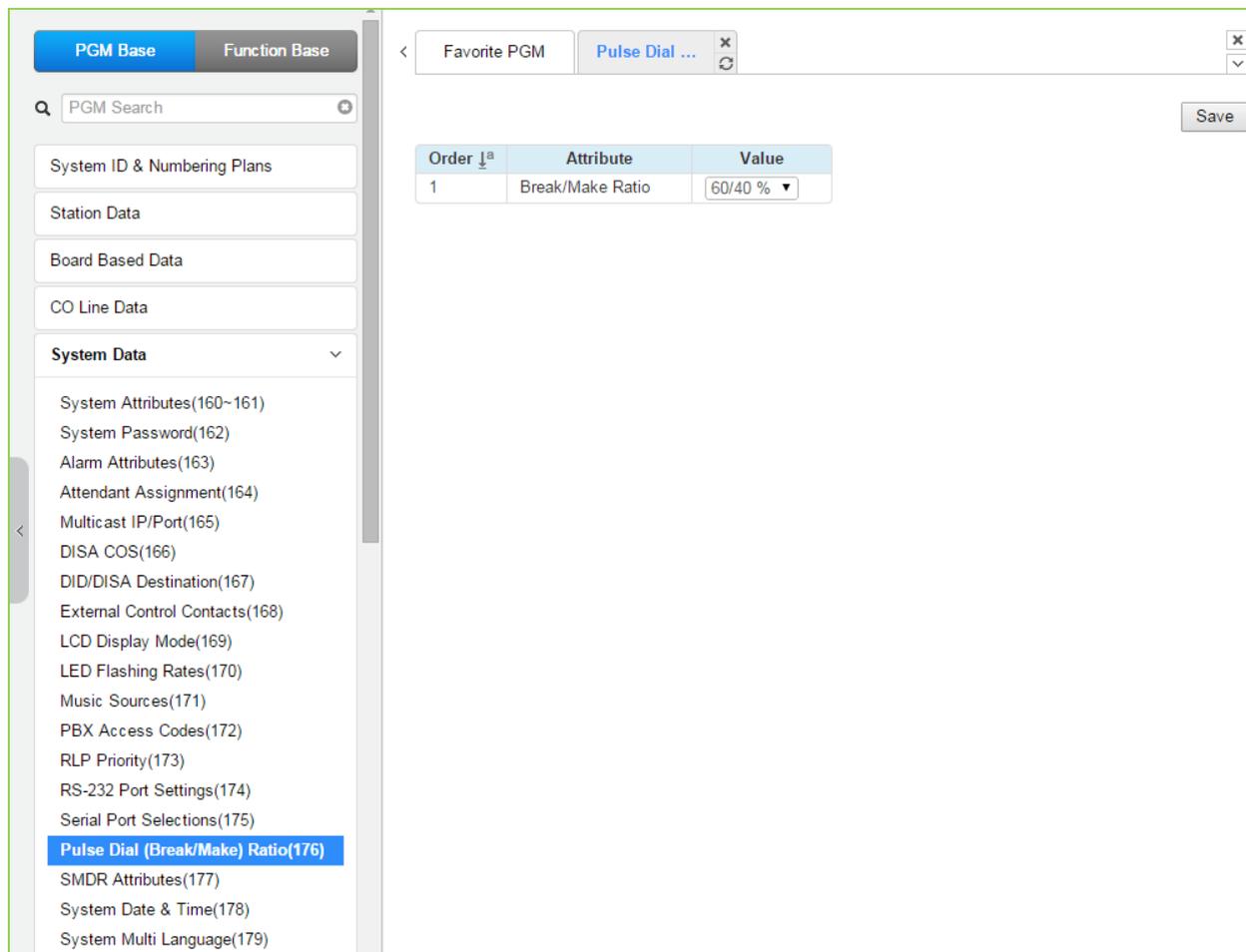
Figure 6.5.16-1 Break/Make Ratio

For Pulse dial CO Lines, the system supports 10pps and break/make ratios of 60/40% or 66/33%.

6.5.17 SMDR Attributes - PGM 177

Selecting SMDR Attributes will display the SMDR Attributes data entry page. Click **[Save]** button after changing Value.
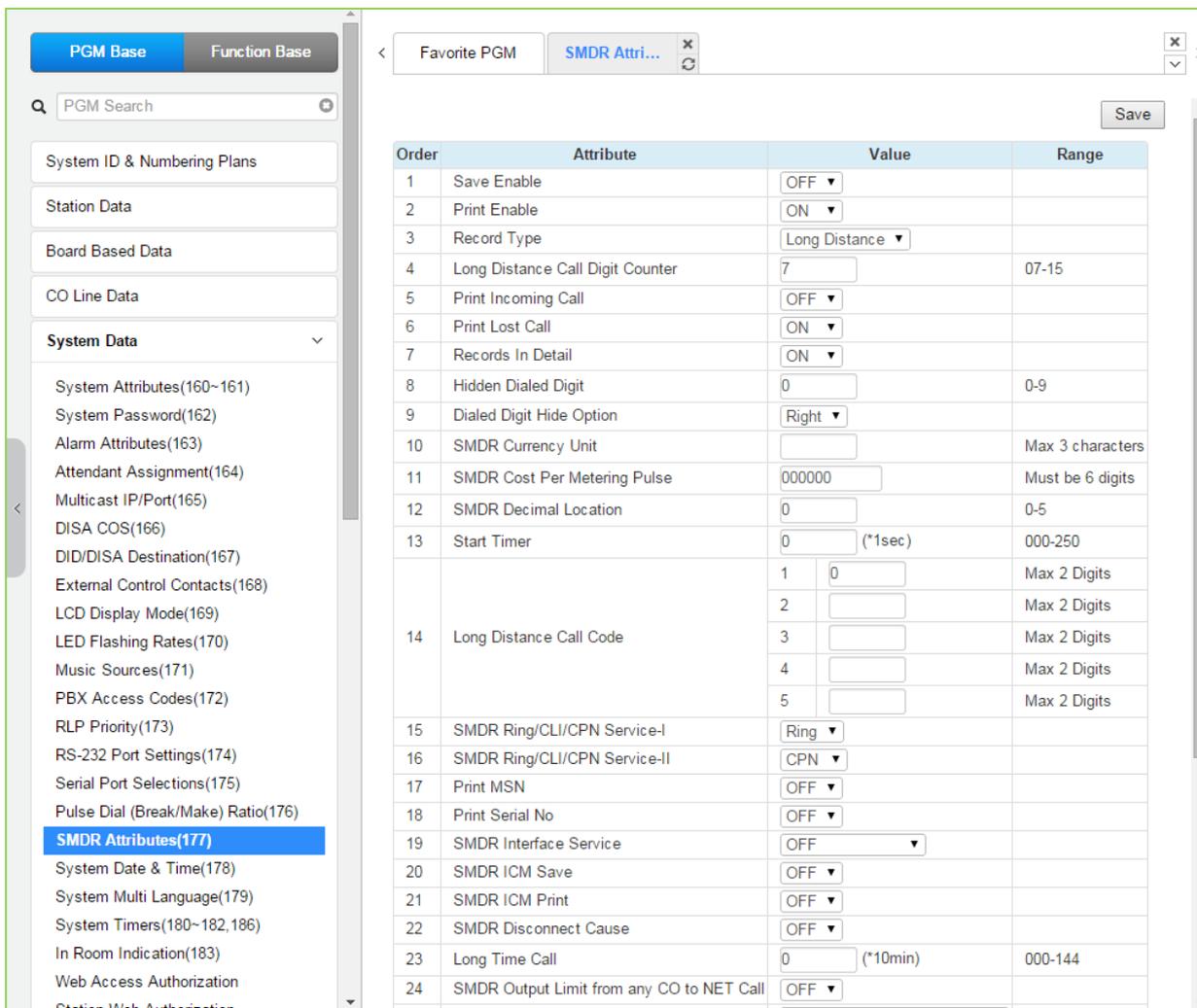
Figure 6.5.17-1 SMDR Attributes

Station Message Detail Recording (SMDR), which is output over an RS 232 port or 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. Refer to the following table for a description of each Attribute and the data entries required.

Table 6.5.17-1 SMDR ATTRIBUTES

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). For SMDR Timer settings refer to "Start Timer", below.	OFF ON	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 is sent at call completion.	OFF ON	ON

Table 6.5.17-1 SMDR ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
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 code".	Long Distance, ALL call	Long Distance
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	7
Print Incoming Call	The system can print the records for Incoming calls as well as outgoing calls. If enabled, incoming as well as outgoing calls are recorded.	OFF ON	OFF
Print Lost Call	When incoming call records are enabled, the system can also provide records for unanswered incoming (abandoned) calls.	OFF ON	ON
Records In Detail	The system can print the detailed call records (ON) or summary call information (total number of calls, cost and cost for each station).	OFF ON	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 Dialed Digit Hide Option below defines whether leading or trailing digits are hidden The station must be assigned for SMDR Hidden digits in '*Common Attributes section*'.	0~9	0
Dialed Digit Hide Option	When "HIDDEN DIALED DIGIT" is enabled above, this field determines if leading or trailing digits are hidden or not.	Right Left	Right
SMDR Currency Unit	The unit of currency for call charge can be selected as up to 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. The SMDR cost is also employed to estimate the call cost when metering is not available.	6-digits	000000
SMDR Decimal Location	This filed determines the position of the decimal in the Cost per Pulse entry above, starting from the right most digits.	0~5	0
Start Timer	To allow for call set-up times through the PSTN, a "Valid call timer" can be set.	000~250 (msec)	0
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)	
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	CPN
Print MSN	For an ISDN call involving an MSN number, the MSN number Information can be included in the SMDR Record.	OFF ON	OFF
Print Serial No	Each SMDR Record can include a record number starting at 1 and incrementing until the records are deleted. The record number will reset to 1 when SMDR capacity is reached or SMDR Mail Auto Delete Set is enabled under System Attributes.	OFF ON	OFF

Table 6.5.17-1 SMDR ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
SMDR Interface Service	When enabled, the system stores SMDR data to send to applications including NMS (Network Management System) upon request.	OFF, CO CALL, CO & ICM CALL	OFF
SMDR ICM Save	When enabled, intercom call data is stored as part of the SMDR data.	OFF ON	OFF
SMDR ICM Print	When enabled, intercom call data is printed as part of the On-line SMDR.	OFF ON	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.	OFF ON	OFF
Long time call	To monitor external calls for long duration, a "Long Time Call" can be set. If CO call duration exceeds this value, a notification will be sent to NMS server and an alarm generated. If set to "000" the feature is disabled.	000 ~ 144 (10 min.)	0
SMDR Output Limit from any CO to NET call	When a CO call is transferred to a Net transit-out CO, the local SMDR record is deleted.	OFF ON	OFF
Transfer Call Charge Rate	When a call is transferred by a station, the SMDR record can be charged based on the following options. 1. INDIVIDUAL: When a call is transferred to another station, the transferred call is charged to both stations based on the time for the call. 2. INTEGRATE XFERING: When a call is transferred to another station, the call is charged to the transferring station. 3. INTEGRATE XFERED: When a call is transferred to another station, the call is charged to the station receiving the transfer.	Individual Station, Integrate Transferring Station, Integrate Transferred Station	Individual Station
Attendant Transfer Charge Rate	When a call is transferred by an Attendant, the SMDR record can be charged based on the following options. 1. INDIVIDUAL: When the Attendant places an outgoing call and transfers the call to a station, the transferred call is charged to both stations based on the time for the call. 2. ATD CHARGING: When Attendant places an outgoing call and transfers this call to a station, the call is charged to the Attendant. 3. XFERED CHARGING: When the Attendant places an outgoing call and transfers this call to a station, the call is charged to the receiving station.	Individual Charging, Attendant Station charging, Transferred Station Charging	Individual charging
International Access Code	The system will recognize the digits assigned as the International access code digit sequence when dialed as the first digits.	Max. 4 digits	
Mobile Access Code	The system will recognize the digits assigned as the Mobile network access code digit sequence when dialed as the first digits.	Max. 4 digits	
VSF Voice mail indication	Calls to the built-in Voice Mail may be shown as "I" for Incoming call or "V" for Voice Mail in the SMDR record.	I (Incoming call), V (Voice Mail call)	I(Incoming call)
Display Nxxxx for Net number	An "N" can be included in the SMDR to identify the call as a Network call.	OFF ON	OFF

6.5.18 System Date & Time - PGM 178

Selecting System Date & Time will display the System Date & Time and DST data entry page. Click **[Save]** button after changing Value.

Figure 6.5.18-1 System Date & Time

The System Date and Time are established by the [Time & Date] menu. The date and time are employed for several features and functions including; LCR, LCD displays, SMDR outputs, Auto Ring mode Selection, Wake-up Alarm etc.

If Daylight Savings Time is enabled, the system time will be adjust one-hour forward or back at the DST start and end times, respectively.

6.5.19 System Multi Language - PGM 179

Selecting System Multi Language will display the System Multi Language data entry page. Click **[Save]** button after changing Value.
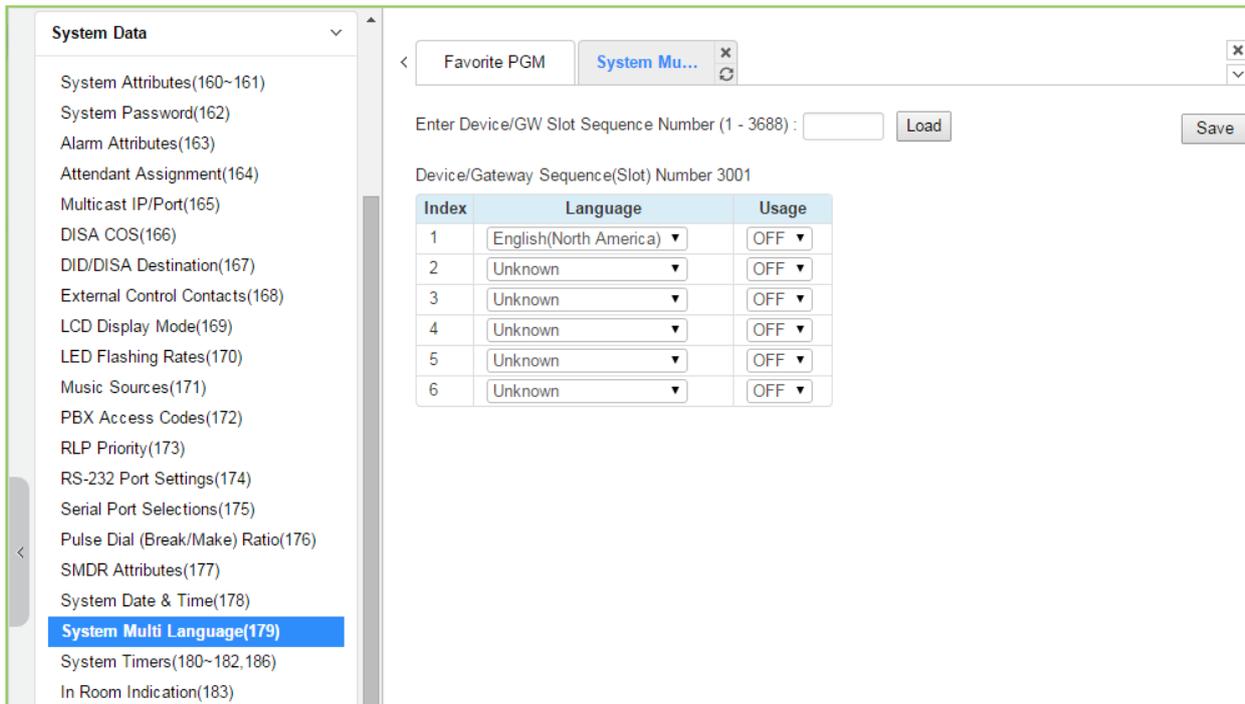
Figure 6.5.19-1 System Multi-Language

The VSF and UVM support multiple languages (18 languages); up to six languages may be supported simultaneously. Once the prompts are downloaded to the VSF and UVM, the caller receives the Language selection announcement for DISA and CCR calls as well as preceding a Station Group announcement or DID error announcement. The Language selection announcement will only affect the language prompts enabled for use with the device indicated by the Sequence number.

6.5.20 System Timers - PGMS 180 ~ 182 & 186

Selecting System Timers will display the System Timers data entry page. Click **[Save]** button after changing Value.
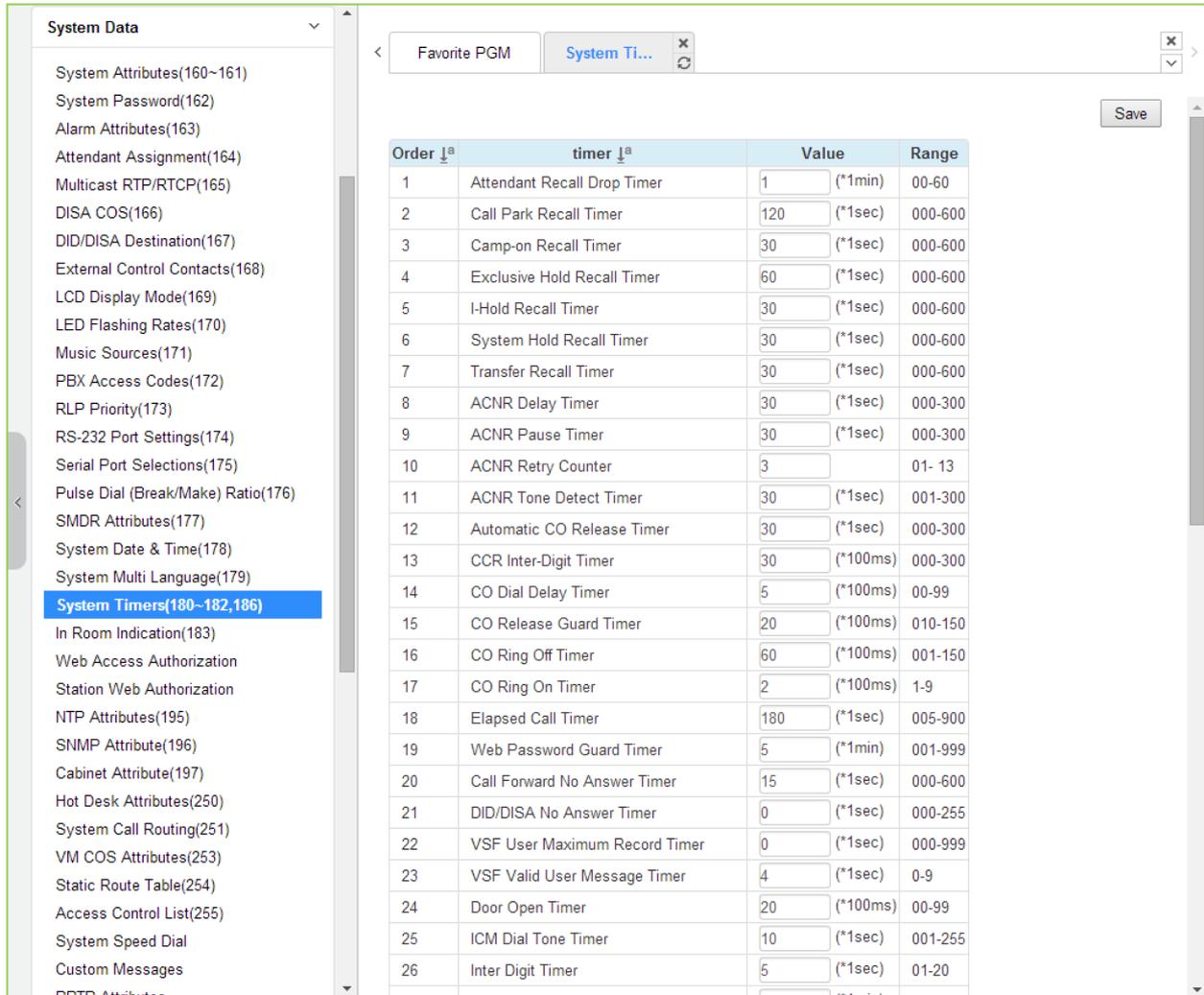
Figure 6.5.20-1 System Timers

A number of timers can be assigned to control and affect many features and functions. Refer to the following table for a description of the timers and the input required.

Table 6.5.20-1 SYSTEM TIMERS

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Attendant Recall Timer	Enter the desired time that Attendant receives recall after the system will disconnect the call.	00~60 (minutes)	1
Call Park Recall Timer	Enter the desired recall time for call park. Parked call will recall the station at a specified time.	000~600 (seconds)	120
Camp-on Recall Timer	When a call is transferred using Camp-On, this field determines the desired recall time for Camp-on. The station receives the call again with a specified time.	000~600 (seconds)	30

Table 6.5.20-1 SYSTEM TIMERS

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Exclusive Hold Recall Timer	Enter the desired time for Exclusive hold. The station receives the call again after a specified time.	000~600 (seconds)	60
Hold Recall Timer	Enter the desired time for Hold recall. The station receives the call again after a specified time.	000~600 (seconds)	30
System Hold Recall Timer	Enter the desired time for System hold recall time. The station receives the call again after a specified time.	000~600 (seconds)	30
Transfer Recall Timer	Enter the desired time for transferring the call again to the receiving station.	000~600 (seconds)	30
ACNR Delay Timer	If the ACNR Pause Timer expires and no CO/IP Line is available for ACNR recall, in this filed, you can set the delay time before ACNR attempts to access a CO/IP line again. This feature doesn't affect the ACNR retry counter.	000~300 (seconds)	30
ACNR Pause Timer	Enter the desired time for pause between ACNR recall attempts.	000~300 (seconds)	30
ACNR Retry Counter	Enter the number of ACNR retry attempts. ACNR will finish after a specified times. Regarding CIS country, the range is from 1 to 9.	1~13	3
ACNR Tone Detect Timer	If call progress tones are not available for ACNR, the system will wait the specified time after dialing before considering the called party as "busy/no answer".	000~300 (seconds)	30
Automatic CO Release Timer	If a user accesses a CO/IP Line and takes no action, the system will automatically release the CO/IP Line when the specified time is over or expired.	000~300 (seconds)	30
CCR Inter-digit Timer	This field defines the inter-digit time used with Customer Call Routing function.	000~300 (100 ms)	30
CO Dial Delay Timer	To prevent dialing when CO/PBX has slow response, dialing by the system can be delayed using this timer.	00~99 (100 ms)	5
CO Release Guard Timer	When a CO/IP Line is returned to idle, the system will deny access for the specified time to assure the PSTN returns the CO/IP Line circuitry to be idle.	010~150 (100 ms)	20
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 ms)	60
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 ms)	2
Elapsed Call Timer	Users can receive a periodic tone indicating the length of an outgoing call. This field sets the time before and between the tones. Note CO Warning Tone must be enabled for the station in '*Station Data section*'.	005~900 (1 sec)	180
Web Password Guard Timer	If no data packet is received during a Web connection, after this guard time a password check will be initiated by the system.	001~999 (minutes)	5
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 Forward timer section 6.2.12 will take precedence.	000~600 (seconds)	15
DID/DISA No Answer Timer	A DID/DISA call to a busy station will forward to the DID/DISA Destination assigned under section 6.5.7 should this timer expires.	000~255 (seconds)	0

Table 6.5.20-1 SYSTEM TIMERS

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
VSF User Maximum Record Timer	This timer sets the maximum duration allowed for a Voice Message in the built-in VSF/UVM Voice Mail.	000~999 (seconds)	0
VSF Valid User Message Timer	This timer sets the minimum duration allowed for a Voice Message in the built-in VSF/UVM Voice Mail	0~9 (seconds)	4
Door Open Timer	This timer sets the minimum time the contact assigned as a door open contact must be active to be recognized.	00~99 (100 msec)	20
ICM Dial Tone Timer	If a user goes off-hook to receive Intercom dial tone and takes no action for this timer, the user will receive error tone.	001~255 (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 or, for a SIP trunk, the digits are dialed.	01~20 (seconds)	5
MSG Wait Reminder Tone Timer	An iPECS IP or LDP Phone user will receive periodic reminder tones of a message waiting at intervals of this timer.	00~60 (minutes)	0
Paging Timeout Timer	Determines the maximum duration of a page after which the caller and Page Zone are released.	000~255 (seconds)	15
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
Soft auto RLS Timer	When a Soft Key is used on the 6000 or 7000 series iPECS IP or LDP Phone, after expiration of this timer, the display will return to the previous display.	01-30 (seconds)	10
VM Pause Timer	When the system sends a "Pause" to Voice Mail using in-band signals, this timer defines the Pause duration.	01-90 (1 msec.)	30
SLT Hook Switch Bounce Timer	This timer determines the duration the system considers an actual state change in the hook-switch and not a spurious contact bounce.	01~25 (100 msec.)	1
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.)	10
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.)	30
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)	60
Unsupervised Conference Timer	This timer determines the duration of an "Unsupervised Conference" before the conference is dropped. 00 means 10 minutes.	00~99 (minutes)	10
Prime Line Delay Timer	This timer sets the delay (no action duration) for delayed (Warm) Prime Line operation.	01~20 (seconds)	5
Wink Signal Timer	This timer sets the duration of the "Seize Acknowledge Signal" (Wink) sent to the PSTN on a DID line.	010~200 (10 msec.)	10
Enblock Inter Digit Timer	When an ISDN Line is assigned to send digits En-block, CO Attributes section 6.4.1, the system will send digits if the user dials "#" or this En-block inter-digit timer expires.	01~20 (seconds)	5
DTMF Duration Timer	This timer establishes the duration of DTMF tones sent on an analog 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 DID digits.	01~99 (100 msec)	30

Table 6.5.20-1 SYSTEM TIMERS

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
R2 Out Manage Timer	Reserved for future usage for R2 timers.	01~50 (seconds)	14
R2 In Manage Timer	Reserved for future usage for R2 timers.	01~50 (seconds)	14
R2 Disappear Timer	Reserved for future usage for R2 timers.	01~50 (seconds)	14
R2 Pulse Timer	Reserved for future usage for R2 timers.	01~30 (*20ms)	7
R2 Ready Timer	Reserved for future usage for R2 timers.	000~500 (*20ms)	7
Dial Tone Delay Timer	Reserved for future usage for R2 timers.	01~30 (*20ms)	20
Wake Up Fail Timer	At expiration of this timer, the system will notify the Attendant when a user does not respond to a Wake up alarm.	00~99 (seconds)	20
VSF Cut Error Tone Timer	The duration of Voice Messages in the built-in Voice Mail are reduced by this timer to remove error tone that may be sent by the carrier after disconnect.	00~90 (seconds)	0
On Hook Auto Idle Timer	When an iPECS IP or LDP Phone receives a disconnect message or signal from CO line the phone goes to idle after this timer.	00~99 (seconds)	0
IP Watch Timer	When Local redundancy is implemented, should the LAN cable fail, both call servers may attempt to be active. To protect against this "dual active" case, the system can periodically check for an active back-up UCP module.	0~250 (seconds)	0
Prepaid Call Drop Warning Timer	When the Prepaid funds are exhausted, the user will receive a warning tone indicating the call will be dropped after this timer expires.	00-99 (seconds)	10
Emergency retry timer	The system attempts to complete an Emergency call using the defined Emergency group. If, after this timer, a CO/IP Line is not available in the group, the system will select any available CO/IP Line to complete the call.	00~99 (seconds)	0
Record Warning Repeat Timer	When Call Record Warning tone, PGM 161 is ON, the tone can be repeated at intervals of this timer for the duration of the call.	000~999 (seconds)	0
Error Tone Timer	This timer sets the duration for Error Tone.	5~180 (seconds)	30
Howling Tone Timer	This timer sets the duration for Howling Tone.	0~180 (seconds)	30
VM Notify Play Delay Over LCO Timer	When VM notification to the mobile uses an analog CO Line, the system will treat the call as answered after this timer and play the new message prompt.	1~99 (1 sec.)	10
Fax Detect timer	It is maximum fax tone detection time to deliver FAX call to fax destination.	1~20 (1 sec.)	10
Auto Pause Release timer	IPCR Mute function will be released by this auto pause released timer.	0~255 (1 sec.)	0
Short Modem Timer	If {Short modem} of a SLT is ON, the SLT maintains the modem mode for this time.	01~60 (1 sec.)	10

6.5.21 In-Room Indication - PGM 183

Selecting In-Room indication will display the In Room data entry page. Click **[Save]** button after changing Value.
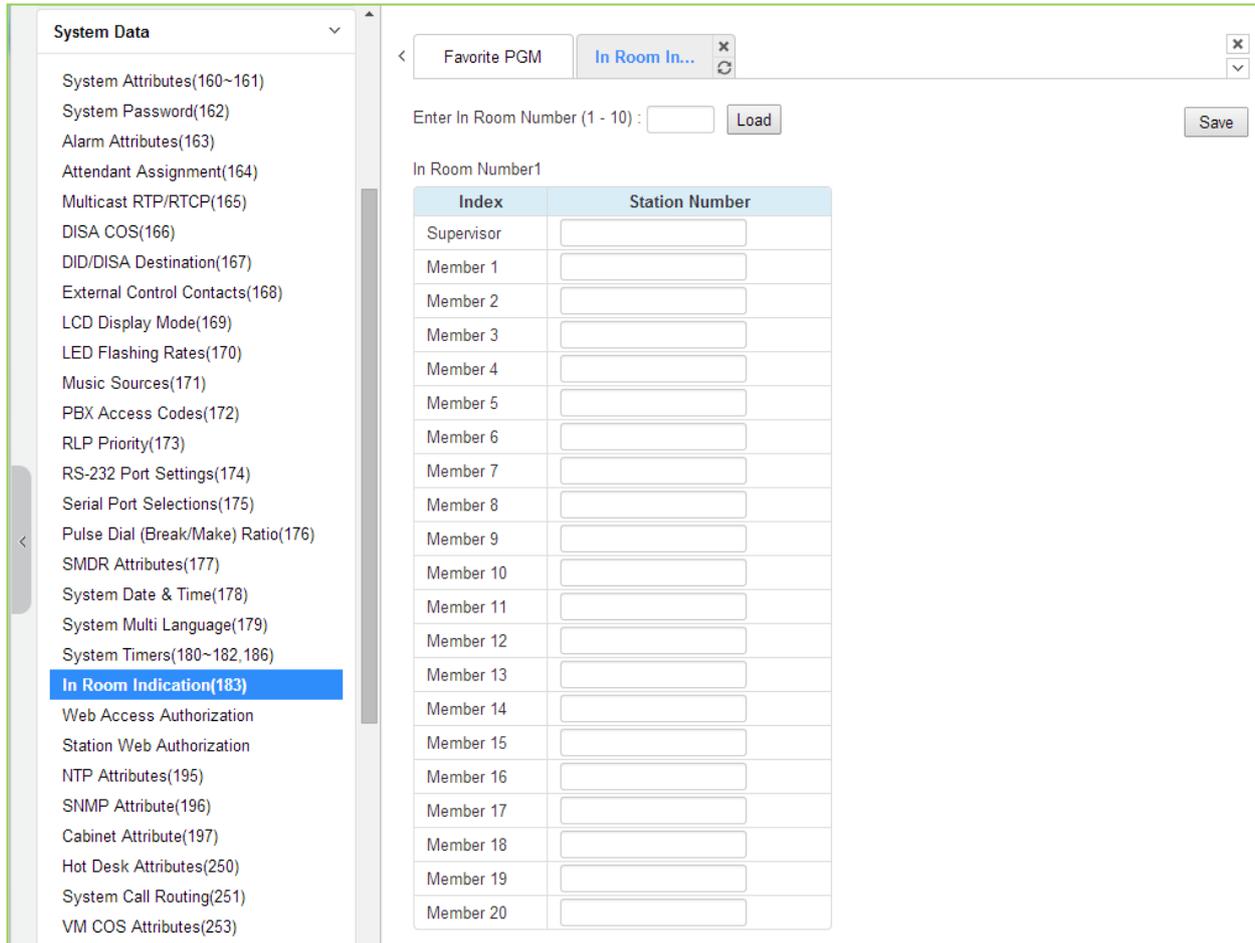
Figure 6.5.21-1 In-Room Indication

The Supervisor Station can set the In-Room Indication for all members in the In-Room indication group. Up to 10 Supervisors (groups) can be configured, and each can have up to 20 members in the group, excluding the Supervisor.

Table 6.5.21-1 IN-ROOM INDICATION

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Supervisor	This entry assigns the Station number for the In-Room Group Supervisor.	Station number	
Member 01~20	This entry assigns stations as members of the In-Room Group.	Station number	

6.5.22 Web Access Authorization

Selecting Web Access Authorization will display the Web Access Authorization data entry page. This page is only displayed when a password is defined. Click **[Save]** button after changing Value.

A user can select three authorization option (N/A, Read, Read/Write) only in Web Access & Station Web Authorization PGM of Web Admin.
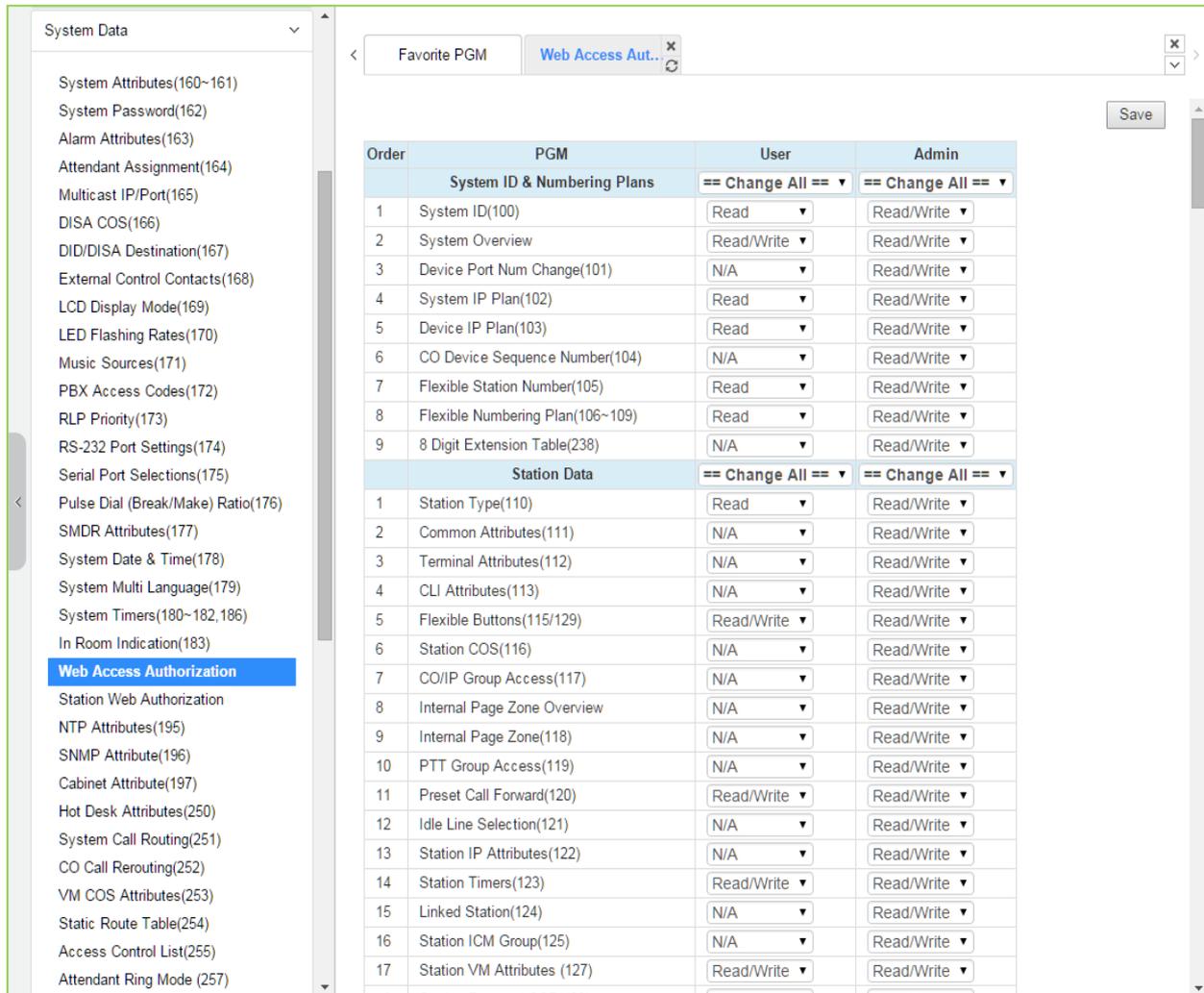
Figure 6.5.22-1 Web Access Authorization

Three different passwords can be assigned for the access to the iPECS Web administration so that the different levels of access to the Admin Web pages can be allowed. Two levels, the User and Admin level, have access to the assigned Web Admin pages and the Database, SMDR, and Voice Mail Delete pages under the Maintenance tab. The Maintenance password has access to all the Web pages and the Maintenance pages including trace settings, device log view, gain & cadence control, lock key install and device delete feature. In addition, the Maintenance level user can assign the authorities of the Admin and User levels.

6.5.23 Station Web Access Authorization

Selecting Station Web Authorization will display the Station Web Access Authorization data entry page. Click **[Save]** button after changing Value.

Figure 6.5.23-1 Station Web Access Authorization

Three access levels can be assigned to each station for access to the Station Web pages in Station Data (Common Attributes (111): 'Station Web Level'). Level 1 has access to all Station pages and attributes. The pages and attributes for Levels 2 and 3 are programmable.

6.5.24 NTP Attributes - PGM 195

Selecting NTP Attributes will display the System NTP Attributes entry page.
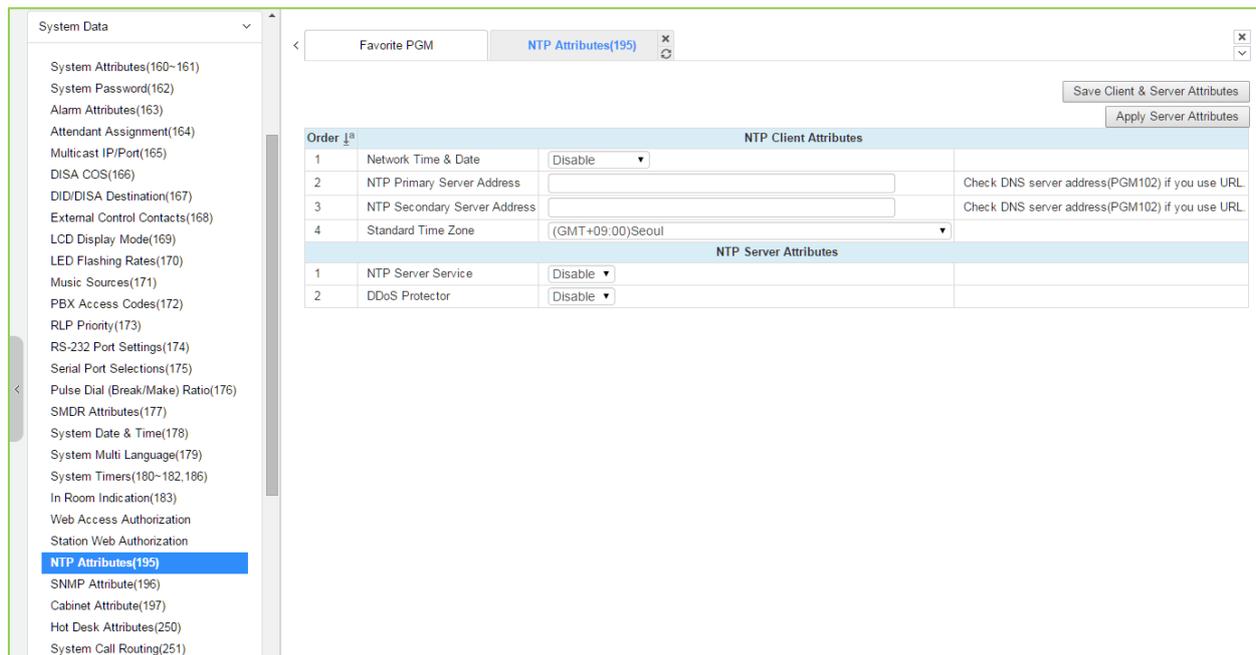
Figure 6.5.24-1 NTP Attributes

The system can employ the Network Time Protocol (NTP) to synchronize the system time with an NTP server. The system requests the time from the NTP server at 10-minute intervals and then determines the time differential. If the system time is more 2 seconds off the NTP time, the system time is adjusted to synchronize with the NTP server time.

6.5.25 SNMP Attribute - PGM 196

Selecting SNMP Attribute will display the SNMP Attributes entry page. Click **[Save]** button after changing Value.
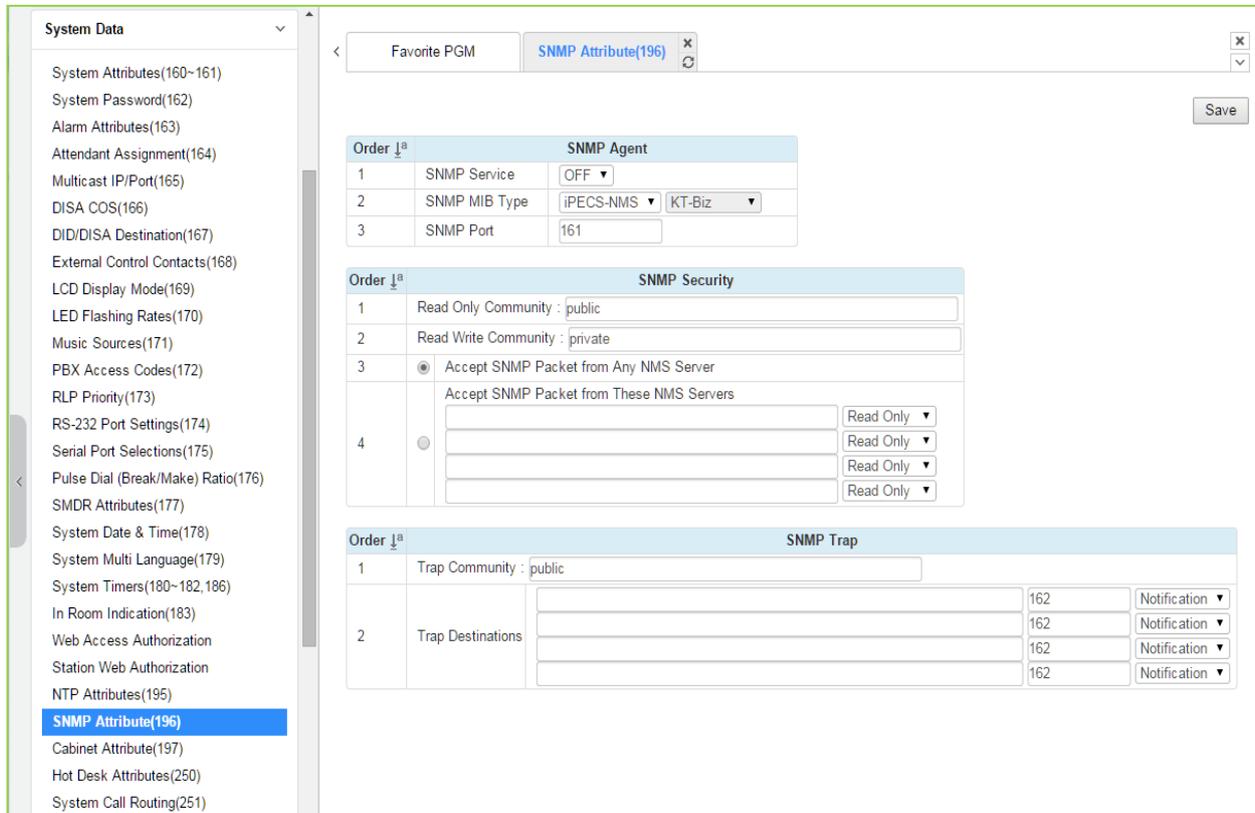
Figure 6.5.25-1 SNMP Attribute

SNMP Attributes, as shown on the screen, are divided into three categories: SNMP Agent, SNMP Security, and SNMP Trap. The SNMP Service field enables the SNMP agent running in the iPECS call server. The SNMP port field defines the UDP port used for communications from the iPECS UCP for SNMP messages. This port should not be changed.

In SNMP Security are the Read Only and Read Write SNMP Community fields, 4 to 16 characters. The SNMP community designates an SNMP communication group to which an SNMP message belongs, and is a logical relationship between the SNMP agent (iPECS UCP) and SNMP manager (iPECS NMS). The SNMP community settings must be the same for the iPECS UCP and the iPECS NMS server.

- Read Only Community (default=Public)—Defines a community string used when the iPECS NMS reads data from the iPECS UCP.
- Read Write Community (default=Private)—Defines the community string used when iPECS NMS reads or writes data to the iPECS UCP.

Although the iPECS UCP can accept packets from any SNMP manger such as iPECS NMS, for improved security, the IP address of specific servers can be defined and allowed Read only or Read Write access. It is recommended that the system be assigned with the IP address of only one NMS server with Read Write access.

The SNMP Trap configuration defines the Trap Community, and the Trap Destination, which

includes the IP Address of the SNMP manager, iPECS NMS, and the .message type. The Trap Community designates a communication group to which a Trap message belongs, and is a logical relationship between the SNMP agent (iPECS UCP) and SNMP manager (iPECS NMS). This 4 to 16--character string should be the same as the Trap community string defined in the iPECS NMS. The Trap community should be the same for all the iPECS UCP systems registered to an iPECS NMS server whereas the SNMP community may be defined with different strings for each iPECS UCP.

The Trap Destination defines the IP address of the iPECS NMS server and the port, 162. Enter the IP address of the NMS server but do not change the port. The pull down menu next to the address is used to define the message type. Three values are available:

- Trap – message type is defined in SNMPv1, but because iPECS-NMS and the iPECS UCP use SNMPV2, the Trap type message is not recommended.
- Notification – message type sent from the SNMP agent once without checking the reception of the message.
- Inform – message type requires an acknowledgement from the SNMP manager. If the agent does not receive a response, the message is resent. Inform messages are intended for use in environments with high packet loss however, use of the Inform message type may detrimentally affect the iPECS UCP performance.

The iPECS SNMP attributes are defined here. Refer to the following table for description and values that can be entered.

Table 6.5.25-1 SNMP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
SNMP service	SNMP Service field is used to set the SNMP agent in the iPECS ON or OFF.	OFF ON	OFF
SNMP MIB Type	Select SNMP MIB specification. U-CEMS is for KOREA telecom speciation.	iPECS-NMS, U-CEMS	iPECS-NMS
SNMP Port	SNMP Protocol port number		161
Read Only Community	Read only community should be used when SNMP manager (NMS) is trying to read data from SNMP agent (UCP)	4 ~ 16 characters	public
Read Write Community	When the SNMP manager (NMS) needs to read and write data to the agent (iPECS UCP). This attribute should be enabled.	4 ~ 16 characters	private
Trap Community	For the SNMP agent (UCP), this field defines the destination IP address to receive trapped messages (Alarm/fault events).	4 ~ 16 characters	public
Trap Destination	IP address of iPECS NMS server, port 162 should not be changed.	IP address	Public
Message Type	Defines how the agent sends the Message.	Notification, Inform, Trap	Notification

6.5.26 Cabinet Attribute - PGM 197

Selecting Cabinet Attribute will display the Cabinet Attributes entry page. Click **[Save]** button after changing Value.

Figure 6.5.26-1 Cabinet Attribute

This Web page displays system cabinet configurations and alarm status.

6.5.27 Hot Desk Attributes - PGM 250

Selecting Hot Desk Attributes will display the Hot Desk Attributes data entry page. Click **[Save]** button after changing Value.
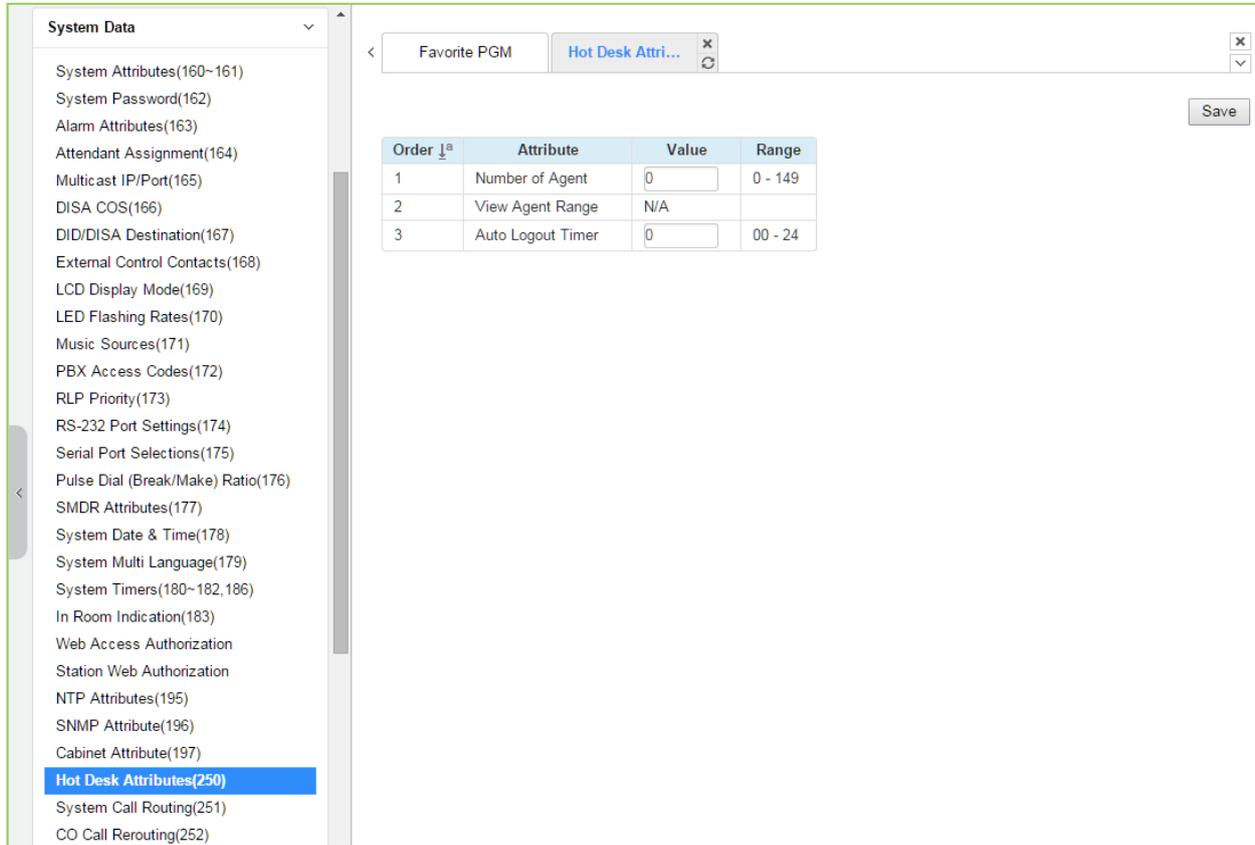
Figure 6.5.27-1 Hot Desk Attributes

A Hot Desk station allows a user to login for access to the system features and resources. Once logged in, the user is provided access to system features and resources employing the database for the user's assigned station.

User station numbers, which are used as the Agent Id, are assigned automatically by the system. The system assigns station numbers to each agent starting at the highest station number available.

Table 6.5.27-1 HOT DESK ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Number of Agents	Assign number of hot desk agent.	UCP100: 0-199 UCP600/1200:0-300	0
View Agent Range	View the assigned station numbers for agents.		N/A
Auto Logout Timer	A Hot desk station will return to inactive if the logged in user takes no action for the Auto Logout timer.	00~24 Hrs.	00

6.5.28 System Call Routing - PGM 251

Selecting System Call Routing will display the System Call Routing data entry page. Enter a valid Index range for the scenarios and click **[Load]** to enter Call Routing data. Click **[Save]** button after changing Value.

Figure 6.5.28-1 System Call Routing

System Call Routing establishes scenarios with criteria to route calls. Criteria include time of day, day of week, Caller and Called numbers, etc. System Call Routing takes precedent over other system based call routing. However, Station Call Routing scenarios take precedence over System Call Routing scenarios.

Table 6.5.28-1 System Call Routing Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Caller ID	This field defines the Caller Id for the scenario.	Max. 23 Digits	N/A
Called Num	This field defines the Called number for this scenario.	Max. 23 Digits	
Time condition (Start Day and End Day, weeks, start time and end time)	The time and day for activation of the scenario can be defined.	YYYY-MM-DD hhmm (Must be 4 digits)	
Destination (Type and Value)	This field defines the destination type and value for call routing when the scenario criteria are met.	STA Station Group, SPD, PABX, VSF, VSF(#), Net Station, Company Directory-	N/A

Table 6.5.28-1 System Call Routing Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
		Last name, INT Page, EXT Page, All Page, VM, ICLID Table	
Scenario Priority	Each scenario can be assigned a priority. The highest priority scenario meeting the criteria is used to route the call.	0-9 (Highest priority)	
Scenario Active	A scenario must be active to be employed. If not active, the scenario is ignored.	ON OFF	OFF
Scenario VMID	A Voice Mail Id can be associated with the scenario for routing to a Voice Mailbox.	Voice mail ID	
Scenario COS	If DISA is active for the scenario, the COS can be assigned for the call.	0-11 (COS level)	0
Scenario DISA Active	The scenario can employ DISA for the incoming call routing.	OFF, ON	OFF
Scenario ICM Group number	The scenario can route calls to a specific Tenancy group.	0-100	0
Scenario Zone	A zone can be assigned so that only CO calls to a CO/IP Line in the Zone will route based on the scenario.	0-32	0
Scenario Start CO and End CO	This field defines a range of CO/IP Lines that will employ the scenario.	Start CO:0-998, End CO: 0-998	0
Scenario Group: this is used for scenario group by attendant.	Scenarios can be group allowing the Attendant to select a group of scenarios to route calls.	001~100 (00: Unused)	0
Zone Holiday	A Zone can be assigned for routing calls using the scenario during Holiday periods defined for the Zone.	0-32	0

6.5.29 CO Call Rerouting - PGM 252

Selecting CO Call Rerouting will display the CO Call Rerouting data entry page. Enter a valid Index range and click **[Load]** to enter Call Routing data. Click **[Save]** button after changing Value.
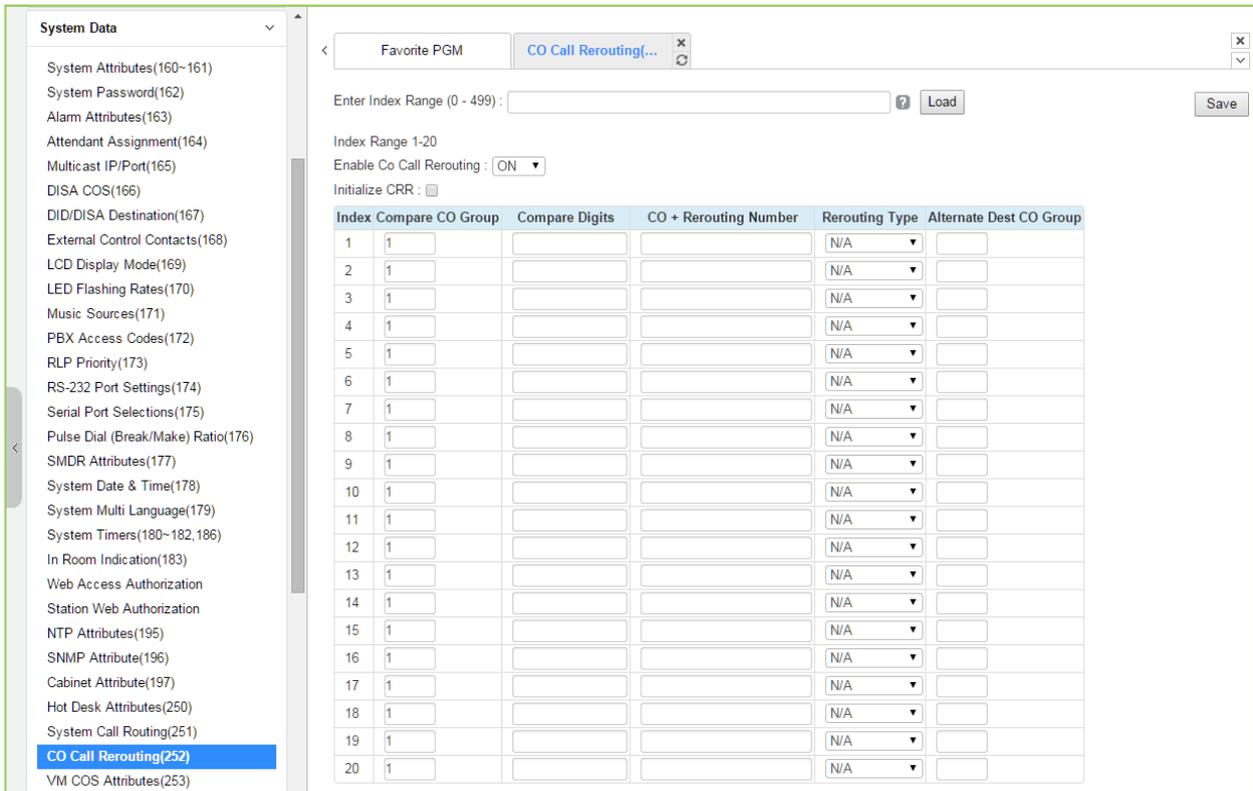

Figure 6.5.29-1 CO Call Rerouting

CO Call Rerouting establishes routing for CO/IP calls with a specified caller id on CO/IP Lines from a specified group. The rerouting sends calls out over another CO/IP Line or group, a network destination, a DISA call or to another station. The incoming CO group and compare digits determine if the call should be rerouted and the destination is determined by the CO code and Telephone number, which is dialed when the incoming group and compare digits are matched. The routing type determines if the call is routed normal (N/A) or if the call is routed over a network or employs DISA.

Example CRR chart

Index	Incoming CO group	Compare Code	CO Code + Tel number	Routing Type
0	1	454	88005123456	N/A
1	2	456**	8901123456	N/A
2	1	42*555	9123456	N/A
3	5	353	8901123456	NET Type
4	5	401		DISA Type

- Index 0: If an incoming call on a CO/IP Line from group has digits "454" then seize CO 5

and send digit 123456.

- Index 1: If an incoming call on a CO/IP Line from group 2 has digits "456**" then seize CO group 1 and send digit 123456.

- Index 2: If an incoming call on a CO/IP Line from group1 has digits "42*555" then seize the first CO/IP Line and send digit 123456.

- Index 3; if an incoming call on a CO/IP Line from group 5 has digits "353" then seize CO group 1 and send digit 123456 as a transit-out call over the Network.

- Index 4: If an incoming call on a CO/IP Line from group 5 has digits "401" then activate DISA and await digits from the caller.

6.5.30 VM COS Attributes – PGM 253

Selecting VM COS Attributes will display the VM COS Attributes data entry page. Select a valid VM COS and click **[Load]** to enter VM COS data. Click **[Save]** button after changing Value.
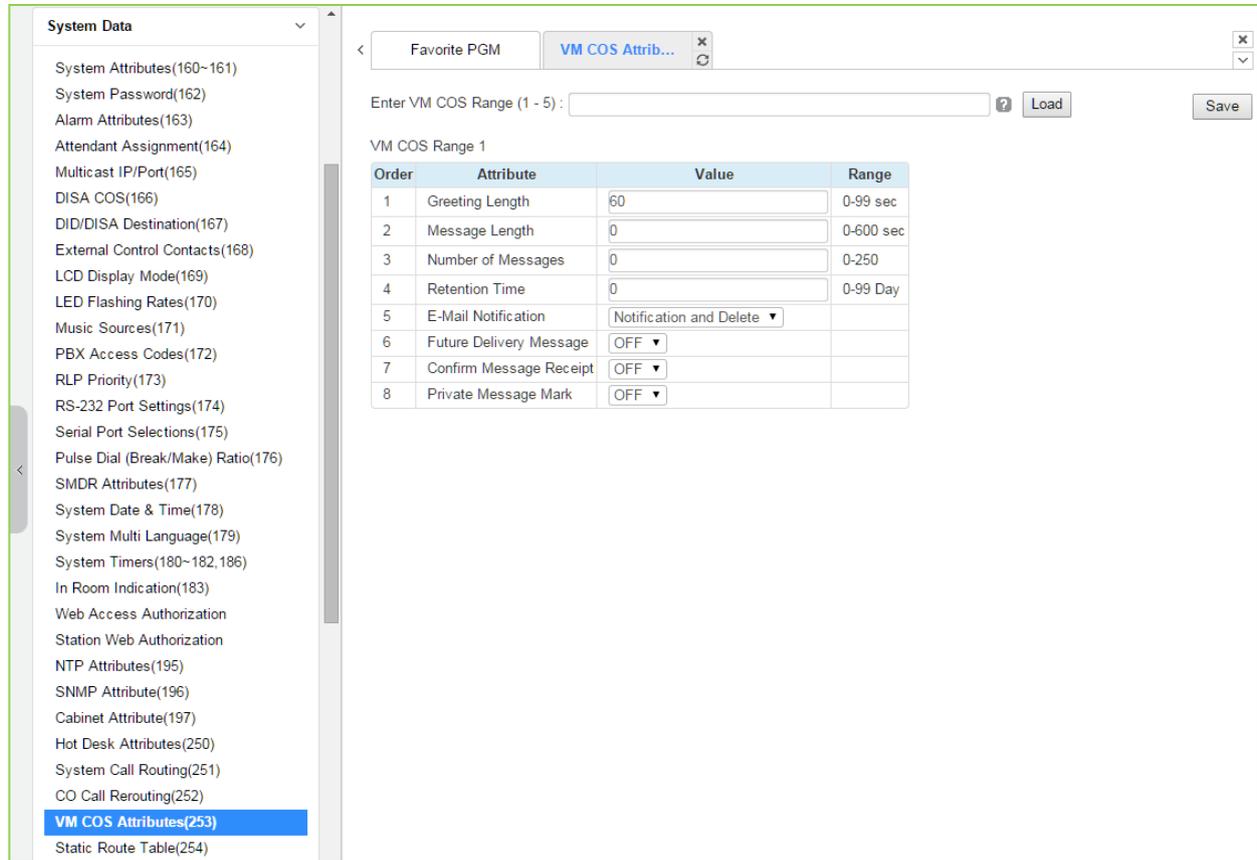
Figure 6.5.30-1 VM COS Attributes

VM COS establishes various common characteristics of the user's Voice Mailbox including greeting and message length, E-mail notification, message retention, etc.

Table 6.5.30-1 VM COS ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Greeting Length	This defines maximum user greeting length.	0-99 (Seconds)	60
Message Length	This defines maximum user message recording time.	0-600 (Seconds)	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 number of days.	0-99 Days	0
E-Mail Notification	E-mail notification can be enabled or disabled and, if enabled, the message can be deleted after notification.	Disable, Notification Only, Notification & Delete	Notification & Delete
Future Delivery Message	Future Delivery of messages can be enabled or disabled.	OFF ON	OFF
Confirm Message Receipt	Confirm message receipt can be enabled or disabled.	OFF ON	OFF
Private Message Mark	Private message mark can be enabled or disabled.	OFF ON	OFF

6.5.31 Static Route Attributes – PGM 254

Selecting Static route Attributes will display the static route attributes data entry page. Click **[Save]** button after changing Value.
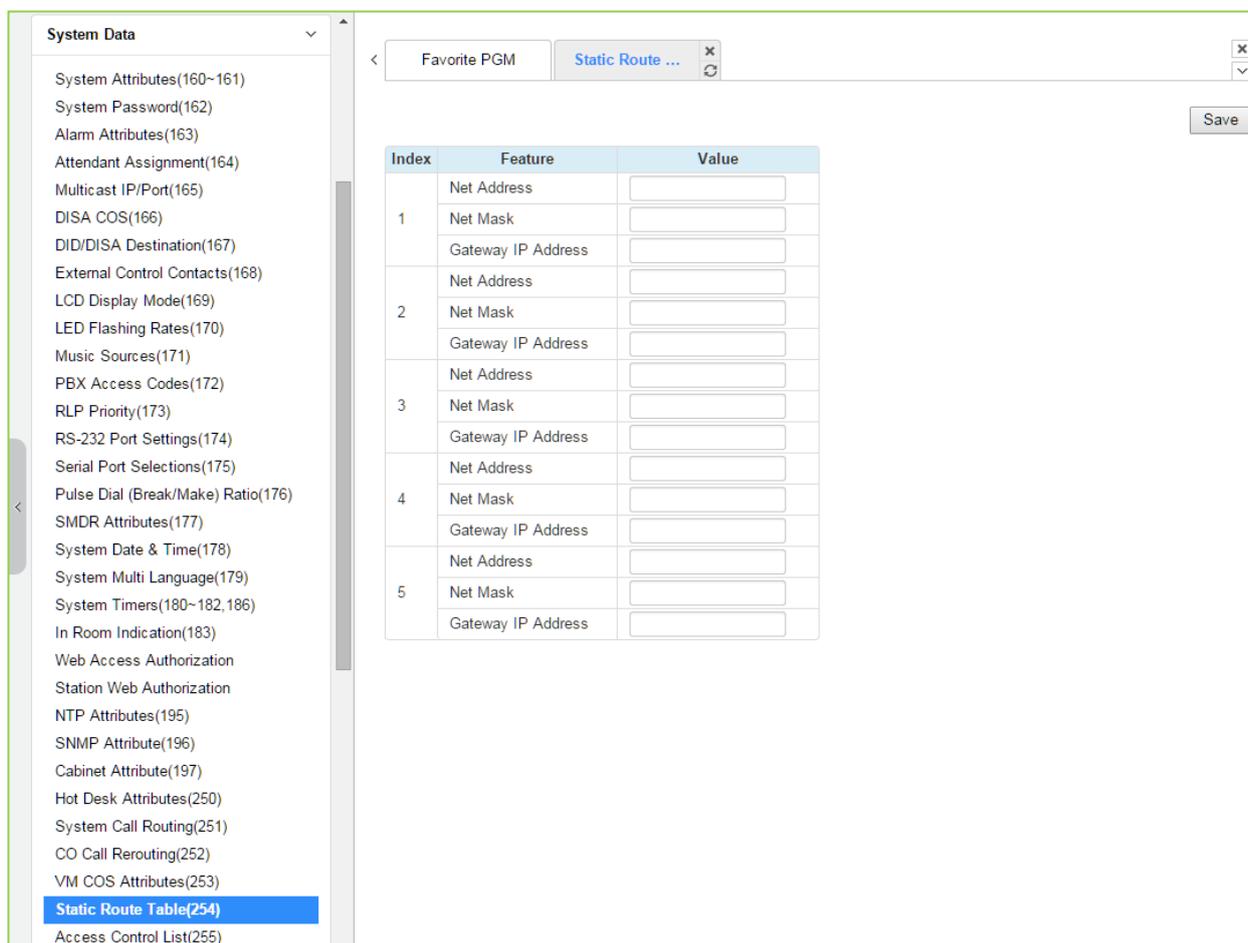
Figure 6.5.31-1 Static Route Table Attributes

Table 6.5.31-1 STATIC ROUTE TABLE ATTITUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Net address	Network IP address	IP address	
Net mask	Net mask		
Gateway IP address	Gateway (route) IP address	IP address	

6.5.32 Access Control List – PGM 255

Access Control List will display the access control attributes data entry page. Enter a valid index range and click **[Load]** to enter ACL data. Click **[Save]** button after changing Value.
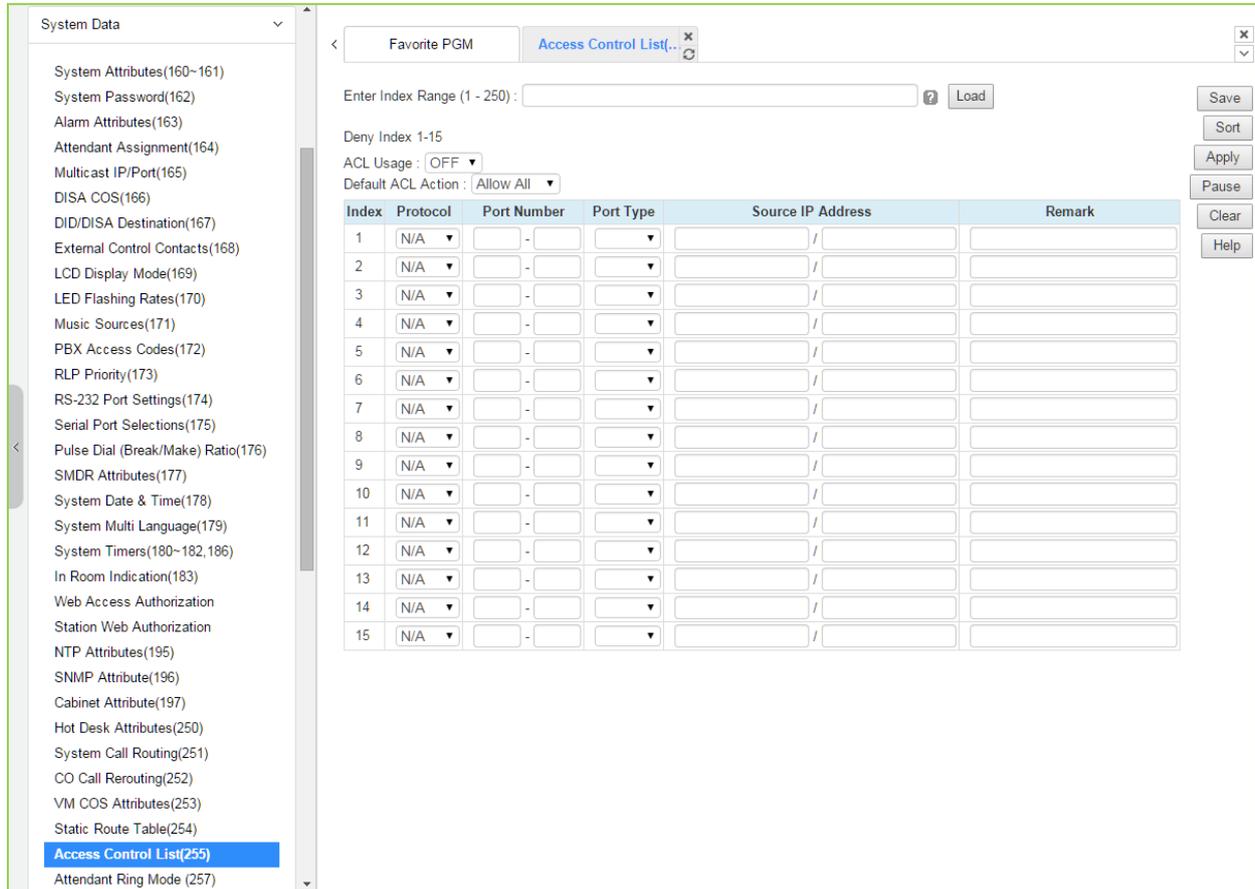
Figure 6.5.32-1 Access Control List

Access Control determines the Source IP addresses that can access the system for specific protocols.

Table 6.5.32-1 ACCESS CONTROL LIST

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Protocol	This field defines the accessing protocol type.	N/A, ALL, TCP, UDP, ICMP	N/A
Port number	This field further defines the protocol TCP/IP port number.		
Port type	The port number can be configured as the source or destination port number.	DEST SRC	
Source IP address	The source IP address and net mask allowed access is defined by this field.		
Remark	This is a general remark field.		

6.5.33 Attendant Ring Mode – PGM 257

Selecting Attendant ring mode will display Attendant ring mode entry page. Click **[Save]** button after changing Value.
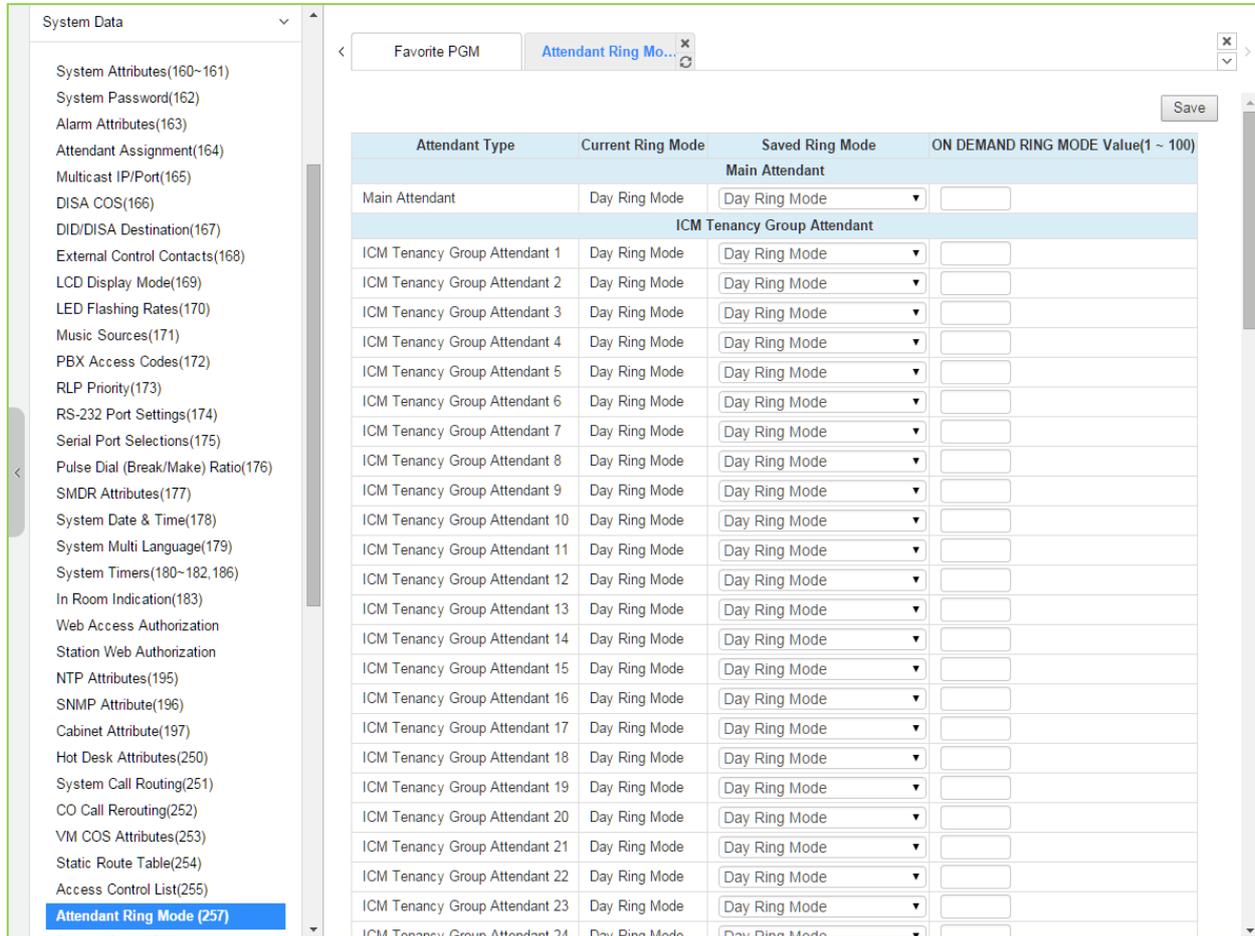
Figure 6.5.33-1 Attendant Ring Mode

Administrator assigns the ring mode to 1 Main attendant and 15 ICM Tenancy Group Attendant.

Five ring modes are supported as Day, Night, Timed, Auto, On demand ring mode (1~100).

The Attendant controls the system Ring mode changing from Auto ring Mode to Day, Night, Timed or On demand ring mode. Based on the ring mode selected in the field of Saved Ring Mode, different ring assignments, COS (Class of Service) and answering privileges are invoked for the system users.

In case Main attendant select the other ring mode of Day ring mode in the field of Saved Ring Mode, ICM Tenancy group attendant controls the system ring mode instead of Main attendant. So Current Ring Mode and Saved Ring Mode of ICM Tenancy group attendant may be different.

6.5.34 System Speed Dial

Selecting System Speed Dial will display the System Speed Dial entry page. Enter a valid range of System Speed Dial numbers and click **[Load]** to enter Speed Dial data. Click **[Save]** button after changing Value.
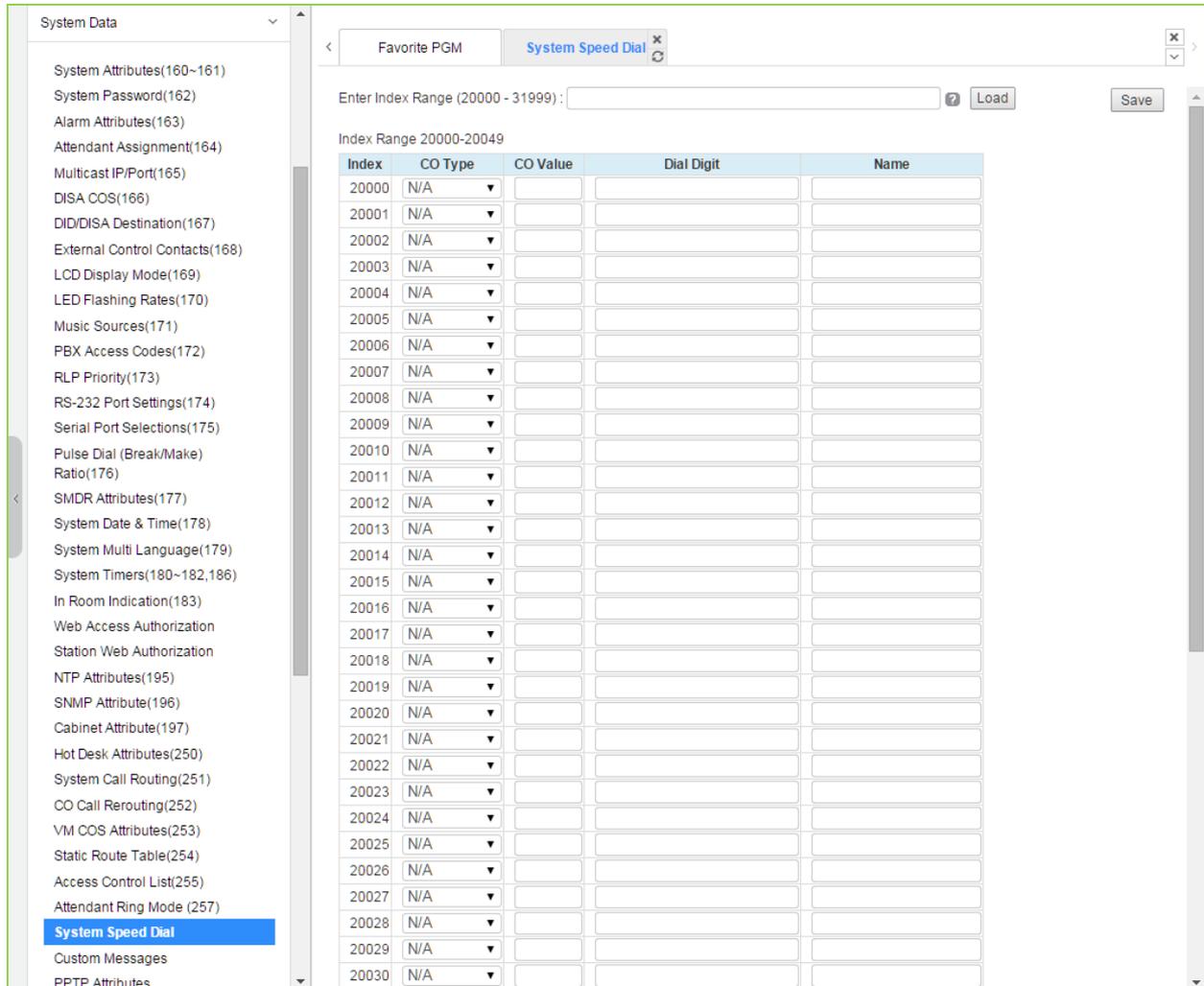
Figure 6.5.34-1 System Speed Dial List

The UCP has memory for 12,000 Speed Dial numbers of up to 25 digits each. Each System Speed bin (index) is assigned the CO/IP Line for the Speed Dial, the number to be dialed and a name for Dial-by-Name.

6.5.35 Custom Messages

Selecting Custom Messages will display the Custom Message Table data entry page. Click **[Save]** button after changing Value.
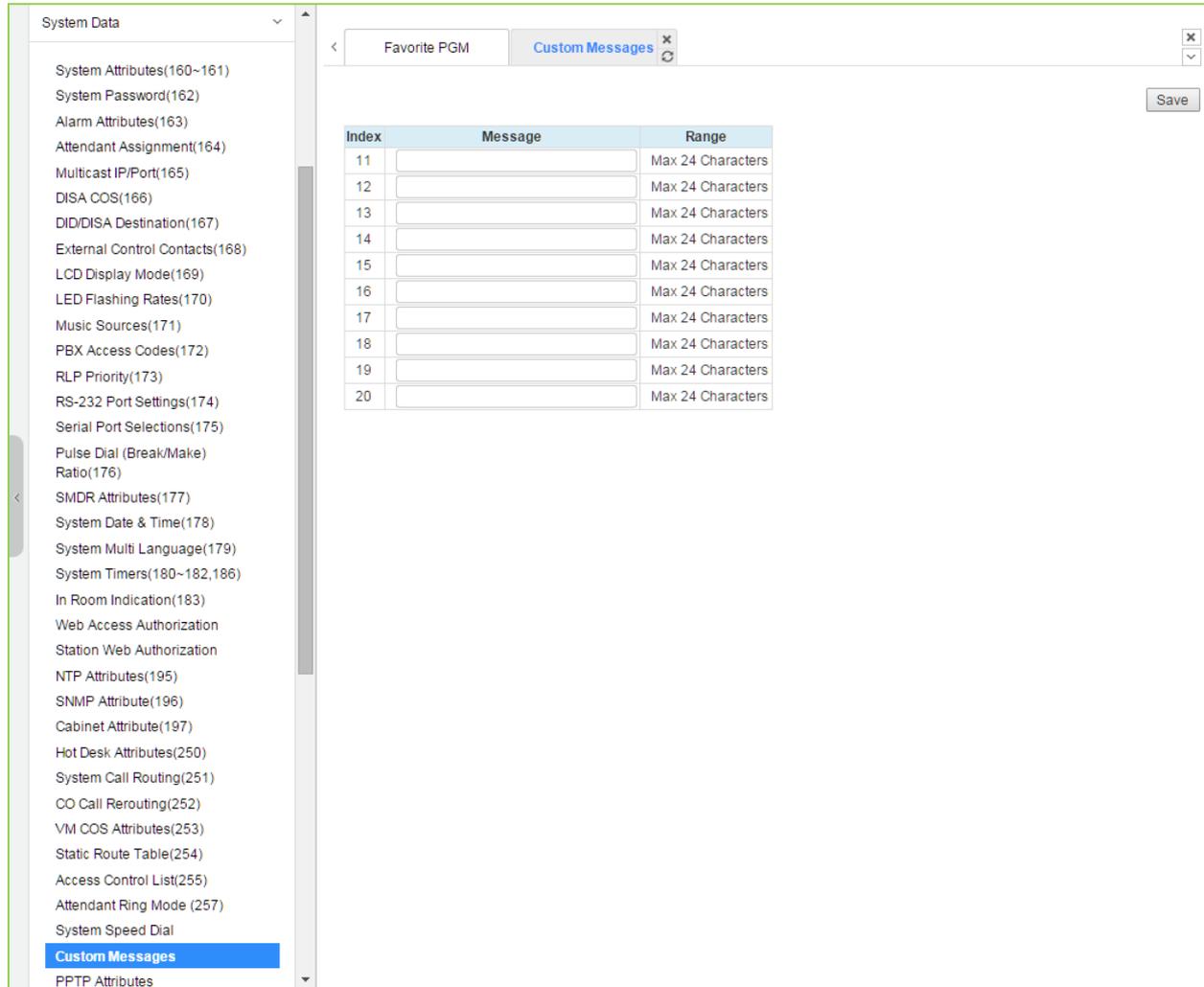
Figure 6.5.35-1 Custom Message

The system's 10 Custom messages can be defined with up to 24 characters each.

6.5.36 PPTP Attributes

Selecting PPTP Attributes will display the PPTP Attributes Table data entry page. Click **[Save]** button after changing Value.

Figure 6.5.36-1 PPTP Attributes

When required, the system supports Point to Point Tunneling Protocol (PPTP). PPTP must be enabled, values for the PPTP server IP address, Id and password must be entered, and a server name may be entered.

6.6 Station Group Data

Selecting the Station Group Data program group returns the sub-menu displayed in the left frame as shown in the following figure.
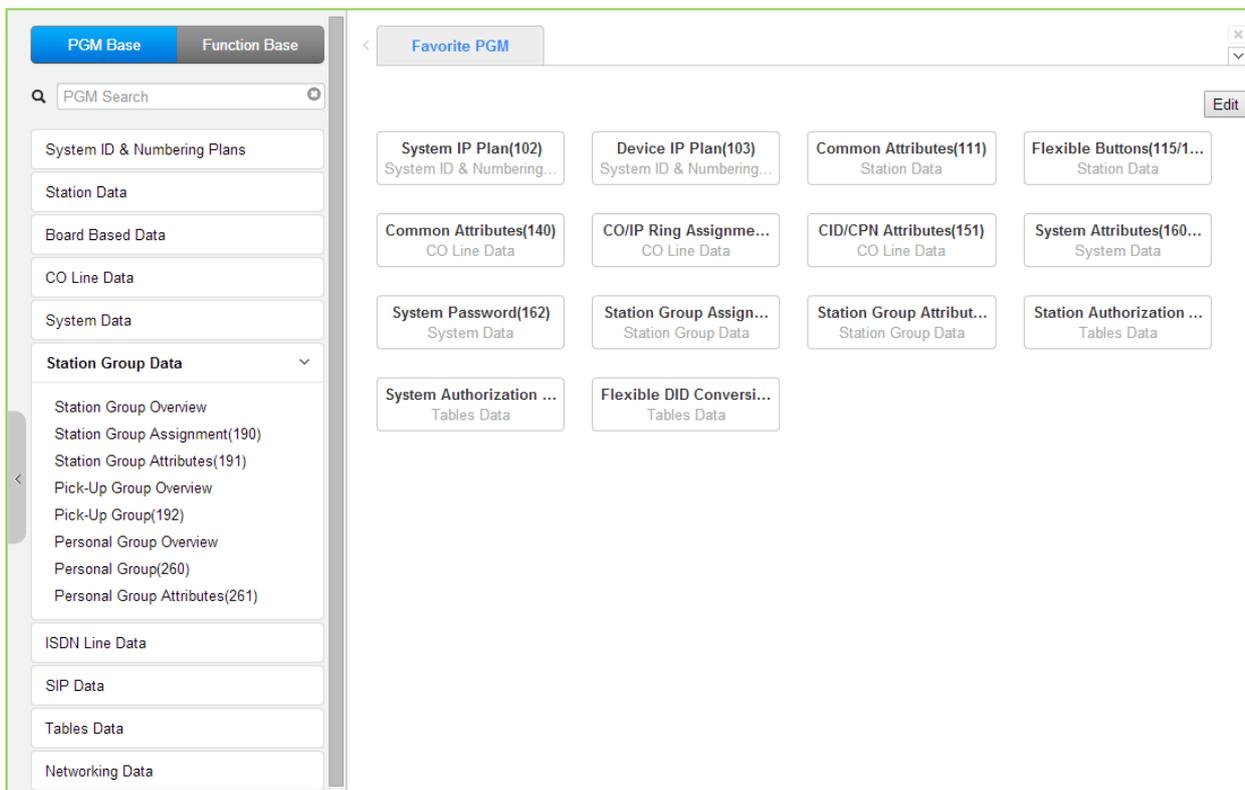
Figure 6.6-1 Station Group Data

Stations can be grouped so that incoming calls will search (Station) for an idle station in the group. The system allows assignment of three Station processes, Circular, Terminal and ACD. In addition, there are eight (8) functional groups available: ACD (Automatic Call Distribution) based on ACD, Ring, Call Pick-Up, external Voice Mail (SLT or SIP), VSF-Voice Mail, iPECS Feature Server (UMS-VM), Network Voice Mail, and UCS Groups.

Certain types of groups can incorporate announcements, which are given to the calling party. The system's VSF can store up to two hundreds (200) announcements for use with Station Groups.

Note that a station can belong to multiple groups if the groups are all of the same type. Also note that when a station group is assigned to a group type (Circular, Terminal, ACD, VM, FS VM, VSF-VM, Net VM, UCS and Ring), the group attributes are initialized to the default values.

6.6.1 Station Group Overview

Selecting the Station Group Overview item will return the Station Group Overview page. This page displays the Station Group attributes (type, pick-up attribute and member stations) for all the Station Groups. Note that data cannot be entered on this page.

Figure 6.6.1-1 Station Group Overview

6.6.2 Station Group Assignment - PGM 190

Selecting Station Group Assignment will display the Station Group data entry page. Enter the desired Station Group number and click **[Load]** to display the Group Assignment.
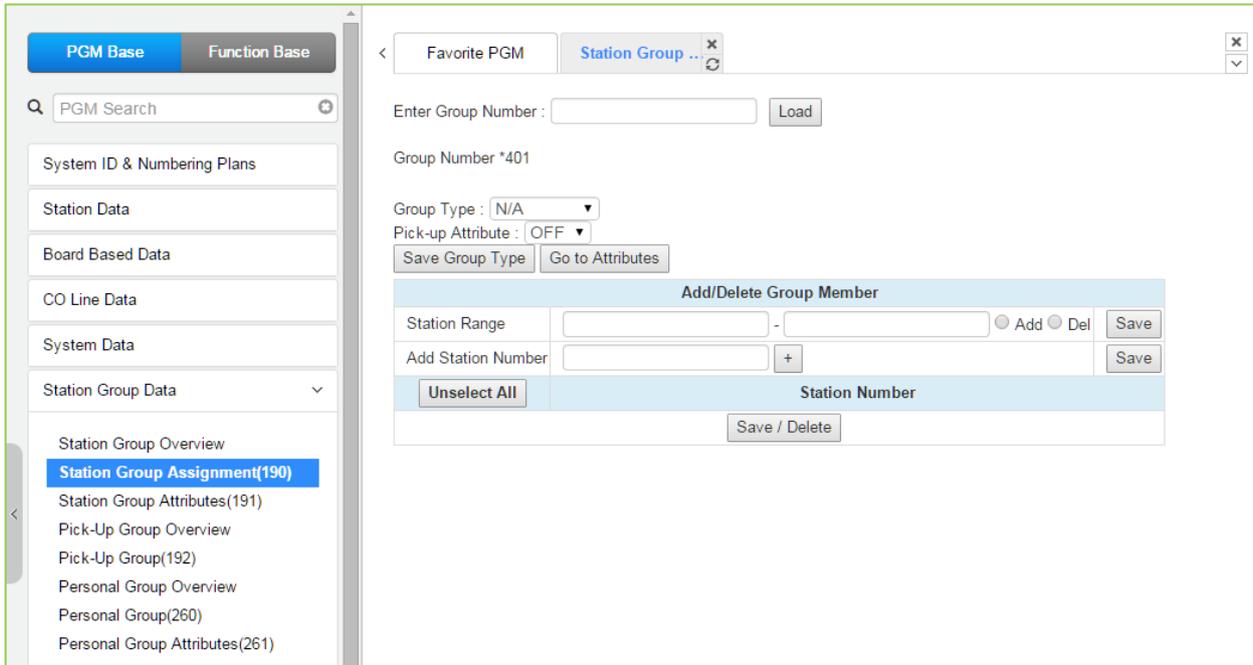
Figure 6.6.2-1 Station Group Assignments

Under Station Group Assignments the type, members and Pick-Up attributes are assigned to the Station Group. Note for the Net VM group, the network number must be assigned as the Net VM group member station.

Table 6.6.2-1 STATION GROUP ASSIGNMENT

ATTRIBUTE	– DESCRIPTION	RANGE	DEFAULT
Group Type	Defines the type of Station group	N/A, Circular, Terminal, ACD, Ring, Voice Mail, Pick-Up, VSF-VM, UMS VM, NET VM, UCS	N/A
Pick-up Attribute	Stations can pick-up group calls ringing at other stations in the group. This does not apply to the VM groups.	OFF ON	OFF
Member	Assigns stations as members of a station group or, for the Net VM group type, defines the Net Number of the group.		-

6.6.3 Station Group Attributes - PGM 191

Selecting Station Group Attributes will display the Station Group Attributes data entry page. Enter the Station Group number and click **[Load]**, the Web page for the selected group will be displayed as in Figure 6.6.3-1 to Figure 6.6.3-8 based on the Group type.

Each type of group has a different set of available attributes relating to announcements, timers, overflow, etc. Table 6.6.3-1 through Table 6.6.3-8 provides descriptions for the attributes and data entries required. Note that the attributes for the Circular and Terminal Station groups are given in Table 6.6.3-1 and the ACD attributes include the ACD functions Table 6.6.3-2.
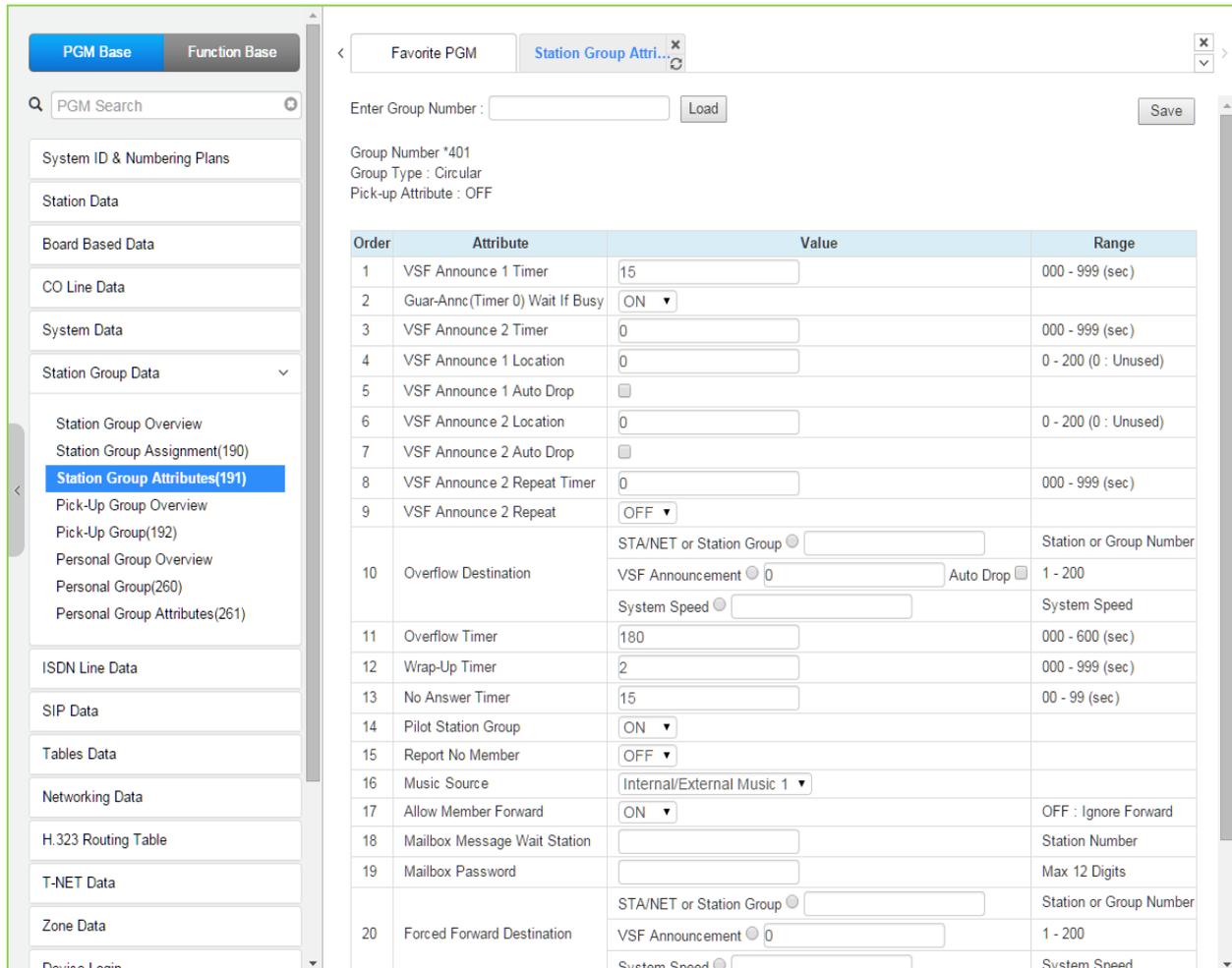
Figure 6.6.3-1 Terminal & Circular Group Attributes

Table 6.6.3-1 TERMINAL & CIRCULAR GROUP ATTRIBUTES

ATTRIBUTE	– DESCRIPTION	RANGE	DEFAULT
VSF Announce 1 Timer	If all stations in the group are busy when a call is offered, 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 Station process (guaranteed announcement).	000~999 (seconds)	15

Table 6.6.3-1 TERMINAL & CIRCULAR GROUP ATTRIBUTES

ATTRIBUTE	– DESCRIPTION	RANGE	DEFAULT
Guard-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 Ring back 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 caller receives the assigned 2nd VSF announcement.	000~999 (seconds)	0
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. An entry of 00 indicates no announcement.	00~200	0: 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 Station & Station 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. An entry of 00 indicates no announcement.	00~200	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)	0
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.	OFF ON	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 route to the assigned overflow destination. If VSF Announcement is selected, Auto Drop can be checked.	STA/NET or Station Group/ VSF Announcement/ Auto Drop/ System SPD	
Overflow Timer	A call to the group remains 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
Wrap-Up Timer	After terminating a group call, a Group member will be maintained in a busy state for the duration of the Wrap-Up timer.	000~999 (seconds)	2
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 Station process.	00~99 (seconds)	15
Pilot Station	A circular/terminal Station group can be set so that only calls to the pilot number (Station Group number) will hunt.	OFF ON	ON

Table 6.6.3-1 TERMINAL & CIRCULAR GROUP ATTRIBUTES

ATTRIBUTE	– DESCRIPTION	RANGE	DEFAULT
Report No Member	If a call is received and no members are on-duty, an ICM call will return re-order tone, while a CO/IP call will route to the overflow destination.	OFF ON	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, Internal/External Music 1, External Music 2, VSF MOH, SLT MOH1~5, VSF MOH2~3	Internal/External Music 1
Allow Member Forward	A member activating Call forward may be placed in an unavailable state for Station group calls (ON). When OFF, group calls are sent to the member as normal (OFF).	OFF ON	ON
Mailbox Message 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	
Mailbox Password	The password associated with a group Mailbox is defined here. The password is used in conjunction with the group Mailbox as with a normal station.	Max. 12 digits	
Forced Forward Destination	Calls to a Station group may forward directly to a defined destination, bypassing the Station process. "Forced Forward", below, must be enabled.	STA./NET or Station group/ VSF Announcement/ Sys. Speed	
Forced Forward Destination Usage	When Forced Forward is enabled for the group, calls to the group forward directly to a defined destination, see above "Forced Forward Destination".	OFF ON	OFF
Group Name	A name can be designated for the group.	Max. 12 characters	
Maximum Queued Call Counter	When the number of calls queued to the group match this parameter, new calls receive error tone and disconnect after the VSF Announcement 1, if assigned, is played.	00-99	99

Figure 6.6.3-2 ACD Group Attributes

Table 6.6.3-2 ACD GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
VSF Announce 1 Timer	If all stations in the group are busy when a call is offered, 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 Station process (guaranteed announcement).	000~999 (seconds)	15
Guard-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 Ring back until a channel is available (ON) or bypass the announcement. (OFF)	OFF ON	ON
VSF Announce 2 Timer	After the 1st announcement, a 2nd timer is activated. At expiration, if the call remains queued to the group, the caller receives the assigned 2nd VSF announcement.	000~999 (seconds)	0
VSF Announce 1 Location	Each Station & Station Group can be assigned an announcement, which is played if the call remains queued beyond the VSF Announce 1 Timer duration.	000~200	0: none

Table 6.6.3-2 ACD GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	The announcement location is a VSF announcement number. An entry of 00 indicates no announcement.		
VSF Announce 1 Auto Drop	If this attribute is selected, the call will drop after the 1st VSF announcement		
VSF Announce 2 Location	The Station & Station 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.	000~200	0: none
VSF Announce 2 Auto Drop	If this attribute is selected, the call will drop after the 2nd VSF announcement		
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 repeating must be "ON" under VSF Announce 2 Repeat below.	000~999 (seconds)	0
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.	OFF ON	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 queue to the group or route to the assigned Overflow Destination. If VSF Announce is assigned, Auto Drop is available.	STA/NET or Station Group/ VSF Announcement/ Auto Drop/ System SPD	
Overflow Timer	A call to a group remains queued to the group or is sent to the assigned Overflow Destination after expiration of the Overflow Timer	000~600 (seconds)	180
Wrap-Up Timer	After terminating a group call, a Station Group member will be maintained in a busy state for the duration of the Wrap-Up timer.	000~999 (seconds)	2
ACD No Answer Timer	Calls to an agent in the group are directed to the station, if unanswered in the NO ANSWER TIMER, the call can be routed to another agent.	00-99 (seconds)	0
Report No Member	If a call is received and no members are on-duty, an ICM call will return re-order tone, while a CO/IP call routes to the overflow destination.	OFF ON	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 while in queue.	Ring back tone, Internal/External Music 1, External Music 2, VSF MOH, SLT MOH1~5, VSF MOH2~3	Internal/External Music 1
ACD Warning Tone	An ACD supervisor can monitor agent conversations. A warning tone can be provided to the agent and connected party when the supervisor activates the monitor feature.	OFF ON	OFF
Alternate Destination	When a call comes into the group and there are no group members available, the call will be routed to the	STA, NET, Station, System Speed	

Table 6.6.3-2 ACD GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	assigned Alternate Destination.		
Supervisor Timer	When calls have been in queue longer than the Supervisor Timer, the ACD supervisor is notified by a display of the longest queue time.	000~999 (seconds)	30
Supervisor Call Count	When the number of calls in queue exceeds the Supervisor Call Count, the ACD Supervisor is notified by a display of queued calls.	00~99	0
Maximum Queued Call Counter	When the number of calls queued to the group match this parameter, new calls receive error tone and disconnect after the VSF Announcement 1, if assigned, is played.	00-99	99
Supervisor 1 to 5	Any valid iPECS IP or LDP Phone with display can be assigned as a Supervisor, max. 5 ACD Supervisors.	Station Number	
ACD DND	This parameter sets the duration a station will receive ring before the system places the station in ACD DND and unavailable for group calls. A setting of '00' disables automatic ACD DND.	002~200 (Sec.)	10
Entered Caller ID ICLID Usage	Within 5 seconds of a guaranteed announcement, the caller may dial digits as an ICLID. The user-dialed digits are compared to the ICLID Table entries, for routing or, for a single dialed digit, to the ACD CCR table below.	OFF ON	OFF
Allow Member Forward	A member activating Call Forward may be placed in an unavailable state for Station group calls (ON). When OFF, group calls are sent to the member as normal.	OFF ON	ON
Group Name	An ACD group name can be designated.	Max. 12 characters	
CIQ Route 1			Not selected
CIQ Route 2	When an ACD call is queued, the caller may be allowed to dial a digit to exit the queue and route to another destination. The alternate destination is based on the user-dialed digit and can be a station, Station group, system-speed bin, or network station. Dial the digit below for the type of destination and enter the value associated with the destination. 1: Enter a station number. 2: Enter a Station group number. 3: Enter a system speed bin. 4: Enter a network station number		Not selected
CIQ Route 3			Not selected
CIQ Route 4			Not selected
CIQ Route 5			Not selected
CIQ Route 6			Not selected
CIQ Route 7			Not selected
CIQ Route 8			Not selected
CIQ Route 9			Not selected
CIQ Route 0			Not selected
ZAP Tone	Agents using a headset can have ACD calls connected to them automatically preceded by a tone (Zap tone).	OFF ON	OFF
CIQ Announcement	If enabled, queued callers receive the CIQ message (You are # in queue) after the 1st and 2nd announcement.	OFF ON	OFF
Mailbox Message Wait Station	When an ACD call overflows or routes to the VM group, a station number is used to identify the Mailbox for the ACD group messages.	Station Number	
Mailbox Password	The password associated with an ACD group Mailbox is	Max. 12 digits	

Table 6.6.3-2 ACD GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	defined here. The password is used in conjunction with the ACD group Mailbox as with a normal station.		
CIQ Display To Agent - Mode	When an ACD call is in queue, the Call in queue information is displayed in the LCD of agent and supervisor phones.	OFF ON	OFF
CIQ #1 Page Alert - Threshold	If the queued call count exceeds the threshold, the system plays the CIQ #1 Announcement to the CIQ #1 Page Zone after the CIQ #1 Announcement Delay Timer. The announcement is repeated at intervals of the CIQ #1 Announcement Repeat Timer.	00-99	10
CIQ #1 Page Alert – Announcement Location	VSF announcement number for the CIQ #1 Announcement.	000~200	0
CIQ #1 Page Alert – Page Zone	Page Zone to receive CIQ #1 Announcement	0~105	0
CIQ #1 Page Alert - Delay Time	Delay timer for CIQ #1 Announcement	000-180	15
CIQ #1 Page Alert - Repeat Time	Interval for repeating the CIQ #1 Announcement	000-180	45
CIQ #2 Page Alert - Threshold	If the queued call count exceeds the threshold, the system plays the CIQ #2 Announcement to the CIQ #2 Page Zone after the CIQ #2 Announcement Delay Timer. The announcement is repeated at intervals of the CIQ #2 Announcement Repeat Timer.	00-99	20
CIQ #2 Page Alert - Announcement Location	VSF announcement number for the CIQ #2 Announcement.	000~200	0
CIQ #2 Page Alert – Page Zone	Page Zone to receive CIQ #2 Announcement	0~105	0
CIQ #2 Page Alert - Delay Time	Delay timer for CIQ #2 Announcement	000-180	15
CIQ #2 Page Alert - Repeat Time	Interval for repeating the CIQ #2 Announcement	000-180	25
CIQ #3 Page Alert - Threshold	If the queued call count exceeds the threshold, the system plays the CIQ #3 Announcement to the CIQ #3 Page Zone after the CIQ #3 Announcement Delay Timer. The announcement is repeated at intervals of the CIQ #3 Announcement Repeat Timer.	00-99	30
CIQ #3 Page Alert - Announcement Location	VSF announcement number for the CIQ #3 Announcement.	000~200	0
CIQ #3 Page Alert - Page Zone	Page Zone to receive the CIQ #3 Announcement	00~15 or 00-40	0
CIQ #3 Page Alert - Delay Time	Delay timer for the CIQ #3 Announcement	000-180	15
CIQ #3 Page Alert - Repeat Time	Interval for repeating the CIQ #3 Announcement	000-180	5

Table 6.6.3-2 ACD GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Forced Forward Destination	Calls to a Station group may forward directly to a defined destination, bypassing the Station process. "Forced Forward", below, must be enabled.	STA./NET or Station group, VSF Announcement, Sys. Speed	
Forced Forward Dest Usage	When Forced Forward is enabled for the group, calls to a Station group forward directly to a defined destination, see above "Forced Forward Destination".	OFF ON	OFF
Auto Ring Mode	Normally, an ACD call will be routed to the longest idle group member. If desired, the call may be routed using the timing in the assigned Auto Ring Mode Table and the Day, Night, and Timed mode destinations entered below. If Manual Change is selected, the Group supervisor may select the active Ring mode manually.	Manual Change Table 0 ~ 100	Manual Change
Day Destination	When the ACD Ring Mode Table schedule is in the Day mode, ACD group calls route to the destination defined here.	Normal Service, STA/NET or Station group, VSF Announcement, Sys. Speed	Normal Service
Night Destination	When the ACD Ring Mode Table schedule is in the Night mode, ACD group calls route to the destination defined here.	Normal Service, STA/NET or Station group, VSF Announcement, Sys. Speed	Normal Service
Timed Destination	When the ACD Ring Mode Table schedule is in the Timed mode, ACD group calls route to the destination defined here.	Normal Service, STA/NET or Station group, VSF Announcement, Sys. Speed	Normal Service
ACD Group Queuing Call Indication	If there are queued group calls, the queuing indication can be served to group members by Mute Ring and LED button flashing.	OFF ON (RING and LED) ON (LED only)	OFF
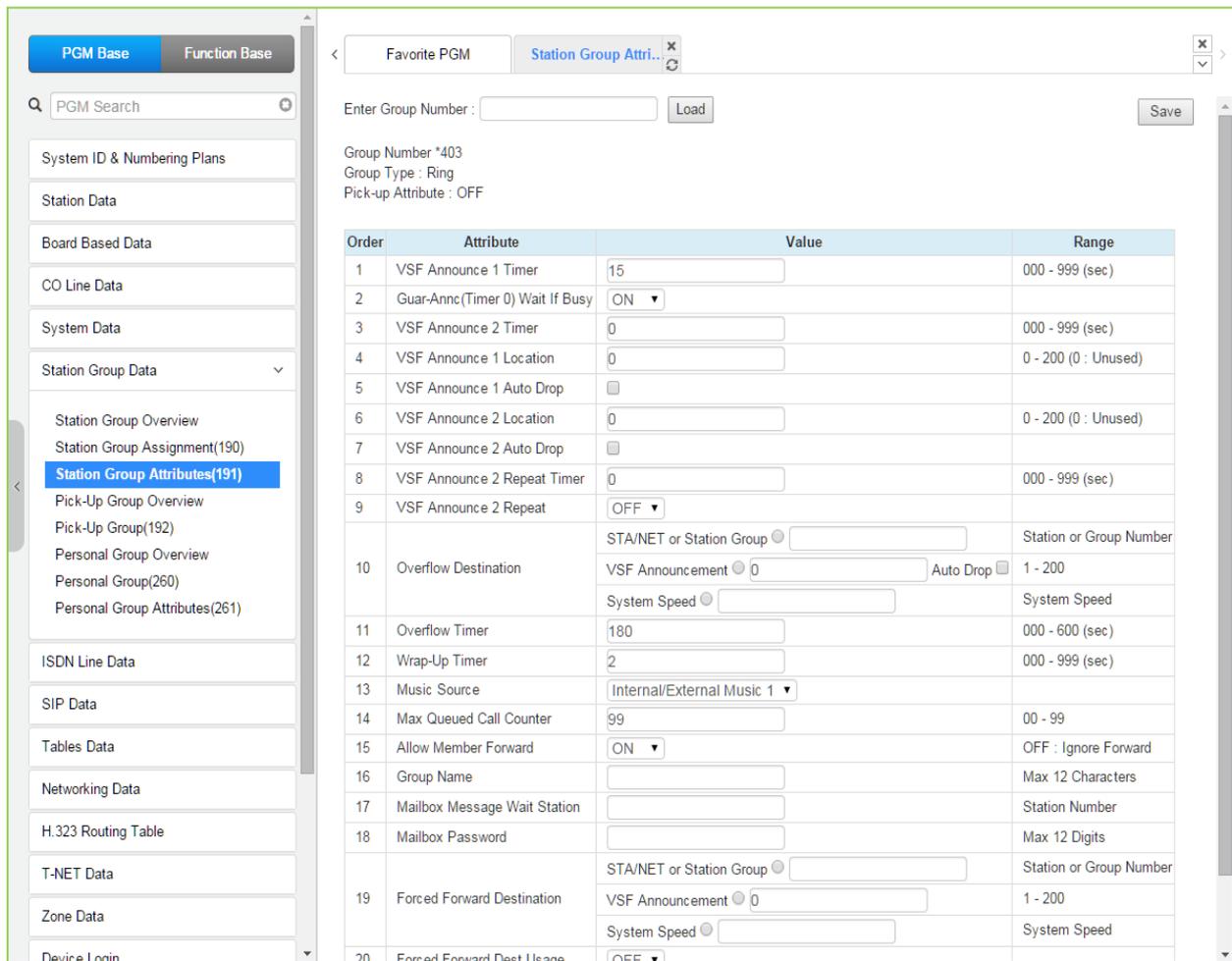
Figure 6.6.3-3 Ring Group Attributes

Table 6.6.3-3 RING GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
VSF Announce 1 Timer	If all stations in the group are busy when a call is offered, 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 Station process (guaranteed announcement).	000~999 (seconds)	15
Guar-Ann(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 Ring back 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 caller receives the assigned 2nd VSF announcement.	000~999 (seconds)	0
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	000~200	0: none

Table 6.6.3-3 RING GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	a VSF Announcement number. An entry of 00 indicates no announcement.		
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.	000~200	0: none
VSF announce 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)	0
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.	OFF ON	OFF
Overflow Destination	A call to the group will continue to ring group member stations until answered. The call will remain at the last station or routes to the assigned Overflow Destination. If VSF Announce is assigned, Auto Drop is available.	STA/NET or Station Group, VSF Announcement, Auto Drop, System SPD	
Overflow Timer	A call to a group remains in the group or routes to the assigned Overflow Destination after expiration of the Overflow Timer.	000~600 (seconds)	180
Wrap-Up Timer	After terminating group call, a Station Group member will be maintained in a busy state for the duration of the Wrap-Up Timer.	000~999 (seconds)	2
Music Source	A Music source can be assigned so that calls to the group receive audio from the assigned source in place of ring-back tone.	Ring back tone, Internal/External Music 1, External Music 2, VSF MOH, SLT MOH1~5, VSF MOH2~3	Internal/External Music 1
Maximum Queued Call Counter	When the number of calls queued to the group match this parameter, new calls receive error tone and disconnect after the VSF AA announcement, if assigned, is played.	00-99	99
Allow Forward Member	A member activating Call Forward may be placed in an unavailable state for Station group calls (ON). When OFF, group calls are sent to the member as normal.	OFF ON	ON
Group name	A group name can be designated.	Max.12 characters	
Mailbox Message 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	
Mailbox Password	The password associated with the group Mailbox is defined here. The password is used in conjunction with the	Max. 12 digits	

Table 6.6.3-3 RING GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	group Mailbox as with a normal station.		
Forced Forward Destination	Calls to a Station group may forward directly to a defined destination, bypassing the Station process. "Forced Forward", below, must be enabled.	STA/NET or Station group, VSF Announcement, Sys. Speed	
Forced Forward Usage	When Forced Forward is enabled for the group, calls to a Station group forward directly to a defined destination, see above "Forced Forward Destination".	OFF ON	OFF
Ring group indication	When a station calls a Ring Group, DSS/BLF buttons assigned for the calling station will flash and muted ring is received.	OFF, Mute, Burst	Mute

Figure 6.6.3-4 External Voice Mail Group Attributes

Table 6.6.3-4 EXTERNAL VOICE MAIL GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Wrap-Up Timer	After terminating a group call, the VM port will be maintained in a busy state for the duration of the Wrap-Up timer allowing the VM time to return the port to idle.	000~900 (seconds)	2

Table 6.6.3-4 EXTERNAL VOICE MAIL GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Put Mail Index	For external analog Voice Mail groups, an index to the Voice Mail Dial Table that contains the "Put Mail" dial code.	1~4	1
Get Mail Index	For external analog Voice Mail groups, an index to the Voice Mail Dial Table that contains the "Get Mail" dial code.	1~4	2
Station Type	The type of Station process applied to the SLT ports connected to the VM can be assigned as Circular or Terminal.	Terminal Circular	Terminal
Overflow Timer	A call to a group remains at the last station in the group or is sent to the assigned Overflow Destination after expiration of the Overflow Time.	000~600 (seconds)	180
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 route to the assigned Overflow Destination. If assigned VSF Announce, Auto Drop is available.	STA/NET or Station Group, VSF Announcement, Auto Drop, System Speed	-
Forced Forward Destination	Calls to a Station group may forward directly to a defined destination, bypassing the Station process. "Forced Forward", below, must be enabled.	STA/NET or Station group, VSF Announcement, Sys. Speed	
Forced Forward Usage	When Forced Forward is enabled for the group, calls to a Station group forward directly to a defined destination, see above "Forced Forward Destination".	OFF ON	OFF
Group Name	A group name can be designated.	Max. 12 character	
Server type	When a third party SIP server is used for AA/VM, or the IPCR or third party SIP recording server is used, the server type must be selected.	IPCR 3rd party Type	3rd party Type
Server number	Each IPCR and third party SIP server must be assigned a server number from 01 ~ 10 which correlates this group with an Agent table.	0-10	0
Member Type	The External AA/VM employs analog SLT protocols or SIP protocols. The protocol (SLT/SIP) must be configured here, and, for SIP, the number of agents defined (1-2400). A server and member licenses may be required.	0-2400 (SLT or SIP Type)	SLT Type
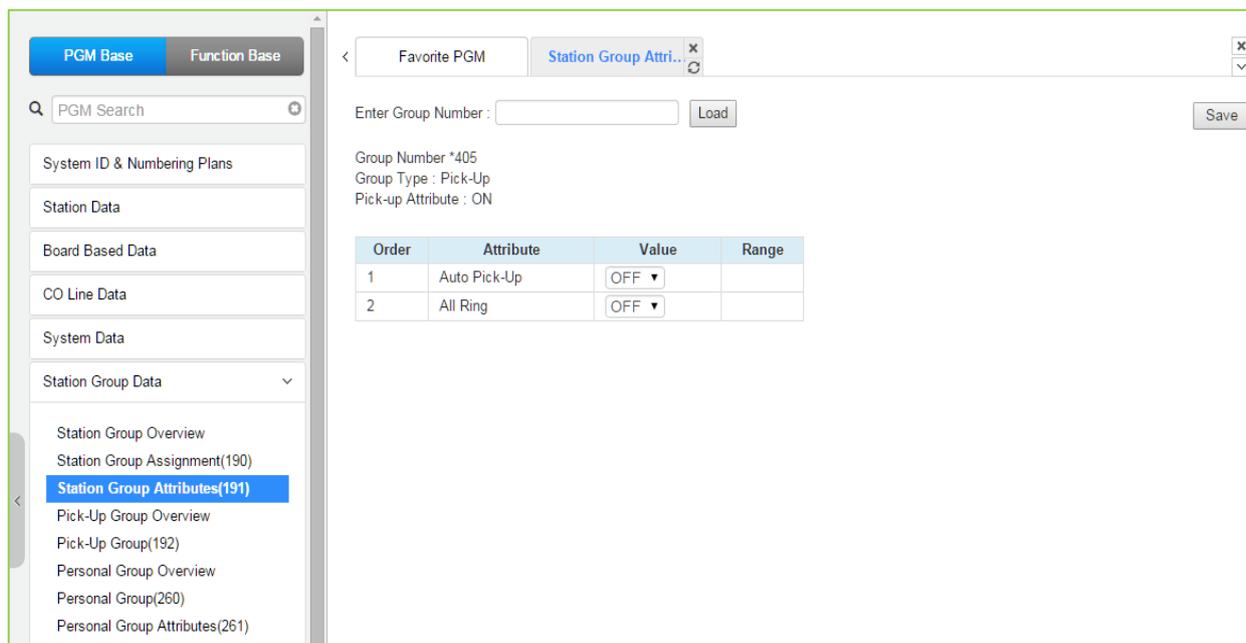
Figure 6.6.3-5 Pick-Up Group Attributes

Table 6.6.3-5 PICK-UP GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Auto Pick Up	If a group member is ringing, other members of the Group can Pick-Up the ringing call by simply going "Off-hook".	OFF ON	OFF
All Ring	When a call is offered to a member of the Pick-Up Group in the Tone Ring mode, all members will ring. Note Auto Pickup above must be "ON".	OFF ON	OFF
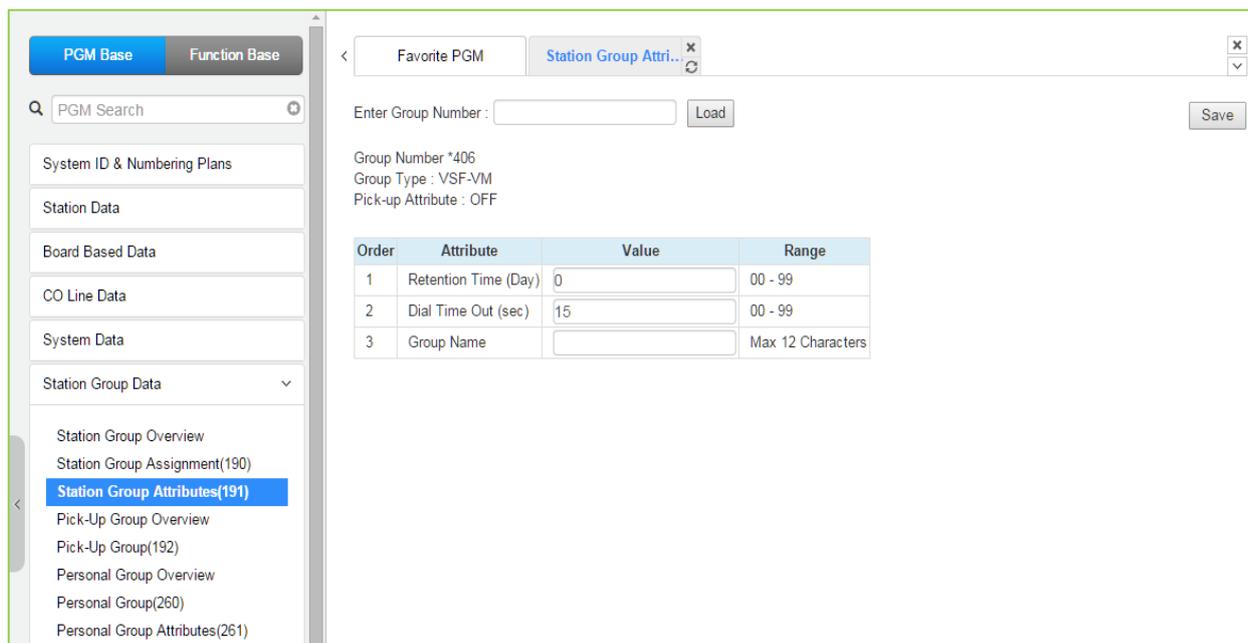
Figure 6.6.3-6 VSF Group Attributes

Table 6.6.3-6 VSF GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Retention Time (day)	When voice messages are stored in the VSF or UVM, the system will maintain (store) the message for the maximum number of days set in this program (0 to 99 days).	00-99 (day)	0
Dial Time Out (sec)	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.	00-99 (seconds)	15
Group Name	A group name can be designated.	Max. 12 characters	

Figure 6.6.3-7 iPECS Feature Server Voice Mail Group (UMS-VM) Attributes

Table 6.6.3-7 FEATURE SERVER VOICE MAIL GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
VSF Announce 1 Timer	If all stations in the group are busy when a call is offered, the call may continue to wait (queue) for an available FS-VM channel. 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 Station process (guaranteed announcement).	000~999 (seconds)	15
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 caller receives the assigned VSF Announce 2 Location.	000~999 (seconds)	0
VSF Announce 1 Location	Each 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.	000~200	0: none
VSF Announce 1 Auto Drop	If this attribute is selected, the call will drop after the 1st VSF announcement.	Check box	

Table 6.6.3-7 FEATURE SERVER VOICE MAIL GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
VSF Announce 2 Location	The 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.	000~200	0: 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 announcement 2 repeat timer. Note VSF Announce 2 Repeat below must be "ON".	000~999 (seconds)	0
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 Repeat timer interval, above.	OFF ON	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 route to the assigned Overflow Destination. If assigned VSF Announce, Auto Drop is available.	STA/NET or Station group, VSF Announcement, Auto drop, System Speed	
Overflow Timer	A call to a group remains at the last station in the group or routes to the assigned Overflow Destination after expiration of the Overflow Timer.	000~600 (seconds)	180
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 Station process.	00~99 (seconds)	15
Pilot Station	A FS-VM Station group can be set so that only calls to the pilot number (station group number) will hunt.	OFF ON	ON
Alternate Destination	When a call comes into the group and there are no group members available, the call will route to the assigned Alternate Destination.	STA/NET or Station group, System SPD	
Station type	The Station process for the FS-VM group can be defined as Circular or Terminal.	Circular Terminal	Circular
Wrap-Up Timer	After terminating any call, the FS port will be maintained in a busy state for the duration of the Wrap-Up Timer.	000~999 (seconds)	2
Forced Forward Destination	Calls to a group may forward directly to a defined destination, bypassing the Station process. "Forced Forward", below, must be enabled.	Sta./NET or Station group, VSF announcement, Sys. Speed	
Forced Forward	When Forced Forward is enabled for the group, calls to a Station group forward directly to a defined destination, see above "Forced Forward Destination".	OFF ON	OFF
Group Name	A group name can be designated.	Max. 12 characters	

Figure 6.6.3-8 iPECS UCS Server Group Attributes

Table 6.6.3-8 UCS GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
UC Server	UCS Server number, this value must be set to 1.	01-16	1

6.6.4 Pick Up Group Overview

Selecting Pick Up Group Overview item will return the Station Pick Up Group Overview page. This page displays the Station Group member stations for all the Station Pick Up Groups. Note that data cannot be entered on this page.
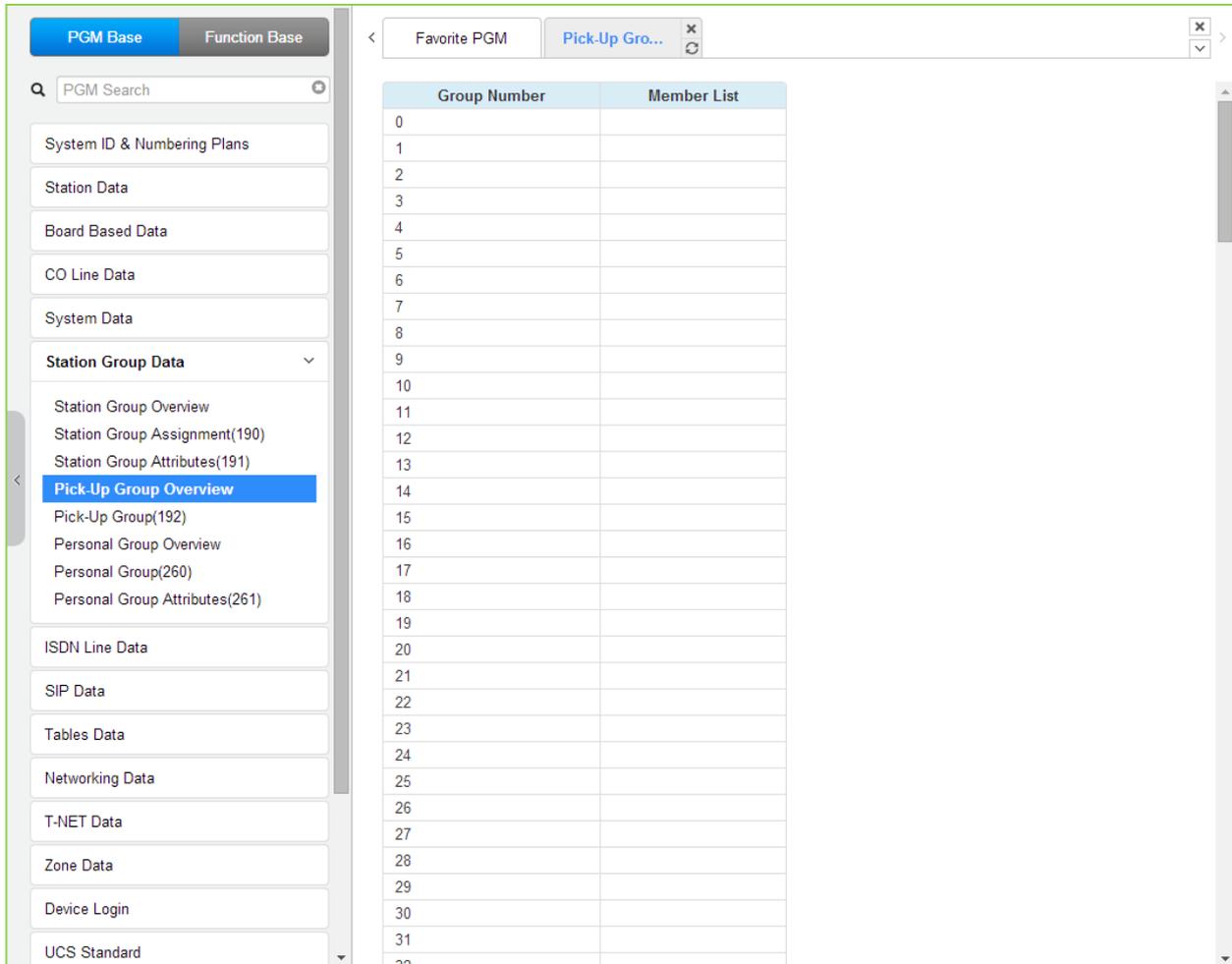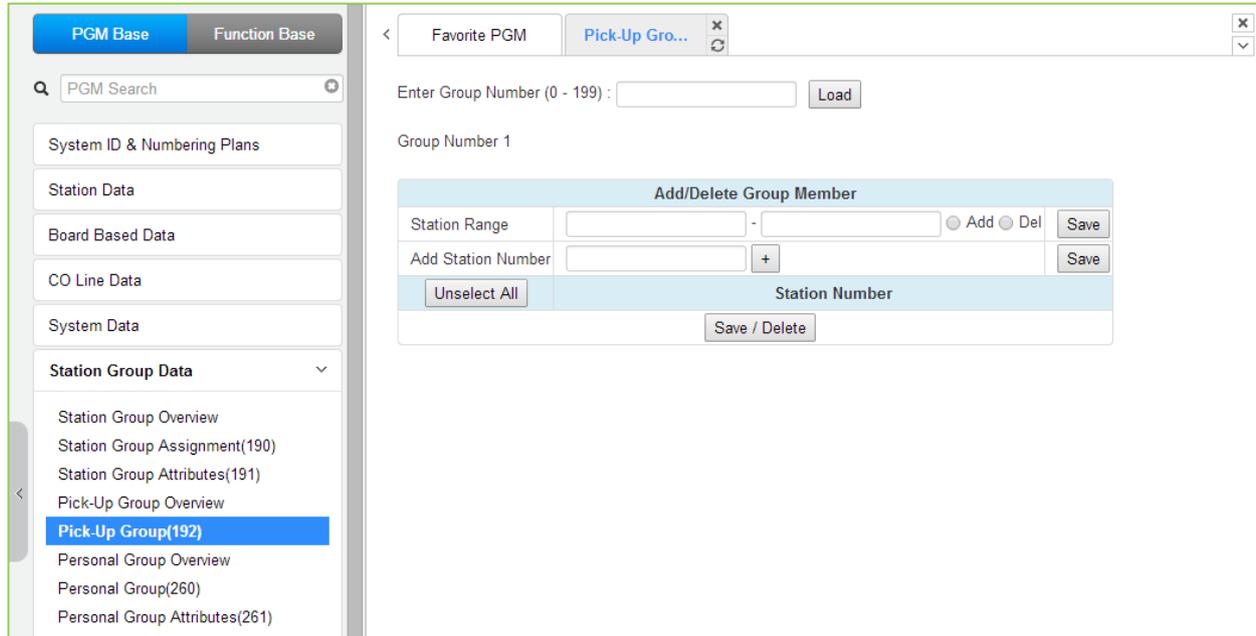

Figure 6.6.4-1 Pick Up Group Overview

6.6.5 Pick Up Group - PGM 192

Selecting Pick Up Group will display the Pick Up Group Assignment entry page. Enter the desired Pick Up Group number and click **[Load]** to display the group member Assignment.

Figure 6.6.5-1 Pick Up Group

Table 6.6.5-1 PICK UP GROUP

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Member	Assign stations as members of the Pick-Up group.		

6.6.6 Personal Group Overview

Selecting Personal Group Overview displays the master station and member list for all the personal groups.
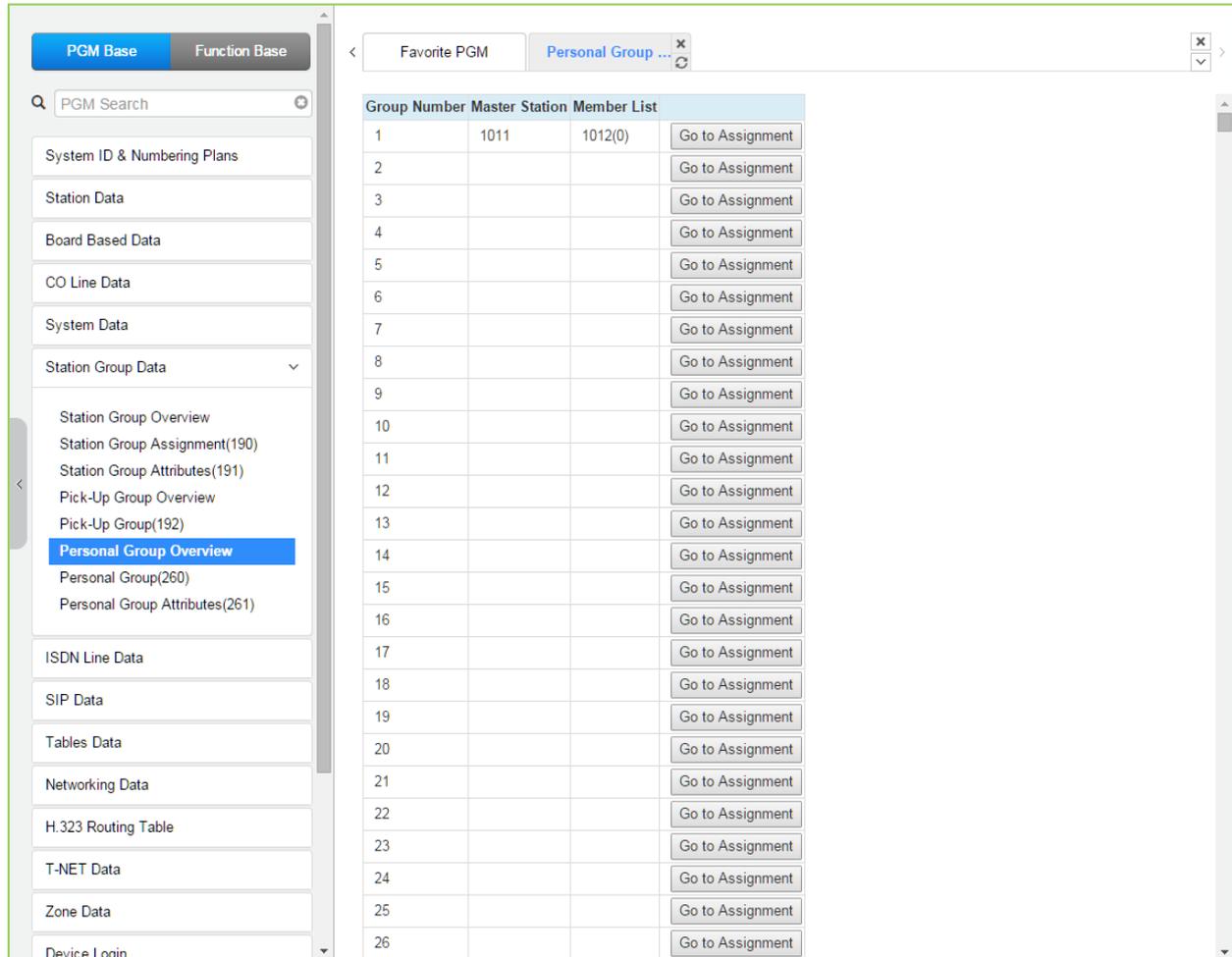
Figure 6.6.6-1 Personal Group Overview

Clicking [Go to Assignment] goes to move Personal group (260) for assigning Group master & member and setting Personal group attributes.

6.6.7 Personal Group - PGM 260

Selecting Personal Group will display the Personal Group entry page. Enter a valid Personal Group number and click **[Load]** to enter group data.
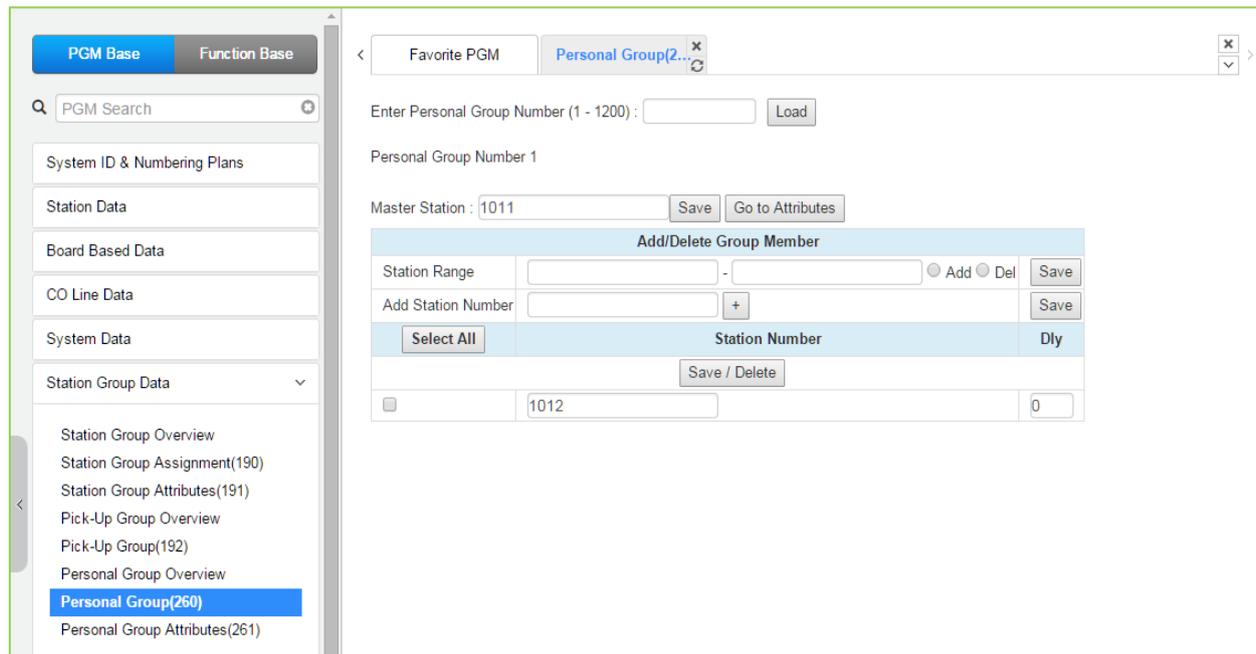
Figure 6.6.6-1 Personal Group

Several stations can share same station number. That means Personal Group is extended feature of Linked Pair.

A Personal Group is composed with a master station and several member stations.

A master station and all member stations share master station's number.

In case of Member station, each member station can be set the delay (Dly) time.

By using this shared number, almost features (Call To xxx / Call From xxx / SMDR / Message Wait…) can be activated.

But, some features can be chosen by PGM 261 attribute about all station activating or individual station working.

In PGM 260, Personal group master and member can be assigned.

In PGM 261, Personal group attribute can be set.

6.6.8 Personal Group Attribute - PGM 261

Selecting Personal Group Attribute will display the Personal Group Attribute entry page. Enter a valid Personal Group number and click **[Load]** to enter the group data.
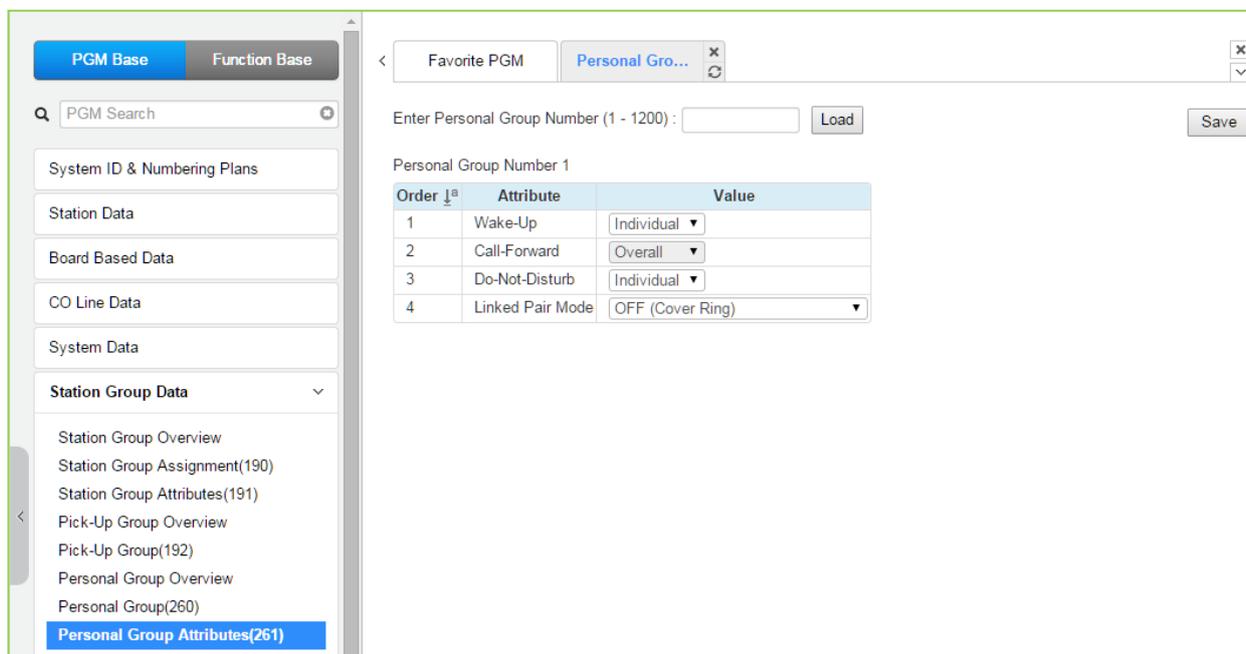
Figure 6.6.7-1 Personal Group Attributes

Table 6.6.7-1 PERSONAL GROUP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Wake-Up	Wake-up Alarms set by the Master station can also notify the linked stations using the "Overall" setting. With the "Individual" setting each station receives Wake-up alarms only if set by the station.	Individual/ Overall	Overall
Call-Forward	When the Master station activates Call Forward, calls to members also forward using the "Overall" setting. With the "Individual" setting each station controls call forward only for that station.	Individual Overall	Overall
Do-Not Disturb	The Master station can activate DND for all stations when this field is set to "Overall". DND is controlled by the each station separately when the "Individual" setting is selected.	Individual/ Overall	Overall
Linked Pair Mode	The member stations can be linked so all members reflect a single status, when one is busy they are all busy (ON), or stations reflect their own status (OFF).	OFF(Cover Ring)/ ON(Cover Ring and State Sync.)	OFF(Cover Ring)

6.7 ISDN Line & ICLID Routing Data

Selecting the ISDN Line Data program group returns the sub-menu displayed in the left frame as shown in the following figure.
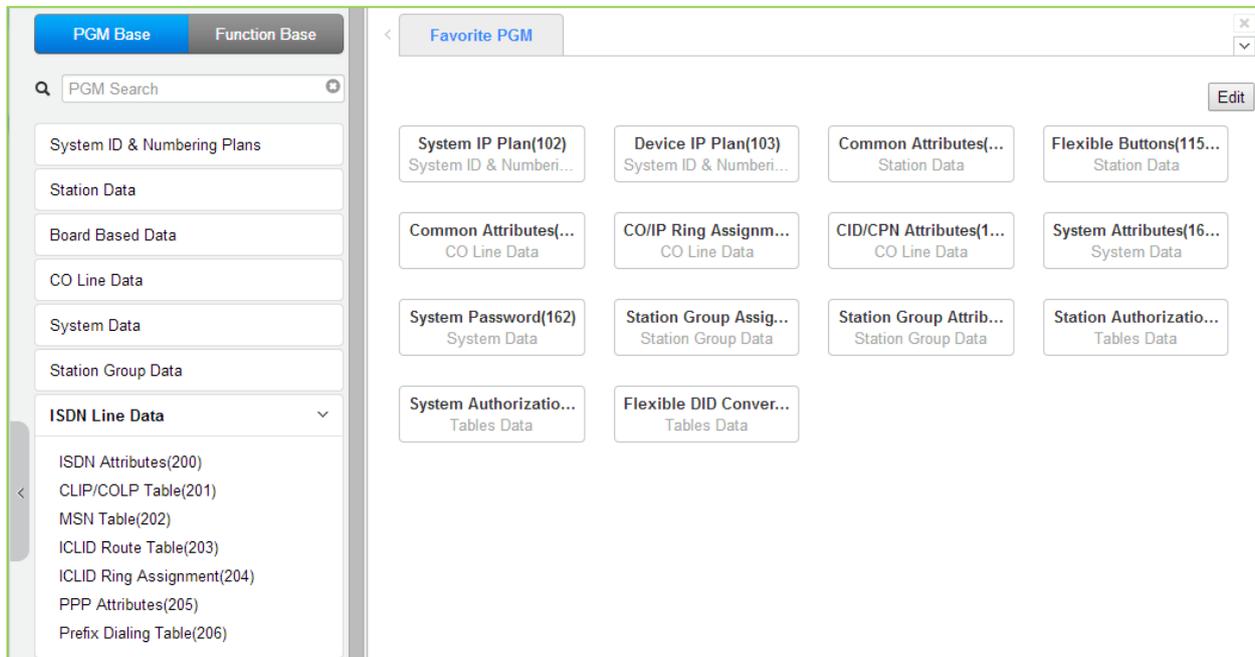
Figure 6.7-1 ISDN Line Data

Each ISDN (Integrated Services Digital Network) Line provides digital services to the end-user. Basic Rate Lines have three (3) channels, 2 B channels and a D channel. The 2 B channels provide 64 Kbps each, a total of 128 Kbps for "Bearer" or voice channels. The D channel provides a 16 Kbps signaling channel. Primary Rate Lines have 23 or 30 64 Kbps 'B' channels and 1 or 2 64 Kbps signaling channels. The 23B + D PRI complies with North American standards and the 30B + 2D complies with ETSI standards. For proper operation, entries are required for various attributes and Tables to match the ISDN circuit and services.

6.7.1 ISDN Attributes - PGM 200

Selecting ISDN Attributes will display the ISDN Attributes data entry page. Click **[Save]** button after changing Value.
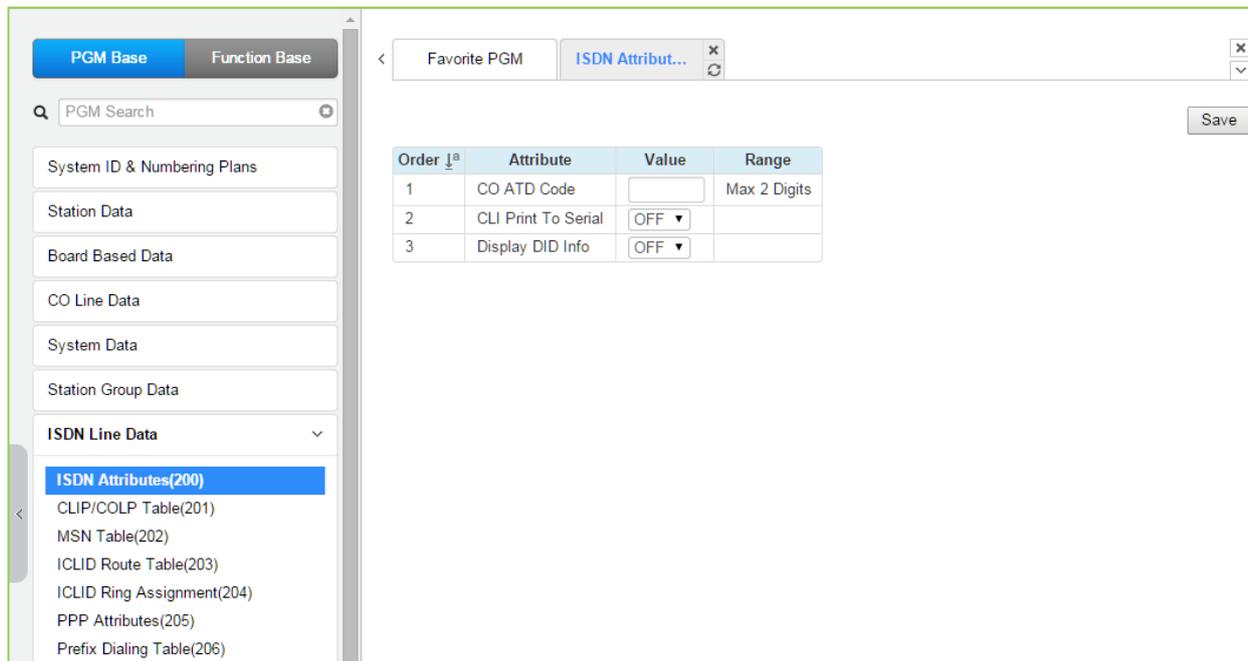
Figure 6.7.1-1 ISDN Attributes

ISDN attributes define several characteristics of the ISDN interface as shown in the following table.

Table 6.7.1-1 ISDN ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CO ATD Code	When the system is configured to send the station number with ISDN CLIP or COLP, either the station number or this ATD code is sent based on Station Data Common Attributes in section 6.2.2, EXT or ATD assignment.	Max.2 Digits	
CLI Print To Serial	The ISDN Calling Line Id may be included in call records, refer to SMDR Attributes section 6.5.17.	OFF ON	OFF
Display DID Information	The received DID digits can be shown in the iPECS IP or LDP Phone LCD and output to the defined SMDR port.	OFF ON	OFF

6.7.2 CLIP/COLP Table - PGM 201

Selecting CLIP/COLP Table will display the CLIP/COLP Table Attributes data entry page. Click
[Save] button after changing Value.
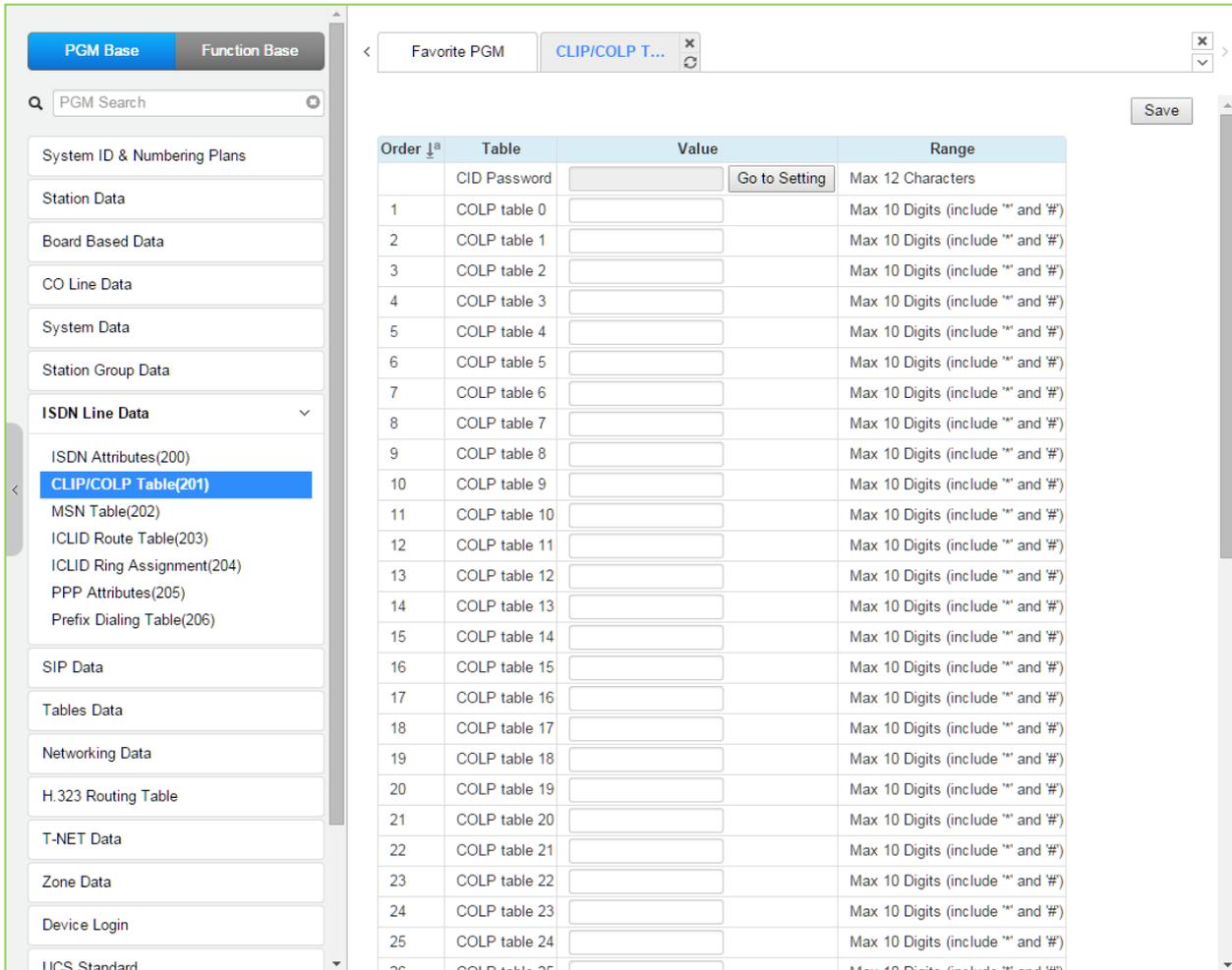
Figure 6.7.2-1 CLIP/COLP Table

Normally, the system will send the primary Directory Number of the ISDN Line in the ISDN call
SETUP and CONNECT messages to identify the caller (CLIP) or the answering (COLP) party
respectively. Under certain circumstances, it may be desirable to provide a secondary or DID
number for the ISDN Line. In these cases, the CLIP/COLP Table may be used to define the digits
sent. The number sent is selected based on the index assigned for the CO/IP Line under
'CID/CPN Attributes section'.

The Station CLI is sent as a suffix to the number in the Table. Note that this number is sent only if
CLIR/COLR is disabled under the CLIR Service and COLR Service assignments in the Station
ISDN Attributes.

Setting CID Password directly

You can set the CID password to click "Go to Setting" button. After clicking it, you will move to the
following PGM 162 and set the CID password, and then save CID password to mark tick on the
save box and click the Save button.

6.7.3 MSN Table - PGM 202

Selecting MSN Table will display the MSN Table data entry page. Enter a valid MSN Table index range and click **[Load]** to enter the MSN Table data. Click **[Save]** button after changing Value.
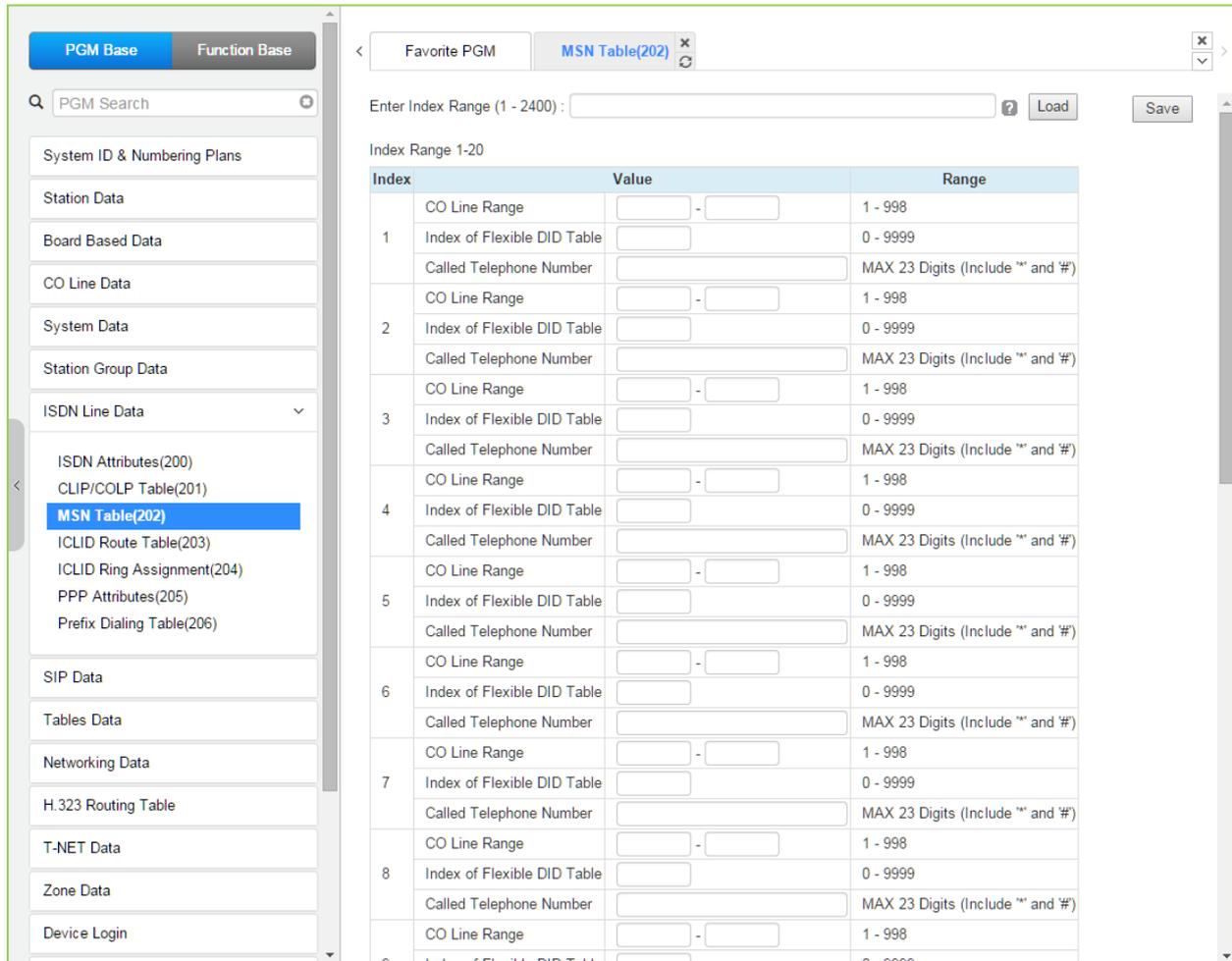
Figure 6.7.3-1 MSN Tables

When a Line assigned for DID operation receives an incoming call, the call will be routed to a station based on the Flexible DID Table Index assigned in the following table, which configures the expected Multiple Subscriber Numbers associated with the ISDN line.

Table 6.7.3-1 MSN TABLE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CO Line Range	The ISDN line associated with the MSN Table entry can be defined to reduce the number of entries searched	1-998	None
Index of Flexible DID Table	This field defines the index to the Flexible DID Table, section 6.9.12, for routing the call.	0~9999	None
Called Telephone Number	When the received MSN number matches this entry, the call is routed based on the DID Table index entered above.	Max. 23 Digits (Include * and #)	None

6.7.4 ICLID Route Table - PGM 203

Selecting ICLID Route Table will display the ICLID Route Table data entry page. Enter the Index range and click **[Load]** to open the ICLID Route Table. Click **[Save]** button after changing Value.
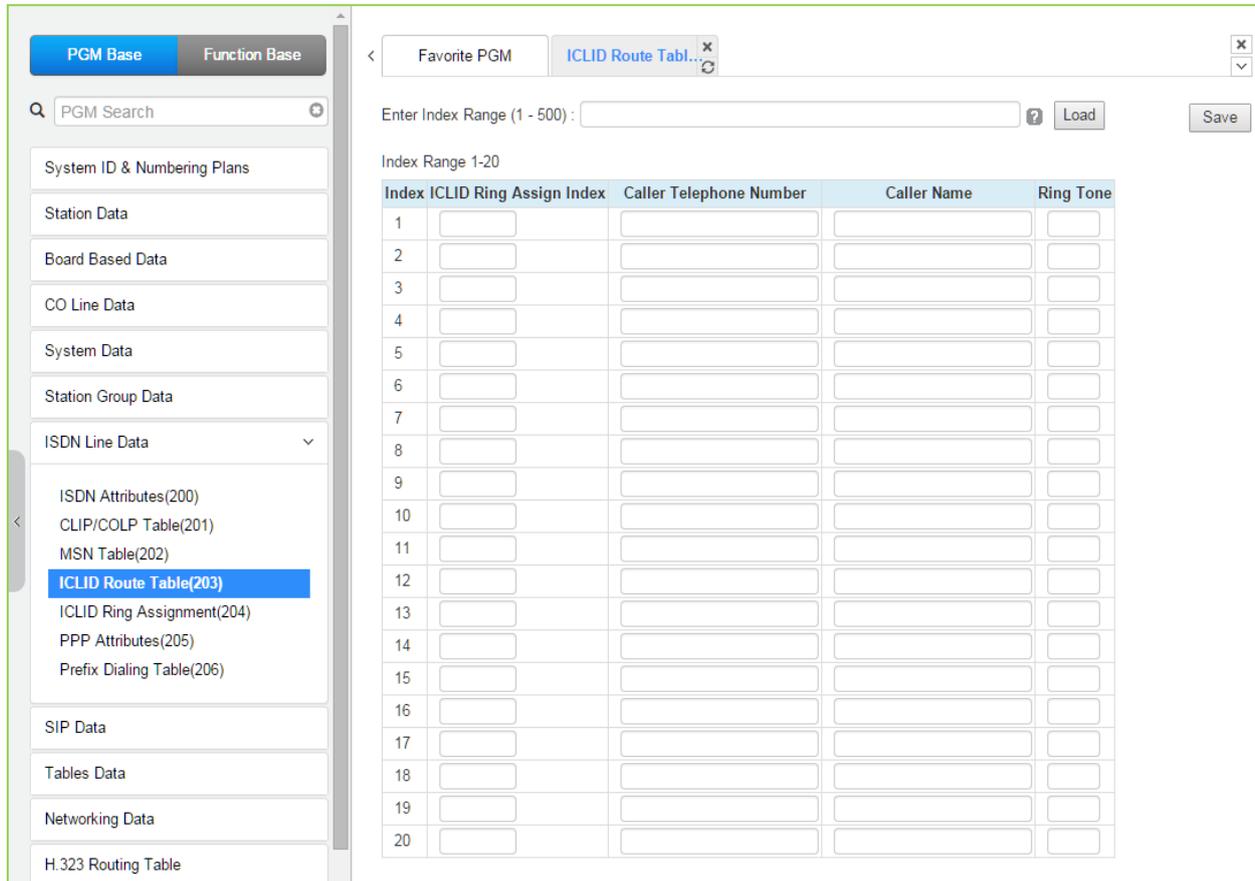
Figure 6.7.4-1 ICLID Route Table

The system can employ ICLID (Incoming Calling Line Id) to determine the routing of incoming external calls. Each CO/IP Line, including DID Lines and ACD group calls may be assigned to employ ICLID routing. The system will compare the received ICLID to entries in the ICLID Route Table and, if a match is found, will route the call to the destination defined in the ICLID Ring Assignment Table index assigned here.

Table 6.7.4-1 ICLID ROUTE TABLE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
ICLID Ring Assign Index	This field is the index to the ICLID Ring Assignment Table, PGM 204 that determines the call routing.	001~250	None
Caller Telephone Number	If the Caller Id matches this Table entry, the index above is used to select the route from PGM 204.	24 Digits	None
Caller Name	The ICLID name that is sent by the system to the destination for the ICLID routed call.	12 characters	None
Ring Tone	If the received Caller Id matches the Caller Telephone Number, the Ring tone selected here is employed for the call alerting.	01 ~ 16	Ring Tone

6.7.5 ICLID Ring Assignment Table - PGM 204

Selecting ICLID Ring Assignment Table will display the ICLID Ring Assignment Table data entry page. Enter the desired index and click **[Load]** to enter the ICLID Ring Assignment. Click **[Save]** button after changing Value.


Figure 6.7.5-1 ICLID Ring Assignment Table

If the Incoming Caller ID matches an entry in the ICLID Route Table, the index from the Table is used the ICLID Ring Assignment Table entry to employ to route the call. Separate ring assignments are made for Day, Night, and Timed Ring mode for each index, 001 to 250. When assigned to ring to a VSF announcement, the call can be automatically dropped after the announcement by entering '#' after the announcement number.

Using the AA Ring time, the call is sent to the assigned VSF announcement after the AA Ring Time expires if the call has not been answered by a station. The delay is defined in seconds from 00 to 30.

6.7.6 ISDN PPP Attributes - PGM 205

Selecting PPP Attributes will display the PPP Attributes data entry page. Use the check boxes to indicate which attributes to define; data for checked attributes is stored for the entire range of stations when saved. Click **[Save]** button after changing Value.
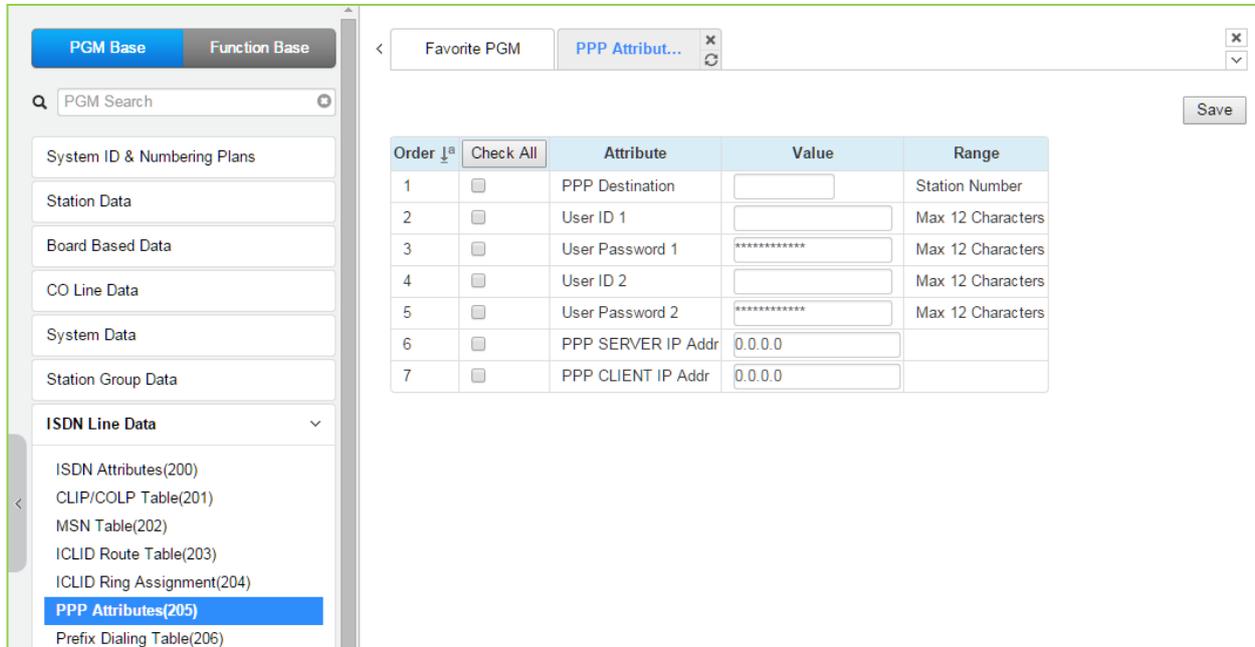
Figure 6.7.6-1 PPP Attributes

In addition to remote access via an IP network connection, the system database may be accessed remotely via an ISDN connection. Placing a call over an ISDN Line to the designated PPP Station will provide a connection to the system database. The system will request a user id and password, which must match one of the User Ids and passwords assigned. After a matching id and password are received, the iPECS Login Home page is provided.

Table 6.7.6-1 PPP ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
PPP Destination	If the incoming capability is 64 Kbps unrestricted digital and the called party number matches the PPP destination, the system will automatically answer the call and request PPP ID and password.	Station number	None
User ID 1	The System accepts this PPP ID 1 as valid.	Max. 12 characters	likppp01
User Password 1	The password entered is used to authorize PPP ID 1.	Max. 12 characters	lpkts01
User ID 2	The System accepts this PPP ID 2 as valid.	Max. 12 characters	likppp02
User Password 2	The password entered is used to authorize PPP ID 2.	Max. 12 characters	lpkts02
PPP Server IP Addr	When configured, the PPP Server IP Address must match this entry. To apply this option the system must be restarted.	IP Address	
PPP Client IP Addr	When configured, the PPP Client IP Address must match this entry. To apply this option the system must be restarted.	IP Address	

6.7.7 ISDN Prefix Dialing Table

Selecting ISDN Prefix Dialing Attributes will display the Prefix Dialing Table Attributes data entry page. Click **[Save]** button after changing Value.
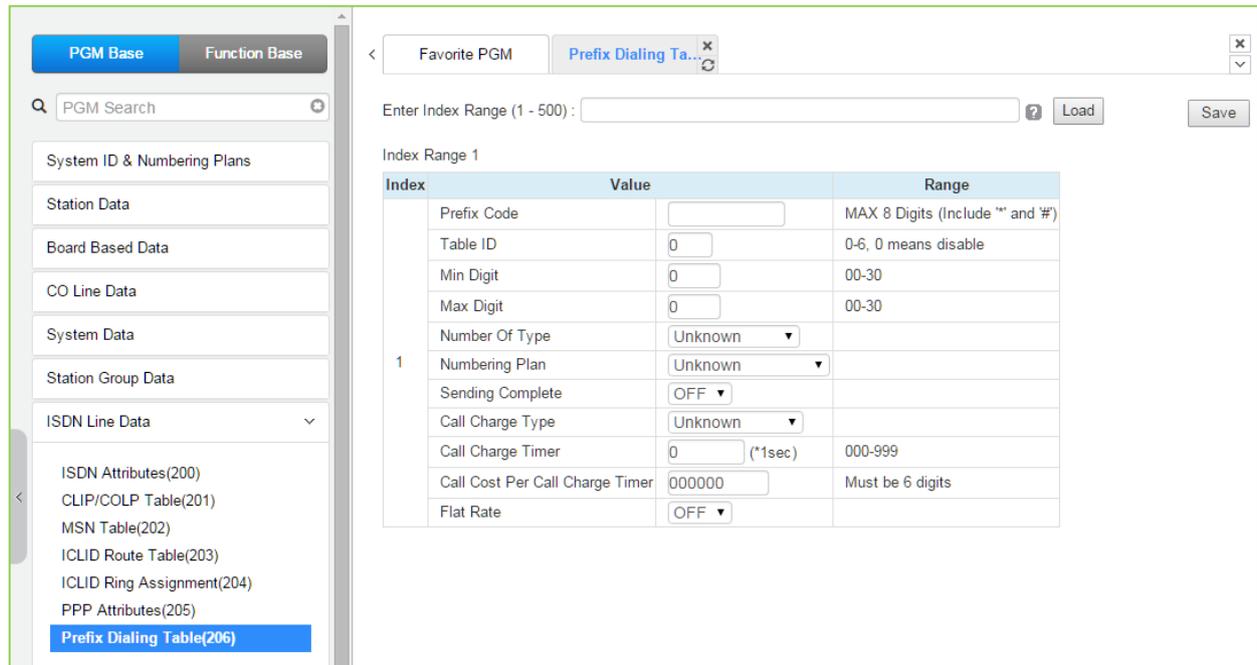
Figure 6.7.7-1 Prefix Dialing Table Attributes

Prefix Dialing Table. With this table, three features can be supported.

1. Analog CO Call Charge with NPR metering.
2. SIP direct dialing with no wait inter-digit timer.
3. ISDN Prefix Call – ISDN en-block Dialing with Prefix Call Setup.

If first some digits (up to 8 digits) of outgoing dial number are matched with Prefix Code of each table, this table can start work. By each Co-line (PGM 142 – F20), Table ID (0-6) can be set. This table ID (PGM 142 – F20) is associated with PGM 206 – each table ID.

Table 6.7.7-1 Prefix Dialing Table Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Prefix Code	The Prefix code is the digits that the user must dial as the first digits in order to activate the Prefix Dialing options.	Max. 8 Digits (Include * and #)	
Table ID	The Prefix Code is assigned to a Table (0~6) that is referenced in the CO Line Common Attributes to define the Prefix codes are associated with the CO/IP Line.	0-6	0
Min Digit	This field defines the minimum number of digits the user must dial to activate Enblock dialing and includes the Prefix digits.	00-30	0
Max Digit	This field defines the maximum number of digits the user must dial to activate Enblock dialing and includes	00-30	0

Table 6.7.7-1 Prefix Dialing Table Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	the Prefix digits.		
Number Of Type	Select Number of Type (0~6).	Unknown, International, National, Network spec, Subscriber, Abbreviated, Reserved	Unknown
Numbering Plan	Select Numbering Plan (0~6).	Unknown, ISDN Telephony, Data numbering, Telex, National standard, Private, Reserved	Unknown
Sending Complete	When the user-dialed digits match the Prefix digits, the system will send the digits Enblock to the ISDN or SIP Call server.	ON/OFF	OFF
Call Charge Type	This field defines the Call type for the charges. The Call Charge Type can be assigned as Unknown, Local, Long Distance, International, Mobile, or reserved	Local, Long distance, International, Mobile, Reserved	Unknown
Call Charge Timer	When the user dialed digits match the assigned Prefix code on an analog CO Line, the call charge is calculated using this timer as Call Duration/Call Charge Timer * Cost per pulse.	000-999	0
Call Cost	Call Cost is calculated by CALL TIMER. (ex : timer is 1 min, cost is 000020, then after 3 minute call, total call cost is calculated to 000060)	000000-999999	000000
Flat Rate	If Flat Rate is ON, Flat Rate is applied by CALL COST per a call.	ON/OFF	OFF

6.8 SIP Data

Selecting the SIP Data program group returns the sub-menu displayed in the left frame as shown in the following figure.
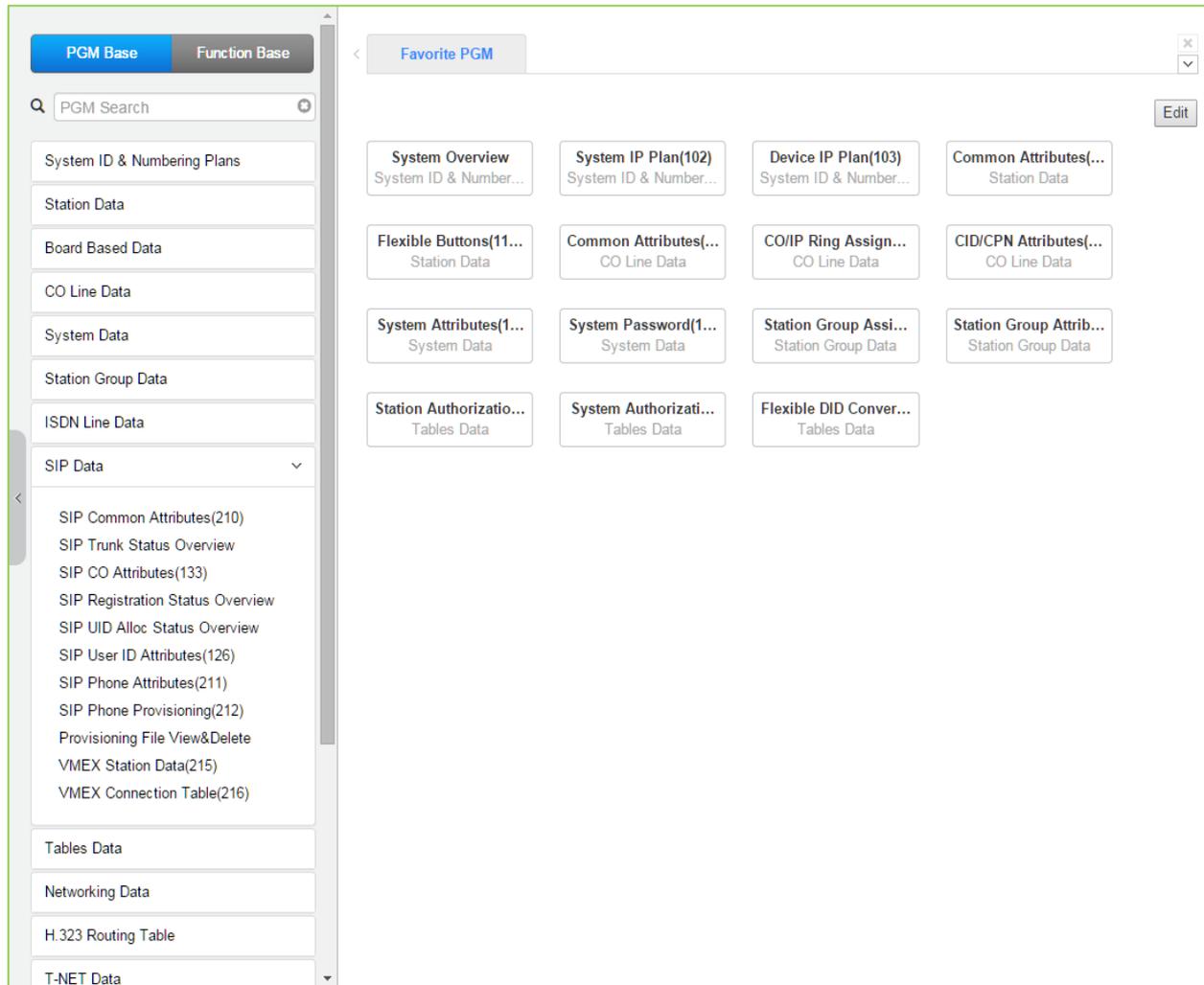
Figure 6.8-1 SIP Data

6.8.1 SIP Common Attributes - PGM 210

Selecting SIP Common Attributes will display the SIP System based Attributes data entry page. The attributes are system based SIP server data that running on UCP.

DNS server address is where system can get IP address of external party that was written in Name in system. Local UDP/TCP/TLS Port is UCP's SIP signaling port number.

'Signal TLS Option' is for the SIP signaling by TLS configuration.

'SIP Status' the status of is running of SIP server in UCP.
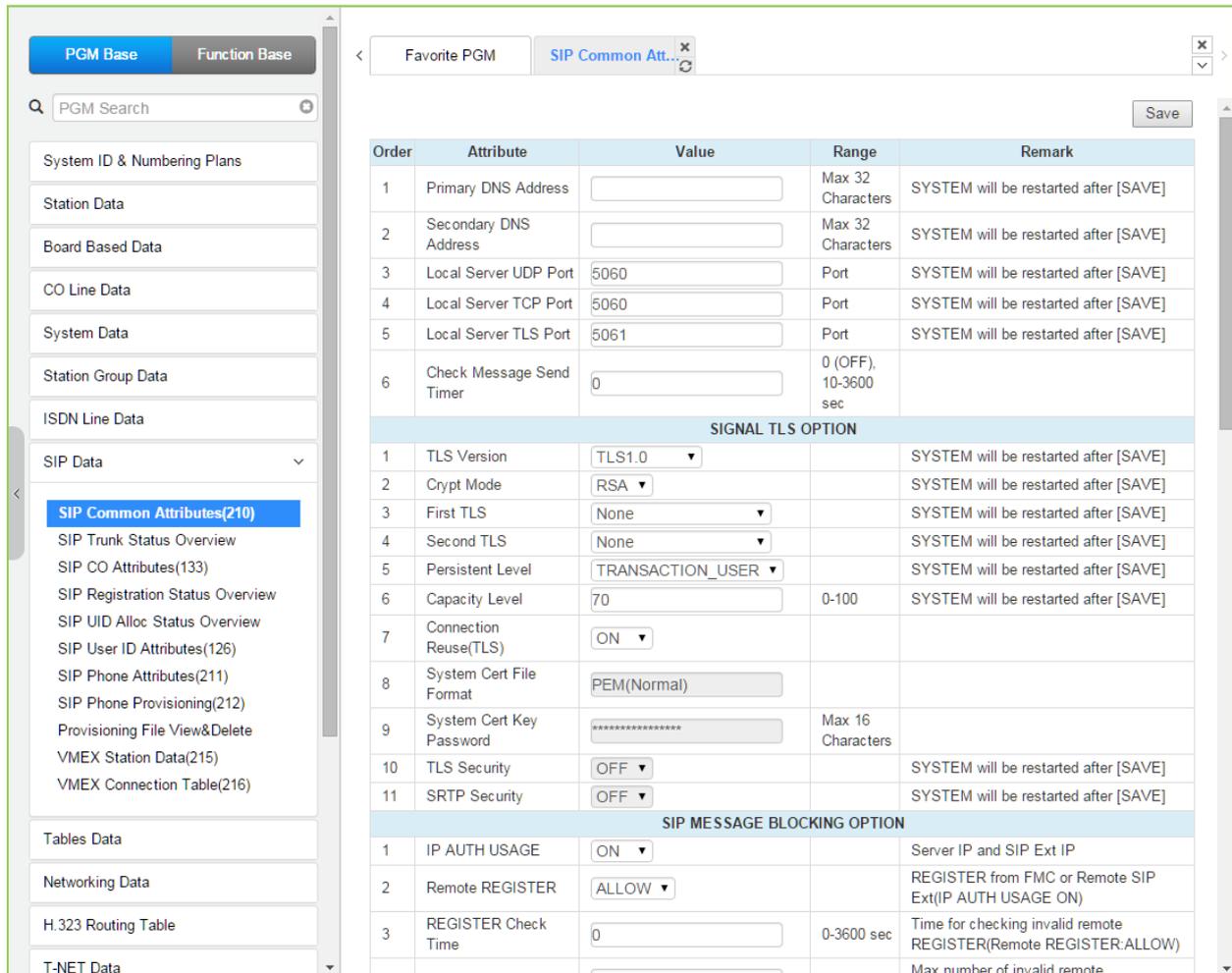
Figure 6.8.1-1 SIP Common Attributes

Check Message Send Timer – This is Keep Alive Message (OPTIONS) frequency from SIP server (UCP) to SIP Phone. If a SIP Phone does not respond to system's Keep Alive Message then system will make the status of SIP Phone to 'disconnected' in system.

Keep Alive Message (OPTIONS) programming for a SIP station is as below:
- Frequency: SIP Data / SIP Common Attributes (210) - Check Message Send Timer.
- Usage ON/OFF for a SIP Extension: SIP Data / SIP Phone Attributes (211) – Keep Alive Usage.
- Retry Count: IP Data / SIP Phone Attributes (211) – Retry Count.

Table 6.8.1-1 SIP Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Primary DNS Address	The system will contact this IP address as the primary Name Resolution Server. System will restart after Save.	IP Address (Max. 32 characters)	
Secondary DNS Address	The system will contact this IP address as a secondary Name Resolution Server. System will restart after Save.	IP Address (Max. 32 characters)	
Local Server UDP Port	The UDP port for SIP signaling packets is configurable. System will restart after Save.		5060
Local Server TCP Port	The TCP/IP port for SIP signaling packets is configurable. System will restart after Save.		5060
Local Server TLS Port	The TLS port for SIP signaling packets is configurable. System will restart after Save.		5061
Check Message Send Timer	To assure a registered SIP Phone is still functioning, the system sends a SIP OPTION message to the phone at this interval. OPTIONS Usage must be enabled for the SIP Trunk (PGM133) and Keep Alive Usage (PGM 211) for the SIP phone must be enabled.	10 ~ 3600 (Sec.)	0
Signal TLS option			
TLS Version	The Transport Layer Security (TLS) version options are: TLS1.0: TLS1.0 is used for TLS connection. TLS1.2: TLS1.2 is used for TLS connection. SSL3 (Auto): TLS1.0/TLS1.2 with auto negotiation System will restart after Save.	TLS1.0, TLS1.2, SSL3(Auto)	TLS1.0
Crypt Mode	The RSA or ECC cryptography modes are available. System will restart after Save.	RSA, ECC	RSA
First TLS	The primary TLS key encryption method for SIP signaling packets can be configured. System will restart after Save.	None, ARIA-128, AES-128, ARIA-128-SHA2, AES-128-SHA2, AES-256-SHA2(RSA)	None
Second TLS	The secondary TLS encryption key method for SIP signaling packets can be configured. System will restart after Save.	None, ARIA-128, AES-128, ARIA-128-SHA2, AES-128-SHA2, AES-256-SHA2(RSA)	None
Persistent Level	The TLS path for SIP signaling packets can be selected as: Transaction: different path than other signaling Transaction_user: same path as other signaling System will restart after Save.	TRANSACTION, TRANSACTION_USER	TRANSACTION_USER
Capacity Level	The maintenance rate or capacity of the TLS session can be limited, maximum 70%. System will restart after Save.	0 - 100	70

Table 6.8.1-1 SIP Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Connection Reuse (TLS)	The TLS session can be maintained and reused for another session.	OFF ON	ON
System Cert File Format	The system supports two certification formats Privacy-Enhanced Electronic Mail (PEM) or Distinguished Encoding Rules (DER).	PEM(Normal) DER (Normal)	PEM (Normal)
System Cert Key password	Password to encrypt private key.	Max. 16 characters	
TLS Security	Change password that is used to encrypt TLS certification periodically.	OFF ON	OFF
SRTP Security	Allow only SRTP call (blocking none SRTP call)	OFF ON	OFF
SIP Message blocking option			
IP AUTH USAGE	The system will authenticate the origin of SIP Requests (INVITE, REGISTER, NOTIFY, OPTIONS, etc.). If the origin IP address is a registered SIP phone or configured SIP Trunk (PGM 133) the system will respond to the request otherwise, the request is ignored.	OFF ON	ON
Remote Register	When IP Authentication above is enabled, remote devices must be allowed to register.	ALLOW DENY	ALLOW
REGISTER Check Time	Enabling IP Authentication and Remote registration may result in attempts to flood the system with registration requests. The system can check for and lock out registration if flooding occurs. This time sets the interval the system will use to check for flooding.	0~3600 (Sec.)	0
REGISTER Threshold	Enabling Remote registration may result in attempts to flood the system with registration requests. The system can check for and lock out registration if flooding occurs. This value sets the number of registration requests for the flooding threshold allowed within the Check Time.	0~60000	0
REGISTER Lock Time	With IP Authentication enabled and registration allowed, should a registration flood be detected, the system will ignore registration requests for this timer period.	0~250 (Min.)	0
SIP SMS Option			
SMS Domain	When SIP SMS is available, this field configures the Domain Name used for sending messages.	Max. 32 characters	
Request URI	For Korea Telecom, the "Request URI" employed when sending a SIP SMS can be defined.	Max. 32 characters	1549
SMS Mode	For Korea Telecom, this value assigns the SMS mode, Normal or External.	Normal, External	Normal
SIP FAX Option			
Start w/ G.711 Fax	When a SIP Trunk call is directed to a Fax port, the system will request use of the G711 Codec.	OFF ON	OFF
G.711 Fax method	When using G711 for Faxing, the VBD (Voice Band Data) encoding can be assigned for use with	711A/711U/711A(VBD). 711U(VBD)	711A

Table 6.8.1-1 SIP Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	the G.711 Codec for improved Fax performance.		
T38 FAX Failover(711)	When T38 is employed for Faxing, should negotiation fail, the system can Fail-over to use the G711 Codec for Fax Transmission.	OFF ON	OFF
Miscellaneous Option			
OCS Prefix Code	When the server type assigned for a SIP Trunk is OCS, the system will send these digits as a prefix to the number in the SIP "To:" header.	Max. 8 Digits	
SIP Pound Use	SIP employs Enblock dialing where the user dials all digits before they are sent to the carrier for processing. When the user completes dialing of a SIP call, '#' is used to indicate end of dialing. If users must be able to dial '#', SIP Pound Use can be disabled and the system will automatically send digits at expiration of the inter-digit time.	OFF ON	OFF
BLF SYNC NOTIFY Timer	When the system reboots, the button LEDs of SIP phones may indicate erroneous status. To display proper status LEDs, the system sends a Notify message to synchronize the LED states at expiration of this timer.	10-360 seconds	10
SRTP PATH(SIPEXT)	For SIP extensions connected to the same LAN as the system, the SRTP path can be established through a VoIP channel (Packet Relay) or directly between the end-points.	VOIM RELAY, DIRECT	VOIM RELAY
DNS SRV Usage	The system can be configured to query the DNS for the SRV resource record, which defines domains for various services.	OFF ON	OFF
Out OF Rsc Response	When the system has no resources available for the SIP Request, the system will respond with this code.	503, 486	503
Error Code For Trunk Rerouting	If the iPECS UCP receives this Error Code in response to a request, the call will be rerouted. The semi-colon (;) is used to separate multiple SIP message codes.		
SIP QOS Option			
SIP QOS Method Selection	For SIP messages, the system can employ Diff Serv or TOS to implement QoS.	DSCP, TOS	TOS
SIP Signal DSCP value	When Diff Serv is selected as the QoS method, the, DSCP value for SIP signaling messages is defined.	0~63	0
SIP Signal TOS value	When TOS is selected as the QoS method, the TOS value for SIP signaling messages is defined.	0~7	5
RTP DSCP value	When DSCP is selected as the QoS method, the DSCP value for RTP packets is defined.	0~63	0
RTP TOS value	When TOS is selected as the QoS method, the TOS value for RTP packets is defined.	0~7	5

Table 6.8.1-1 SIP Common Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
SIP T-NET Option			
CM Prefix	Korea Telecom only, when employing an iPECS-CM as the CM, iPECS UCP will require a Prefix to set-up a SIP trunk call.	Max. 4 Digits	
CM Prefix Method	Korea Telecom only, enables the Prefix method for processing a call with a SIP trunk through iPECS-CM.	Normal, With prefix	Normal
SIP TRUNK REGISTER Option			
Register Retry Timer	When registration fails, the UCP attempts to register periodically at intervals of this timer.	20~3600	60
Option Check number	Korea Telecom only, a SIP Options message is used for redundancy. If the system does not respond to the Option message after the number of attempts, the redundant server becomes active	1~20	
Option Check Interval	Korea Telecom only, a SIP Options message is used for redundancy. The Option message is sent at intervals of this timer.	20~3600	
SIP ALARM SERVER Option			
Alarm Server Usage	If this value is changed, all WTIM will restart.	OFF/ ON	OFF
Alarm Server Address	Enter the alarm server address up to 32 characters.	Max. 32 characters	
Alarm Server UDP Port	Default UDP port for Alarm server.	Port	5060
Alarm Server Check Usage	If IPECS doesn't receive reply for this option 3 consecutive times, will not send Message and Information to Alarm server. PGM 210 check message send timer.	OFF/ ON	OFF
Terminal State Announcement	Information (Terminal state) is sent to Alarm server if this value is ON, not sent to Alarm server if this value is OFF.	OFF/ ON	OFF
600BE Channels for Alarm	The number of 600BE channels reserved for Alarm SMS.	0~1	0

6.8.2 SIP Trunk Status Overview

Selecting SIP Trunk Status Overview displays the overview page. The page displays the Proxy, Domain, etc. for the SIP Trunks configured in PGM 133.

Figure 6.8.2-1 SIP trunk status overview

6.8.3 SIP CO Attributes - PGM 133

Selecting SIP CO Attributes will display the SIP CO Attributes data input page, Figure 6.8.3-1. Enter the CO Range and click **[Load]** to enter attribute values. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of stations when saved.
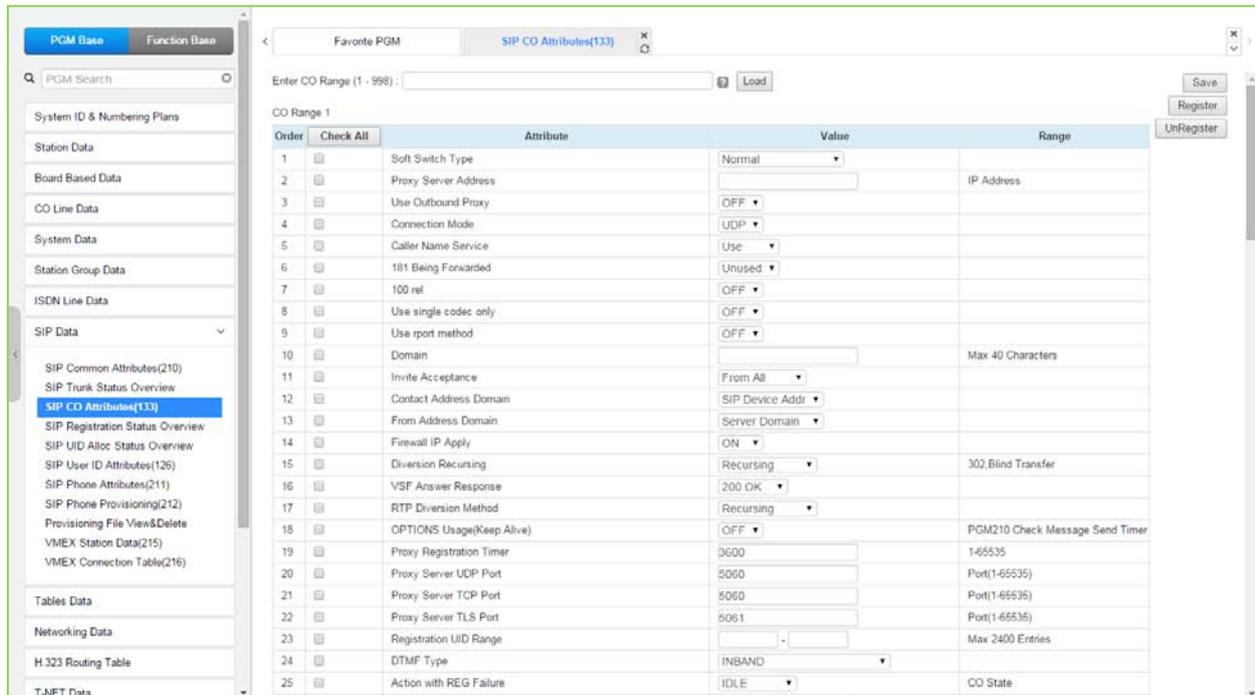
Figure 6.8.3-1 SIP CO Attributes

Various parameters must be entered for proper operation of SIP Trunks including the SIP proxy and Registrar as outlined in Table 6.8.3-1.

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Soft Switch Type	The Soft Switches that provide SIP Trunking service each may have slightly different extended capabilities and protocol implementations. While "Normal" will function with standard SIP Call Servers, extended functions of specific servers mentioned are available.	Normal, Broadsoft, KT. SK TELINK, KT-C, MS OCS, SKYPE CONNECT, SIP-CC, TI PK0, ERICSSON IMS, DNS REDUNT, MS LYNC KT CENTREX	Normal
Proxy Server Address	This field assigns the IP address or URL of the Service Provider's SIP Proxy server.	IP address or URL	

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Use Outbound Proxy	With this field enabled, the system will send all SIP messages to the Proxy server defined above.	OFF ON	OFF
Connection Mode	This field establishes the transport protocol for SIP signaling packets as UDP, TCP or TLS.	UDP, TCP, TLS	UDP
Caller Name Service	The caller name may be included in SIP messages. When included, the name can display in the LCD of iPECS IP and LDP Phones. In addition, the Contact Display Name assigned to the SIP User Id (PGM 126) will be sent in the SIP message.	Unused Use	Use
181 Being Forwarded	A SIP 181 Message is sent when a call is being redirected or forwarded, if enabled.	Unused Use	Unused
100 rel	To improve reliability of Provisional SIP messages, the system is configured to send and expect to receive an ACK response to such messages.	OFF ON	OFF
Use single codec only	During capabilities negotiation, the system sends the first priority codec id or the prioritized list of codecs as defined in the Codec priority settings below.	OFF ON	OFF
Use rport method	When employed behind a NAPT server, the system can use the Rport parameter in the SIP Via header to request the SIP server respond to the IP address and port of the originator.	OFF ON	OFF
Domain	Domain name of the Service Provider's SIP Call server that is used in SIP "To:" headers.	Max. 40 characters	
Invite Acceptance	The system can accept SIP INVITE requests from any domain or only from the "Domain" specified above.	Domain Only, From All	From All
Contact Address Domain	The system will populate the SIP "Contact" header Domain with either the iPECS device IP address or the "Domain" specified above.	SIP Device Addr, Server Domain	SIP Device Addr
From Address Domain	The system will populate the SIP "From" header Domain with either the iPECS device IP address or the "Domain" specified above.	SIP Device Addr, Server Domain	Server Domain
Firewall IP Apply	When the iPECS system and VoIP devices are assigned a Firewall IP address, the system can use either the Firewall or local IP address in the Via and Contact headers as well as in SDP messages.	OFF ON	ON
Diversion Recursing	When a SIP call is redirected by a 3xx Diversion response such as when a call forwards, the SIP message can be forked (recursing) or forwarded (non-recursing).	Recursing, Non-Recursing	Recursing
VSF Answer Response	The system can respond to a SIP Invite with a SIP 183 Session Progress message. This allows a VSF announcement to play and user dialed digits to be analyzed by CCR and, if the CCR destination is an external number, the system can send a SIP 3xx call diversion message to forward the call via the SIP network.	183 Msg. 200 OK	200 OK
RTP Diversion Method	Reserved Currently.	Recursing, Non-Recursing	Recursing
OPTIONS Usage	With "OPTIONS Usage" ON, an Option message is	OFF	OFF

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
(Keep Alive)	sent at intervals of the Check Message Send Timer assigned in PGM 210 to assure a connection with the SIP CO. SIP COs generally provide an Option message. In this case, the UCP should not be enabled here.	ON	
Proxy Registration Timer	Periodically, the system must re-register with the SIP Registrar. While this timing is often negotiated with the Registrar, the system can be configured with this timer to establish the re-register interval.	1-65535	3600
Proxy Server UDP Port	When employing UDP transport, this port number is employed for messages to the SIP proxy.	Port	5060
Proxy Server TCP Port	When employing TCP transport, this port number is employed for messages to the SIP proxy.	Port	5060
Proxy Server TLS Port	When employing TLS transport, this port number is employed for messages to the SIP proxy.	Port	5061
Registration UID Range	The User Id indices from the SIP User Id Attributes Table PGM 126 that will register with the SIP Service Provider's SIP Registrar must be configured.	Max. 2400 Entries	
DTMF Type	DTMF dialing signals can be sent from the system using in-band or various Info messages. The method or type must match the SIP Call Server.	Inband, 2833, Info (DTMF), Info (DTMF relay), Info (tel. event, Info (Nortel Networks)	Inband
Action with REG Failure	When registration fails, the link is down to the SIP Call server, or the system receives no response to an Invite message in the "Call Setup No-response" timer below, the call will return error tone (Wait Idle), or Fail-over to a Line from the Fail-over CO Group specified below (Idle).	IDLE WATI IDLE	IDLE
Media Port	The UDP ports used for RTP (media) packets can be limited to a fixed range.	UDP Port	6000-19972
Secondary Proxy Server			
Secondary Proxy Server Address	For Korea Telecom, a secondary SIP proxy server can be assigned for redundancy.	IP Address	
Secondary Domain	For Korea Telecom, a secondary Domain can be assigned for redundancy.	Max. 32 Characters	
Secondary Proxy Server UDP Port	For Korea Telecom, a UDP port for the secondary SIP proxy server can be assigned for redundancy.	Port	
ID Presentation Option			
ID Usage			
P-Asserted-ID	The system normally provides a P-Asserted ID in SIP messages. The system can be configured not to provide the header.	Unused Use	Use
Remote-Party-ID	The system normally provides a Remote-Party-ID in	Unused	Use

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	SIP messages. The system can be configured not to provide the header.	Use	
Privacy(CLIR) Presentation	SIP employs various headers that include a User Id and Name. In some cases, it may desirable to restrict the called party from receiving this information. Several options for Caller Id restriction can be applied.	Anonymous Name & Anonymous Number, Anonymous Name, Privacy: user, Privacy: id, Privacy: user;id;critical, Privacy: id & anonymous & P-Preferred-ID	Anonymous Name & Anonymous Number
ID Individuality			
CID Password	Enter CID Password.		

From ID	The Id in the "From" header of SIP messages can be based on the calling station, the User Id or a fixed User Id.
	    Extension SIP User Id: one of three SIP User Ids assigned to the station in PGM 111. Select which of the three indices to use in the SIP UID Selection below.
	    Extension outgoing CLI: the CLI configured for the station through Web PGM 113 and 151.
	    Authorized Representative Id: the User Id of the "Authorized Representative Id" assigned to the SIP User Id that is indexed to the station in PGM 111. The specific index is selected below as the SIP UID Selection.
	    Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below
		Extension SIP-User-ID Table, Extension outgoing-CLI, Authorized Representative ID, Fixed Table	Extension SIP-User-ID Table

From Display	The "Display" field of the "From" header can be configured to use the below:
	    System Rule:
	     a. From ID = Extension SIP User-ID Table, display Contact Display Name from PGM 126, otherwise display Station Name from PGM 111 or blank
	     b. From Id = Extension Outgoing CLI, display Station Name from PGM 111 or blank
	     c. From Id = Authorized Rep Id, display Contact Display Name of Authorized Rep Id from PGM 126 otherwise display Station Name from PGM 111 or blank
	     d. From Id = Fixed Table display Contact Display Name from PGM 126 or Station Name from PGM 111 or blank
	    Extension outgoing CLI: the CLI configured for the station through Web PGM 113 and 151
		SYS RULE, Extension outgoing-CLI	SYS RULE

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT

P-Asserted-ID	The Id in the "P-Asserted Id" header of SIP messages can be based on the calling station, the User Id or a fixed User Id.
	    Ext SIP User Id: one of three SIP User Table indices assigned to the station in PGM 111. Select which of three in the SIP UID Selection below.
	    Extension outgoing CLI: the CLI configured for the station through Web PGM 113 and 151.
	    Authorized Representative Id: the User Id of the "Authorized Representative Id" assigned to the SIP User Id that is indexed to the station in PGM 111. The specific index is selected below as the SIP UID Selection.
	    Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.
		Extension SIP-User-ID Table, Extension outgoing-CLI, Authorized Representative ID, Fixed Table	Extension SIP-User-ID Table

P-Asserted-ID Display	The "Display" field of the "P-Asserted Id" header can be configured to use the below:
	    System Rule:
	     a. P-Asserted ID = Extension SIP User-ID Table, display Contact Display Name from PGM 126, otherwise display Station Name from PGM 111 or blank.
	     b. P-Asserted Id = Extension Outgoing CLI, display Station Name from PGM 11 or blank.
	     c. P-Asserted Id = Authorized Rep Id, display Contact Display Name of Authorized Rep Id from PGM 126 otherwise display Station Name from PGM 111 or blank.
	     d. P-Asserted Id = Fixed Table display Contact Display Name from PGM 126 or Station Name from PGM 111 or blank.
	    Extension outgoing CLI: the CLI configured for the station through Web PGM 113 and 151.
		SYS RULE, Extension outgoing-CLI	SYS RULE

Contact ID	The Id in the "Contact ID" header of SIP messages can be based on the calling station, the User Id or a fixed User Id.
	    Ext SIP User Id: one of three SIP User Table indices assigned to the station in PGM 111. Select which of three in the SIP UID Selection below.
	    Extension outgoing CLI: the CLI configured for the station through Web PGM 113 and 151.
	    Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.
		Extension SIP-User-ID Table, Extension outgoing-CLI, Fixed Table	Extension SIP-User-ID Table

Remote-Party-ID	The Id in the "Remote-Party ID" header of SIP messages can be based on the calling station, the User Id or a fixed User Id. Ext SIP User Id: one of three SIP User Table indices assigned to the station in PGM 111. Select which of three in the SIP UID Assignment section	Extension SIP-User-ID Table, Extension outgoing-CLI, Fixed Table	Extension SIP-User-ID Table

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	below. Extension outgoing CLI: the CLI configured for the station through Web PGM 113 and 151. Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.		
Offnet Call Route ID Transit			
CO to Offnet Direct Call Route **The below applies to calls routed from a CO/IP Line to an Off-net location by the system over a SIP Trunk.**			
From/Contact ID	The Id in the "From" and "Contact" headers of SIP messages employ System Attendant CLI, the original CLI or a fixed User Id. Sys Atd: the CLI configured for the System Attendant through Web PGM 113 and 151. Original CLI: the CLI received by the system for the original incoming call. Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.	SYS ATD, Original CLI, Fixed Table	SYST ATD
From Display	The "Display" field of the "From" header can be configured to use the below: System Rule: a. From ID = System Atd, display Contact Display Name from PGM 126, otherwise display Attendant Station Name from PGM 111 or blank. b. From Id = Original CLI, display the Name in received by the system for the original call. c. From Id = Fixed Table display Contact Display Name from PGM 126 or Station Name from PGM 111 or blank. Original CLI: the CLI received by the system for the original incoming call.	SYS RULE, Original CLI	SYS RULE
P-Asserted-ID	The Id in the "P-Asserted Id" header of SIP messages can be based on the System Attendant, the Original CLI or a fixed User Id. Sys Atd: CLI: the CLI configured for the System Attendant through Web PGM 113 and 151. Original CLI: the CLI received by the system for the original incoming call. Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.	SYS ATD, Original CLI, Fixed Table	SYST ATD
P-Asserted-ID Display	The "Display" field of the "P-Asserted Id" header can be configured to use the below: System Rule: a. P-Asserted ID = System Atd, display Contact Display Name from PGM 126, otherwise display Attendant Station Name from PGM 111 or blank. b. P-Asserted Id = Original CLI, display the Name in received by the system for the original call. c. P-Asserted Id = Fixed Table display Contact Display Name from PGM 126 or Station Name from	SYS RULE, Original CLI	SYS RULE

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	PGM 111 or blank. 　　Original CLI: the CLI received by the system for the original incoming call.		
Remote-Party-ID	The Id in the "Remote-Party ID" header of SIP messages can be based on the System Attendant, the Original CLI or a fixed User Id. 　　Sys Atd: the CLI configured for the System Attendant through Web PGM 113 and 151. 　　Original CLI: the CLI received by the system for the original incoming call. 　　Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.	SYS ATD, Original CLI, Fixed Table	Original CLI
Diversion	The Id in the "Diversion" header of SIP messages can be based on the System Attendant, the Original CLI or a fixed User Id, or Unused. 　　Unused: no Diversion header provided. 　　Sys Atd: the CLI configured for the System. Attendant through Web PGM 113 and 151. 　　Original CLI: the CLI received by the system for the original incoming call. 　　Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.	Unused, SYS ATD, Original CLI, Fixed Table	Unused
<td colspan="4" align="center">**Offnet Call Forward by Station** **The below apply to calls routed from a CO/IP Line to an Off-net location by a station over a SIP Trunk.**</td>
From/Contact ID	The Id in the "From" and "Contact" headers of SIP messages employ the forwarding Station, the original CLI or a fixed User Id. 　　Extension: the Extension SIP User ID or Extension Outgoing CLI as assigned for the From ID under ID Individuality. 　　Original CLI: the CLI received by the system for the original incoming call. 　　Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.	Extension, Original CLI, Fixed Table	Extension
From Display	The "Display" field of the "From" header can be configured to use the below: 　System Rule: 　　a. From ID = Extension, display Contact Display Name from PGM 126, otherwise display Station Name from PGM 111 or blank. 　　b. From Id = Original CLI, display the Name in received by the system for the original call. 　　c. From Id = Fixed Table display Contact Display Name from PGM 126 or blank. 　　Original CLI: the CLI received by the system for the original incoming call.	SYS RULE, Original CLI	SYS RULE
P-Asserted-ID	The Id in the "P-Asserted Id" header of SIP messages employ the forwarding Station, the original CLI or a fixed User Id.	Extension, Original CLI, Fixed Table	Extension

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT

	Extension: the Extension SIP User ID or Extension Outgoing CLI as assigned for the From ID under ID Individuality.
	     Original CLI: the CLI received by the system for the original incoming call.
	     Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.		

P-Asserted-ID Display	The "Display" field of the "P-Asserted Id" header can be configured to use the below:
	     System Rule:
	     a. P-Asserted ID = Extension, display Contact Display Name from PGM 126, otherwise display Station Name from PGM 111 or blank.
	     b. P-Asserted Id = Original CLI, display the Name in received by the system for the original call.
	     c. P-Asserted Id = Fixed Table display Contact Display Name from PGM 126 or blank.
	     Original CLI: the CLI received by the system for the original incoming call.
		SYS RULE, Original CLI	SYS RULE

Remote-Party-ID	The Id in the "Remote Party Id" header of SIP messages employ the forwarding station, the original CLI or a fixed User Id.
	     Extension: the Extension SIP User ID or Extension Outgoing CLI as assigned for the From ID under ID Individuality.
	     Original CLI: the CLI received by the system for the original incoming call.
	     Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.
		Extension, Original CLI, Fixed Table	Extension

Diversion	The Id in the "Diversion" header of SIP messages can be based on the forwarding station, the Original CLI or a fixed User Id, or Unused.
	     Unused: no Diversion header provided.
	     Extension: the Extension SIP User ID or Extension. Outgoing CLI as assigned for the From ID under ID Individuality.
	     Original CLI: the CLI received by the system for the original incoming call.
	     Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.
		Unused, Extension, Original CLI, Fixed Table	Unused

Mobile Extension External Call **The below applies to calls routed to a Mobile Extension over a SIP Trunk**			

From/Contact ID	The Id in the "From" and "Contact" headers of SIP messages employ the calling station, the original CLI or a fixed User Id.
	     Extension: the Extension SIP User ID or Extension Outgoing CLI as assigned for the From ID under ID Individuality.
	     Original CLI: the CLI received by the system for the
		Extension, Original CLI, Fixed Table	Extension

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	original incoming call or Station number for ICM call. Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.		

From Display	The "Display" field of the "From" header can be configured to use the below:
	    System Rule:
	      a. From ID = Extension, display Contact Display Name from PGM 126, otherwise display Station Name from PGM 111 or blank.
	      b. From Id = Original CLI, display Station Name from PGM 111 for ICM call or the Name in received by the system for the original outside call.
	      c. From Id = Fixed Table display Contact Display Name from PGM 126 or blank.
	    Original CLI: the CLI received by the system for the original incoming call.
		SYS RULE, Original CLI	SYS RULE

P-Asserted-ID	The Id in the "P-Asserted Id" header of SIP messages employ the calling station, the original CLI or a fixed User Id.
	    Extension: the Extension SIP User ID or Extension Outgoing CLI as assigned for the From ID under ID Individuality.
	    Original CLI: the CLI received by the system for the original incoming call or station number for ICM call.
	    Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.
		Extension, Original CLI, Fixed Table	Extension

P-Asserted-ID Display	The "Display" field of the "P-Asserted Id" header can be configured to use the below:
	    System Rule:
	      a. P-Asserted ID = Extension, display Contact Display Name from PGM 126, otherwise display Station Name from PGM 111 or blank.
	      b. P-Asserted = Original CLI, display Station Name from PGM 111 for ICM call or the Name in received by the system for the original outside call.
	      c. P-Asserted = Fixed Table display Contact Display Name from PGM 126 or blank.
	    Original CLI: the CLI received by the system for the original incoming call.
		SYS RULE, Original CLI	SYS RULE

Remote-Party-ID	The Id in the "Remote Party Id" header of SIP messages employ the Station CLI, the original CLI or a fixed User Id.
	    Extension: the Extension SIP User ID or Extension Outgoing CLII as assigned for the From ID under ID Individuality.
	    Original CLI: the CLI received by the system for the original incoming call or station number for ICM call
	    Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.
		Extension, Original CLI Fixed Table	Extension

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT

Diversion	The Id in the "Diversion" header of SIP messages can be based on the calling station, the Original CLI or a fixed User Id, or Unused.
	      Unused: no Diversion header provided.
	      Extension: the Extension SIP User ID or Extension. Outgoing CLII as assigned for the From ID under ID Individuality.
	      Original CLI: the CLI received by the system for the original incoming call or the station number for ICM call.
	      Fixed Table: the SIP User Id from the SIP UID Fixed Table Index attribute assigned below.
		Unused,
		Extension,
		Original CLI,
		Fixed Table
			Unused

SIP UID Assignment			
SIP User ID Fixed Table Index	When a header is assigned to use "Fixed Table", the Id from this SIP User ID (PGM 126) Table index is used.	Index	
SIP User ID SELECTION	When a header is assigned to use the "Extension SIP-User-ID Table", the SIP User ID is selected using this SIP UID index in the Station Attributes (PGM 111).	Index, Index 2, Index 3	SIP User Table Index
External CODEC Priority Configuration			
1st ~5th priority	When the system negotiates with the SIP Trunk, the Codec priority specified is employed. If only the 1st. priority Codec is defined, only that Codec is available from the system. If the Codec is not supported by the remote party, the call will fail.	None, g.711-u, g.711-a, g.723.1, g.729, g.729-a, g.722	None
SIP Call Setup Failover Option			
Call Setup No Response Time	When the system initiates a SIP Trunk call and receives no response from the SIP proxy server, after expiration of this timer the SIP call is canceled and a Fail-over call placed on the Fail-over CO/IP Line group specified below. Note the timer can be set at 3 to 15 seconds and '0', which disables Fail-over.	0, 3 ~ 15 sec	5
Failover CO Group Number	When the system attempts to initiate a SIP Trunk call and the SIP Trunk is in an OOS state or the SIP proxy server does not respond in the No Response time above, the system will cancel the SIP call and place a call over a CO/IP Line from this group.	CO/IP Line Group	
SIP Session Timer			
Session Timer Usage	During a SIP call or "session", there are no signaling packets sent or received from the SIP Call server. In order to assure a session is still active, the system can periodically send an Option message that the SIP Call server should acknowledge.	OFF ON	OFF
Session Timer Value	When "Session Timer Usage" is enabled, the system will verify the session at this timer interval.	90~ 3600	360
Min SE	During negotiation with "Session Timer Usage" enabled, the system will use this value as the minimum Session expiration timer and will not respond to a SIP	90~ 3600	90

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	Option message prior to expiration of this timer.		
URI Formatting and Rules			
General Formatting			
To Field Method	The SIP "To:" header is formatted using the SIP or Telephony method as shown below. SIP method To: < sip:[Number]@[Domain];user=phone > Telephony method To: < tel:+[Number] >Domain	sip: method tel: method	SIP: method

Numbering Format	When assigned the Telephony method for the "To" header, the number format can be:
	    Local - [tel:+Number]@[Domain] or
	    Global (+E164) - [tel:+E.164 Address]@[Domain]
	    E.164 Address: Nation + Area Code + Number
		Local
		Global
			Local

Local: include Area Code	The Area code (PGM 143) can be added as a prefix if the 'Numbering Format' is set as Local. Example: user dials '8701234' and the Area code is '042' The resulting "To" field URI is tel:+0428701234@[Domain]	Yes No	No
Global: include phone-context	If the 'Numbering Format' is Global and the 'To Field Method' is Telephony, the 'phone-context' can be added as below. user dials '0011428701234' from Country code 82 The resulting "To" field URI is, tel:+0011428701234@[Domain];phone-context=+82	Yes No	No
Specific Formatting by Conversion (example)			
From 4 digits	To 6 digits	User Dial	Result
	The first digits, up to four (4), dialed by the user can be converted to a digit string of up to six (6) digits and may include a plus (+) sign. For example if the "From" digits are 00 and the "To" digits are +820, if the user dials 002233432, the system converts the number to +820 2233432.		
SRTP Setting			
SRTP Usage	When implemented by the carrier, the system can encrypt media (RTP) packets employing SRTP (Secure Real-Time Protocol). But VoIP Virtual switching channel does not support SRTP.	OFF ON	OFF

1st CRYPTO	The first priority cryptographic method for SRTP is selected from the below:
	    • None
	    • ARIA_CM_192_HMAC_SHA1_80
	    • AES_CM_128_HMAC_SHA1_80
	    • ARIA_CM_128_HMAC_SHA1_80
	    • AES_CM_192_HMAC_SHA1_80
	    • ARIA_CM_256_HMAC_SHA1_80
			None

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	• AES_CM_256_HMAC_SHA1_80		
2nd CRYPTO	The second priority cryptographic method for SRTP is selected from the below: • None • ARIA_CM_192_HMAC_SHA1_80 • AES_CM_128_HMAC_SHA1_80 • ARIA_CM_128_HMAC_SHA1_80 • AES_CM_192_HMAC_SHA1_80 • ARIA_CM_256_HMAC_SHA1_80 • AES_CM_256_HMAC_SHA1_80		None
Caller/Called ID			
ID Option			
Caller ID Selection	For the purposes of display and ICLID call routing, the iPECS employ this header as the "Caller ID".	P-Asserted-ID, Remote-Party-D, From ID	P-Asserted-ID
Display Caller Name(Though Id is Anonymous)	Even though the User Id is Anonymous, the system can display the SIP "From" header "Display Name" field for the call.	No Yes	No
Called ID Selection	For the purposes of call routing, the "SIP Request" or "To" header will be employed by the system as the "Called Party ID".	Request URI, To ID	Request URI
Miscellaneous set			
Drop Busy Station	While busy, the system can be configured to terminate a station call, and accept and connect any new call from the SIP Trunk. For special use in Italy only.	No Yes	No
Ignore INBAND DTMF	In some situations, DTMF tones from the connected party may be received along with the DTMF Type specified above. This can cause errors in detection such as double digits. In this case, the system can be configured to ignore Inband DTMF signals. Note do not disable Inband signals if selected as the "DTMF Type" above as this may disable DTMF detection.	No Yes	No
SIP Trunk Group	Multiple SIP User Id ranges may register with the SIP CO, for example, SIP COs from different providers. When different SIP User ID ranges are required on a SIP CO, a different SIP Trunk Group should be assigned to each range. Note this has no relationship with the CO/IP Line Group.	0~71 (0:Normal/1-71:Check To header)	0
Send Refer for Transfer	The system can employ the SIP Refer method to forward or transfer incoming calls to an "Off-net" location if supported by the SIP Service Provider.	No Yes	No
CN Payload Insert	When Comfort Noise generation is desired, the system will provide a Comfort Noise Payload during periods of silence.	No Yes	No
Ignore 180 after 183	If the option is Yes, an appropriate ring back tone may be played to the calling party when the called party's phone is alerting without changing the internal or system ring back tone. For example, when placing a	No Yes	No

Table 6.8.3-1 SIP CO ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	call from USA to Korea, a Korean ring back tone is heard by the caller.		
Add "user=phone" param	For outgoing SIP call, "user=phone" will be added in Request URI of INVITE.	No Yes	No

6.8.4 SIP Registration Status Overview

Selecting SIP Registration Status will display the Registration Overview page. The page displays a list of configured SIP User Ids and the status.

Figure 6.8.4-1 SIP Registration status overview

6.8.5 SIP UID Allocation Status Overview

Selecting SIP User ID Allocation Status will display the allocation overview page. The SIP User Id Allocation Status Overview page displays the station(s) assigned to each SIP User Id index in the Station Common Attributes PGM 111.

Figure 6.8.5-1 SIP UID Allocation Status Overview

6.8.6 SIP User ID Attributes – PGM 126

Selecting SIP User ID Attributes will display the SIP User ID input page. Enter a valid SIP User ID Index Number range, and click **[Load]** to view the SIP User ID Attributes for the first index in the range. Enter new data and click **[Save]** to modify the attributes for the index range.
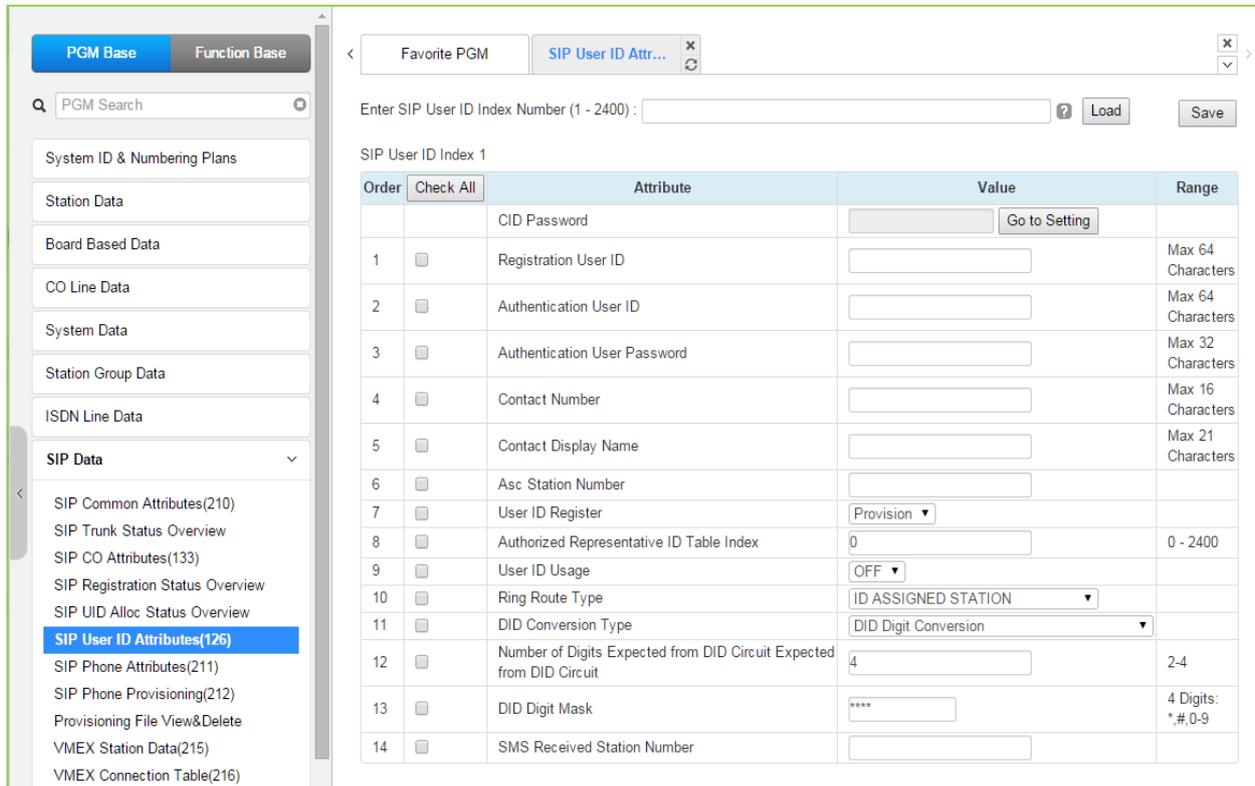
Figure 6.8.6-1 SIP User ID Attributes

For each station, an index to the SIP User Id Attributes Table is defined in PGM 111 Station Attributes. The SIP User Id Attributes Table defines SIP characteristics associated with the index including User ID, Authentication name, etc. These characteristics are required for proper operation of the system and registration of the terminals when employed with SIP trunks. See also PGM 133. Note PGM 126 and PGM 133, which are accessible only via Web Admin, apply to SIP Trunks.

Table 6.8.6-1 SIP User ID ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Registration User ID	This field defines the SIP User Id from by the SIP Service provider. The User Id has the format User ID@Domain. Note the domain is commonly the system IP address.	Max. 64 characters	
Authentication User ID	The SIP Service Provider may require authentication of the user for registration and at other times during call setup. The Authentication name from the provider must be entered for proper SIP registration	Max. 64 characters	
Authentication User Password	The SIP Service Provider may require authentication of the user for registration and at other times during call setup. The Authentication password from the provider must be assigned for proper SIP registration.	Max. 32 characters	

Table 6.8.6-1 SIP User ID ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Contact Number	The Contact header of SIP messages is populated with the specified SIP User Id.	Max. 16 characters	
Contact Display Name	The Display Name specified is used for the Contact header Name field.	Max. 21 characters	
Associative Station Number	Not used.	Station number	

User ID Register	The SIP Trunks are provided in two formats
	    1) Registered: the system must register for service often using Authentication
	    2) Provision: the provider sends all SIP messages to a fixed IP; the system does not register for service.
		Register
		Provision
			Provision

Authorized Representative ID Table Index	The User Id can be assigned an index of another User Id. When the SIP CO is configured to employ the Authorized Representative Id as the SIP "From" or "Contact" header, the indexed User Id is employed.	0-2400	0
User ID Usage	If registration is enabled (User ID Register above) the iPECS can send the User ID or Authorized Representative ID to the SIP Proxy to register the ID. Otherwise, only the Authentication Name and password are used.	OFF ON	OFF
Ring Route Type	Incoming calls from a SIP trunk can be routed 1) to the ID assigned Station (any station with any of the SIP User ID Table Indices in PGM 111 matching the incoming SIP User Id), 2) based on CO/IP Ring assignments (PGM 144), 3) using DID treatment defined below, or 4) follow the MSN Table routing (PGM 145).	ID Assigned Station, Ring Assignment, DID Conversion, MSN-DID Conversion	ID assigned Station
DID Conversion Type	When the Ring Route above is defined as DID, the system will send the call to a destination based on the DID conversion selected here. The digits in the SIP User ID may be used "as is" to identify the desired station, modified based on the DID Digit mask below and routed to the resulting station or the modified DID number can be used as an index to the Flexible DID Conversion Table, PGM 231 to route the call.	DID Digit Conversion, Use 'as is', Modify Using Flexible DID Conversion Table	DID Digit Conversion
Number of Digits (2-4) Expected from DID Circuit	When DID Digit Conversion or Flexible DID Conversion Table routing is used, the number of digits received is defined in this field.	2-4	4
DID Digit Mask	When DID Digit Conversion or Flexible DID Conversion Table routing is used, the digit conversion is defined in this field. For each of the four (4) digits, use "*" to accept any digit, "#" to delete the digit, or a digit 0-9 to replace the digit.	4 Digits: *, #, 0-9	****
SMS Received Station Number	When an SMS is received for this User Id, the Station to receive the SMS must be defined.		

6.8.7 SIP Phone Attributes - PGM 211

Selecting SIP Phone Attributes will display the SIP Phone input page. Enter a valid SIP Station Number or range, and click **[Load]** to view the SIP Phone Attributes for the first Station Number in the range. Use the check boxes to indicate the data to modify for the Station range. Enter new data and click **[Save]** to modify the attributes for the Station Range.

Figure 6.8.7-1 SIP Phone Attributes

< Registration >

For a new registration of SIP station, input ID/PWD & Desired Station Number in PGM 443 of Station User Login Table. This SIP Phone Attributes are for Stations that are already registered to system.

- − Register Mode - Register/ Manual : Set Registration Time Out or Not
- − Registration Status : View connection status (Disconnected or Not) for a station
- − IP Address : SIP Phone's IP address
- − IP Port : SIP Phone's IP Port Number
- − Transport Mode : SIP signaling method
- − SIP Phone Type : Automatically Assigned by System
- − Device NAT Usage : Automatic Detection
- − Registration Timer Usage: OFF – Assign (Re-) Registration Timer by Provisioning

(212), ON – Assign (Re-) Registration Timer by SIP Phone Attributes (211).
- Registration Timer: more than 10 minute recommended.
- 407 Authentication: Authentication of Registration (and Call Setup). To implement authentication, user login Password should be available in PGM 443 for the Station.

< Keep Alive / NAT Resolution >

To keep stable information of SIP Phone's Connection, IP address and Port number that is under NAT environment, system uses 'OPTIONS' message to implement Keep Alive and assist NAT resolution - effort to maintain IP address of SIP Phone by sending message so often from system to SIP Phone. SIP Phone should be capable to answer for 'OPTIONS' message
- Check Message Sending Timer in [SIP Data / SIP Attributes (210)] : 120 seconds
- Keep Alive Usage for a SIP Station in [SIP Data / SIP Phone Attributes (211)] : ON
- Retry Count for a SIP Station in [SIP Data / SIP Phone Attributes (211)] : 3

< System Firewall Resolution >

In case of firewall routed with MPB, to distinguish remote SIP Phone that is outside of firewall from system local area a check bit is required per a SIP Station. With this check bit, system can determine whether to serve communication using firewall mapped WAN IP address of MPB or serve communication using LAN IP address of MPB.
- SIP Phones that are outside of system protect firewall : [SIP Data / SIP Phone Attributes (211)] – 'Same Zone with MPB' to 'OFF'

< Session Timer >

To confirm talk state frequently during in talk state, system sends 'UPDATE' message to SIP Phone. If there is no response for the UPDATE message with in Maximum session timer, system will disconnect the talking call.
- [SIP Data / SIP Phone Attributes (211)] – Session Timer Support : ON
- [SIP Data / SIP Phone Attributes (211)] – Max Session Timer : if exceed, disconnect talking call
- [SIP Data / SIP Phone Attributes (211)] – Min Session Timer: minimum guard timer for session timer negotiation.

< SRTP >

Voice & Video Data Encryption requires synchronization of CRYPTO method between system and SIP Phone side. If system specifies SRTP information then same information should be in SIP Phone side by Phone user programming.

SRTP usage requires a SRTP relay channel via eMG system VOIU and VOIB.
- [SIP Data / SIP Phone Attributes (211)] – SRTP Usage: ON
  →SIP Phone self-programming is required, too – SRTP ON
- [SIP Data / SIP Phone Attributes (211)] – 1st CRYPTO key generation type: one of ARIA_CM_192_HMAC_SHA1_80, AES_CM_128_HMAC_SHA1_80, ARIA_CM_128_HMAC_SHA1_80

→SIP Phone self-programming is required, too – 1st/2nd CRYPTO method
- [SIP Data / SIP Phone Attributes (211)] – 2nd CRYPTO key generation type: one of ARIA_CM_192_HMAC_SHA1_80, AES_CM_128_HMAC_SHA1_80, ARIA_CM_128_HMAC_SHA1_80
  →SIP Phone self-programming is required, too – 1st/2nd CRYPTO method

< DTMF >

1) INFO (OUT BAND) type DTMF

   - Presented in SIP signaling message.

   - INFO (SIMPLE DTMF) / INFO (NORTEL NETWORKS) / INFO (DTMF RELAY) / INFO (TELEPHONE EVENT)

   - Default: INFO (DTMF RELAY)

2) TONE (INBAND) type DTMF

   - Presented in RTP packet

   - Additional VOIU/VOIB' DSP channel is required to detect DTMF in RTP

   - INBAND / 2833

<CO DIAL TONE>

This is to avoid double play of CO dial tone

'Set' if SIP phone plays CO dial tone by itself. If not, there will be another CO dial tone from external.

<Request URI Type>

Some SIP Phone will reject Request-URI if IP and port in domain field is different from its contact IP and port.

'Normal': IP and port number in Request URI domain field will be the real IP and port number of the SIP phone.

'KT-FMC': IP and port number in Request URI domain field will be system IP and port

<Busy Serve>

System Busy Tone: there will be a 'busy-tone' on busy state that is presented by system.

Additional VOIB/VOIU DSP channel is required.

486 Busy Message: there will be '486 busy' SIP signaling response on busy state.

<Call Initiation Mode>

For a SIP station, system can establish multiple or single call sessions. With this option system can control 'call-wait' option in system side. Normally, the SIP Phone has its 'call-wait ON/OFF' option by itself.

Options are as below:

'Multiple': for a new additional call to SIP Phone, system initiates every call for the SIP

Phone regarding it is on idle state. The 'call-wait allow/deny' is decided by SIP Phone itself.

'Single': system initiates only one call for a SIP Phone. The 'call-wait' is denied by system side. So, a call to a busy SIP station will be implemented on a busy state call-control.

Table 6.8.7-1 SIP PHONE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Registration Mode	Initial registration of a SIP phone is accomplished employing the parameters set in Station User Login PGM 443. The SIP phone can be required to register with the system periodically based on the Registration Timer or the registration can be maintained without the need for the SIP phone to reregister with the system (Manual).	Manual Register	Register
Registration Status	The system will display the status of the SIP phone registration.		
IP address	The system will display the IP address of the registered SIP phone.		
IP Port	The system will display the IP port used for the registered SIP phone.		
Transport Mode	The system will display the IP transport used by the registered SIP phone for signaling messages (UDP, TCP or TLS). In case of TLS, the options configured in the SIP Common Attributes (PGM 210) apply.		UDP
System SIP Port	The system will display the system SIP Port.		
SIP Phone Type	The type of SIP phone is generally determined by the system and can be an Ericsson-LG standard SIP phones or the 3rd party SIP phone type.		3rd SIP
Device register mode	The Register Mode determines if the SIP phone is behind a NAT server. When set in the Auto mode, the system will determine if the phone is behind a NAT server.	AUTO, NO NAT, NAT	AUTO
Registration Timer usage	When the Registration Mode is "Register", the phone must register with the system periodically. The timer that determines the period can be the Registration Timer below (ON) or, when OFF, the timer is assigned through provisioning (PGM212).	OFF ON	OFF
Registration Timer	When the Registration Timer is enabled above, the system informs the SIP phone that registration is required at intervals of this Registration Timer. If the phone does not register within the timer, the phone is placed in an Out-of-Service state until the phone registers. Note shorter times will increase LAN traffic.	30-3600	3600
Keep Alive Usage	The system will periodically send an Option message to assure a connection with the registered SIP phone. Note when separated by a NAT server, Keep-Alive should be employed to maintain the NAT table record. The Check Message Sending Timer in PGM 210 determines the frequency.	OFF ON	OFF
Retry Count	The system periodically sends an Option message to assure a connection with the registered SIP phone. If	3-10	3

Table 6.8.7-1 SIP PHONE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	there is no response, the system sends additional Option messages, and, after the retry count, considers the SIP phone Out-of-Service.		
407 Authentication	The system can challenge the SIP phone during registration and outgoing call set-up with a SIP 407 Authentication message requiring the SIP phone to provide the Authentication Id and password assigned in the Station Login PGM 443.	OFF ON	ON
181 Being Forwarded	N/A	OFF ON	OFF
100 rel	N/A	OFF ON	OFF
Session Timer Support	During a SIP call or "session", there are no signaling packets sent or received from the SIP Call server. In order to assure a session is still active, the system can periodically send an Update message that the SIP Call server should acknowledge.	OFF ON	OFF
Max Session Timer	When "Session Timer Support" is enabled, the system will verify the session at this timer interval.	180-3600	1800
Min Session Timer	During negotiation with "Session Timer Support" enabled, the system will use this value as the minimum Session expiration timer and will not respond to a SIP Option message prior to expiration of this timer.	60-150	90
Within same firewall with UCP	For a remote SIP phone, the system must communicate with the phone employing the system's "Firewall" address (OFF). Otherwise, the system employs the LAN address to communicate with the SIP phone.	OFF ON	ON
SRTP Usage	When supported by the SIP phone, the system can encrypt media (RTP) packets employing SRTP (Secure Real-Time Protocol).	OFF ON	OFF
1st CRYPTO	The first priority cryptographic method for SRTP is selected from the below: • None • ARIA_CM_192_HMAC_SHA1_80 • AES_CM_128_HMAC_SHA1_80 • ARIA_CM_128_HMAC_SHA1_80 • AES_CM_192_HMAC_SHA1_80 • ARIA_CM_256_HMAC_SHA1_80 • AES_CM_256_HMAC_SHA1_80		None
2nd CRYPTO	The second priority cryptographic method for SRTP is selected from the below: • None • ARIA_CM_192_HMAC_SHA1_80 • AES_CM_128_HMAC_SHA1_80 • ARIA_CM_128_HMAC_SHA1_80 • AES_CM_192_HMAC_SHA1_80 • ARIA_CM_256_HMAC_SHA1_80 • AES_CM_256_HMAC_SHA1_80		None
DTMF Type	DTMF dialing signals sent by the SIP phone must be	INBAND,	INFO(DTMF

Table 6.8.7-1 SIP PHONE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	defined for the system to detect the tones properly. For Inband DTMF, a VoIP channel is required.	2833, INFO(SIMPLE DTMF), INFO(NORTEL NETWORKS), INFO(DTMF RELAY), INFO(TELEPHONE EVENT)	RELAY)
SMS Type	The Short Message Service Protocol (type) must be selected to support SMS.	AUTO, Text/plan, Text/plan(KR), Xnipm+xml	AUTO
CO Dial Tone	When the user of a SIP phone dials a CO/IP Line access code with Enblock dialing, the system can provide virtual dial tone to the user.	OFF ON	OFF
MWI NOTIFY	For compatible SIP phones, the system supports SIP Subscribe/Notify. When enabled here, the system sends Message Waiting notifications to the SIP phone.	OFF message-summary	OFF
Request URI Type	The SIP Request header Domain field can use the SIP phone's IP address and port (Normal) or for 'KT-FMC' the Request URI Domain field will be system IP and port.	Normal KT FMC	Normal
Busy Serve	To indicate a busy condition to the SIP phone, the system can provide RTP packets with busy tone or the SIP 486 Busy message. Providing busy tone requires a VoIP DSP channel in the system.	System Busy Tone, 486 Busy Message	System Busy Tone
Call Initiation Mode	The system can route calls to the SIP phone while busy (Multiple). In this case, the SIP phone determines if Call Waiting is supported. Otherwise, if the SIP phone is busy, the system routes calls based on the busy treatment (Single).	Multiple, Single	Multiple
Pre Audio Connection For DTMF	The system normally provides the 183 Session Progress SIP message to establish a "Pre-audio" connection. The "Pre-audio" connection permits the system to send tones (CO dial tone or ringback tone) to the SIP phone. In addition, the SIP phone can send DTMF tones the user dials in response to CO dial tone or a remote IVR message. Some SIP phones may require the 200 OK message, which "answers" the call, to allow dialing after the call has been initiated.	183 Session Progress, 200 OK	183 Session Progress
Do Not Overwrite Station Name	This feature can be set in case of the following Range: - OFF(REG): Extension name will be updated with displayed in REGISTER message. - ON: Extension name will not be updated. - OFF(INV): Extension name will be updated with displayed in INVITE message. - OFF(REG+INV): Extension name will be updated with displayed in REGISTER or INVITE message.	OFF(REG), ON, OFF(INV), OFF(REG+INV)	OFF(REG)
Follow CO Enblock Process	Skip "dial tone by system" phase and deliver transparent message of CO trunk to SIP phone.	OFF ON	OFF

Table 6.8.7-1 SIP PHONE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Suffix DID Tbl to CLI	If this option is ON, CLI is published by suffixing DID Conversion table index.	OFF ON	OFF

6.8.8 SIP Phone Provisioning - PGM 212

Selecting SIP Phone Provisioning will display the SIP Phone Provisioning Data input page. In the CONFTYPE, select the Ericsson-LG Enterprise SIP Phone type to view/configure the Provisioning data. To view/configure a phone specific configuration file, enter the SIP phone MAC in the Private MAC field. After modifying data, press **[Save]** button to save the configuration.

To store provisioning files that have been uploaded to the iPECS UCP, press **[Store uploaded Provision files]**. To view the stored configuration files, press **[View Provision files]**. To download Provision files, press **[Download Provision files]**. To view TLS Cert. files, press **[View TLS Cert files]** button.
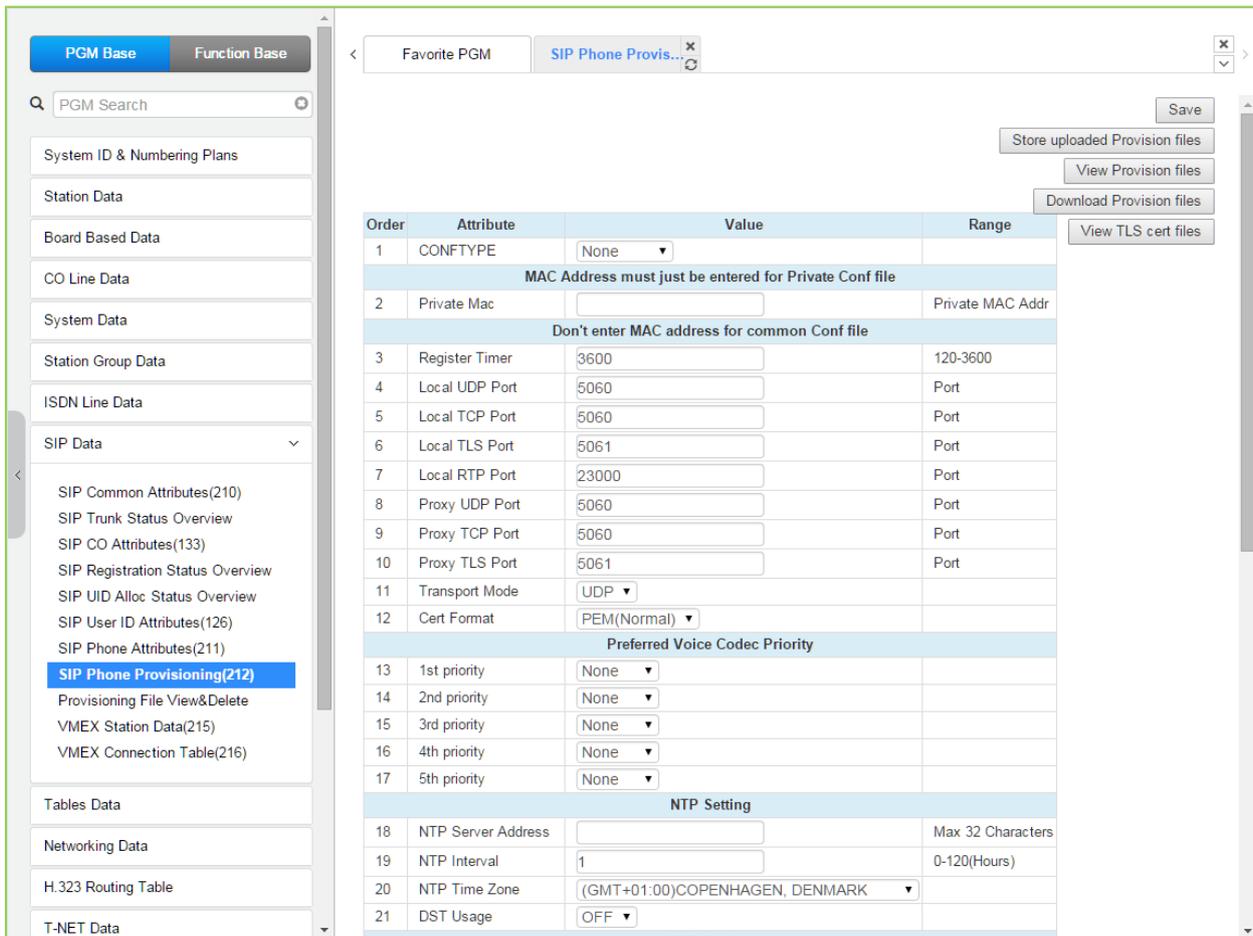
Figure 6.8.8-1 SIP Phone Provisioning

The IP8800 series SIP phones from Ericsson-LG Enterprise may be configured automatically by the system. The configuration files are compacted using the Linux/Unix Tar command to create a sip_conf.tar file. The file is then uploaded to the iPECS UCP File system using File Upload under S/W upgrade in the Maintenance Web services. Once uploaded to the File System, the configuration files are unpacked and stored in the provisioning file directory using the [Store] button.

Table 6.8.8-1 SIP PHONE PROVISIONING

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CONFTYPE	To modify or view a Configuration file, from the drop-down, select an Ericsson-LG Enterprise SIP Phone type. The WIT-400HE is for future use.		None
MAC address must just be entered for Private Conf file			
Private MAC	There are two types of Config files, a global file for all phones of a specific type and a phone specific file. To view or modify a Phone specific file, enter the MAC address of the SIP phone.	Private MAC address	
Don't enter MAC address for common Conf file			
Registration Timer	When the SIP User ID Attributes are configured with Registration Mode=Register and the Registration Timer Usage=OFF, this timer is used to determine the valid registration period.	120-3600	3600
Local UDP Port	When employing UDP transport, this port number is employed for messages from the SIP phone.	Port	5060
Local TCP Port	When employing TCP transport, this port number is employed for messages from the SIP phone.	Port	5060
Local TLS Port	When employing TLS transport, this port number is employed for messages from the SIP phone.	Port	5061
Local RTP Port	The system sends RTP packets to the SIP phone using the first port available starting with this port number.	Port	23000
Proxy UDP Port	The SIP phone will send messages to the UCP system over this SIP Server port.	Port	5060
Proxy TCP Port	The SIP phone will send messages to the UCP system over this SIP Server port.	Port	5060
Proxy TLS Port	The SIP phone will send messages to the UCP system over this SIP Server port.	Port	5061
Transport Mode	The IP packet transport mode used between the system and SIP phone for signaling messages is defined as UDP, TCP or TLS. In case of TLS, the options configured in the SIP Common Attributes (PGM 210) apply.	UPD TCP TLS	UDP
Cert Format	The UCP system supports two certification formats Privacy-Enhanced Electronic Mail (PEM) or Distinguished Encoding Rules (DER).	PEM(Normal), DER (Normal)	PEM(Normal)
Preferred Voice CODEC PRIO			
1st priority ~ 5th priority	When the system negotiates with the SIP phone, the Codec priority specified is employed. If only the 1st. priority Codec is defined, only that Codec is available from the phone. If the Codec is not supported by the remote party, the call will fail.	G.711-u G.711-a G.723.1 G-729 G.722	None
NTP Setting (Need for TLS)			
NTP Server Address	The IP-8800 series phones employ NTP to determine the time of day. The NTP server id must be configured for the phone.	Max. 32 Characters	
NTP Interval	The phone periodically checks the NTP server to determine the proper time of day.	0-120 (Hours)	1
NTP Time Zone	The phone must be configured with the local Time Zone for proper operation and time display.	GMT	
DST Usage	The phone must be configured to support Daylight Saving	OFF	OFF

Table 6.8.8-1 SIP PHONE PROVISIONING

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	Time.	ON	
DSP Setting			
Speaker Volume	The default volume level of SIP phone speaker can be set through the Config file.	1-11, 1-7 (LIP-8850)	6
Handset Volume	The default volume level of SIP phone handset can be set through the Config file.	1-11, 1-7 (LIP-8850)	6
Headset Volume	The default volume level of SIP phone headset can be set through the Config file.	1-11, 1-7 (LIP-8850)	6
Max Handset Volume	The default maximum volume level of SIP phone handset can be set through the Config file.	1-11, 1-7 (LIP8850)	11
Digit Map			
Dial Tone Digit	A digit string can be defined in the Config file so that the SIP phone will provide a "Second Dial Tone" to the user.	Max. 256 Digits	
Pause Timer	Dial Pause timer in SIP Phone.	01-20 (Sec.)	3
Digit Map	Send setup to system numbering plan.	Max. 1000	
Emergency Code	The Emergency dial code for the SIP phone can be set through the Config file.	Max. 1000	
System Setting			
Feature Sync.	With Do Not Disturb and Call Forward feature synchronization ON, when SIP Phone changes the DND or Call Forward state, the system is informed of the change in status.	ON OFF	ON
Auto Idle Timer	Phone goes to idle after this timer when the phone receives disconnect message or signal from system.	00-99 (Sec.)	5
Check Domain	The IP8800 SIP Phones can be configured to verify the domain in SIP messages match the registered proxy server.	OFF ON	ON
Telnet Usage	The IP8800 series phones can support Tenet access to the CLI.	Enable Disable	Disable
WEB Usage	The IP8800 series phones incorporate a Web server for access to the phone configuration files.	Enable Disable	Disable
Crypt Mode	The system provisioning cryptosystem is configured to employ either RSA or ECC (Elliptic Curve Cryptography)..	RSA ECC	RSA
Save button: save provisioning for the specified common CONFTYPE or specific SIP Extension with Private MAC. Store uploaded Provision files button: Store the uploaded provision files. View Provision files button: display all of saved provisioning information. Download Provision files button: download the provision files. View TLS Cert files button: display all of saved TLS Cert files information.			

6.8.9 Provisioning File View & Delete

Selecting Provisioning File View & Delete displays a list of all the Provisioning files stored in the provisioning files directory as shown in Figure 6.8.9-1. Note this does not included files that have been uploaded to the iPECS UCP file system but not "stored" under PGM 212. To delete a file from the provisioning file directory, check the box in front of the file to be deleted then click **[Delete]**.
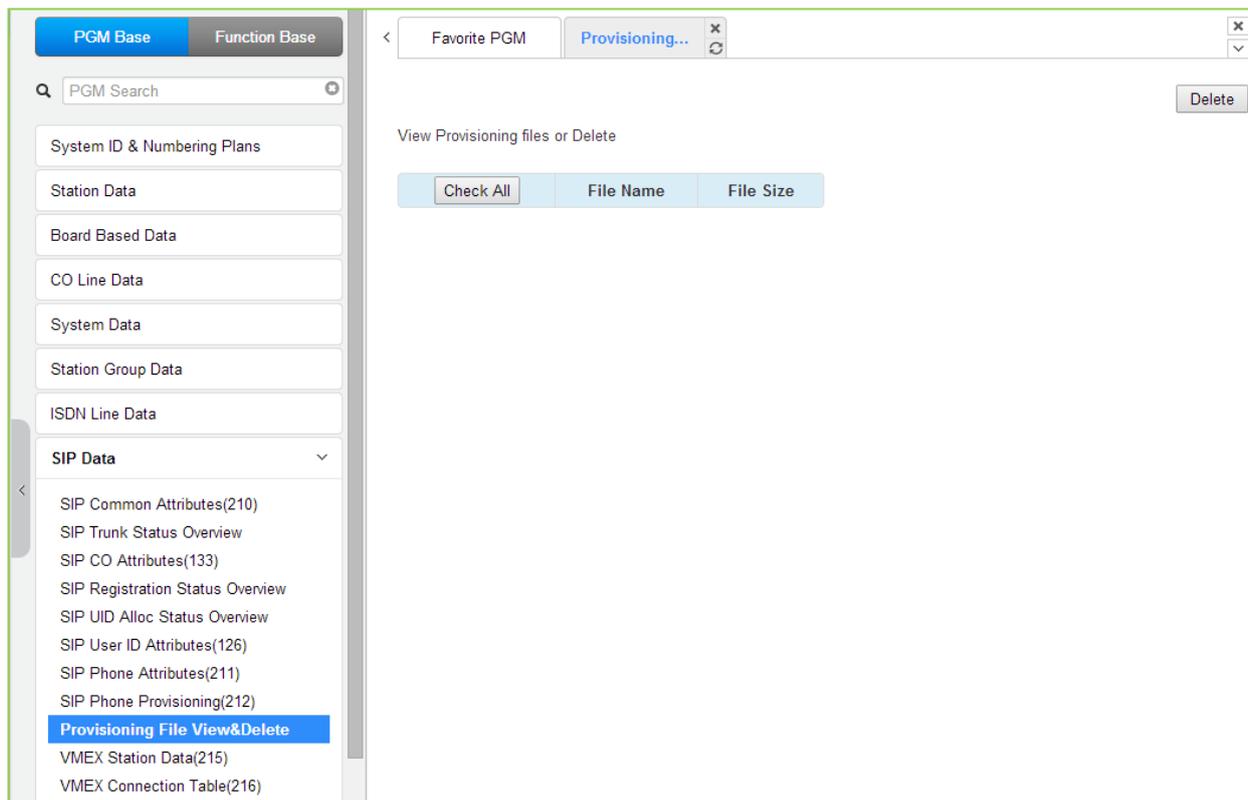
Figure 6.8.9-1 Provisioning File View & Delete

6.8.10 VMEX Station Data - PGM 215

Selecting VMEX Station data will display the VMEX Station Data input page. Enter a range, and click **[Load]** to view the VMEX Data for all stations in the range. Use the check boxes to indicate the VMEX Station Data to modify. After modifying the data, click **[Save]** to store the VMEX Station Data.
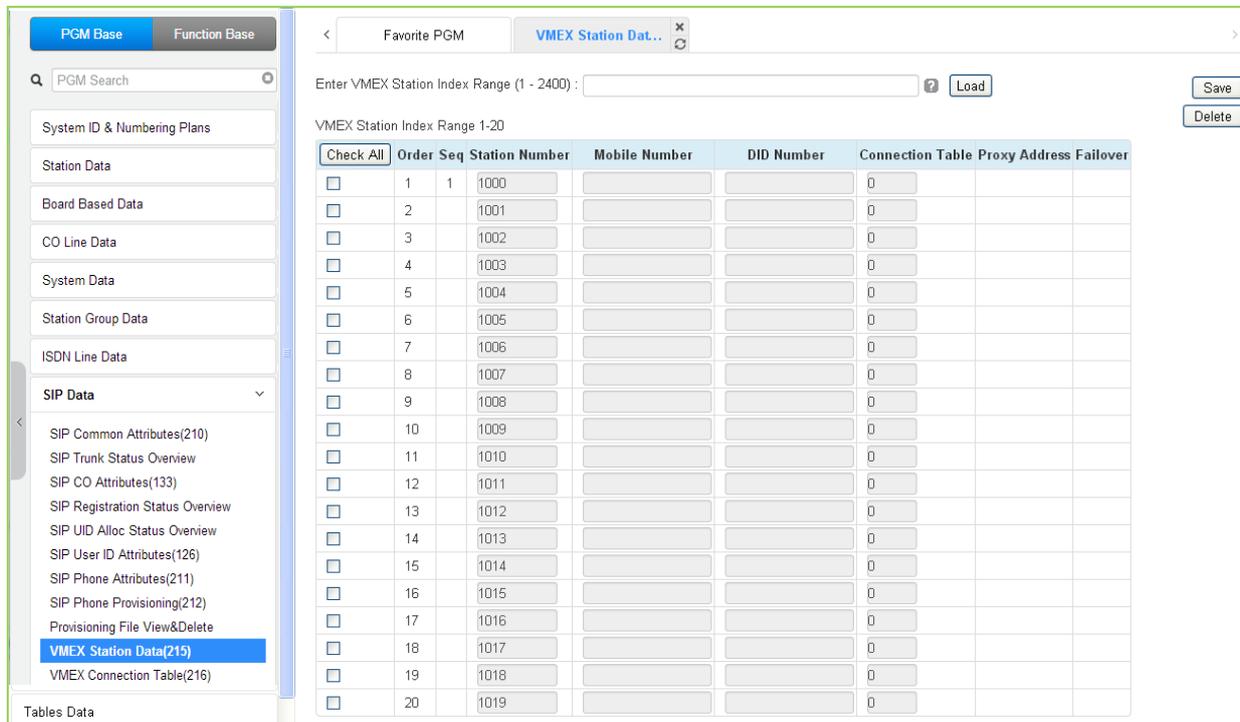
Figure 6.8.10-1 VMEX Station Data

The Virtual Mobile Extension feature permits a remote mobile phone to act as a station within the system employing SIP. For proper operation, various parameters as shown in Table 6.8.10-1 must be configured for the Station data.

Table 6.8.10-1 VMEX Station Data

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Station Number	The station number of the VMEX can be modified. The VMEX station number must not conflict with other numbering plans. The station number can be modified here or in PGM 105 and is affected by the 8 Digit Table	Station Number	
Mobile Number	The Caller Id of the Mobile phone must be entered to allow the system to recognize the VMEX phone. The mobile number here must match the number in the SIP From header.		
DID Number	The DID number associated with the VMEX must be configured to allow for recognition of calls to the VMEX.		
Connection Table	An index to the Connection Table PGM 216 must be entered. One of ten indices can be assigned permitting the use of multiple SIP Call servers.	0 ~ 10	
Proxy Address	The SIP Proxy for the VMEX must be configured for access to		

Table 6.8.10-1 VMEX Station Data

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	the VMEX.		
Fail-over	The index (Station Number) of the Mobile Extension Table (PGM 236) can be assigned for Fail-over operation. In this case, the Table entry must be configured for fail-over operation.	Station Number	

6.8.11 VMEX Connection Table - PGM 216

Selecting VMEX Connection table will display the VMEX Connection Data input page. Use the check boxes to indicate the Table index data to modify.
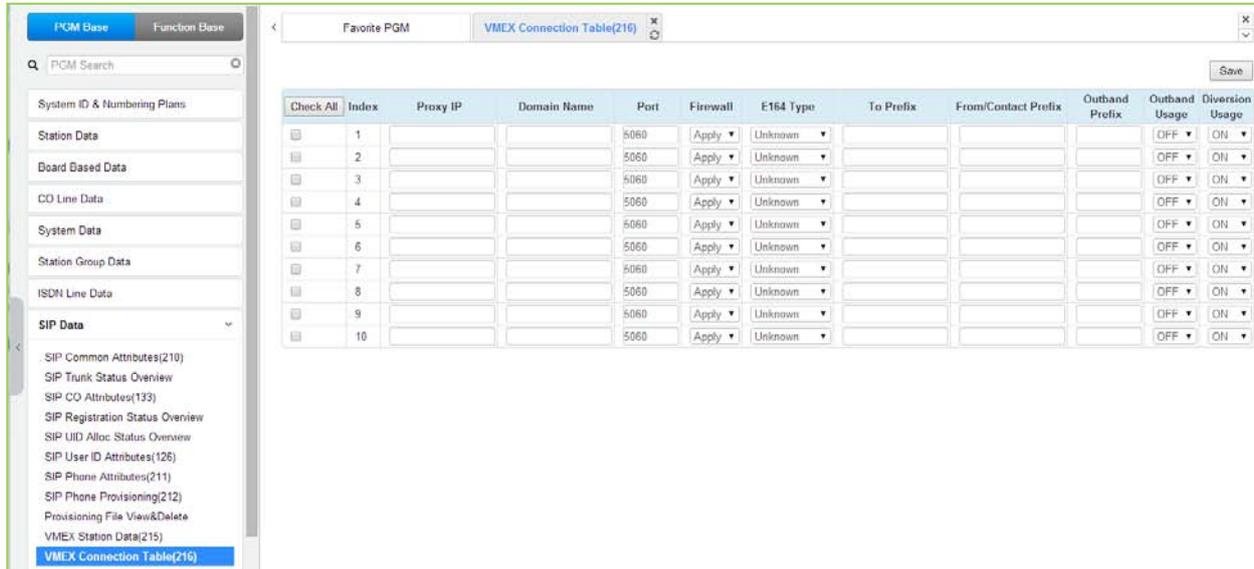
Figure 6.8.11-1 VMEX Connection Table

With VMEX, the carrier's Mobile network sends calls to and from the Mobile phone directly to the system employing a SIP Call server. For proper operation, the SIP Call server must be configured and special prefix codes must be defined for calls from and to the mobile phone.

Table 6.8.10-1 VMEX Connection Table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Proxy IP	This field assigns the IP address or URL of the Service Provider's SIP Proxy server.		
Domain Name	Domain name of the Service Provider's SIP Call server that is used in SIP "To:" headers.		
Port	Normally SIP signaling messages are sent on port 5060. If desired a different port can be defined.		5060
Firewall	The iPECS UCP can use the Firewall address or LAN address of the UCP module when communicating with the VMEX.	Not Apply	Apply
E164	The structure of the received SIP invite To header from the VMEX can be defined for proper conversion. This field is not presently used.	Unknown International National Local	Unknown
To Prefix	A special prefix code is included in the SIP invite "To" header sent from the iPECS UCP to the carrier's SIP Call server to identify VMEX service	3 to 5 digits	
From Contact Prefix	A special prefix code is included in the SIP invite "From" and "Contact" header sent from the iPECS UCP to the carrier's SIP Call server to identify VMEX service	3 to 5 digits	
Outbound Prefix	A special prefix code that must be in the SIP invite "To" header received by the iPECS UCP for calls from the VMEX. If the prefix received does not match this entry, the call is rejected.	8 digits	

Table 6.8.10-1 VMEX Connection Table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Outbound Usage	Not presently used.	OFF ON	OFF
Diversion Usage	Not used	OFF ON	ON

6.9 Tables Data

Selecting the Tables Data group returns the sub-menu displayed in the left frame as shown in the following figure.
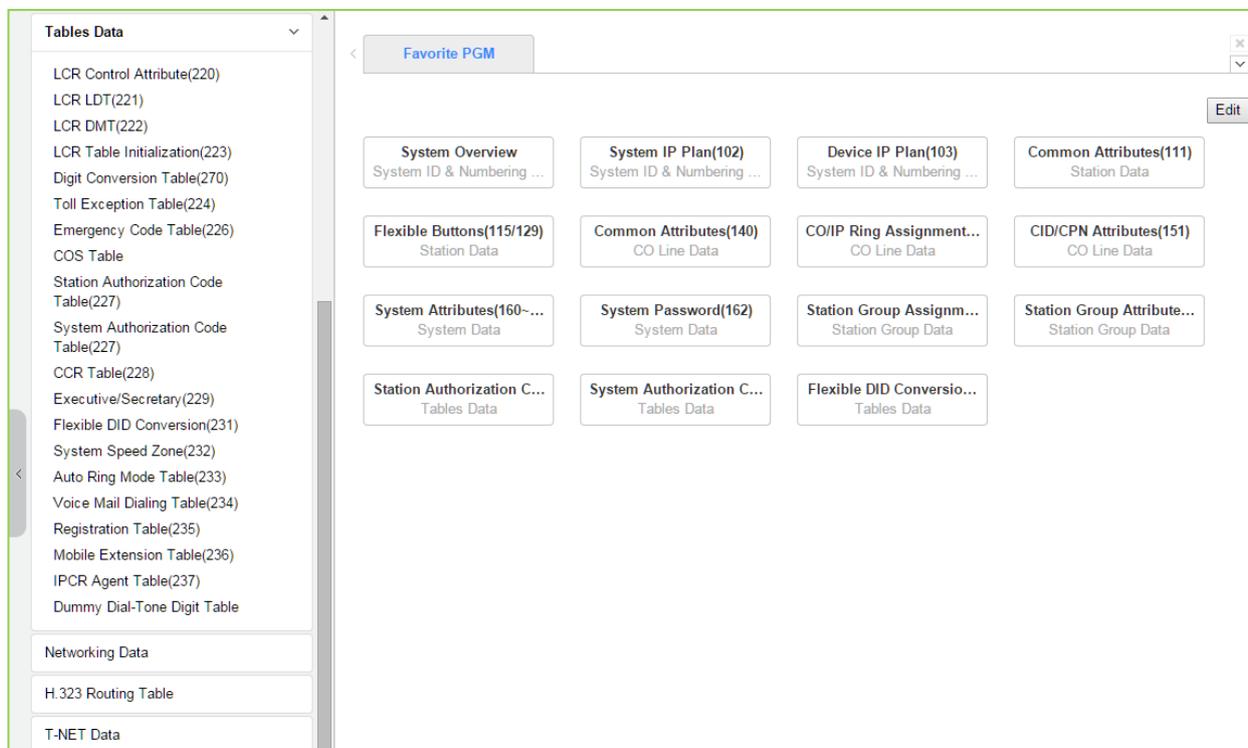
Figure 6.9-1 Tables Data

6.9.1 LCR Control Attributes - PGM 220

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. Click **[Save]** button after changing Value.

Figure 6.9.1-1 LCR Control Attributes

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/IP Group code (8xx), or presses a Loop button)
- Loop and Internal (user dials digits without a CO/IP 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 that the Zone is active is defined (Time Zones). Table 6.9.1-1 provides general descriptive information and input ranges.

Table 6.9.1-1 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.	Disable, Only Loop LCR, Internal and Loop LCR, Loop and Direct CO LCR, Internal, Loop and Direct CO LCR, Internal, Loop, Direct CO and Direct Loop LCR	Disable
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.	Zone1~3	Zone 1
Time Zone1	This entry defines the hours of the day during which each Zone is active for Time Zone 1. 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 each Zone 2 is active for Time Zone 2. Note hours not defined in Time Zone 2 and 3 are automatically part of Time Zone 1.	00~24	00~24
Time Zone3	This entry defines the hours of the day during which Zone 3 is active for Time Zone 3. Note hours not defined in Time Zone 2 and 3 are automatically part of Time Zone 1.	00~24	00~24

6.9.2 LCR LDT (Leading Digit Table) - PGM 221

Selecting LCR-LDT (Leading Digit Table) will display the LCR-LDT data entry page. Select the LDT Index (1 – 32) and the Index range in the LDT Table then click **[Load]** to edit the LDT Table data. Click **[Save]** button after changing Value.
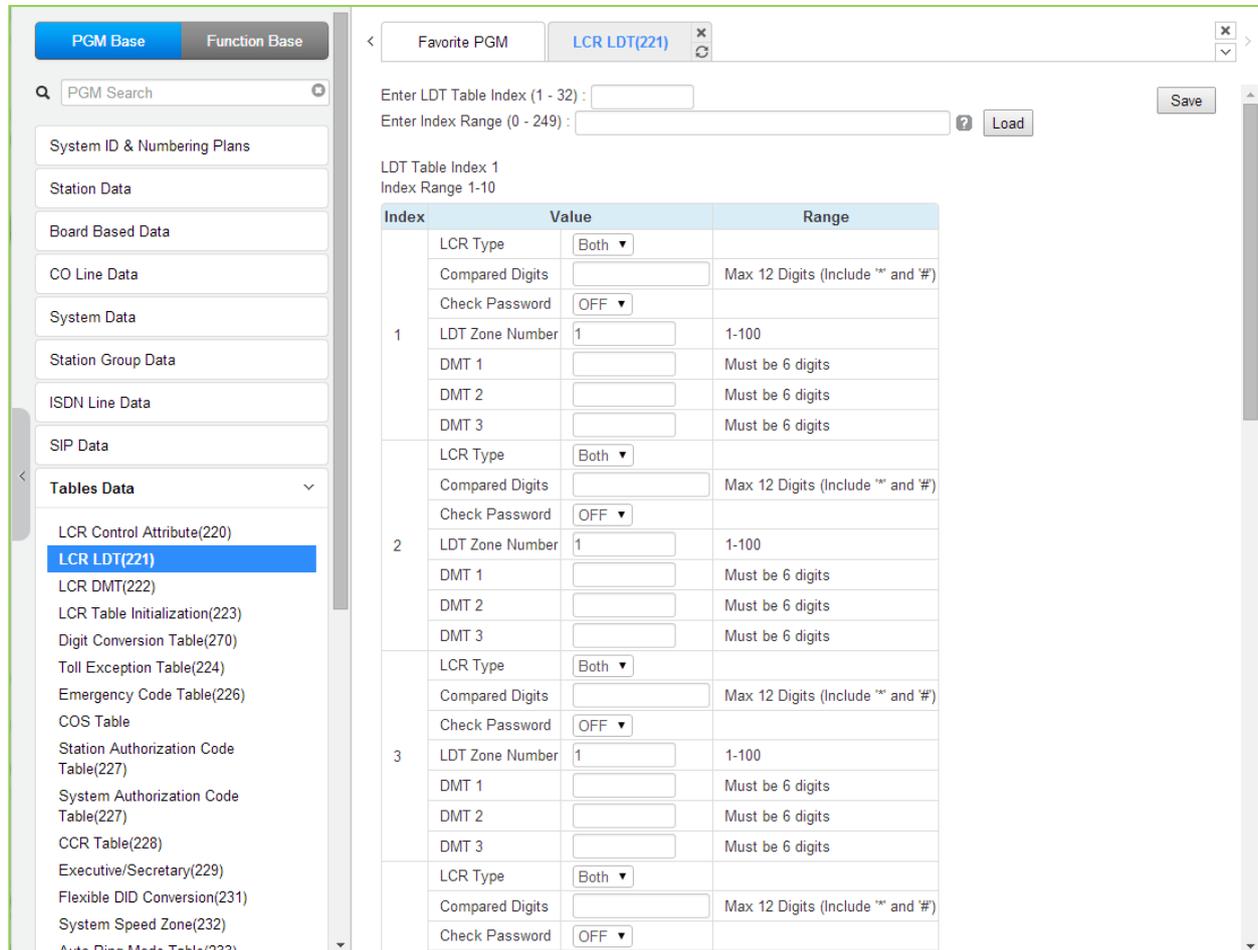
Figure 6.9.2-1 LCR Leading Digit Table

The Leading Digits Table is used to analyze the user-dialed digits to determine an appropriate Digit Modification Table Index. There are 32 LDT Tables each with up to 249 entries. The applicable LCR Access Modes (LCR Type) and 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 of the three (3) Zones and Time Zones configured in the LCR Control Attributes. Table 6.9.2-1 provides a brief description and entries for the Leading Digit Table.

Table 6.9.2-1 LCR LEADING DIGITS

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
LCR Type	This entry defines the LCR modes that will apply to this LDT index.	Internal, CO Line, Both	Both
Compared Digits	Up to 12 digits that, if matched by the user dialed digits, will	Max. 12 digits	

Table 6.9.2-1 LCR LEADING DIGITS

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	access the DMT Indices of the associated Leading Digit Table bin.	(Include * and #)	
Check Password	If enabled (ON), when the dialed digits match the LDT digits, the system will send second dial tone to request the user input a valid Authorization code.	OFF ON	OFF
LDT Zone Number	If the LDT Zone Number of a station/co line is equal to this value, this LDT table is available to the station/co line. (Station Data->Common Attributes(111), CO Line Data->Common Attributes(140))	1-100	1
DMT1	This entry defines the Digit Modification Table index (00~99) for Day Zone 1. One entry (DMT index) is made for each Time Zone (1~3), six (6) digits. The appropriate index is selected for the current Day and Time Zone.	Must be 6 digits 3 DMT indices	
DMT2	This entry defines the Digit Modification Table index (00~99) for Day Zone 2. One entry (DMT index) is made for each Time Zone (1~3), six (6) digits. The appropriate index is selected for the current Day and Time Zone.	Must be 6 digits 3 DMT indices	
DMT3	This entry defines the Digit Modification Table index (00~99) for Day Zone 3. One entry (DMT index) is made for each Time Zone (1~3), six (6) digits. The appropriate index is selected for the current Day and Time Zone.	Must be 6 digits 3 DMT indices	

6.9.3 LCR DMT (Digit Modification Table) - PGM 222

Selecting LCR DMT (Digit Modification Table) will display the LCR-DMT data entry page. Enter the DMT Table Index range then click **[Load]** to modify the DMT data. Click **[Save]** button after changing Value.

Figure 6.9.3-1 LCR Digit Modification Table

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/IP 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/IP path assigned. If the assigned path is not available, the "Alternate DMT index" is used to determine the number and CO/IP path to be used.

Table 6.9.3-1 LCR DIGIT MODIFICATION

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	Max. 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
CO/IP Group	This entry defines the CO/IP Group that the system will attempt to use for the call.	01-21	1
Alternative DMT Index	This entry defines an Alternate Digit Modification Table Index to use if no path is available in the assigned CO/IP Group.	00~99	
Networking Number Plan Bin	This entry defines the Net Number Plan Table bin that the system will attempt to use for a transit out call.	001-251	
SMDR Code	This code is sent to the CM of a T-Net to notify the CM of the local SMDR code required for the CM to complete a call.	Max. 4 digits (Include * and #)	

6.9.4 LCR Table Initialization - PGM 223

Selecting LCR Table Initialization will display the LCR Table Initialization data entry page. Use the check boxes to identify the data items to initialize then click **[Initialize]**.
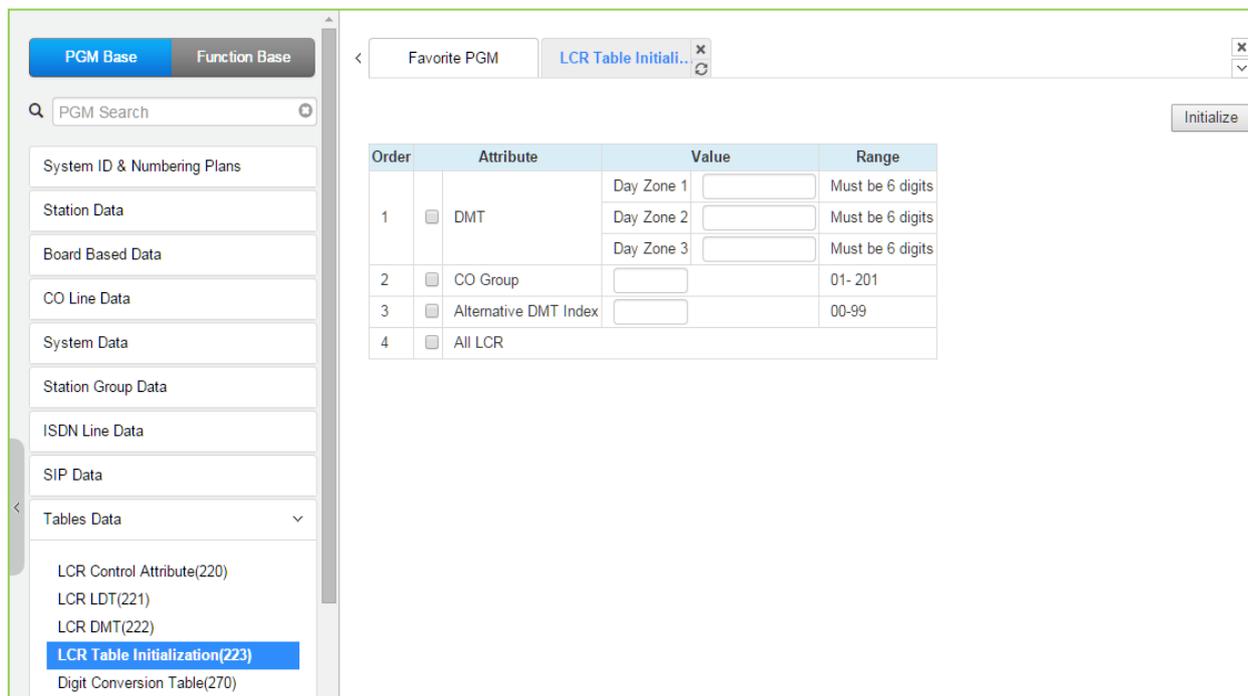
Figure 6.9.4-1 LCR Table Initialization

The LCR Table Initialization allows global values to be assigned to the Digit Modification Table entries for each Day Zone, the CO Group and DMT. In addition, all LCR data can be initialized to the default (global) values.

6.9.5 Digit Conversion Table - PGM 270

Selecting Digit Conversion Table will display the following page. Enter the Table number (1 through 100), and the index of the Table then click **[Load]** modify the Digit Conversion data. Click **[Save]** button after changing Value.
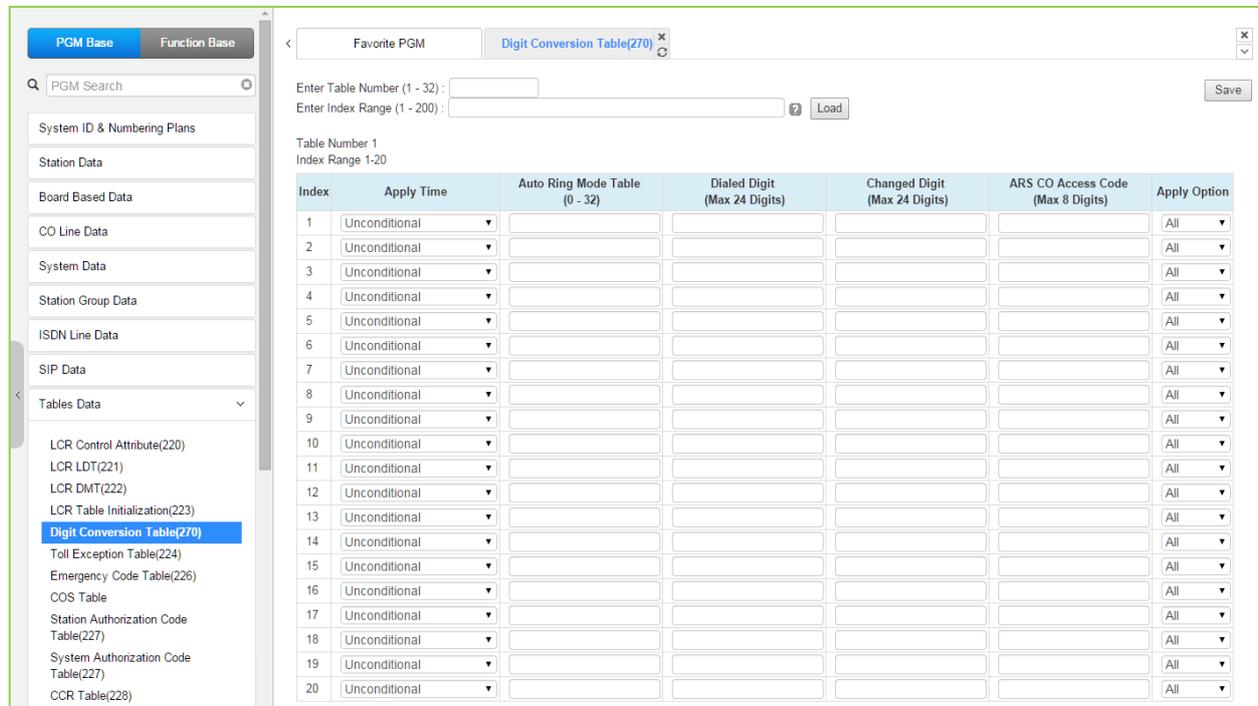
Figure 6.9.5-1 Digit Conversion Table

A Digit Conversion Table is assigned to Stations and CO/IP lines. Digit conversion can be applied according to the Apply Time Type (Unconditional, Day, Night, Timed, or LCR Time) as necessary. Each Table has 200 entries of up to 24 digits. Entries in the Tables can be any digit (0-9), "*", "#","X" (Mask Digit), "F" (Ignore digit). Digit conversion applies to the types of calls assigned for the Apply Option.

Table 6.9.5-1 Digit Conversion table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Apply time	Digit conversion from this entry is applied for the period indicated by this setting. In addition to the individual Day/Night/Timed mode, LCR and Unconditional settings are available.	Unconditional, Follow Day/Night/Timed, Follow LCR time	Unconditional
Auto Ring Mode table	Digit conversion applies to calls which ring based on the specific Auto Ring Mode table selected.	N/A, 00-32	N/A
Dialed digit	Digit conversion applies to the specific user-dialed digits.	Max. 24 Digits	
Changed digit	Digits are converted to the assigned digit string.	Max. 24 Digits	
ARS CO Access Code	If a selected path is not available for some reason (All Busy, Line Fault etc.) after digit	Max. 8 Digits	

Table 6.9.5-1 Digit Conversion table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	conversion, Alternative Route Selection (ARS) will connect calls using predefined designated ARS digit in digit conversion table.		
Apply option	Digit conversion applies to the types of calls defined in the Apply Option. The Apply Option is reserved for future use.	All Reserved CO line Disable	All

6.9.6 Toll Exception Table - PGM 224

Selecting Toll Exception Table will display the Toll Table data entry page. Select the desired Allow or Deny Table then click **[Load]** to modify the data. Click **[Save]** button after changing Value.
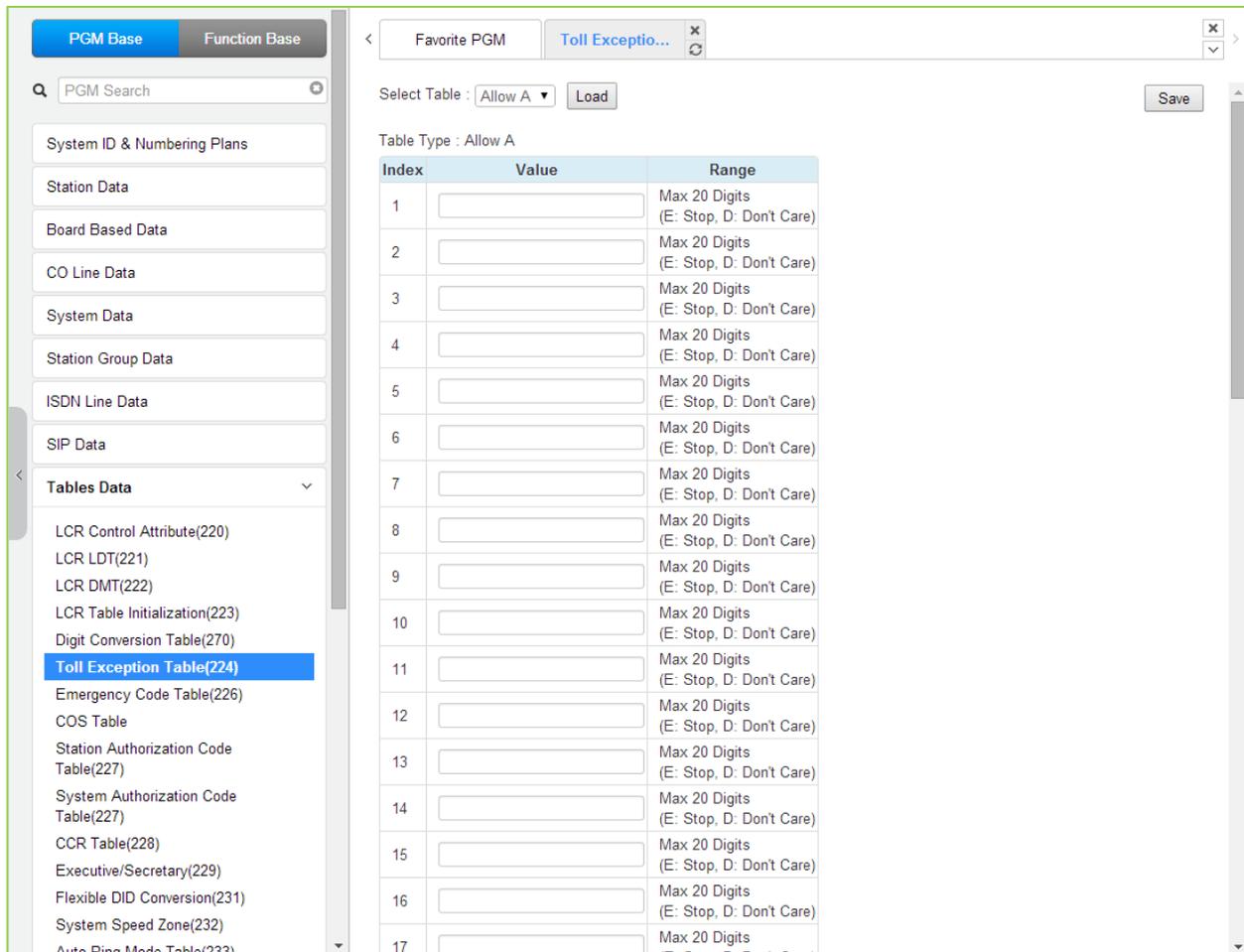
Figure 6.9.6-1 Toll Exception Table

There are ten Toll restriction Tables arranged in pairs. Each pair consists of an Allow Table and a Deny Table. Allow and Deny entries for Table `A' apply to Station and DISA Class of Service 2/4 and 11. Allow and Deny entries for Table `B' apply to Station and DISA Class of Service 3/4 and 11. Allow and Deny entries for Table `C' apply to Station and DISA Class of Service 5 and 6. Allow and Deny entries for Table `D' apply to Station and DISA Class of Service 8/10 and 11. Allow and Deny entries for Table `E' apply to Station and DISA Class of Service 9 to 11.

The Allow and Deny Tables each permit up to 50 entries of up to 20 digits. Entries in the Tables can be any digit (0-9), "#" as a wild card (don't care) digit, or "*" as an end of entry digit. Based on Table entries, stations or DISA users are allowed or denied dialing specified numbers. The following rules apply to establishing restrictions based on the Table entries:

- If the appropriate Allow/Deny Table pair has no entries and COS is 2 to 4, no restrictions are applied. If the COS is 5 or 6, no Long Distance dialing is allowed.

- 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.

- If there are both Allow and Deny entries, the Deny entries are searched. If the dialed number matches a Deny entry, and it is not part of an Allow entry the call is restricted; if no match is found the call is allowed.   For example, if 2223 is assigned in the Allow Table and 222 is assigned in the Deny Table numbers starting with 222 are denied except for 2223.

6.9.7 Emergency Code Table - PGM 226

Selecting Emergency Code Table will display the Emergency Code Table data entry page. Click **[Save]** button after changing Value.

Figure 6.9.7-1 Emergency Code Table

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.

6.9.8 COS Table

Selecting COS Table will display the COS Table data entry page. Data can be filtered to display specific modes and the values to change can be limited to a specific mode and type, Station or System Authorization codes.

Figure 6.9.8-1 COS Table

Assignments for COS are made for the Day, Night, and Timed Ring for each station and system Authorization code. The standard Station COS 1 to 11 is assigned to each Authorization code. Note the COS may also be defined in the Station and System Authorization Code Tables.

6.9.9 Authorization Codes Table - PGM 227

There are two Authorization Code Tables, the System Authorization Code Table and Station Authorization Code Table. Selecting Station Authorization Code Table will display the data entry page in Figure 6.9.9-1 and selecting System Authorization Code Table will display the page in Figure 6.9.9-2. Enter a Station or System Index range and click **[Load]** to modify the data. Click **[Save]** button after changing Value.
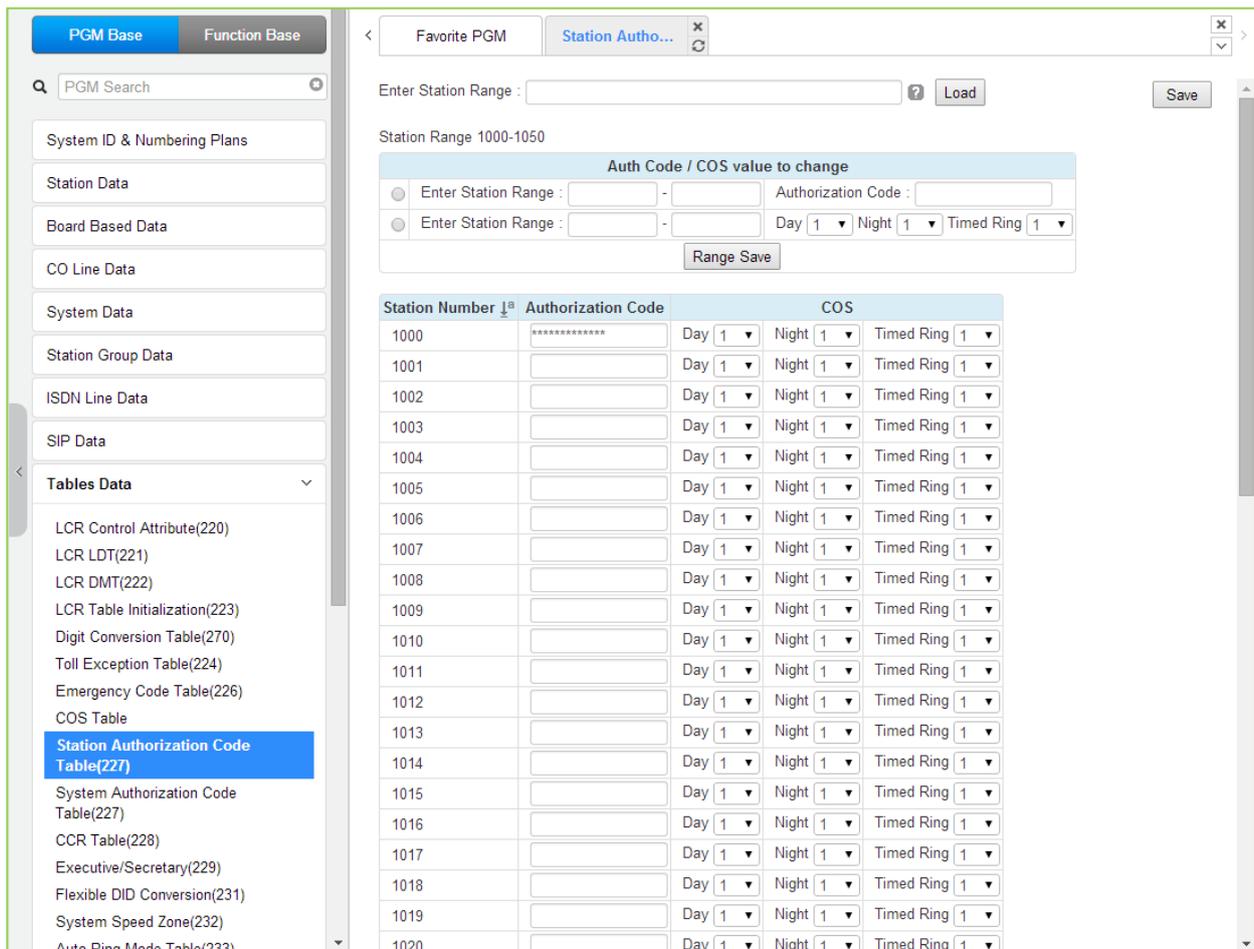
Figure 6.9.9-1 Station Authorization Code Table
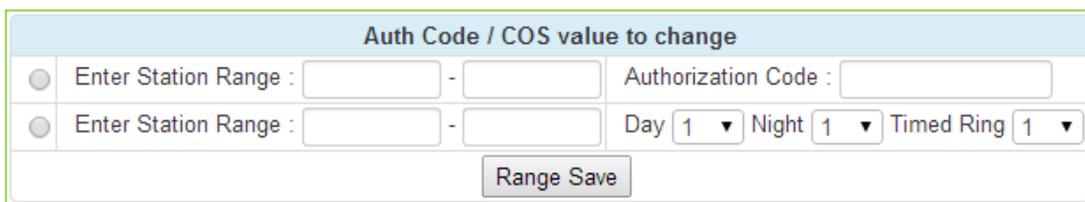
User can change the authorization code and COS for the station range across the board and click the Range save button to save.

Figure 6.9.9-2 System Authorization Code Table

User can change COS for the system index range across the board and click the Range save button to save.

Authorization codes are employed to control access to system resources and facilities. Walking COS, CO/IP Group access, DISA calls and certain Call Forward types may require input of a valid Authorization code. Codes up to 12 digits may be entered into the system database.

The Station Authorization Code Table entries are associated with individual stations. The number of system Authorization codes varies based on the configuration; the number of available codes is provided in Table 2.1-1. Each Authorization code is assigned a separate COS for Day, Night and Timed Ring mode operation.

The system will allow the station associated Authorization codes to be duplicated. However, the iPECS will not allow duplicate or conflicting system level codes unless the '*' and Authorization Table indexing is used to enter codes. Conflicting codes occur when a shorter code contains the first digits of a longer code, i.e. 12 conflicts with 1234.

Table 6.9.9-1 STATION CLASS-OF-SERVICE

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 dialed cannot be a Long Distance code, default "0", and further denied/allowed based on Exception Table C.
6	The leading digits dialed cannot be a Long Distance code & digit count cannot exceed the LD digit counter, default 8 digits, and further denied/allowed based on Exception Table C.
7	Intercom and paging calls are allowed. No outgoing dialing except for emergency calls is allowed on CO Lines.
8	The assignments in the Exception Table D are monitored for allow and deny numbers.
9	The assignments in the Exception Table E are monitored for allow and deny numbers.
10	The assignments in the Exception Table D & E are monitored for allow and deny numbers.
11	The assignments in the Exception Table A & B and D & E are monitored for allow and deny numbers.

Table 6.9.9-2 STATION/CO LINE COS TOLL RESTRICTIONS

	CO COS 1	CO COS 2	CO COS 3	CO COS 4	CO COS 5
STA COS 1	No Restriction	No Restriction	No Restriction	Only Local Call(LD code/counter) and Table C	No Restriction
STA COS 2	Exception Table A governs the dialing	Exception Table A governs the dialing	No Restriction	Only Local Call(LD code/counter) and Table C	No Restriction
STA COS 3	Exception Table B governs the dialing	No Restriction	Exception Table B governs the dialing	Only Local Call(LD code/counter) and Table C	No Restriction
STA COS 4	Exception Table A & B governs the dialing	Exception Table A governs the dialing	Exception Table B governs the dialing	Only Local Call(LD code/counter) and Table C	No Restriction
STA COS 5	Local Call only (LD Code, "1" or "0") and Table C	Local Call only (LD Code, "1" or "0") and Table C	Local Call only (LD Code, "1" or "0") and Table C	Only Local Call(LD code/counter) and Table C	No Restriction
STA COS 6	Only Local Call (LD code/counter) and Table C	Only Local Call (LD code/counter) and Table C	Only Local Call (LD code/counter) and Table C	Only Local Call(LD code/counter) and Table C	No Restriction
STA COS 7	In-house dialing only	In-house dialing only	In-house dialing only	In-house dialing only	In-house dialing only

Table 6.9.9-2 STATION/CO LINE COS TOLL RESTRICTIONS

	CO COS 1	CO COS 2	CO COS 3	CO COS 4	CO COS 5
STA COS 8	Exception Table D governs the dialing	Exception Table D governs the dialing	No Restriction	Only Local Call(LD code/counter) and Table C	No Restriction
STA COS 9	Exception Table E governs the dialing	Exception Table E governs the dialing	No Restriction	Only Local Call(LD code/counter) and Table C	No Restriction
STA COS 10	Exception Table D & E governs the dialing	Exception Table D & E governs the dialing	No Restriction	Only Local Call(LD code/counter) and Table C	No Restriction
STA COS 11	Exception Table A & B and D & E governs the dialing	Exception Table A & B and D & E governs the dialing	No Restriction	Only Local Call(LD code/counter) and Table C	No Restriction

6.9.10 Customer Call Routing Table - PGM 228

Selecting CCR Table will display the Customer Call Routing data entry page. Enter an index to select the appropriate CCR Table and click **[Load]** to modify the table. Click **[Save]** button after changing Value.
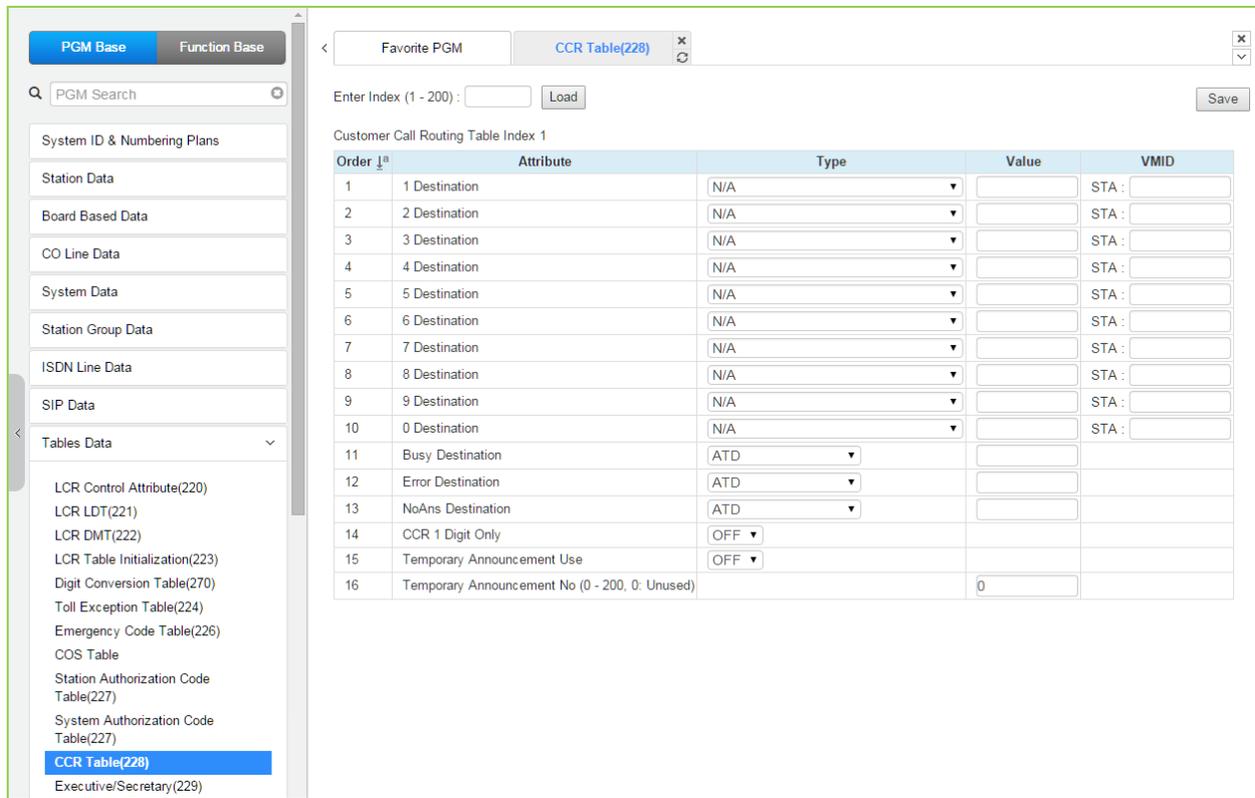
Figure 6.9.10-1 Customer Call Routing Table

The system incorporates IVR (Interactive Voice Response) capabilities called CCR (Customer Call Routing). After, or during a System 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 specified Announcement (01-70). The available destinations are shown in Table 6.9.10-1 below.

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.

Each CCR announcement has a re-route destination for Busy, Error, or No Answer routing.

The Re-route destinations available are Tone, Attendant, Station Group, and a VSF System Announcement.

The table includes a "CCR 1 Digit Only" option. When this option is enabled, the system will accept only a single digit as the entry. When the option is OFF, the caller can dial multiple digits that are accepted as DISA dialing.

Temporary announcement can be used in holiday, or emergency case.

When Temporary Announcement Usage is set to ON, programmed Temp Announcement is played instead of CCR table announcement.

This Temporary Announcement can be set also by Remote Access - by using [Remote Access to

Mobile Extension] feature.

After access to Mobile Extension,

to turn On Temp Announcement, 8 + CCR Announcement Number.

to turn Off Temp Announcement, 9 + CCR Announcement Number.

Table 6.9.10-1 CCR DESTINATIONS (PGM 228)

TYPE	DESCRIPTION
01	Route to a Station
02	Route to a Station Group
03	Route with System Speed Dial
04	Route as PBX Transfer with System Speed Dial (Flash then dial speed dial digits)
05	Route to VSF Announcement
06	Route to VSF Announcement and disconnect
07	Route to Networked Station
08	Conference Room
09	Internal Page
10	External page
11	All Call Page
12	Route to voice mail (station group/station number)
13	Company Directory (USA Only)
14	Record VM Greeting (USA Only)
15	Room type Conf Group Join

6.9.11 Executive/Secretary Table - PGM 229

Selecting Executive/Secretary will display the Executive/Secretary Table data entry page. Click **[Save]** button after changing Value.
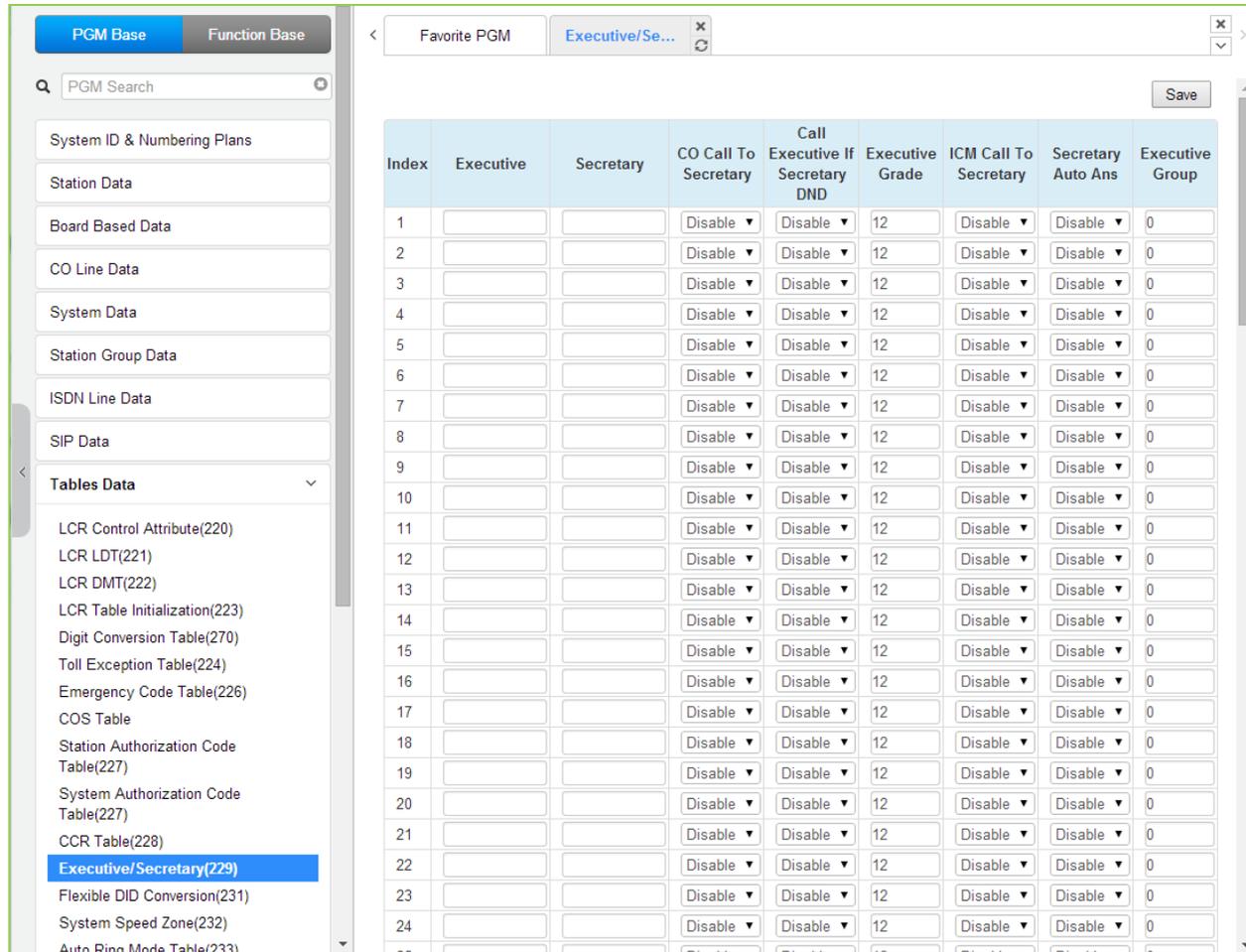
Figure 6.9.11-1 Executive/Secretary Table

Stations can be paired as Executive/Secretary pairs so that when the Executive enters DND, intercom and transferred calls are automatically routed to the Secretary. An Executive may have only one Secretary however, 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. In addition, when active, the Secretary can be assigned to receive the Executive's voice messages, refer to Common Attributes section 6.2.2.

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 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 executives, to the defined Secretary's station regardless of the Executive's station status.

Refer to Table 6.9.11-1 for a description of the Executive/Secretary parameters and the input required

Table 6.9.11-1 Executive/Secretary Table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Executive	The Station numbers of the Executive is assigned here.	Station Number	
Secretary	The Station number of the Secretary is assigned here.	Station Number	
CO Call to Secretary	If desired, all incoming CO calls to the Executive route to the Secretary's station regardless of the Executive's status.	Enable/ Disable	Disable
Call Executive if Secretary is in DND	If the Secretary is in DND, Executive calls can be routed to the Executive.	Enable/ Disable	Disable
Executive grade	Higher or equal grade Executives can override the Executive/Secretary Forward feature to call a lower grade Executive. Highest grade: 01, Lowest grade: 12	01~12	12
ICM Call to Secretary	If this option is ON, all internal calls to the Executive station, except for calls from the same or higher executives, are routed to the Secretary's station regardless of the Executive's status. The default value for Korea, India, Israel, Turkey, and Thailand is Enable	Enable Disable	Disable
Secretary Auto Answer	When enabled, calls from the Executive to the secretary employ the HF ICM signaling mode.	Enable Disable	Disable
Executive Group	When assigned to the same group, lower grade Executives can call higher grade Executives overriding the Executive/Secretary feature.	00-50	0

6.9.12 Flexible DID Conversion Table - PGM 231

Selecting Flexible DID Conversion Table will display the Flexible DID Table data entry page. Enter an index range and click [Load] to modify the DID Conversion data. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of DID indices when saved. Click **[Save]** button after changing Value.
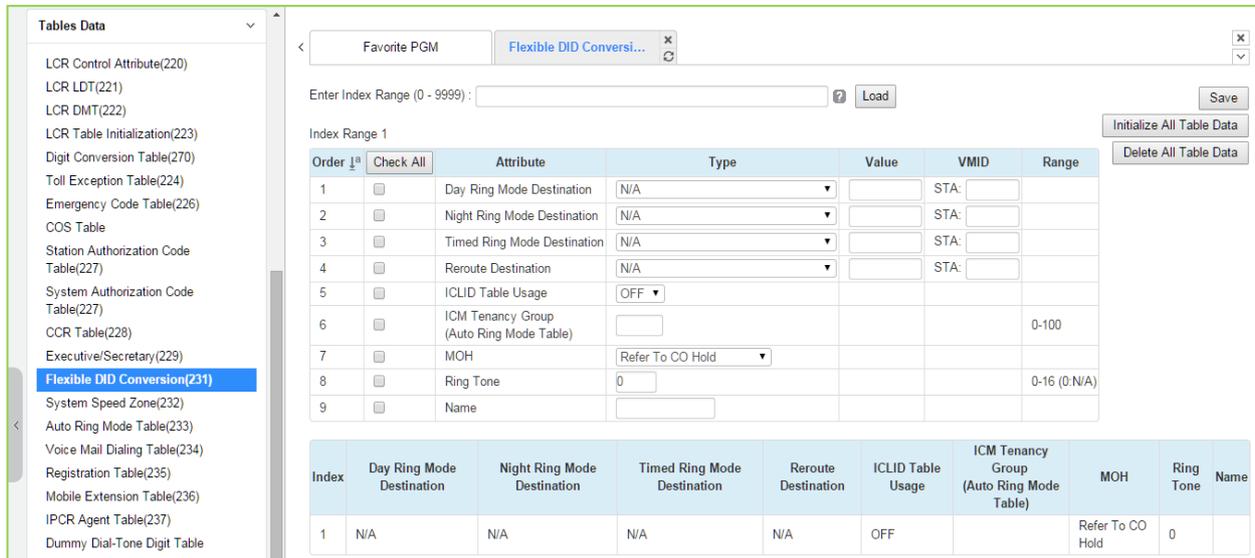
Figure 6.9.12-1 Flexible DID Conversion Table

When the received DID digits are converted, the resulting four-digit number may be used as an index to the Flexible DID Conversion Table. The Flexible DID Table index is used when DID Line is assigned a Conversion type 2, refer to the DID Service attributes section 6.4.6. Using the index from the digit conversion a destination for the DID call is determined by a Look-up in the Flexible DID Conversion Table. The destination for the call is generally defined as a type and a value. The type selects options such as station, station group, VSF, etc. The value specifies the particular station, station group, etc. In addition, ICLID routing can be enabled for DID lines or can be assigned an index to the Auto Ring Mode table.

Table 6.9.12-1 FLEXIBLE DID CONVERSION

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Day Ring Mode Destination	The destination for Day Ring mode is entered here.	N/A, Station, Station Group, System Speed, PBX Transfer with System Speed, VSF Announcement, VSF Announcement and disconnect, Networking, Conference Room, Internal Page,	
Night Ring Mode Destination	The destination for Night Ring mode is entered here.		
Timed Ring Mode Destination	The destination for Timed Ring mode is entered here.		
Reroute Destination	The destination for calls that need reroute service is entered here		

Table 6.9.12-1 FLEXIBLE DID CONVERSION

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
		External page, All Call Page, Voice Mail, ICLID Table, Company Directory, Record VM Greeting, Room type Conf Group Join	
ICLID Table	A DID Conversion Table index can be assigned to employ ICLID routing, section 6.7.4.	OFF ON	OFF
ICM Tenancy Group (Auto Ring Table)	A DID Conversion Table Index can be assigned to employ an ICM Tenancy Group Auto Ring Table to determine Day/Night/Timed operation mode.	0~100	0
MOH	A Music source is assigned so that calls to the destination receive audio from the source while on hold.	Refer to CO Hold, Internal/External Music 1, External Music 2, VSF MOH, SLT MOH 1, SLT MOH 2, SLT MOH 3, SLT MOH 4, SLT MOH 5, VSF MOH 2, VSF MOH 3	Refer to CO Hold
Ring Tone	A call routed with the DID Conversion Table will employ the selected Ring tone to alert the destination.	0-12	0
Name	An eleven character Name can be configured for the Table bin	11 characters	

6.9.13 System Speed Zone Table - PGM 232

Selecting System Speed Zone will display the System Speed Zone data entry page. Click **[Save]** button after changing Value.
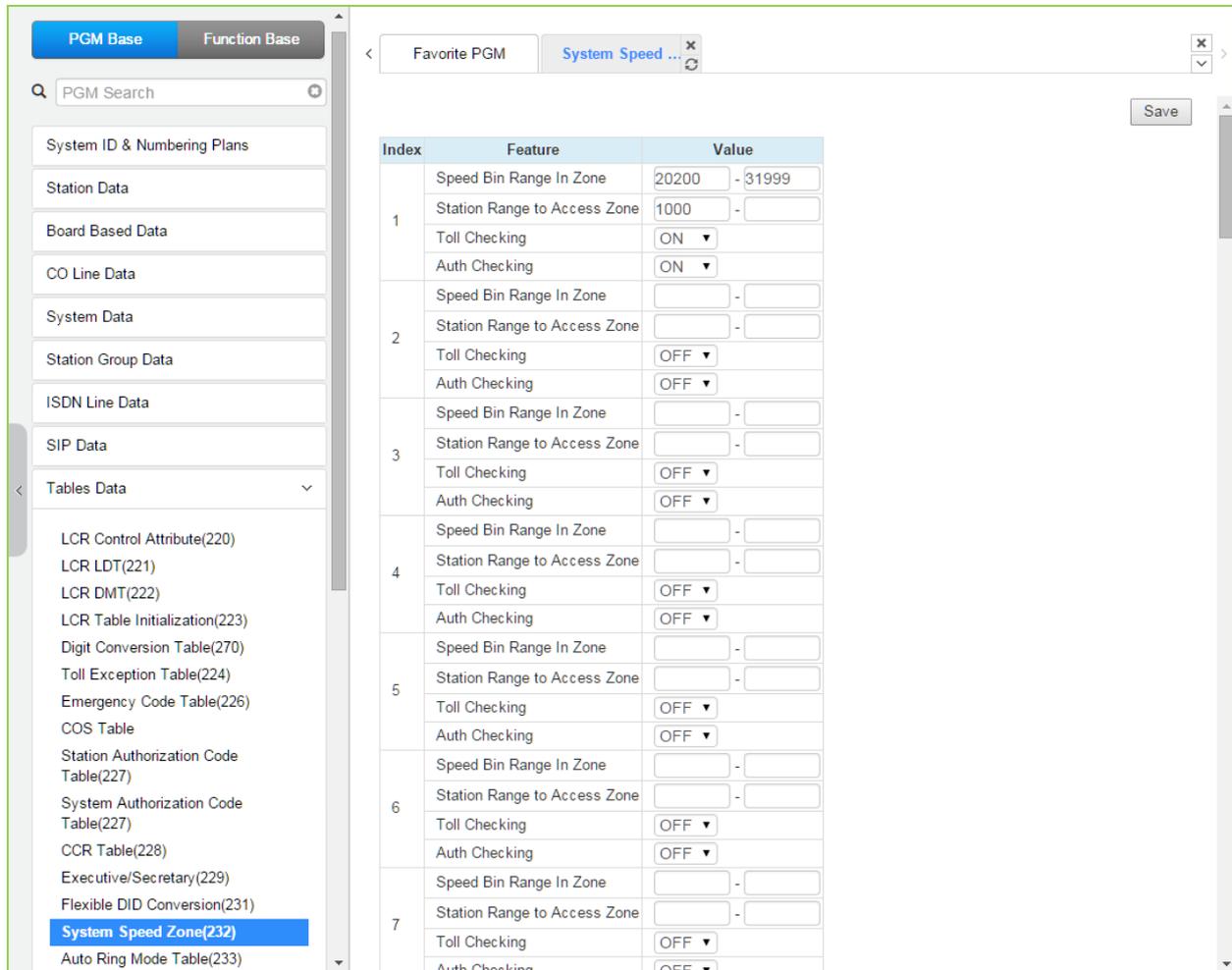
Figure 6.9.13-1 System Speed Zone

System 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 and CO Line COS for the speed dial number prior to dialing. Available parameters and values are shown in Table 6.9.13.

Table 6.9.13-1 SYSTEM SPEED ZONES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Speed Bin Range in Zone	A range of Speed Dial Bins is assigned to a zone.	20000-31999	20200-31999
Station Range to Access Zone	A range of Stations is permitted access to the System Speed dials in the Zone.	1000~	1000~3399
Toll Checking	Toll restriction can be applied to Speed Dials in the Zone.	OFF ON	ON
Auth Checking	The user can be required to enter a valid Authorization code to use Speed Dials in the zone.	OFF ON	ON

6.9.14 Auto Ring Mode Table - PGM 233

Selecting Auto Ring Mode Table will display the Auto Ring Mode Table data entry page. Enter the desired index (Tenancy Group) and click **[Load]** to enter data. Click **[Save]** button after changing Value.

Figure 6.9.14-1 Auto Ring Mode Table

The system can be programmed to automatically select the Ring and COS based on time of day and day of the week. Three Ring & COS modes are available, Day, Night, and Timed. The Ring assignments are as defined in CO Ring Assignments, section 6.4.5. COS assignments are made in the Station COS and DISA COS programs discussed in sections 6.2.6 and 6.5.6, respectively.

The start times for Day, Night and start and end times for Timed modes are entered for each day of the week. After the Timed mode "End time", the mode returns to previous mode. The Attendant can override the Automatic selection and select the desired mode (Day, Night, and Timed) manually. A separate Auto Ring Table is established for each ICM Tenancy Group in (indices 1 ~ 100) and for the system (index 00).

6.9.15 Voice Mail Dialing Table - PGM 234

Selecting Voice Mail Dialing Table will display the Voice Mail Dialing data entry page. Click **[Save]** button after changing Value.
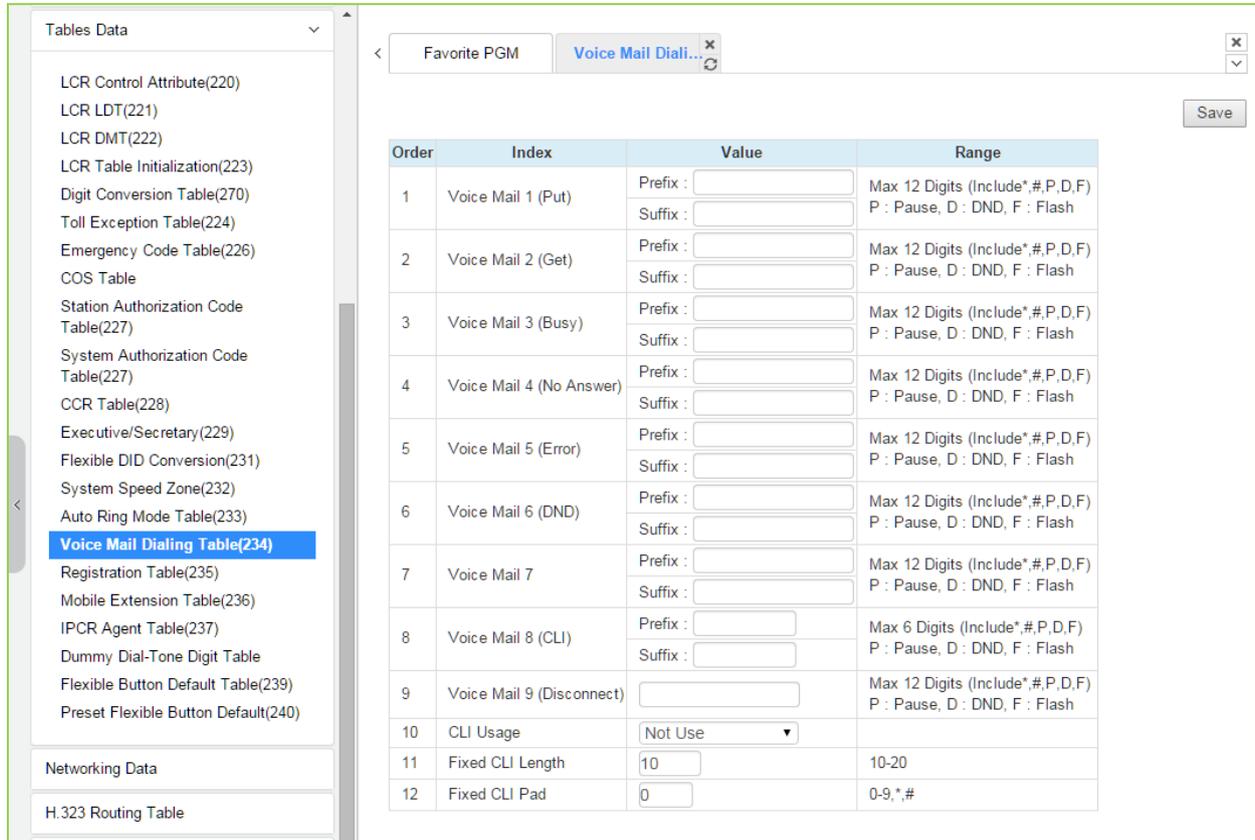
Figure 6.9.15-1 External Voice Mail Dialing Table

When an external Voice Mail system is used that connects to an SLT port, a digit sequence must be defined for the system to signal various call characteristics to the Voice Mail system. The external Voice Mail uses the sequences to determine appropriate announcements or further call routing. The Table permits the definition of digits as either a prefix or suffix to other digits (station number for mailbox identification). Sequences are defined for such call characteristics as Put Mail, Get Mail, No Answer call, etc. as described in Table 6.9.15-1.

Table 6.9.15-1 VOICE MAIL DIAL FUNCTIONS

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Voice Mail 1 (Put)	Code sent when the Voice Mail is to receive a call to record a message. Put Mail	Prefix Suffix (Max. 12 Digits- Include *, #, P, D, F)	
Voice Mail 2 (Get)	Code to send when the voice mail is to playback a recorded message. Get Mail	Prefix Suffix (Max. 12 Digits/ Include *, #, P, D, F)	
Voice Mail 3 (Busy)	Code to send when the voice mail is to receive a	Prefix	

Table 6.9.15-1 VOICE MAIL DIAL FUNCTIONS

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	call when the user is busy. Busy Mail	Suffix (Max. 12 Digits/ Include *, #, P, D, F)	
Voice Mail 4 (No Answer)	Code to send when the voice mail is to receive a call when the user did not answer. No Answer Mail	Prefix Suffix (Max. 12 Digits/ Include *, #, P, D, F)	
Voice Mail 5 (Error)	Code to send when the voice mail is to receive a call when a user dialing error exists. Error Mail	Prefix Suffix (Max. 12 Digits/ Include *, #, P, D, F)	
Voice Mail 6 (DND)	Code to send when the voice mail is to receive a call when the user is in DND. DND Mail	Prefix Suffix (Max. 12 Digits/ Include *, #, P, D, F)	
Voice Mail 7	When you enter the Digit, the following Digit means as below: P : Pause, D : DND, F : Flash	Prefix Suffix (Max. 12 Digits/ Include *, #, P, D, F)	
Voice Mail 8 (CLI)	When you enter the Digit, the following Digit means as below: P : Pause, D : DND, F : Flash	Prefix Suffix (Max. 12 Digits/ Include *, #, P, D, F)	
Voice Mail 9 (Disconnect)	Code to send when the call disconnects. Disconnect Mail. When you enter the Digit, the following Digit means as below: P : Pause, D : DND, F : Flash	Prefix Suffix (Max. 12 Digits/ Include *, #, P, D, F)	
CLI Usage	Not use: Do not send CLI Real CLI length: Send CLI without padding before prefix code. Fixed CLI length: Send CLI with fixed length	Not use, Real CLI length, Fixed CLI length	Not use
Fixed CLI Length	Define length of CLI	10-20	10
Fixed CLI Pad	Define padding character for CLI	0-9,*,#	0

6.9.16 Registration Table - PGM 235

Selecting Registration Table will display the Registration data entry page. Click **[Save]** button after changing Value.
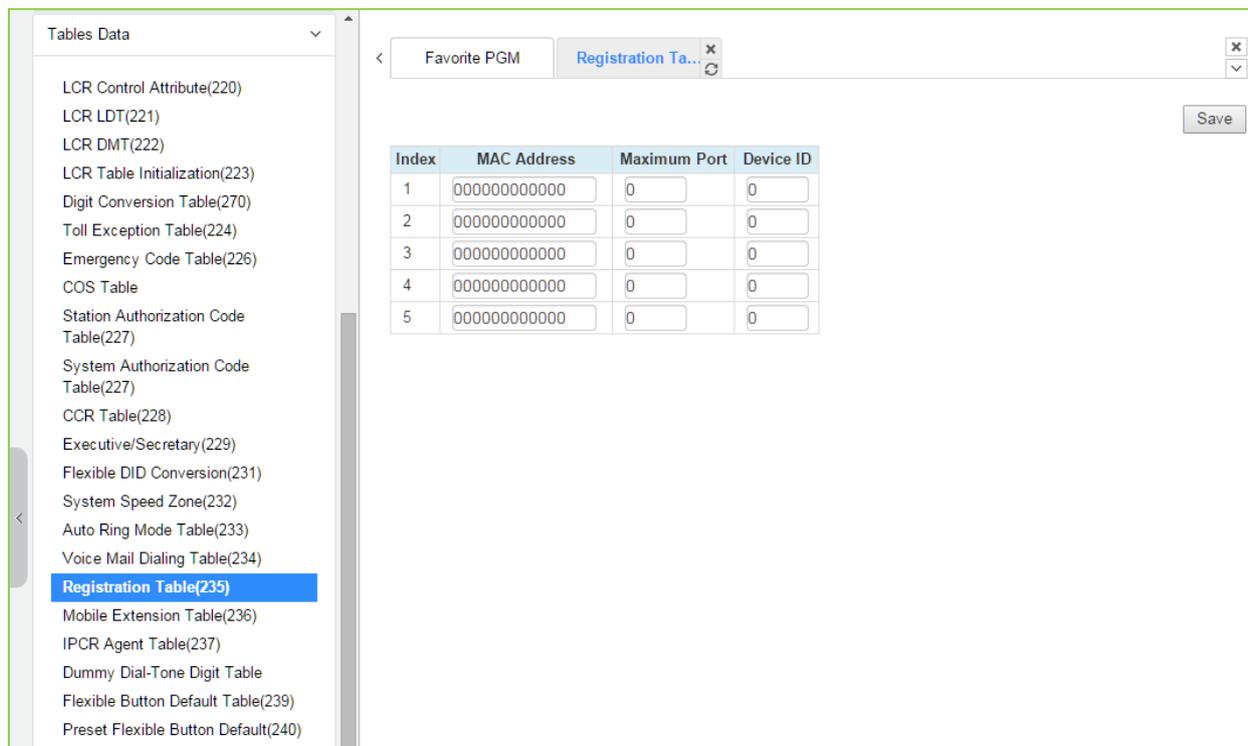
Figure 6.9.16-1 MAC Registration Table

When multiple iPECS UCP are located on the same LAN, it may be desirable to register add-on devices employing the Registration Table.

Table 6.9.16-1 MAC REGISTRATION

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
MAC Address	Enter the MAC address of a device to register		000000000000
Maximum Port	Enter the maximum number of ports (channels) for the device.	00-99	0
Device ID	A device Id may be entered for registration. The Device Ids available are shown at the bottom of the System Overview Web page.	0-255	0

6.9.17 Mobile Extension Table - PGM 236

Selecting Mobile Extension Table will display the Mobile Extension data entry page. Enter a Station Range then click **[Load]** to modify data. Click **[Save]** button after changing Value.
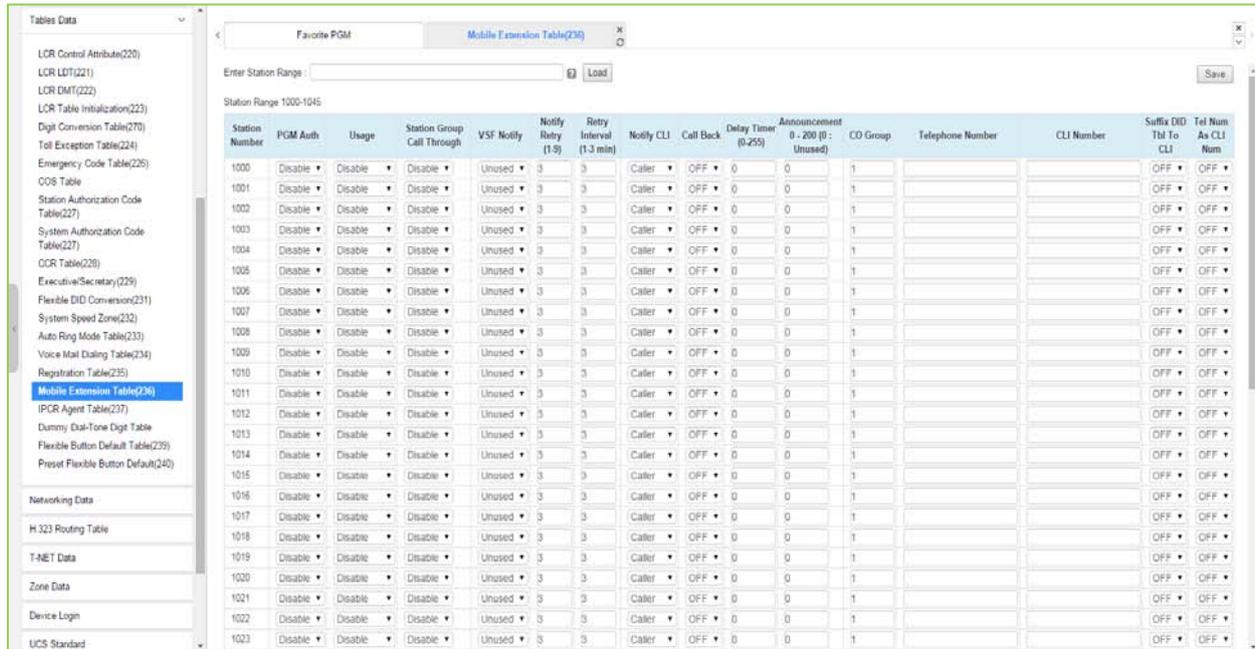
Figure 6.9.17-1 Mobile Extension Table

A mobile phone can be used in conjunction with an iPECS Phone. The Mobile phone can access system resources available to the user's wired phone and will receive ring for incoming iPECS 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/IP Line Group to place calls to the Mobile phone. In addition, the mobile phone can be assigned to receive Station group calls to the primary extension. In addition, parameters for notification of new VSF voice mails can be defined.

Table 6.9.17-1 MOBILE EXTENSION TABLE

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
PGM Authority	The user may be allowed to activate the Mobile Extension feature.	Disable Enable	Disable
Usage	The Mobile Extension feature can be enabled or the Table can be used to configure Fail-Over operation.	Disable Mobile Ext Fail Over	Disable
Station Group Call through	When the associated station is a member of a Station group (ACD, Circular, Ring or Terminal), group calls can be sent to the active Mobile Extension.	Disable Enable	Disable
VSF Notify	When enabled the active Mobile Extension will receive notification by the system when the user has a new message in the built-in Voice Mail.	Unused Use	Unused
Notify Retry	The number of attempts the system will make to complete a notification when receiving busy/no-answer is defined.	1 – 9 times	3
Retry Interval	This field defines the time between notification attempts. If a notification fails, the system will retry after the timer expires.	1 – 3 minutes	3

Table 6.9.17-1 MOBILE EXTENSION TABLE

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
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 the station.	Caller My Ext.	Caller
Call Back	If "ON", an incoming Mobile Extension call will be released before answered and system places a call to Mobile Extension. After the Mobile answers, the dial tone is provided and the Mobile Extension can place internal or external calls.	OFF ON	OFF
Delay Timer	When the Mobile Extension requests a Call Back, the system will place the Call Back after expiration of this Delay timer.	0 ~ 255 seconds	0
Announcement	A system announcement can be recorded to provide the Mobile caller with a menu of dialing commands available to the remote Mobile user, remote control.	0-200	0
CO Group	The CO/IP Line group used to call (ring) the Mobile Extension is defined.	0~201	1
Telephone Number	The telephone number of the Mobile extension or Fail-over number of the station must be assigned for proper operation.		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
Suffix DID table index to the CLI	An incoming DID call is passed to the Mobile Extension with the original caller's CLI and the Flexible DID Conversion Table index. (format: 'original CLI' + '*' + 'DID Table Index')	OFF ON	OFF
Tel Num As CLI Num	If this option is set to ON, Telephone Number is used the same as CLI Number. That means, when the mobile user calls to his station, if Telephone Number or CLI Number is matched with his mobile number, then the mobile user hears system dial tone, and calling to outside is available. If this option is set to OFF, only CLI Number is matched, system dial tone is served.	OFF/ON	ON

6.9.18 IPCR Agent Table - PGM 237

Selecting IPCR Agent Ta ble will display the IPCR Agent data entry page. Click **[Save]** button after changing Value.

Figure 6.9.18-1 IPCR Agent Table

This table correlates an Object Type (Station or CO/IP Line) to the IP CR (Call Recording) Agent index in the IP CR server. The iPECS UCP can support up to 2400 agents on up to ten different IPCR serves. The table is also employed with third-party servers for call recording. When a call is placed or received by the station or CO/IP Line, the call is recorded in the IP CR server under the Agent Id (Order number).

6.9.19 Dummy Dial-Tone Digit Table

Selecting Dummy Dial-Tone Digit Table will display the Dummy Dial-Tone Digit data entry page. Click **[Save]** button after changing Value.
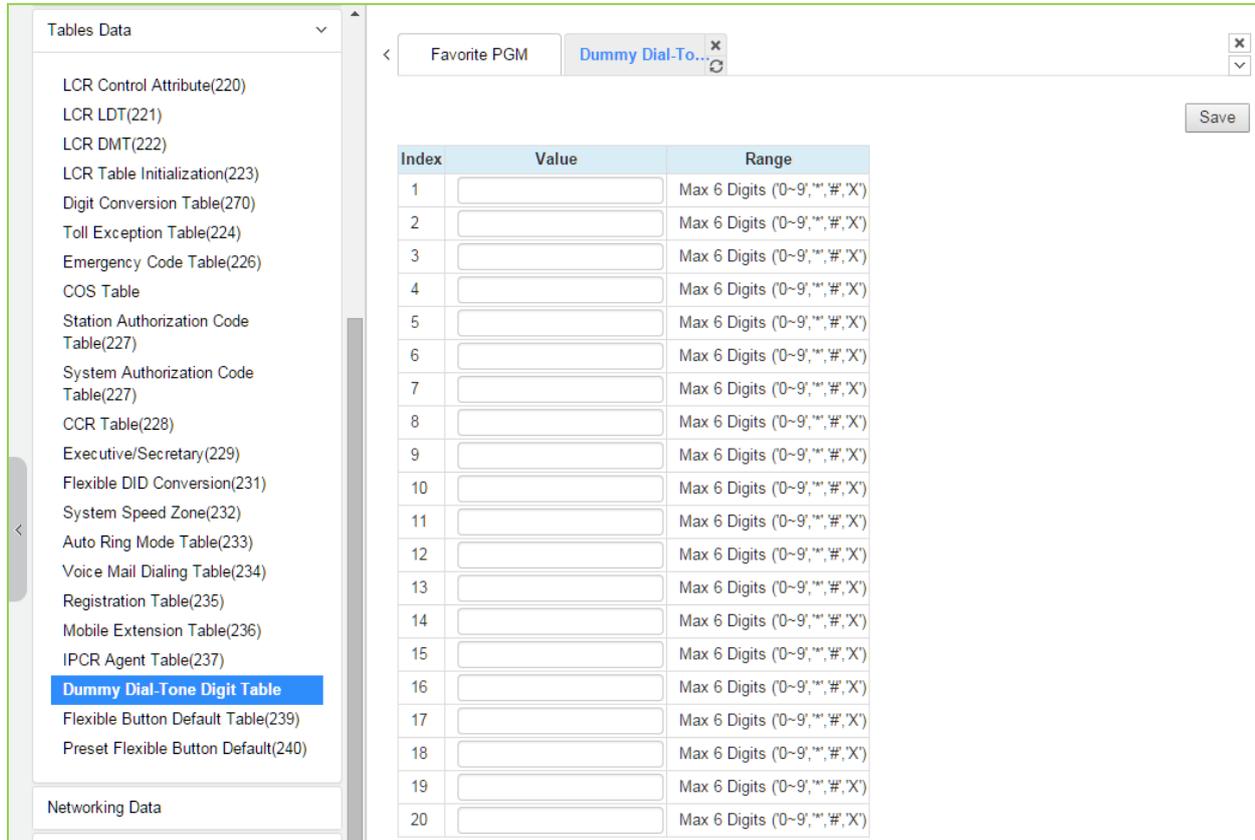
Figure 6.9.19-1 Dummy Dial-Tone Digit Table

When digit conversion is programmed, the CO line is seized after digit conversion is completed and therefore user cannot hear the CO dial tone from PX until completing digit conversion. For this case, a dummy dial tone can be programmed. Pressing one of pre-programmed digits ('0–9', '*', '#', X') will provide CO dial tone to the user regardless of CO line seizure.

6.9.20 Flexible Button Default Table – PGM 239

Selecting Flexible button default table will display the below page. Use the check mark to indicate which attributes to modify; data for checked attributes is stored by clicking **[Save]** button.
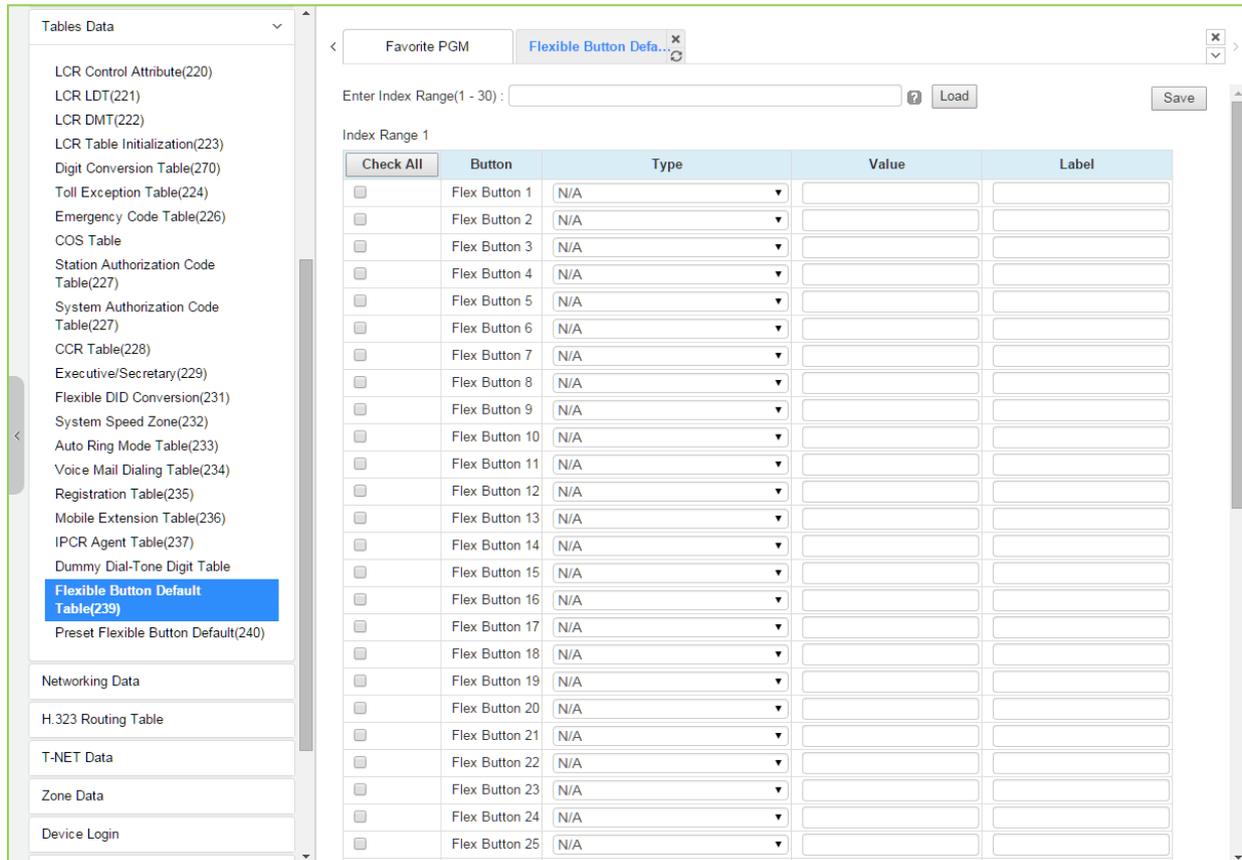
Figure 6.9.20-1 Flexible Button Default Table

The system provide 30 default flexible button table so administrator can be configured each table for using 'Preset Flexible Button Default (240).

Administrator can assign the type among the below types on Flexible button.

Table 6.9.20-1 FLEX BUTTON TYPE

TYPE	REMARK
N/A	Empty (unassigned), may be defined by the user.
CO Line	Assigns button to access a defined CO/IP line.
CO Group	Assigns button to access a free line in the CO/IP Group.
Loop	Assigns button to access a loop line.
Station Number	Assigns button as DSS/BLF for the assigned station number.
Programming (Numbering Plan)	Assigns button to dial a code from the Flexible Numbering Plan, see Appendix B section.
Programming (PGM)	Assigns button to perform a User Program function from the Fixed Numbering Plan, Appendix B section.
Station Speed Bin	Station Speed Dial bin.
System Speed Bin	System Speed Dial bin.
Net Station Number	Refer to section Network Numbering Plan Table - PGM 324.
U-Loop	U-Loop button for call wait of internal & external call

6.9.21 Preset Flexible Button Default – PGM 240

Selecting Preset flexible button default will display the below page. Click **[Save]** button after changing Value.

Figure 6.9.21-1 Preset Flexible Button Default

Administrator can assign Preset flexible button table to the station automatically by default. Before connecting the station, administrator has to set this table according to Priority, Station Type, the number of Button, Station Range, and check the preset Index (1~30). The preset index means the range of Flexible Button Default Table (PGM 239).

6.10 Networking Data

Selecting the Networking Data program group will display the sub-menu displayed in the left frame as shown in the following figure.
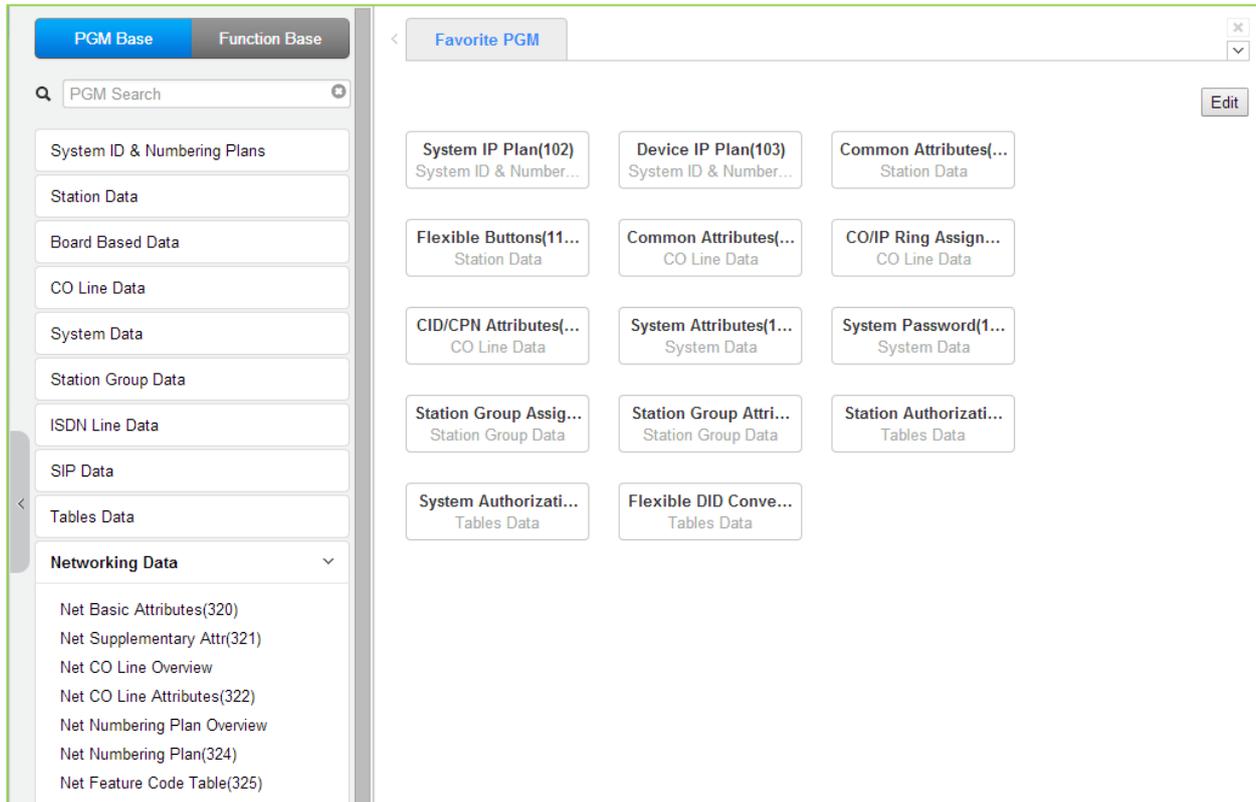
Figure 6.10-1 Networking Data

6.10.1 Network Basic Attributes - PGM 320

Selecting Network Basic Attributes will display the Network Basic Attributes entry page. Click **[Save]** button after changing Value.
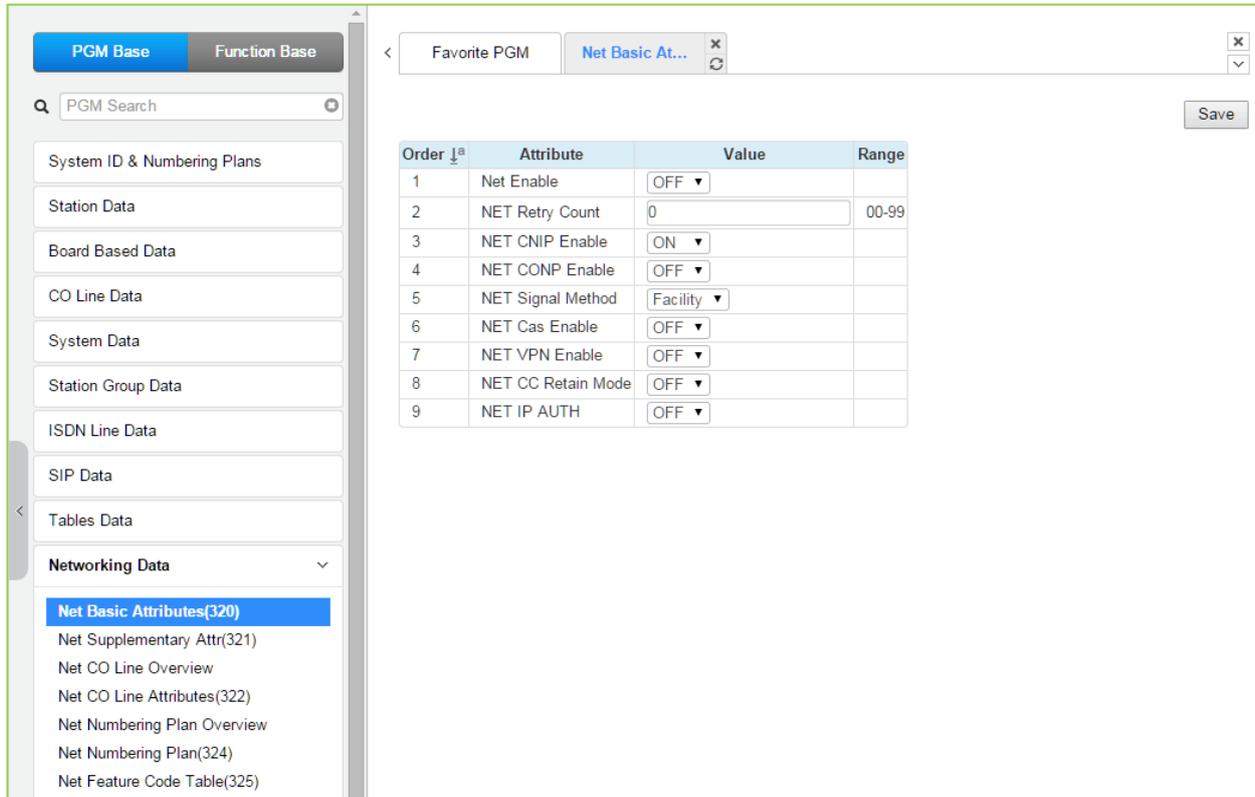
Figure 6.10.1-1 Network Basic Attributes

Basic Network Attributes are used to enable networking and to configure the basic characteristics of the network as detailed in Table 6.10.1-1.

Table 6.10.1-1 NETWORK BASIC ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Net Enable	For operation, the Networking function must be enabled here. Note a license is required.	OFF ON	OFF
Net Retry Count	Not used.	00-99	0
Net CNIP Enable	The name of calling station is sent to the called iPECS UCP. CNIP is shown in the LCD of the called party's station.	OFF ON	ON
Net CONP Enable	The name of the connected station is sent to the calling iPECS UCP system. The CNOP is shown in the LCD of the calling party's station.	OFF ON	OFF
Net Signal Method	Network signaling can be sent in the Facility or User QSIG supplementary service message.	UUS/ Facility	Facility
Net Cas Enable	Not used.	OFF ON	OFF
Net VPN Enable	Not used.	OFF ON	OFF
Net CC Retain Mode	This field defines the Network signaling retain mode for CCBS service.	OFF ON	OFF

6.10.2 Network Supplementary Attributes - PGM 321

Selecting Network Supplementary Attributes will display the Network Supplementary Attributes entry page. Click **[Save]** button after changing Value.

Figure 6.10.2-1 Network Supplementary Attributes

Supplementary attribute configure signaling for supplementary features and characteristics of the BLF Manager.

Table 6.10.2-1 NETWORK SUPPLEMENTARY ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Net Transfer Mode	Call Forward and Transfer over the network employs the selected Rerouting or Join method	REROUT JOIN	REROUT
TCP Port for BLF	The TCP/IP port for BLF message packets to BLF Manager is defined.	9500-9999	9500
UDP Port for BLF	The UDP port for BLF message packets to BLF Manager is defined.	9500-9999	9501
BLF Manager IP	The IP Address of BLF Manager server is required when iPECS UCP is configured with eMG80 systems in the Network		0.0.0.0
Duration of BLF State	The system sends BLF messages to the BLF Manager at intervals of this timer.	01-99 (100 ms)	10
Multicast IP	The multicast IP address for BLF service.		0.0.0.0
Net Trans Recall timer	When a call forward or transfer does not receive a response for the Network system, the call recalls the transferring party at expiration of this timer.	001-300 (msec)	10

Table 6.10.2-1 NETWORK SUPPLEMENTARY ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
NET Reroute CO Group	If an outgoing SIP call receives no response after expiration of this timer, the call is rerouted to the alternate CO line.	1-200	0
BLF Service Usage	BLF Manager support must be enabled for use.	OFF ON	ON
Alternate/Secondary signal port	This signal port is used to add an alternate or secondary receiving signal port. A default receiving signal port is TCP 1720 and an additional signal port will be opened if this field is configured to valid value. When the system is installed behind xDSL modem, the problem of consecutive second call can happen by uncontrolled H.323 ALG function at xDSL modem. This field can be used to solve the problem of consecutive second call.	0, 1024-65535	0

6.10.3 Net CO Line Overview

Selecting Net CO Line Overview will return the overview display page.

Device Type	CO Line	CO Type	Net CO Group	Net CO Type
T1IM GW	1	Normal	0	PSTN
T1IM GW	2	Normal	0	PSTN
T1IM GW	3	Normal	0	PSTN
T1IM GW	4	Normal	0	PSTN
T1IM GW	5	Normal	0	PSTN
T1IM GW	6	Normal	0	PSTN
T1IM GW	7	Normal	0	PSTN
T1IM GW	8	Normal	0	PSTN
T1IM GW	9	Normal	0	PSTN
T1IM GW	10	Normal	0	PSTN
T1IM GW	11	Normal	0	PSTN
T1IM GW	12	Normal	0	PSTN
T1IM GW	13	Normal	0	PSTN
T1IM GW	14	Normal	0	PSTN
T1IM GW	15	Normal	0	PSTN
T1IM GW	16	Normal	0	PSTN
T1IM GW	17	Normal	0	PSTN
T1IM GW	18	Normal	0	PSTN
T1IM GW	19	Normal	0	PSTN
T1IM GW	20	Normal	0	PSTN
T1IM GW	21	Normal	0	PSTN
T1IM GW	22	Normal	0	PSTN
T1IM GW	23	Normal	0	PSTN
T1IM GW	24	Normal	0	PSTN

Figure 6.10.3-1 Network CO Line Overview

6.10.4 Network CO Line Attributes - PGM 322

Selecting Network CO Line Attributes will display the Network CO Line Group entry page. Enter a valid CO/IP Line range and click **[Load]** to enter the Network CO Line data. Click **[Save]** button after changing Value.
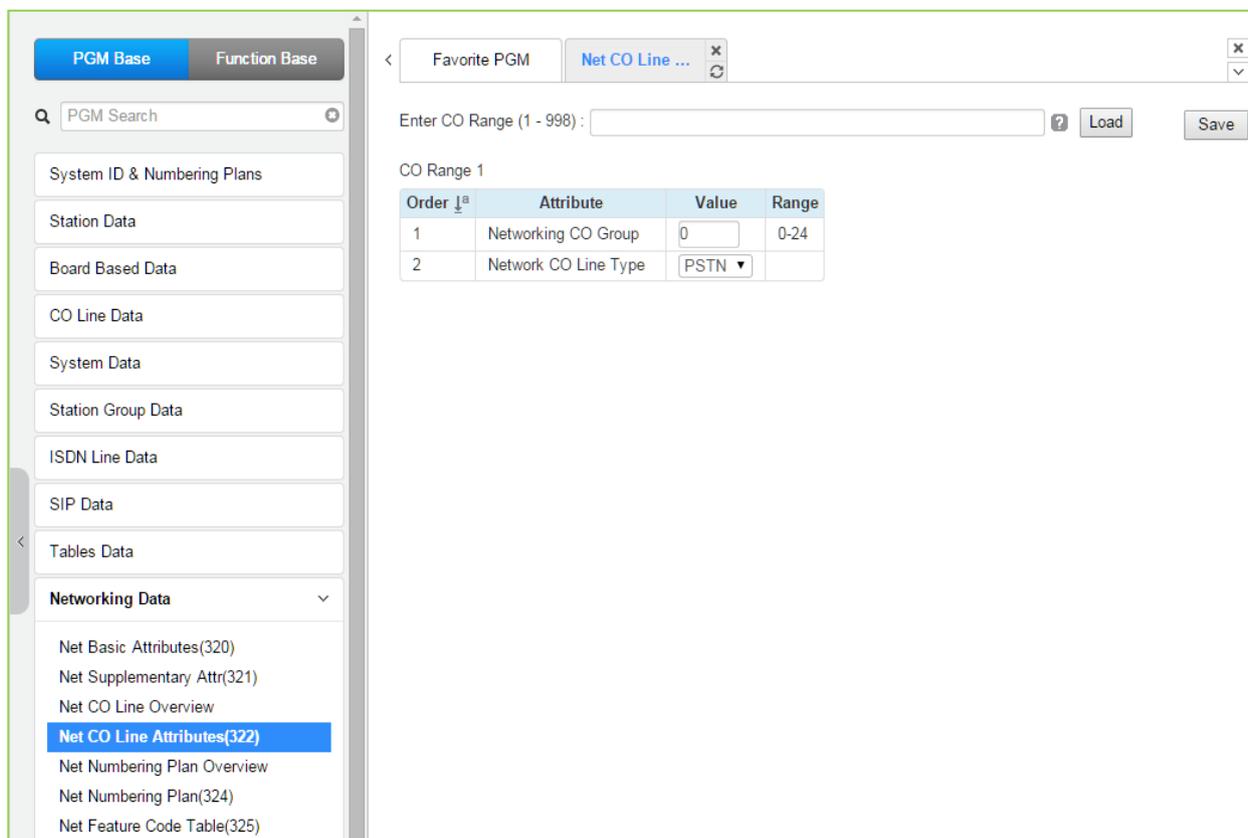
Figure 6.10.4-1 Network CO Line Attributes

Table 6.10.4-1 NETWORK CO LINE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Networking CO Group	The CO/IP Lines in the assigned group are employed as Network CO Lines, connecting to other systems in the network.	00-24	0
Network CO Line Type	The CO/IP Line is assigned for use by the network as a PSTN (carrier) or connection to the network.	NET PSTN	PSTN

6.10.5 Network Numbering Plan Table Overview

Selecting Network Numbering Plan Overview will return the overview display page.

Figure 6.10.5-1 Network Numbering Plan Table overview

6.10.6 Network Numbering Plan Table - PGM 324

Selecting Network Numbering Plan Table will display the Network Numbering Plan Table data entry page. Enter a valid table index and click **[Load]** to enter the Network Numbering Plan data. Click **[Save]** button after changing Value.

Figure 6.10.6-1 Network Numbering Plan Table

The Network Numbering Plan establishes the digit strings that make-up the numbering plan and associated routing for each Plan code.

Table 6.10.6-1 NETWORK NUMBERING PLAN TABLE

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Network Type	The type or use of the code is defined as directed to the PSTN (carrier) or a Networked system.	NET PSTN	NET
Numbering Plan Code	This field defines the digits, 0 ~ 9 that make up a Network Numbering Plan code. An '*' will represent any digit. To assign the code for the stations in another system, enter the common station number digits followed by "#".	Max. 16 digits (Include * and #)	
Networking CO	The Numbering Plan CO Group indicates the Network	00-24	

Table 6.10.6-1 NETWORK NUMBERING PLAN TABLE

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Group	CO/IP Group that is used for the connection. A '00' entry is an internal net station number.		
CPN ISDN Information	When an ISDN Line is used to place a network call, the CPN ISDN Information and the network Number are used as the Called party number.	Max. 16 digits	
CPN Information	When a VoIP channel is used to place a network call, the CPN used is the IP address associated with the VoIP channel. 1: 00 CPN INFORMATION 01 2: 00 CPN INFORMATION 02 3: 00 CPN INFORMATION 03 4: 00 CPN INFORMATION 04	Enter IP address	
Setup Wait Response Time	Networking call-setup failure timer when there is no proceeding message from network (i.e. IP-Network is down)	0, 3~15 Sec.	0
ALT Speed bin	Should the Network path fail, the system can place the call over an alternative path using a System Speed Dial number.`	20000~31999	
UCP IP Address	This field is the IP Address of the destination system for the code		0.0.0.0
UCP Port number	The TCP/IP port number of destination system for the Net Numbering Plan code is defined.	0000-9999	5588
Digit Repeat	When the number plan code, see above, is for PSTN call or transit-call, the number code can be included in SETUP message.	YES NO	NO
PSTN Enblock	When a PSTN receives a Transit-out call from a networked user, the digits can be sent En-block (YES) or Over-lap (NO).	YES NO	NO
PSTN CLI Method	The CLI sent with the call for this code can be set as the NET CLI (station number) or PSTN, which sends the CLI configured for the CO/IP Line and Station CLI.	NET PSTN	NET
CO Attendant Code CLI	When a networked system places a transit out call, the Centralized ATD CLI can be sent.	OFF ON	OFF
Firewall Routing	When the system can communicate with the networked system directly, over a common VPN, the systems Non Firewall address is sent in IP packets (OFF). Otherwise, the Firewall IP address is sent (ON).	OFF ON	ON
Transit Out Auth COS	When a user requests a transit-out call by seizing a CO line, COS may be applied according to the authorization code.	YES NO	NO
SMDR Digit Hide	For a Transit-out call from the iPECS, the networked system that places the call may desire to receive the Authorization code	YES NO	NO
Site name	A twelve-character name can be assigned to the system for use with network calls.	Max. 12 characters	
Emergency reroute timer	When a networked station places an Emergency call and the "Emergency CO or Group" in the Station Common Attributes is a Transit-out CO/IP Line or group, the call will Fail-over to the "Fail-over CO Group" in PGM 133 or the "Net Reroute CO Group" in PGM 321 after this timer expires. The "Fail-over CO Group" will have priority.	0-10 (seconds)	0

Table 6.10.6-1 NETWORK NUMBERING PLAN TABLE

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Tunneled SIG MSG	Send and Receive H323 message that include ISDN QSIG message. It is used to make networking with Panasonic system.	OFF ON	OFF
Alternate/Secondary signal port	This destination signal port is used on calling to remote network system. A default destination signal port is TCP 1720 but the port number is changed if this field is configured to valid value. When the system is installed behind xDSL modem, the problem of consecutive second call can happen by uncontrolled H.323 ALG function at xDSL modem. This field can be used to solve the problem of consecutive second call. Here, the value should be an alternate or secondary receiving signal port which is configured at remote system.	0-65535	
Local route ID	These two fields are used when a remote network system want to check the route ID value to authenticate the system.	Max. 15 characters	
Remote route ID	Here, 'Local Route ID' is the route ID string for local system. And 'Remote Route ID' is the route ID string for remote system.	Max. 15 characters	
Sending Name option	Basically the calling party name is transferred in QSIG messages. This field is an additional way to send the calling party name. In some PBX systems, the Q.931 Display IE field is used to transferring the calling party name and this value can be used as an interoperating solution for displaying the calling party name.	OFF, Display IE(CISCO)	OFF

6.10.7 Network Feature Code Table - PGM 325

Selecting Network Feature Code Table will display the data entry page. Click **[Save]** button after changing Value.
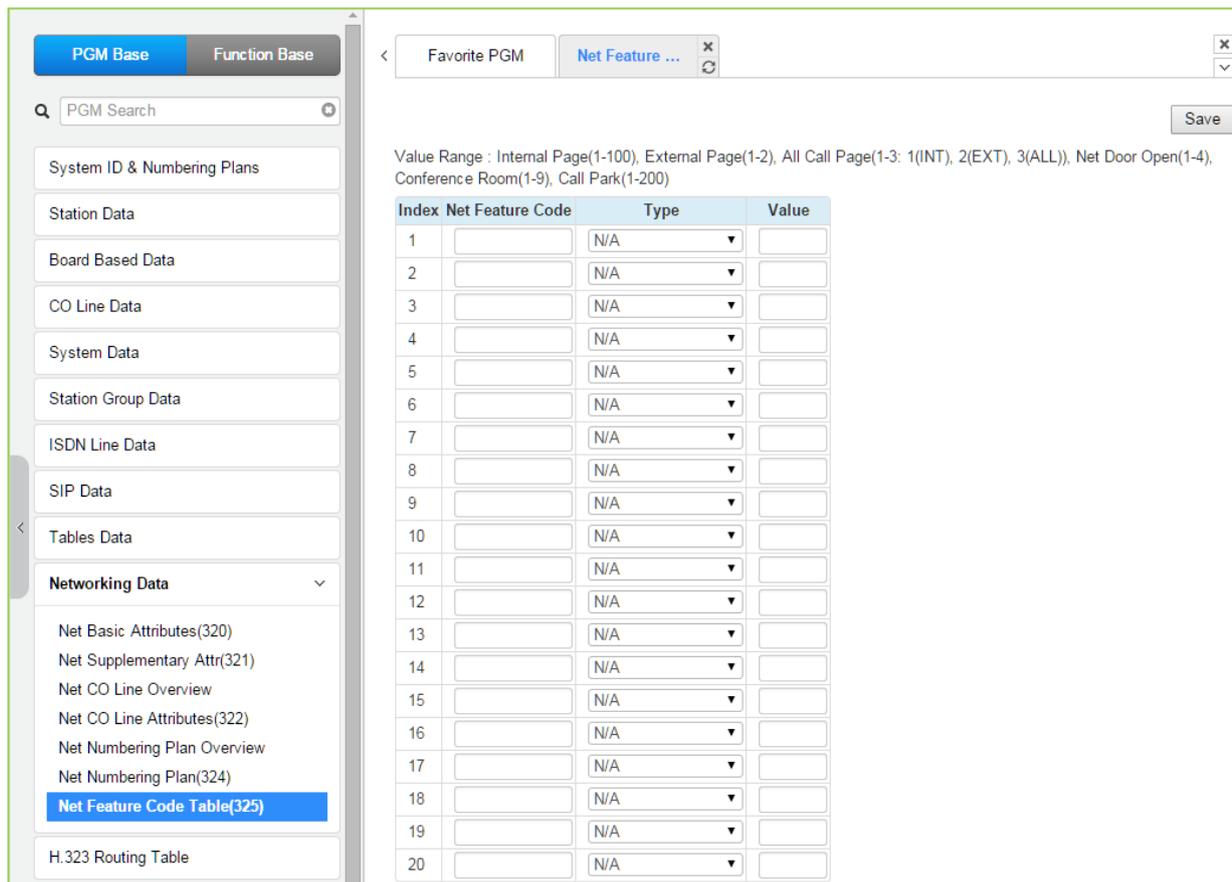
Figure 6.10.7-1 Network Feature Code Table

Codes can be assigned to activate special features over the network.

Table 6.10.7-1 NETWORK FEATURE CODE TABLE

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Net Feature Code	Digit sequence or code used to activate special Network Features.	16 digits	None
Net Feature type	Each Network Feature Code is assigned to activate a feature in the destination system.	N/A, Internal Page, External Page, All Call Page, Net Door Open, Conference Room, Call Park	N/A

6.11 H.323 Routing Table

Selecting the H.323 Data program group returns the sub-menu displayed in the left frame as shown in the following figure.
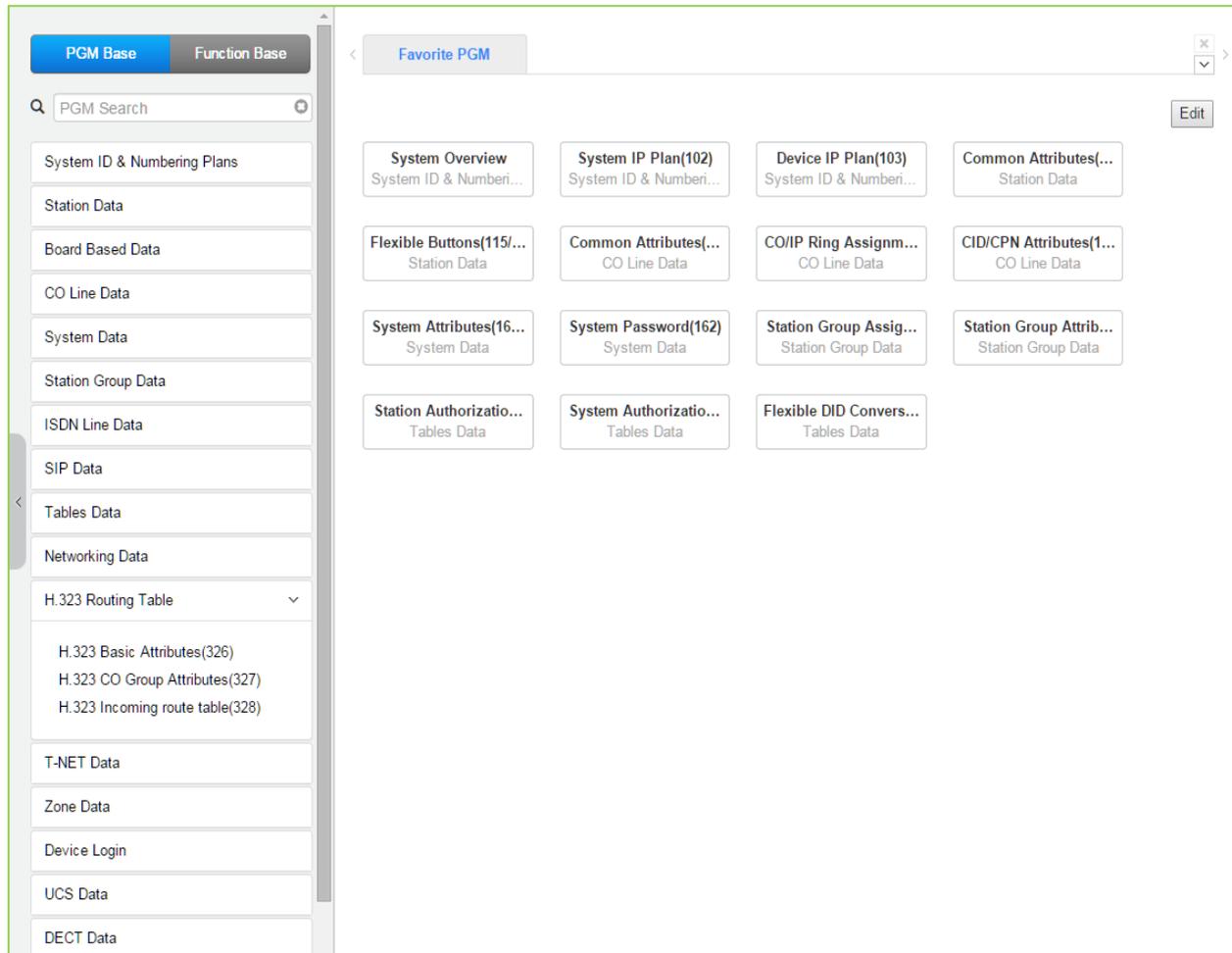
Figure 6.11-1 H.323 Routing Table Main Page

6.11.1 H.323 Basic Attributes -PGM 326

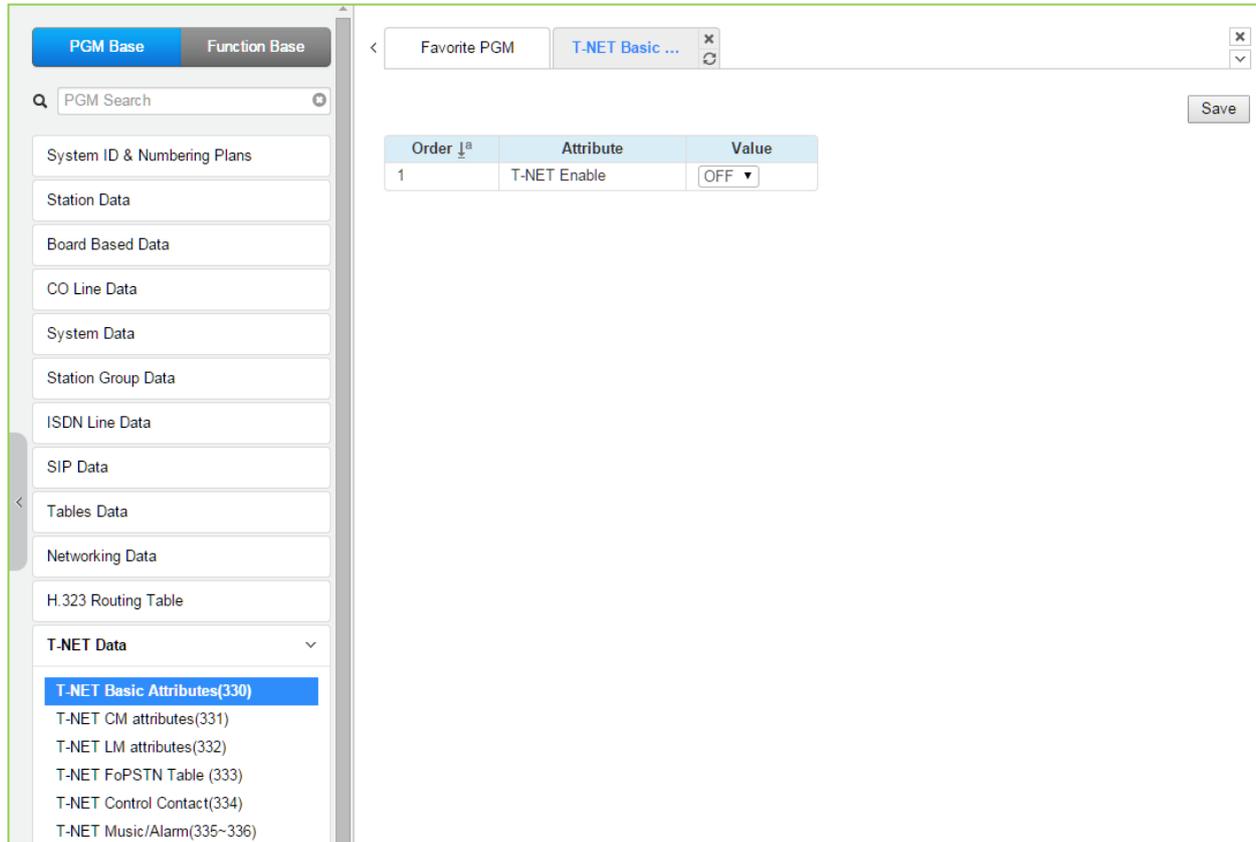Selecting H.323 Basic Attributes will display the data entry page. Click **[Save]** button after changing Value.
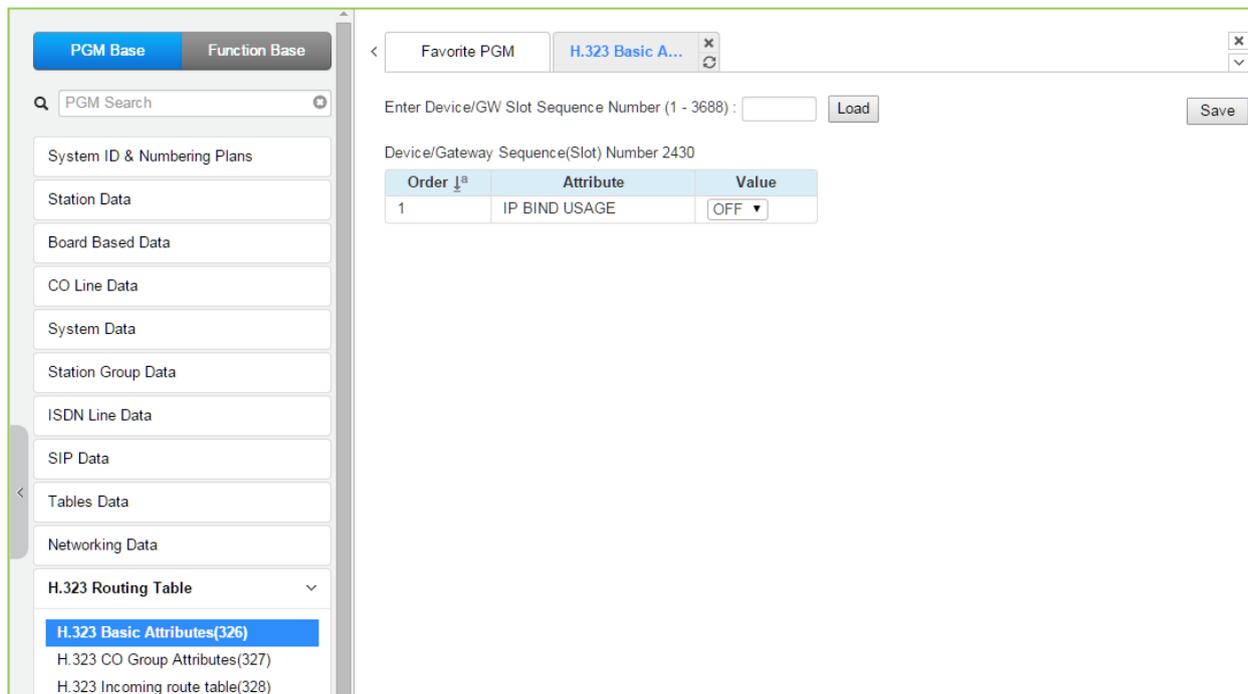
Figure 6.11.1-1 H.323 Basic Attributes

H.323 Signaling can be operated with each VOIM or UCP.

Table 6.11.1-1 H.323 Basic Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
IP Band Usage	If it is set to ON, system allows H.323 signaling with system IP address. Or, each VOIB IP Address can be used.	OFF ON	OFF

6.11.2 H.323 CO Group Attributes -PGM 327

Selecting H.323 CO Group Attributes will display the data entry page. Click **[Save]** button after changing Value.

Figure 6.11.2-1 H.323 CO Group Attributes

H.323 Signaling can be operated with each VOIB board or UCP.

The VOIP channels are used for Distributed Networking, access to SIP or H.323 networks and for remote iPECS devices. When the standard H.323 VoIP protocol is employed for an external VoIP call, several attributes of these channels can be assigned. The H.323 call set-up mode and tunneling (H.245 Encapsulation) can be established.

Also for H.323 support, a RAS (Registration, Admissions and Status) channel can be defined. The RAS channel IP addresses (uni-cast and multi-cast) as well as the IP port Numbering Plan and other H.323 set-up characteristics are defined.

Table 6.11.2-1 H.323 CO Group Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
H.323 Setup mode	H.323 IP calls can be set-up using the H.323 normal or Fast Start mode.	Fast Norm	Fast
H.323 Tunneling mode	H.323 IP calls can be set-up using the H.245 encapsulation (Tunneling).	OFF ON	ON

Table 6.11.2-1 H.323 CO Group Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
H323 Early Media (early H245)	This feature is the ability of two user endpoints to communicate before call is actually established in normal call mode. This feature is not used when 'H323 Setup Mode' is 'Fast'. * Setup: Caller party tries to open early media on receiving the Setup message. * Proceeding: Calling party tries to open early media on receiving the Proceeding message. * Alerting: Calling party tries to open early media on receiving the Alerting message.	Setup, Proceeding, Alerting	
H.323 DTMF Path	During a connection, DTMF Digits can be sent in-band or out of band (H.245).	IN, OUT, 2833	IN
TCP Keep Alive	The system will send a polling message every 75 seconds to assure the status of the TCP connection.	OFF ON	ON
TCP No Delay	Normally small chunks of TCP (H323) messages are combined into one packet and be sent to remote party to improve the efficiency of network. Sometimes this action yields the H323 signaling problem when system is behind NAT router and there is H323 ALG is running. 'OFF' value can be a solution to overcome the problem.	OFF ON	OFF
Sending Setup Ack message	System provides SETUP ACK message when there is no 'sending complete IE' in SETUP message.	OFF ON	OFF
Name Service option	Basically the calling party name is transferred in QSIG messages. This field is an additional way to send the calling party name. In some PBX systems, the Q.931 Display IE field is used to transferring the calling party name and this value can be used as an interoperating solution for displaying the calling party name.	OFF, Display IE (CISCO)	OFF
Gatekeeper Attributes			
RAS Usage	Determine whether VOIU/VOIB Board will be used as a Gatekeeper.	OFF ON	OFF
RAS MultiCast IP Port	Multi-cast IP Port for RAS Information of Gatekeeper.	1~65535	1718
RAS MultiCast IP Address	Multi-cast IP address for RAS Information of Gatekeeper.	IP Address	224.0.1.41
RAS UniCast IP Port	Uni-cast IP Port for RAS Information of Gatekeeper.	1~65535	1719
RAS UniCast IP Address	Uni-cast IP address for RAS Information of Gatekeeper.	IP Address	82.134.80.2
RAS Keep Alive Time	The time between exchange of RAS Information between GK and VOIB/VOIU.	001-999 (SEC)	120
RAS IIR Multiplier Ratio	This feature is used when H323 call is routed by RAS gatekeeper. Keep alive interval time between the system and gatekeeper is provided by gatekeeper on registration time. This value of ratio is used to change the keep alive interval time from gatekeeper.	10~100%	80
RAS Number Plan	The numbering plan for Calling Number in RAS	Max. 23 digits	9

Table 6.11.2-1 H.323 CO Group Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Prefix	Setup.		
RAS Light RRQ usage	The Gatekeeper ID (This can be programmed only via WEB Admin).	OFF ON	OFF
RAS Gateway ID	The system can be assigned to use the simple RRQ (Registration Request) message (ON) or the full RRQ message (OFF).	Max. 128 characters	
Fail over Usage	The H.323 call will be failover to another line (FAIL OVER USAG: ON).	OFF ON	OFF
Call Setup No Response time	The H.323 call will be failover. The time will be set.	0, 3-15 (Sec.)	5
Fail over CO Group number	If the H.323 call will be failover, assign another CO group.	1-200	

6.11.3 H.323 Incoming Route table -PGM 328

Selecting H.323 Incoming Route table will display the data entry page. Click **[Save]** button after changing Value.
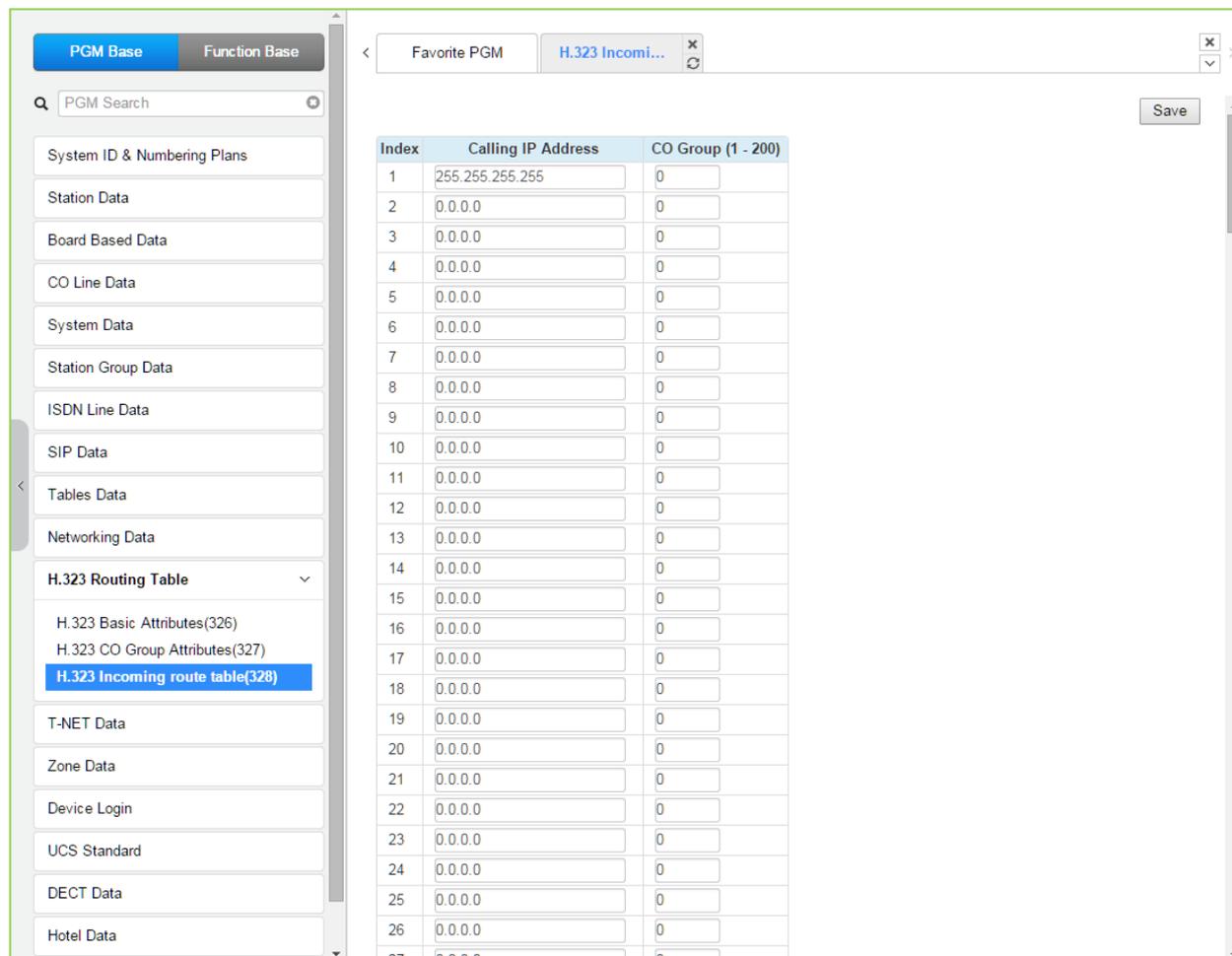
Figure 6.11.3-1 H.323 Incoming Route table

To get the direct H.323, the From IP-Address and 'incoming CO Group number' to be routed should be assigned.

Table 6.11.3-1 H.323 Incoming Route table

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Calling IP Address	IP address associated with H.323 incoming calls. 255.255.255.255 is used when external call cones from unknown IP Address which is not listed in this table entry.		0.0.0.0
CO Group	CO group number associated with H.323 incoming calls.	1~200	0

6.12 T-NET (Central Control Networking) Data

Selecting the T-NET Data program group returns the sub-menu displayed in the left frame as shown in the following figure.
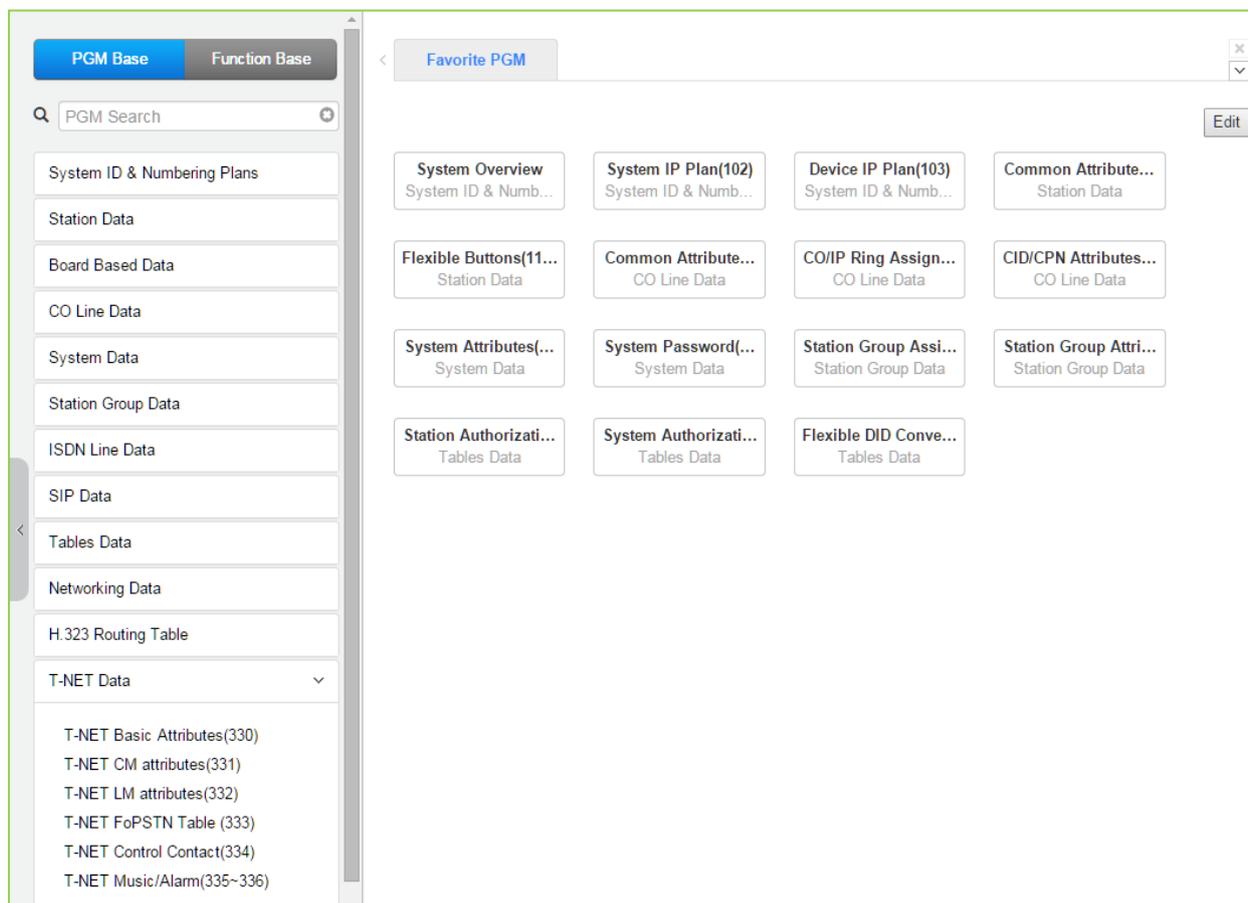
Figure 6.12-1 T-NET Data Main Page

In a Centralized Control T-NET (Transparent Network), remote devices may be registered to a Central UCP Module (CM) and to a Local UCP Module (LM). In this way, the CM maintains control of the remote device. Should the WAN connection between an LM and CM fail (polling error), the LM will initiate operational control of the locally registered devices. Calls between the systems (CM and LM) can automatically shift to PSTN Modules registered with the LM for Fail-over operation. The configuration and characteristics of LMs and CM are configurable as is Fail-over operation.

6.12.1 T-NET Basic Attributes -PGM 330

Selecting T-Net Basic Attributes will display the data entry page. Click **[Save]** button after changing Value.

Figure 6.12.1-1 T-NET Basic Attributes

Each UCP in a Central Control network environment must be enabled for T-NET operation in order to function as part of the network.

6.12.2 T-NET CM Attributes -PGM 331

Selecting T-Net CM Attributes will display the data entry page. Click **[Save]** button after changing Value.
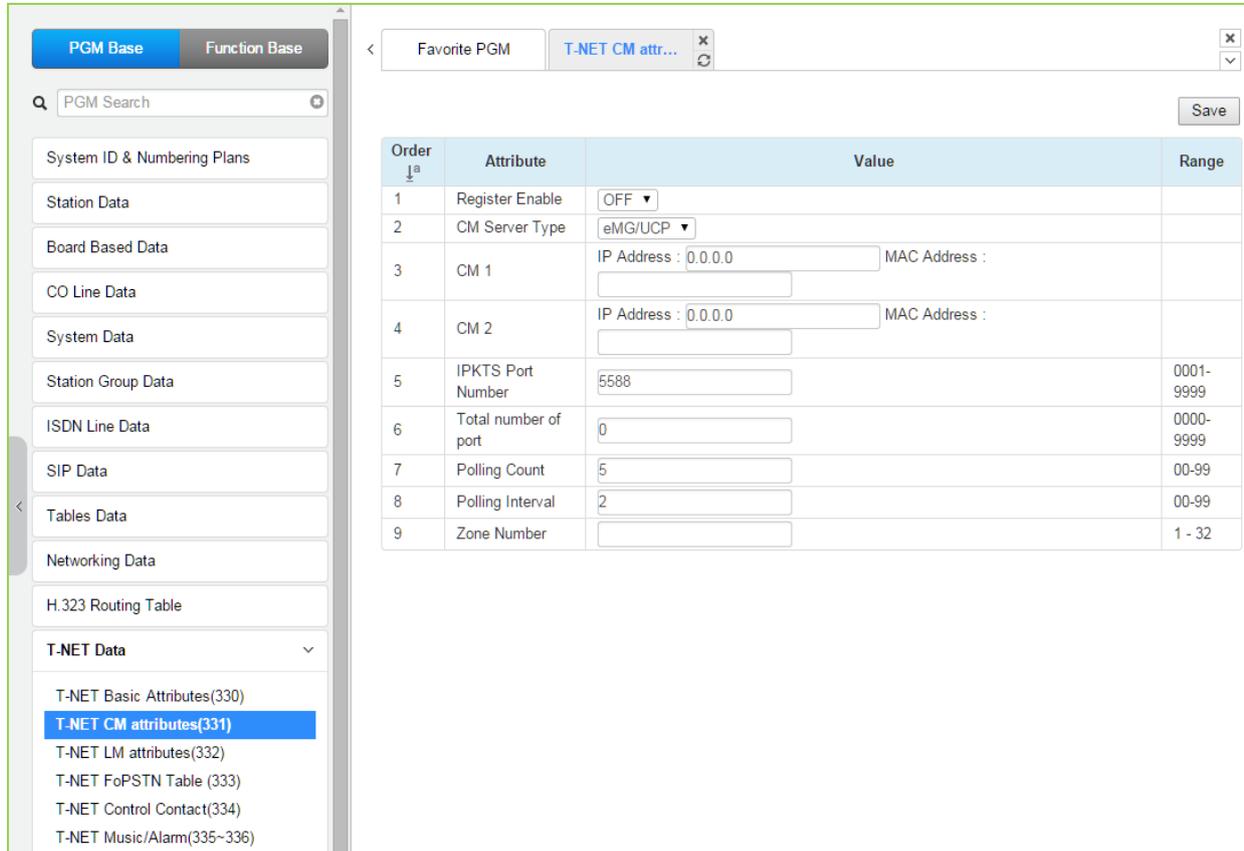
Figure 6.12.2-1 T-NET CM Attributes

Each LM (Local UCP), which is part of a Central Control Network, must be defined with the IP Address of the CM (Central UCP). The LM configuration data is 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 to or less than the ports defined in the CM for the LM, see PGM 332, in order to register properly.

Table 6.12.2-1 T-NET CM 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
CM server type	Assign the type of CM server; iPECS UCP or iPECS CM.	eMG/UCP, CM	eMG/UCP
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

Table 6.12.2-1 T-NET CM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
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	5
Polling interval	This field defines the interval time between LM to CM polling attempts.	00-99	2
Zone Number	Zone number can be assigned to Device or GW.	1-32	

6.12.3 T-NET LM Attributes -PGM 332

Selecting T-Net LM Attributes will display the data entry page. Click **[Save]** button after changing Value.
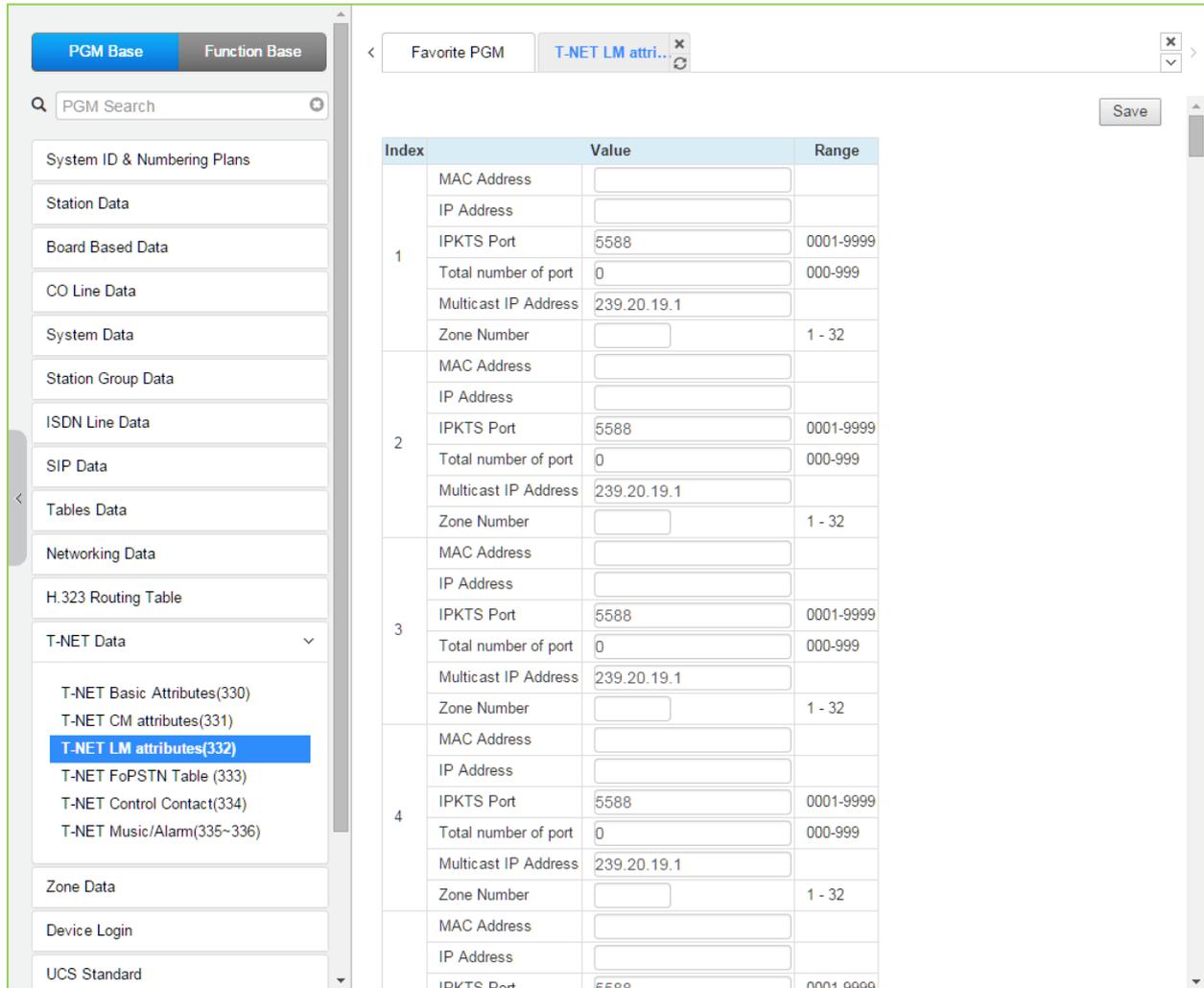
Figure 6.12.3-1 T-NET LM Attributes

The CM (Central UCP) must be programmed with the MAC and IP address of each LM (Local UCP) in the Centralized Control network as well as the maximum configuration of each LM. Up to 100 Local UCPs (LMs) may be defined and configuration entered. The port counts define the ports that are allocated in the CM database for use by devices registered to the LM. The number of ports defined in the database for each LM in PGM 331, must be equal to or less than the ports defined in the CM for the LM, in order to register properly.

Table 6.12.3-1 T-NET LM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Mac Address	This field defines the MAC address of the LM that will be part of the T-NET environment and is used by the CM for authorization.	MAC address	
IP Address	This field is the IP address of the LM.	IPv4 address	

Table 6.12.3-1 T-NET LM ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
IPKTS Port	In the T-NET 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 from the CM for devices attached to the LM. This value must be equal to or more than the port count defined in the LM.	000-999	
Multicast IP address	This field defines the multicast IP address that could be used in T-NET branch site.	IPv4 address	
Zone Number	Zone number can be assigned to Device or GW.	1-32	

6.12.4 T-NET FoPSTN table -PGM 333

Selecting T-NET FoPSTN Table will display the data entry page. Enter an index range then click **[Load]** to modify the data. Click **[Save]** button after changing Value.
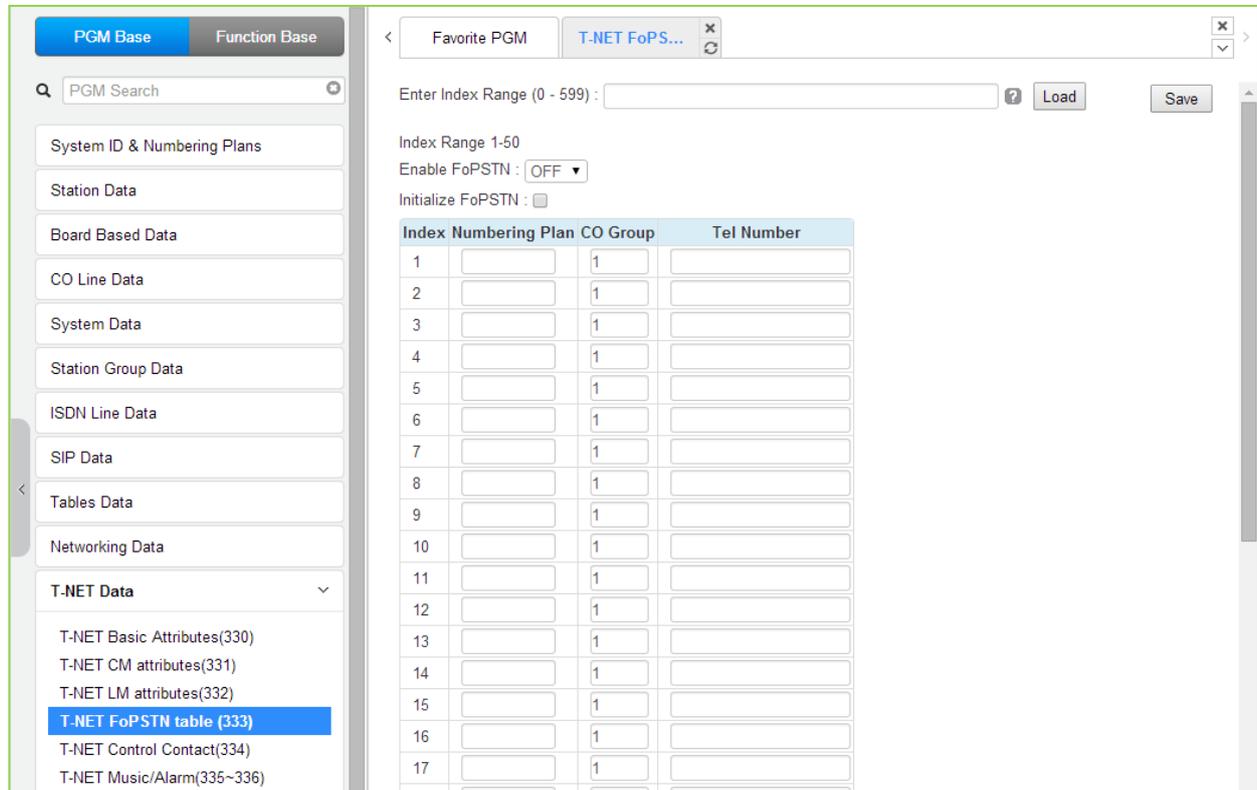
Figure 6.12.4-1 T-NET FoPSTN Table

The Fail-over function allows the systems in T-NET environment to complete calls between systems over a PSTN (analog or digital) line should the WAN connection to the CM fail. A CO gateway Module must be registered to the LM for local CO services. Users may call others in the normal manner and the call is routed over CO facilities to the remote CM. When calls are directed to a DID line at the receiving system, the system will select a line from the assigned CO Group and dial the Tel Number with the station number dialed as the trailing digits.

Table 6.12.4-1 Fail Over to PSTN ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Numbering Plan	Station numbers associated with the remote system. A range can be indicated by using "*" to indicate the range. Example: 21** covers 2100 to 2199.	Station number	
CO Group	This field defines the CO Group of the local system that will be used to place calls to the stations entered in the Fail Over Numbering Plan above, should WAN failure occur.	CO/IP group	
Tel Number	This field defines the telephone number the system should dial to place a call to the stations entered in the FO Numbering Plan, should WAN failure occur. An "*" may be entered as a wild card to indicate insertion of the dialed station number.	24 digits	

6.12.5 T-NET Control Contact -PGM 334

Selecting T-NET Control Contact will display the data entry page. Enter the T-NET range from the T-Net LM Attributes then click **[Load]** to modify the data. Click **[Save]** button after changing Value.
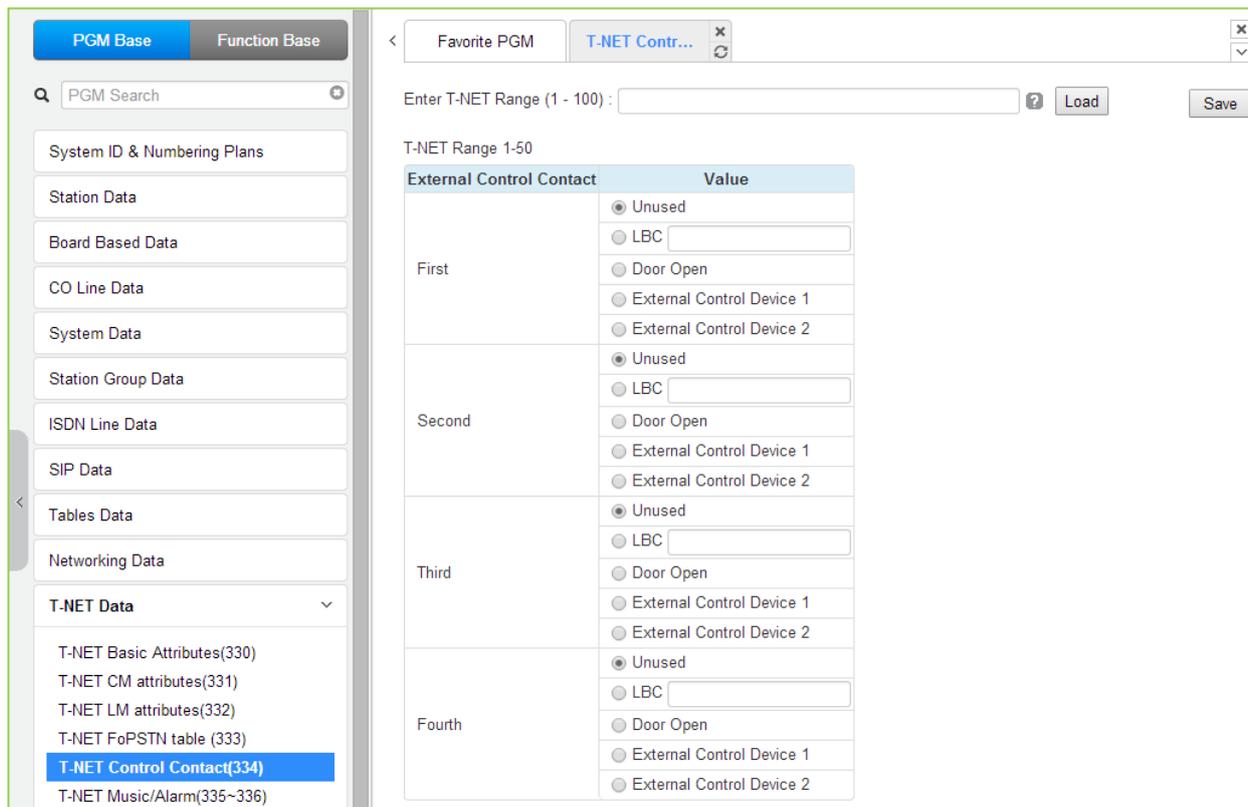
Figure 6.12.5-1 T-NET Control Contact

Each LM incorporates relay contacts that can be employed as a Door Lock Release. The contact activates a third party Door Lock Release mechanism activated by dialing the Door Unlock code at a local station.

6.12.6 T-NET Music/Alarm -PGM 335 ~ 336

Selecting T-NET Music/Alarm will display the data entry page. Enter the T-NET range from the T-Net LM Attributes then click **[Load]** to modify the data. Click **[Save]** button after changing Value.

Figure 6.12.6-1 T-NET Music/Alarm

To minimize WAN traffic, the CM does not provide BGM/MOH to an LM. The LM employs local BGM and MOH facilities to reduce traffic load on the WAN and IP channel processors. The LM uses IP Multicast for local BGM and MOH transport. In addition, the Alarm contacts of the LM can be defined for use as a local alarm or doorbell.

6.13 Zone Data

Selecting the Zone Data program group returns the sub-menu displayed in the following figure.
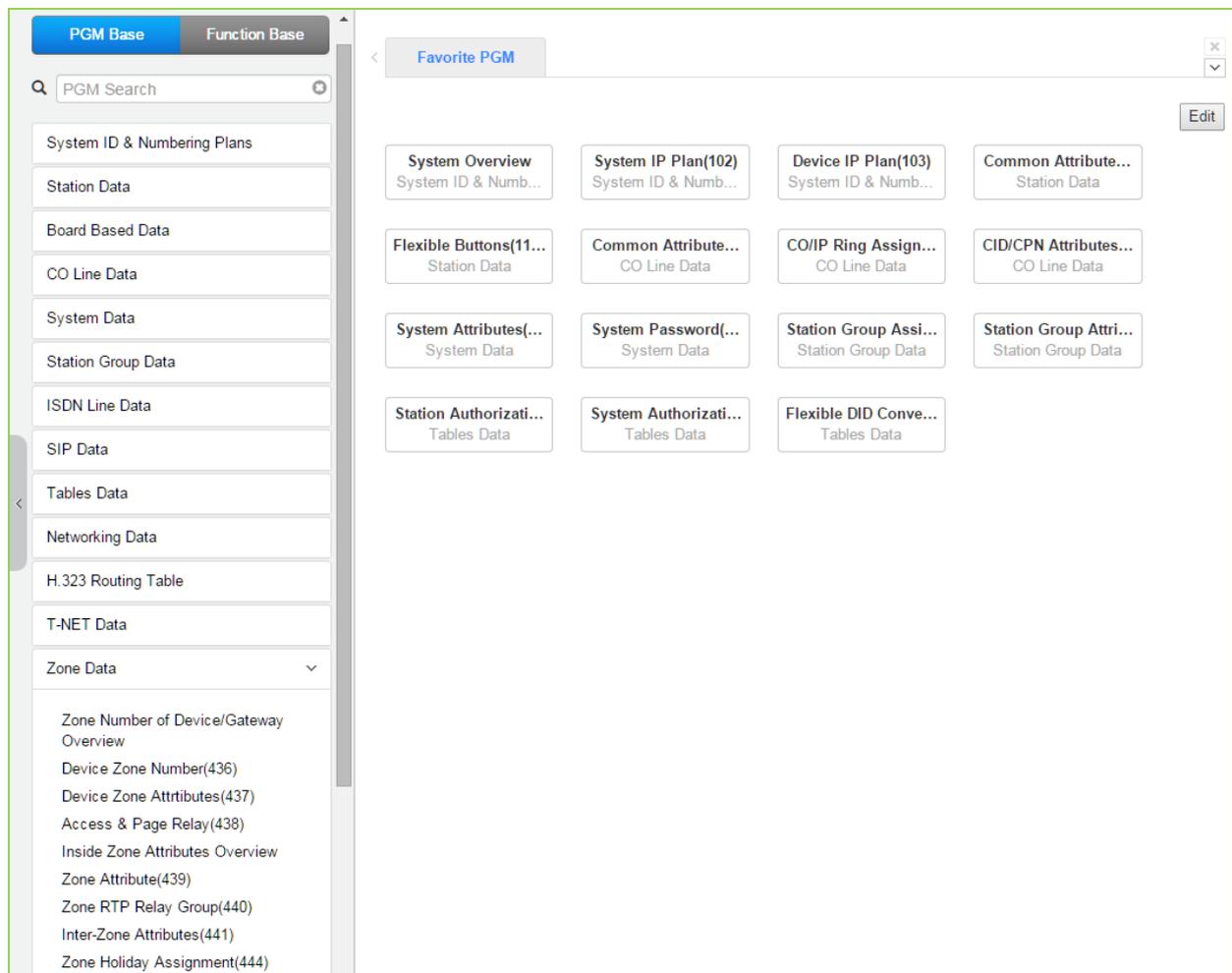
Figure 6.13-1 Zone Data Main Page

6.13.1 Zone Number Overview

Selecting Zone Number Overview will display the data entry page.

Figure 6.13.1-1 Zone Number Overview

Zone data is a tool employed to manage the characteristics of groups of devices under the control of an UCP. Such devices can be grouped to a Zone to define common characteristics including Country Code, DSCP, RTP packet handling, etc. Common attributes are defined at the device, Zone and Inter-zone level. Device settings have priority over system and Zone settings, while Zone settings have priority over system settings.

Generally, transport of RTP packets should be a peer-to-peer communication over either a LAN or VPN. If iPECS devices are separated by a NAPT server or direct peer-to-peer communications is not available, packet relay must be employed to assure communication. In packet relay, RTP packets are received by a local VoIP channel (UCP or VOIM), which is under control of the UCP, and the IP address is translated from a public to the device's private address. The VoIP channel implements a secure channel using IPSec protocol. Devices can be assigned as part of an "RTP Relay group" to use the same VoIP channels to implement relay of RTP packets. Packet relay groups also provide for conversion of multi-cast packets from the UCP to uni-cast and back again at the group level to multi-cast. Note packet relay require a VoIP channel be available locally for each simultaneous call that requires packet relay.

6.13.2 Device Zone Number – PGM 436

Selecting the Device Zone Number will display the Zone Number data input page. Enter the Sequence number range for the devices, refer to section 6.1.5 Device IP Address Plan - PGM 103, and click **[Load]** to assign a Zone number for the device. Click **[Save]** button after changing Value.

Figure 6.13.2-1 Device Zone Number

Device Zone Number assigns a device to one of up to 32 specific Zones.

6.13.3 Device Zone Attributes – PGM 437

Selecting the Device Zone Attributes will display the Device Zone Attributes data input page. Enter the Sequence number Range (refer to section 6.1.5 Device IP Address Plan - PGM 103) and click **[Load]** to assign Zone attributes for the device. Use the check boxes to indicate which attributes to modify; data for checked attributes is stored for the entire range of devices when saved.
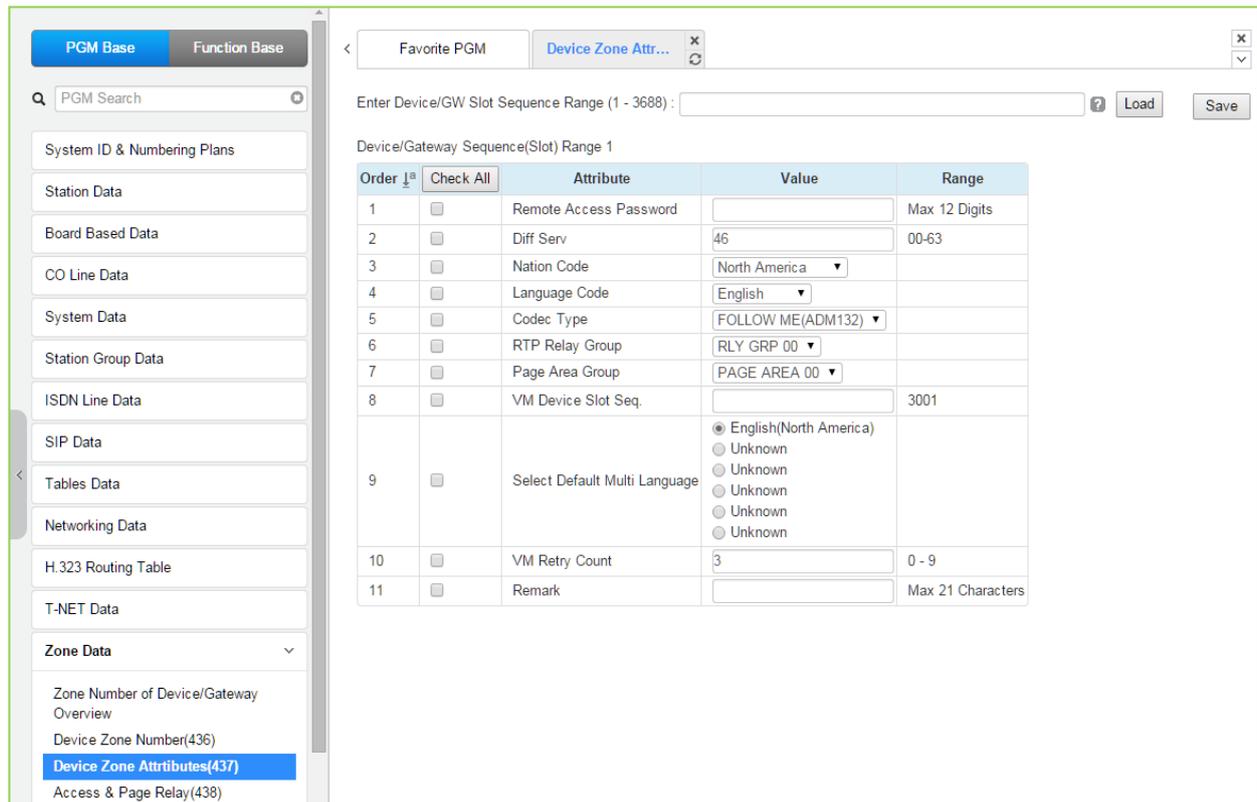
Figure 6.13.3-1 Device Zone Attributes

Device Zone Attributes define characteristics specific to the device including the registration password, Diff Serv Code Point, Nation, etc. In addition, Zone characteristics set at the device level take precedence over characteristics for the Zone Attributes. While a Zone may incorporate up to 15 different RTP packet Relay Groups, for clarity a single RTP Relay Group should be used within a Zone.

Table 6.13.3-1 DEVICE ZONE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Remote Access Password	For a standalone remote device, the password assigned in device for Remote for registration	Max. 12 digits	None
Diff Serv	This parameter defines the Diff Serv Code Point for IP packets from the device. Note the system Code Point will take precedence.	0-63	4
Nation Code	The device can be located in a different country then the system. Available selections are given in Table 6.1.1-1.		
Language Code	Each device can be assigned a local language or English as the LCD Display Language.		English

Table 6.13.3-1 DEVICE ZONE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Codec Type	The codec selection method can be defined as device (board) based (Follow-Me), see PGM 132, or based on the codec type assigned to the Zone.	Follow Me (PGM132 Board Based Follow Zone	Board based
RTP Relay Group	This attribute defines the RTP Relay group employed by the device, 00-no relay. Zone parameters define the VoIP device to employ.	00-15	00
Page Area Group	The Paging Area Group identity is employed to determine when multicast to unicast conversion is needed to relay paging data across the network using a VoIP channel of a VOIU/VOIM or Paging Agent (VoIP channel of an iPECS LIP Phone).	00-15	00 (same paging relay area)
VM Device Slot Sequence	The VSF gateway (VSF or UVM) used to support Voice Mail for a device is defined. The VSF/UVM must be under control of the same UCP as the device.	Max. 4 Digits	
Select Multi Language	The recorded language selection prompt is played to the user when accessing the built-in AA/VM. The system supports up to six languages.	1 ~ 6	1
VM Retry Count	The user may select an available language. If the language is unavailable, the user may attempt to enter a valid language type based on this retry counter. If the user cannot enter a valid language, the announcement is provided in the default language.	0-9	3
Remark	Descriptive information to help installer/programmer in identifying the device Zone, i.e. Branch1.	Max. 21 characters	

6.13.4 Access & Page Relay – PGM 438

Selecting Access & Page Relay will display the Access & Page RTP packet relay data input page. Enter the Sequence number Range (refer to section 6.1.5 Device IP Address Plan - PGM 103) and click **[Load]** to define packet relay characteristics for Paging between Zones. Use the check boxes to indicate which Zones to modify; data for checked Zones is stored for the entire range of devices when saved.
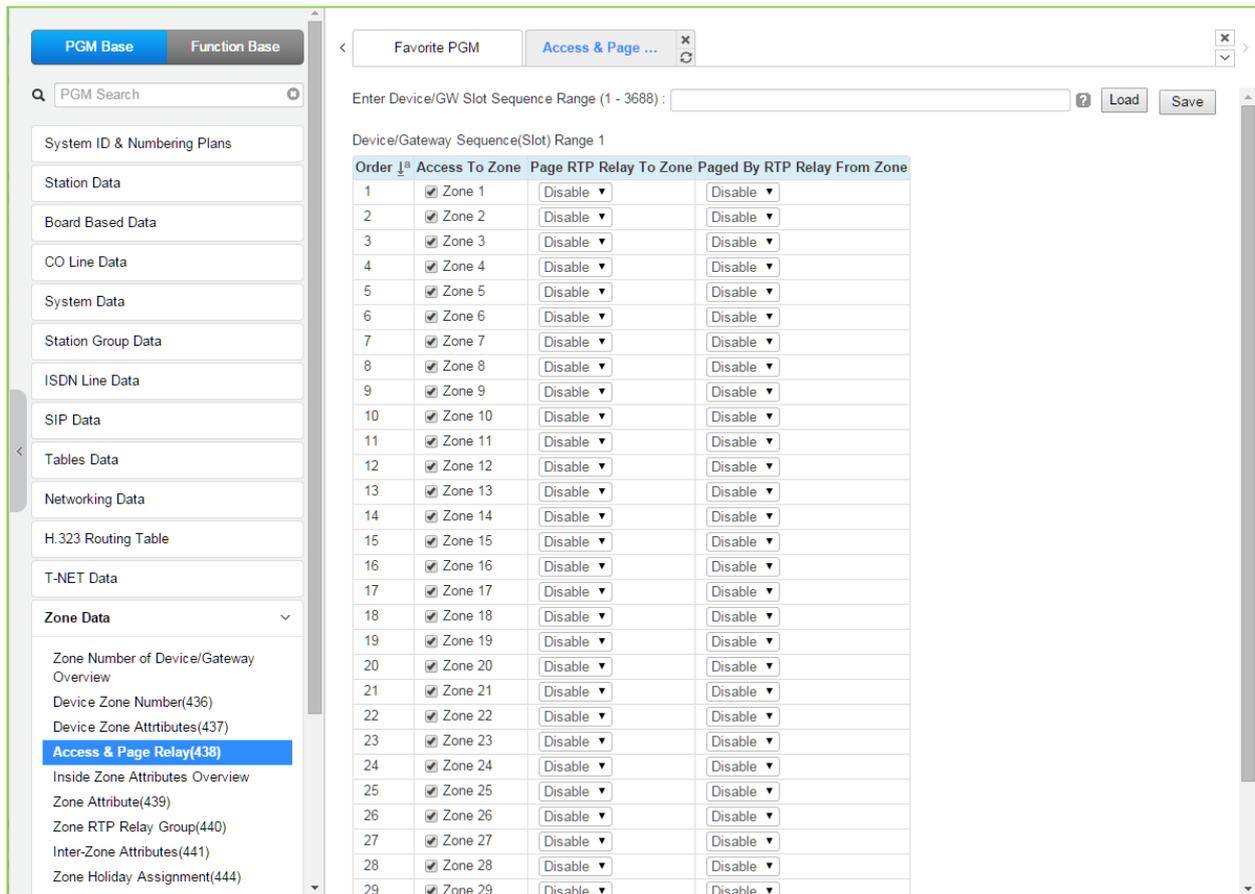
Figure 6.13.4-1 Zone Page RTP Packet Relay

iPECS protocol employs multicast to send RTP packets to multiple devices at one time. This improves efficiency and lowers traffic by sending a single multicast packet to multicast group members. In particular, paging employs this multicast technique. However, since multicast is not commonly supported outside of the LAN, unicast must be used to transport such signals between routers. At the receiving Zone, a local VoIP channel receives the unicast signal from the controlling UCP. Then the VoIP channel converts the packet address to a multicast signal for delivery to devices in the same RTP Page Relay Group.

Table 6.13.4-1 Zone Page RTP Packet Relay

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Page RTP Relay to Zone	The device can be configured to relay RTP packets for paging to other zone members, Paging Rely.	Disable Enable	Disable
Page By RTP Relay From Zone	The device can be configured to receive relayed packets from the assigned Zones(s).	Disable Enable	Disable

6.13.5 Inside Zone Attributes Overview

Selecting Inside Zone Attributes Overview returns the data display page.

Figure 6.13.5-1 Inside Zone Attributes Overview

6.13.6 Zone Attributes – PGM 439

Selecting the Zone Attributes will display the Zone Attributes data input page. Enter the desired Zone number range and click **[Load]** to assign Zone characteristics. Click **[Save]** button after changing Value.
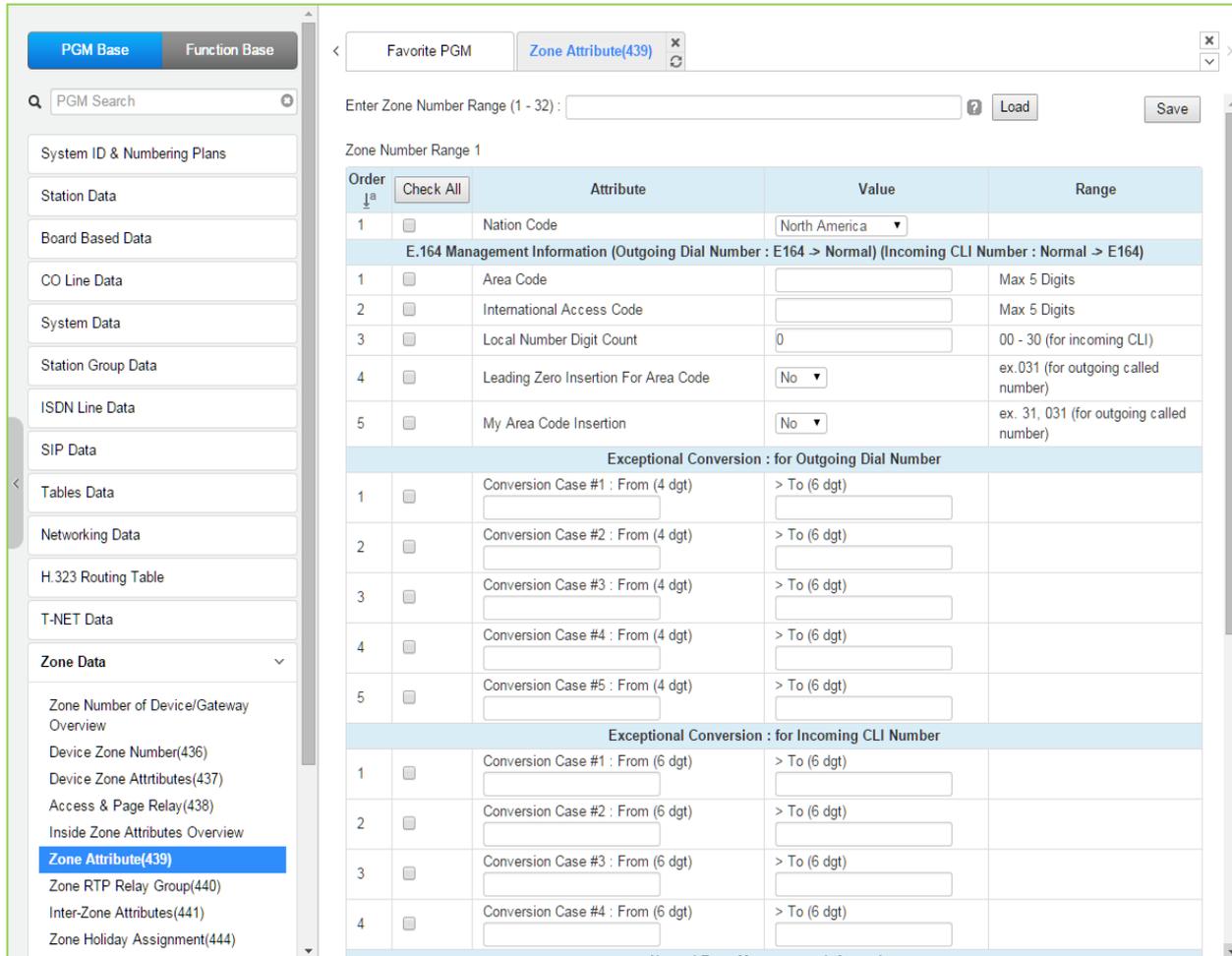
Figure 6.13.6-1 Zone Attributes

The Nation code, codec and VSF/UVM assigned for a Zone will be employed by all devices in the Zone unless a different entry is made in Device Zone data. If the Device Zone data is default or assigned by the UCP, the Zone data will take precedence.

The Zone Attributes define when and which VoIP channels to use for RTP packet relay. Local VoIP channels are assigned to perform the packet relay function and the use can be defined as "if needed" or use the assigned RTP Relay Group. For "if Need", the UCP will employ the IP KTS STUN protocol to determine when packet relay is required. If assigned "RTP Relay Group", packet relay will always be employed for RTP packet receipt.

Table 6.13.6-1 ZONE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Nation Code	Nation code of devices in the zone. Available	Country code	North America

Table 6.13.6-1 ZONE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	selections are given in Table 6.4.1-2.		
E.164 Management Information (Outgoing Dial Number: E164/Normal) (Incoming CLI Number: Normal/164)			
Area Code	Assign the Area Code that is used to convert the CLI format (normal <-> E.164)	Max 5 Digits	
International Access Code	Assign the International call access code that is used to convert the CLI format (normal <-> E.164)	Max 5 Digits	
Local Number Digit Count	If the incoming CLI exceeds this digit length, the call is consider long distance call.	00-30	0
Leading Zero Insertion for Area code	For an outgoing call, the system can insert a zero ('0') in front of area code.	No Yes	No
My Area Code Insertion	For an outgoing call, the system can insert my area code in the CLI.	No Yes	No
Exceptional Conversion : for Outgoing Dial Number			
Conversion Case #1			
Conversion Case #2	The first digits, up to four (4), dialed by the user can be converted to a digit string of up to six (6) digits and may include a plus (+) sign. For example if the "From" digits are 00 and the "To" digits are +820, if the user dials 002233432, the system converts the number to +820 2233432.	From (4-digits) and To (6-digits)	
Conversion Case #3			
Conversion Case #4			
Conversion Case #5			
Exceptional Conversion : for Incoming CLI Number			
Conversion Case #1			
Conversion Case #2	The first digits, up to four (4), received as the CLI can be converted to a digit string of up to six (6) digits and may include a plus (+) sign. For example if the "From" digits are 00 and the "To" digits are +820, if the user dials 002233432, the system converts the number to +820 2233432.	From (4-digits) and To (6-digits)	
Conversion Case #3			
Conversion Case #4			
Normal Zone Management Information			
Language Code	The Language code for devices in the zone is assigned.	Language Code	English
Codec Type	The codec employed by devices in the Zone is defined.	System codec G.711 G.723.1 G.729 G.722	System codec
RTP Relay Rule	Assigns when to use the packet relay function, with "If Need" the UCP will automatically determines when to use packet relay, while "Forced to Relay" will always implement packet relay for RTP packets.	If need Relay Group Forced to Relay	If need
1st RTP Relay Device Slot Seq.	The local VoIP channels that will be used to implement packet relay for devices in the Zone is	Sequence Number	

Table 6.13.6-1 ZONE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	assigned.		
2nd RTP Relay Device Slot Seq.	Back-up VoIP channels for RTP packet relay use in the Zone can be assigned.	Sequence Number	
VM Device Slot Sequence	The UVMU/UVM used to support Voice Mail for devices in the Zone is assigned. The UVMU/UVM must be under control of the same UCP as the device being assigned.	Sequence Number	
Remark	Descriptive information to help installer/programmer in identifying the device Zone, i.e. Branch1.	Max. 21 characters	
Display time zone	The time & date displayed at the station are based on the time zone selected or the system time.	Time zone	System time
Max Total Inter Zone Connections	This parameter may be used to limit the number of maximum Inter Zone connections. When this value is set to 0, there is no limit on the total Inter Zone connections.	0-2400, 0: No limit	0
Max Outgoing CO Calls	This field may be used to limit the number of outgoing CO/IP calls from the Zone. When this value is set to 0, there is no limit on the number of outgoing CO/IP calls.	0-998, 0: No limit	0
Zone MOH Assign	A Music source can be assigned for calls placed on hold by devices in this zone.	Refer to Sys Hold, Internal/External Music1, External Music2, VSF MOH, SLT MOH1~5, VSF MOH2~3	Refer to Sys Hold
Emergency Call Notify Station	This field assigns the station to receive notification of an Emergency call from a device in the Zone.	Station number	
Emergency Call Notify 2nd Station	This field assigns a secondary station to receive notification of an Emergency call from a device in the Zone..	Station number	
DCOB Fault Notify Station	This field assigns the station to receive notification of a DCOB (Digital CO Board) fault.	Station number	
SIP Registration Fault Notify Station	This field assigns the station to receive notification of a SIP Trunk registration fault.	Station number	
Daylight Saving Usage	If option is "System DST mode", the zone will follow the system DST mode (System Date &Time (178)). If option is "OFF", the zone will not apply the Daylight Saving Time.	System DST mode, OFF	System DST mode

6.13.7 Zone RTP Relay Group – PGM 440

Selecting the Zone RTP Relay Group will display the RTP Relay ON/OFF data input page. Enter the desired Zone and Group numbers and click **[Load]** button to assign Zone characteristics. Check the boxes to indicate if RTP Relay is required between the Relay groups in the Zone. Click **[Save]** button after changing Value.

Figure 6.13.7-1 Zone RTP Packet Relay Groups

While it is strongly recommended that a Zone only have a single RTP Relay Group, up to 15 Groups can be assigned to a Zone. Devices in an RTP Relay Group should have common requirements for packet relay use. In some situations, it may be necessary to implement packet relay to groups in a Zone. Note that when "if Need" is assigned as the RTP Relay Rule in the Zone Attributes, assignments here are ignored.

6.13.8 Inter Zone Attribute – PGM 441

Selecting the Inter Zone Attribute will display the data input page. Enter the desired Source and Destination Zone number and click **[Load]** to assign Zone characteristics. Use the check boxes to indicate the attributes to modify. Click **[Save]** button after changing Value.
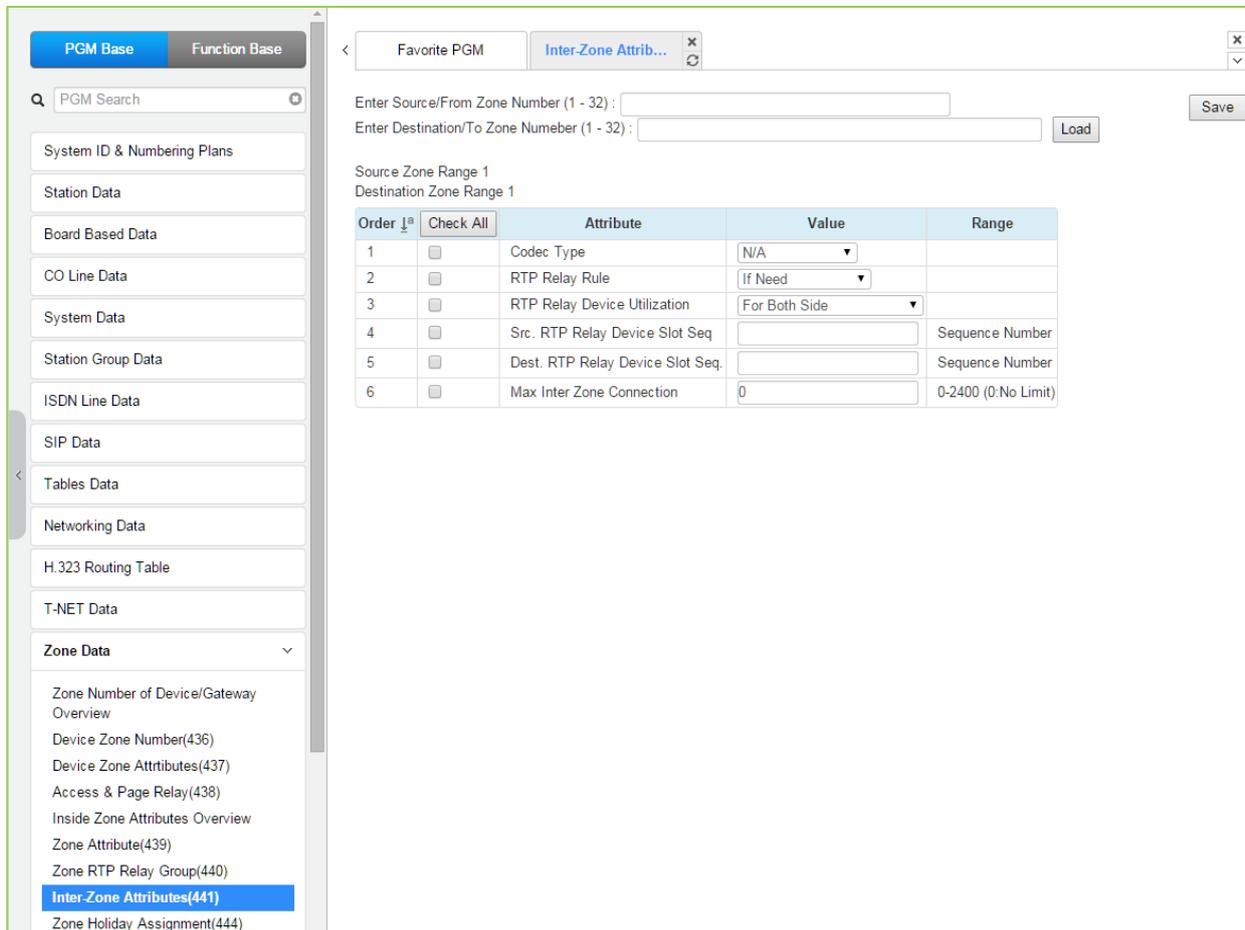
Figure 6.13.8-1 Inter-Zone Attributes

Inter Zone Attributes define RTP packet relay treatment for communications between devices in different Zones.

Table 6.13.8-1 INTER-ZONE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Codec Type	Assigns the codec employed for inter-Zone communications.	N/A, System Codec, G.711, G.723.1, G.729, G.722	NA
RTP Relay Rule	The rule defines when to use RTP relay between the Source and Destination zone. With "If Need", the UCP will automatically determine when to use packet relay, while "Forced to do" will always implement packet relay for RTP packets between the Zones.	If need Always not Forced to do	If need

Table 6.13.8-1 INTER-ZONE ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
RTP Relay Device Utilization	The assigned Source VoIP channels may be used for both sides of the communication or separately only for a device in the Source Zone. The Destination channels are then used as back-up channels or only for devices in the Destination Zone	For Both Side Separate (SRC to DEST)	For Both side
Src. RTP Relay Device Slot Seq.	The Sequence number associated with the VoIP channels in the Source Zone that handle Packet Relay.	Sequence Number	
Dest. RTP Relay Device Slot Seq.	The Sequence number associated with the VoIP channels to handle packet relay for the Destination Zone is defined. When Utilization is Separate the channels are used for devices in the destination Zone, otherwise they are used as back up for both sides.	Sequence Number	
Max Inter Zone connection	This field determines the maximum Inter Zone connections between the source zone and destination zone. When this value is set to 0, there is no limit on the number of connections between the zones.	0-2400 (0: no limit)	0

6.13.9 Zone Holiday Assignment - PGM 444

Selecting the Zone Holiday Assignments will display the data input page. Enter the desired Source and Destination Zone numbers range and click **[Load]** to assign Zone holiday. Click **[Save]** button after changing Value.
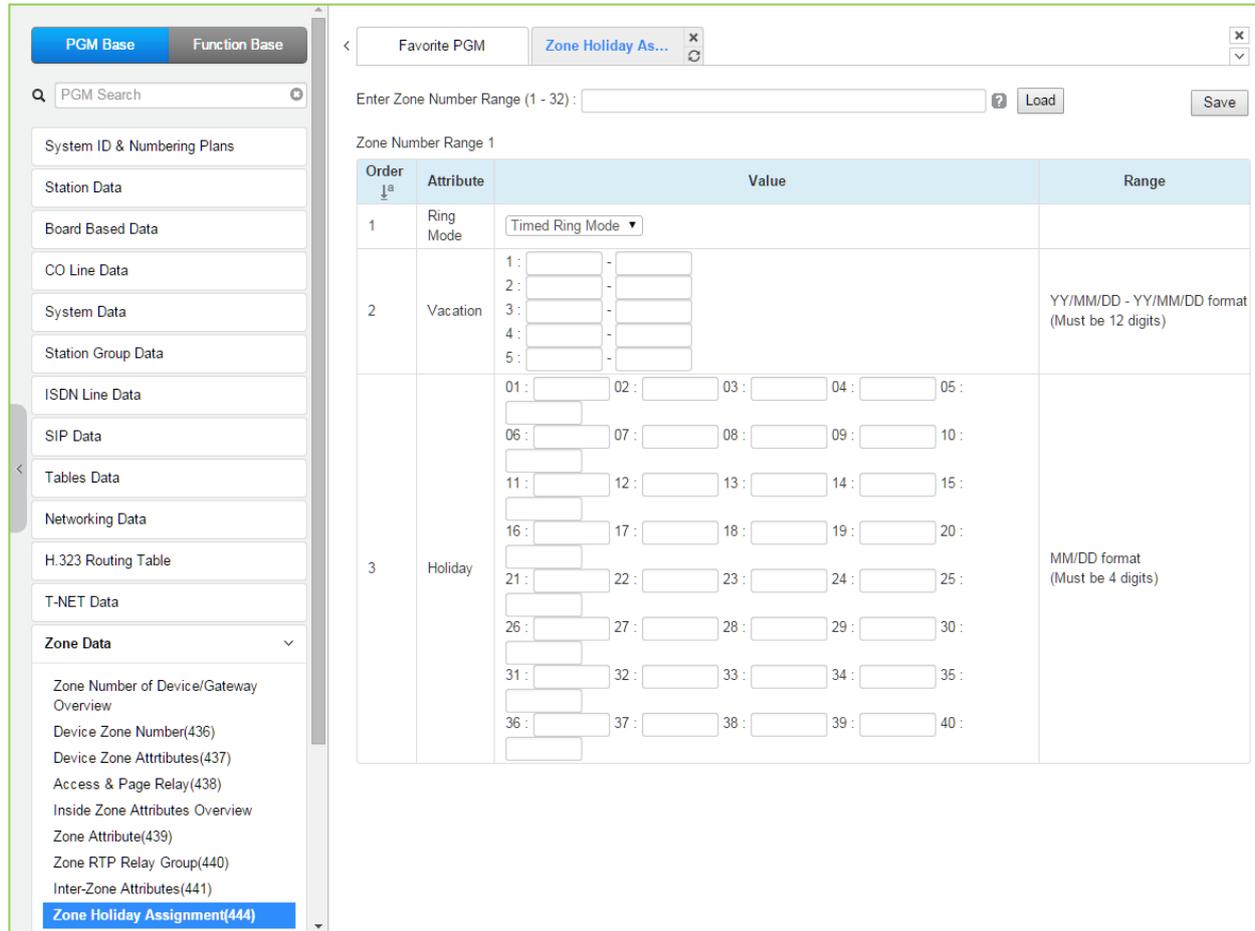
Figure 6.13.9-1 Zone Holiday Assignment

Holidays and vacation day intervals for each Zone can be established to define the Service mode (Day, Night, and Timed). Up to 40 holidays and five (5) vacation intervals can be defined.

Table 6.13.9-1 ZONE HOLIDAY ASSIGNMENT

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Ring Mode	Select the desired Service mode for the Holiday or Vacation.	Day Ring Mode Night Ring Mode Timed Ring Mode N/A	Timed Ring Mode
Vacation	Five ranges may be entered for vacation periods, enter the start and end dates as YY/MM/DD – YY/MM/DD.	Must be 12 digits	None
Holiday	Each Zone can have up to 40 holidays assigned, entering the date as MM/DD.	Must be 4 digits	None

6.14 Device Login

Selecting the Device Login Data program group returns the sub-menu displayed in the left frame as shown in the following figure.
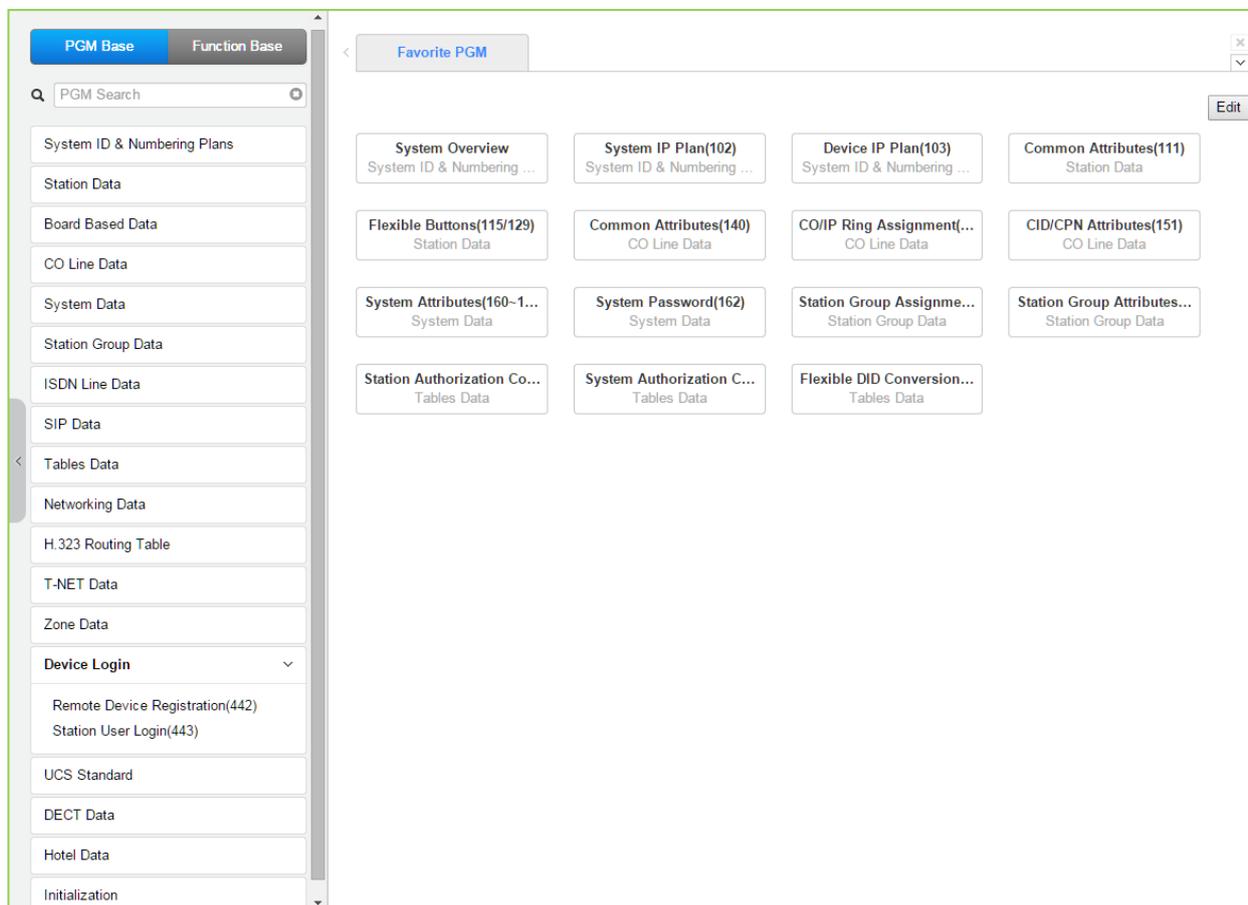
Figure 6.14-1 Device Login Data Main Page

6.14.1 Remote Device Registration – PGM 442

Selecting the Remote Device Registration will display the Remote Registration Table data input page. Enter a Table index range and select **[Load]** to enter MAC address information. Click **[Save]** button after changing Value.
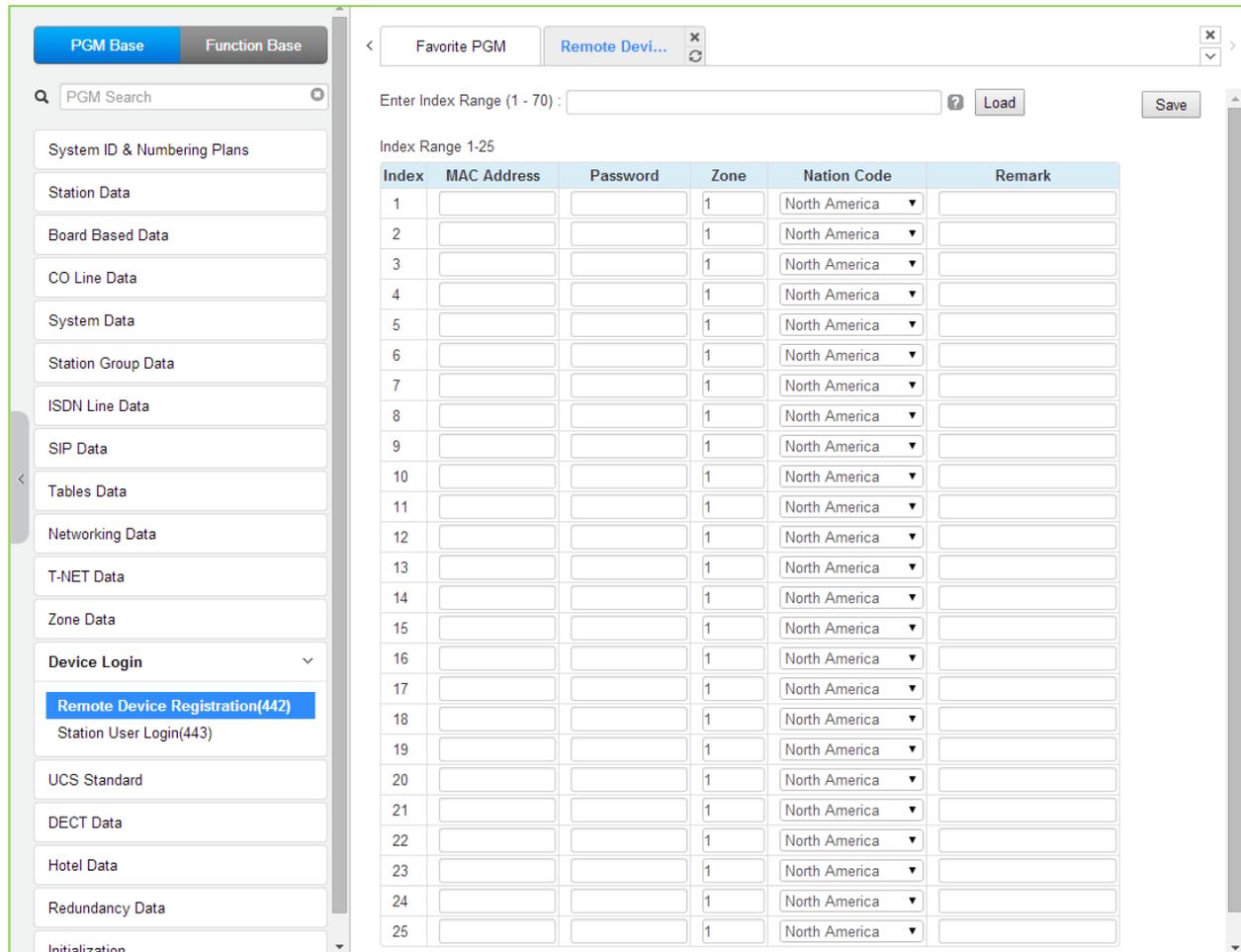
Figure 6.14.1-1 Remote Device Registration

When a device attempts to register with the controlling iPECS, the system will check the device MAC address and password against the Registration table. If a match is found, the device is registered regardless of Dip Switch position 3. The system database is updated, the device receives device specific configuration data from the UCP and the entry in the Registration Table is deleted.

Table 6.14.1-1 REMOTE DEVICE REGISTRATION

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Mac Address	The Mac address of remote device is entered for registration	MAC Address	
Password	A password, if any, is assigned in the devices database via Device Web admin.	Up to 12 digits	
Zone	The remote device can be assigned to a Zone.	01-32	01
Nation code	The Nation code for the device is assigned.	See Table 3.3.1-2	North America

6.14.2 Station User Login – PGM 443

Selecting the Station User Login will display the User Login Registration data input page. Enter an index range then click **[Load]** to modify Station Login data. Click **[Save]** button after changing Value.
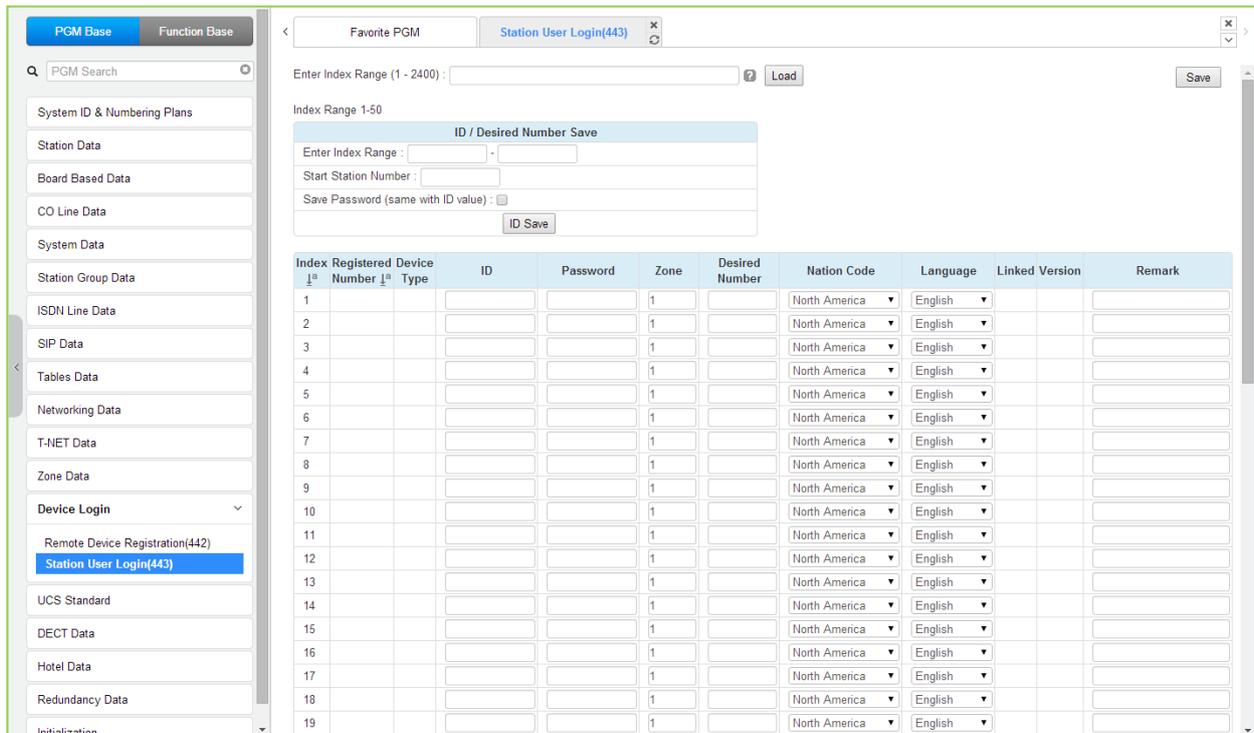
Figure 6.14.2-1 Station User Login

Station User Login configures User credentials for registration of iPECS Communicator Soft phones, UCS clients and initial registration of SIP phones. A station must register with the system each time it is connected to the system. The ID and password are assigned along with other characteristics of the remote station such as Zone, desired station number, country code, Language and a remark can be defined. The iPECS Communicator or UCS Client can be configured as link-paired station by assigning the same Desired-Number as a registered desktop iPECS LIP Phone.

Note for the UCS Client, the UCS Standard Client Login, PGM 446, may be employed in place of PGM 443 as a single point to configure the client information.

Table 6.14.2-1 STATION USER LOGIN

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Registered Number	Station number registered to the station, displayed only after registration.	Station number	
Linked	This field indicates the Linked pair status of the station.		
ID	The User Login ID must be entered for registration	12 Characters	
Password	The User Login password must be entered for registration.	12 digits	

Table 6.14.2-1 STATION USER LOGIN

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Zone	The device can be assigned to a Zone.	1-32	1
Desired Number	The desired number can be entered for the device. To link an iPECS UCS client to an iPECS LIP Phone, enter the station number of the registered iPECS LIP Phone prior to registration of the iPECS UCS. For the UCS Client, PGM 446 may be used as the single administration point.	Station number	
Nation code	The Nation code for the device is assigned.		North America
Language	The default Language type for system voice prompts can be assigned to the device from the available languages.	Language Code 1 ~ 6	English
Linked	This field indicates the Linked pair status of the station.		

6.15 UCS Data

Selecting the UCS data group returns the sub-menu displayed in the left frame as shown in the following figure.
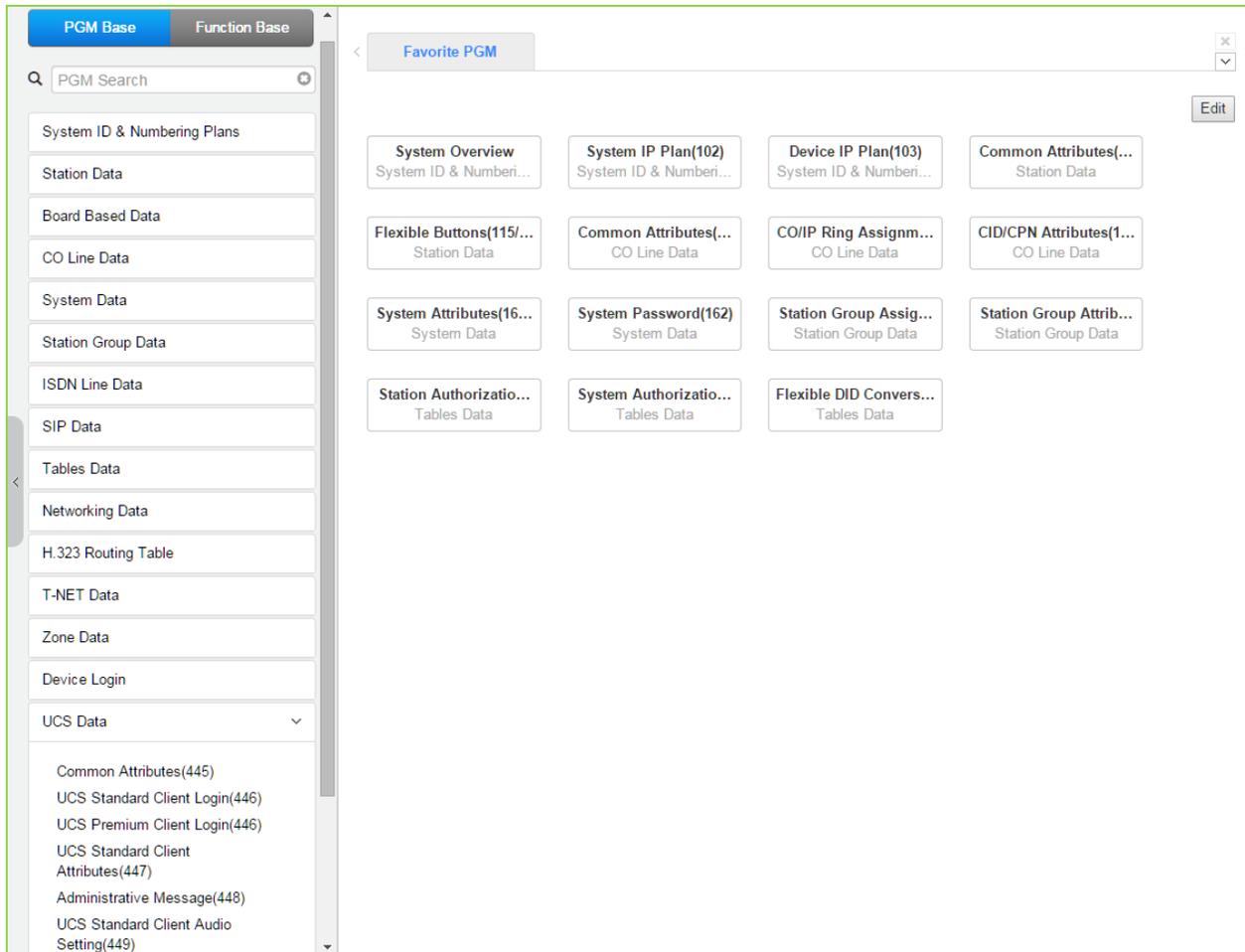
Figure 6.15-1 UCS Data Main Page

6.15.1 Common Attributes – PGM 445

Selecting Common attributes will display the common attributes input page. Click **[Save]** button after changing Value.
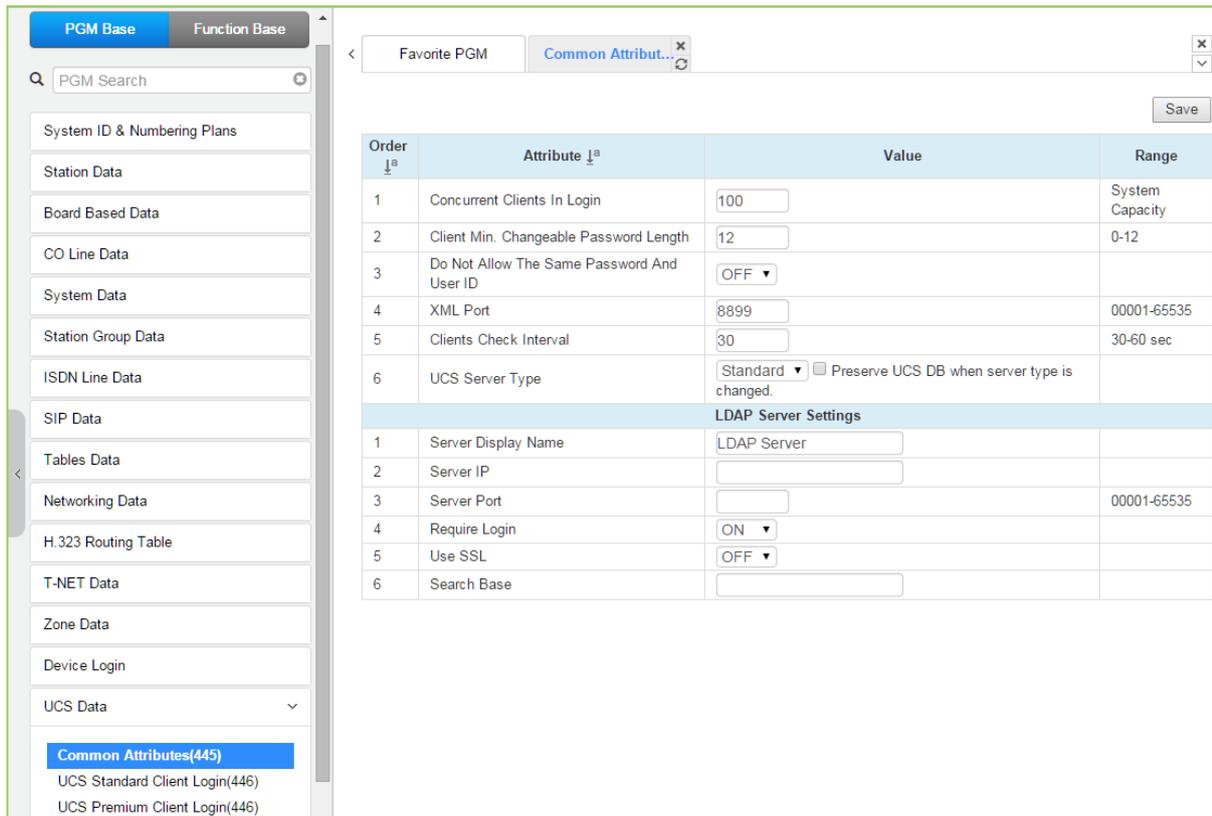
Figure 6.15.1-1 Common attributes

Common Attributes for the UCS Clients include client login characteristics and LDAP server information shared with the clients as shown in Table 6.15.1-1.

Table 6.15.1-1 Common attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Concurrent Clients in login	iPECS UCS employs a per-seat license. The number of UCS Clients that are active at a time cannot exceed the licensed capacity.	System capacity	100
Minimum Password Length	The minimum length of a UCS Client password can be defined up to 12 characters.	Max. 12	12
Do Not Allow the Same Password and User ID	The User Id can be employed as the password if permitted (OFF).	OFF ON	OFF
XML Port	The system employs XML to send or request certain information such as the User picture. The TCP/IP port can be defined.		8899
Clients Check Interval	Periodically, the iPECS system will verify the status of logged in UCS clients.	30~60 seconds	30
UCS Server Type	The server that provides UCS service can be the iPECS UCP (Standard) or the external UCS Server	Standard Premium	Standard

Table 6.15.1-1 Common attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
	(Premium). Additional video and collaboration features are provided by the UCS Server as Premium service. When the UCS Server is employed, the Standard Client Login information (PGM 446) is transferred from the iPECS UCP to the UCS Server after the client is registered. To preserve UCS DB when server type is changed, check the box.		
LDAP Server Settings			
Server Display Name	When the UCS Client requires access to an LDAP server, the system will provide the LDAP server information to the client. The server name is defined in this field.	15 characters	
Server IP	When an LDAP server is employed, the IP address of the server must be defined.		0.0.0.0
Server Port	When an LDAP server is employed, the TCP/IP port of the server must be defined.	00001-65535	
Require Login	The UCS Client ID and Password may be required for log in to the LDAP server.	OFF ON	ON
Use SSL	When supported by the server, the client can employ SSL (Secure Sockets Layer) for added security.	OFF ON	OFF
Search Base	Search base means Search option. You can get the search option from LDAP Server manager. For example, if OU (Organization Unit) is OC and DC (Directory Company) are ucapp and com, you can give the option "OU=OC, DC=ucapp, DC=com" in this field. You will get the desired directory.		

6.15.2 UCS Standard Client Login – PGM 446

Selecting UCS Standard Client Login will display the UCS standard client login input page. Enter the Client index and click **[Load]** to modify the Client data. Use the check boxes to indicate the attributes to modify. Click **[Save]** button after changing Value.
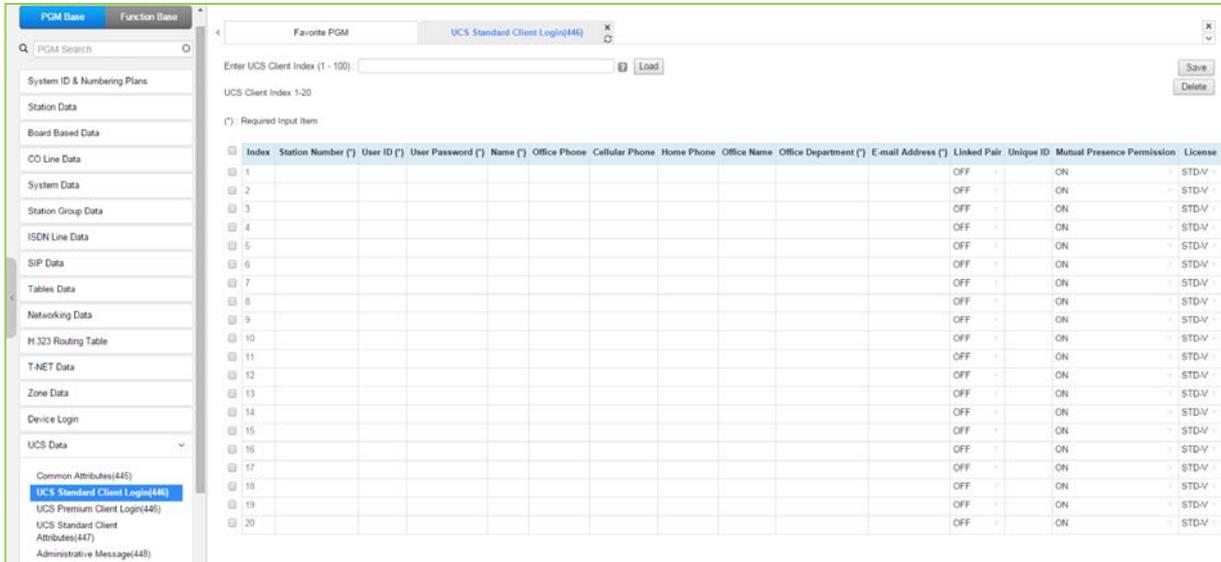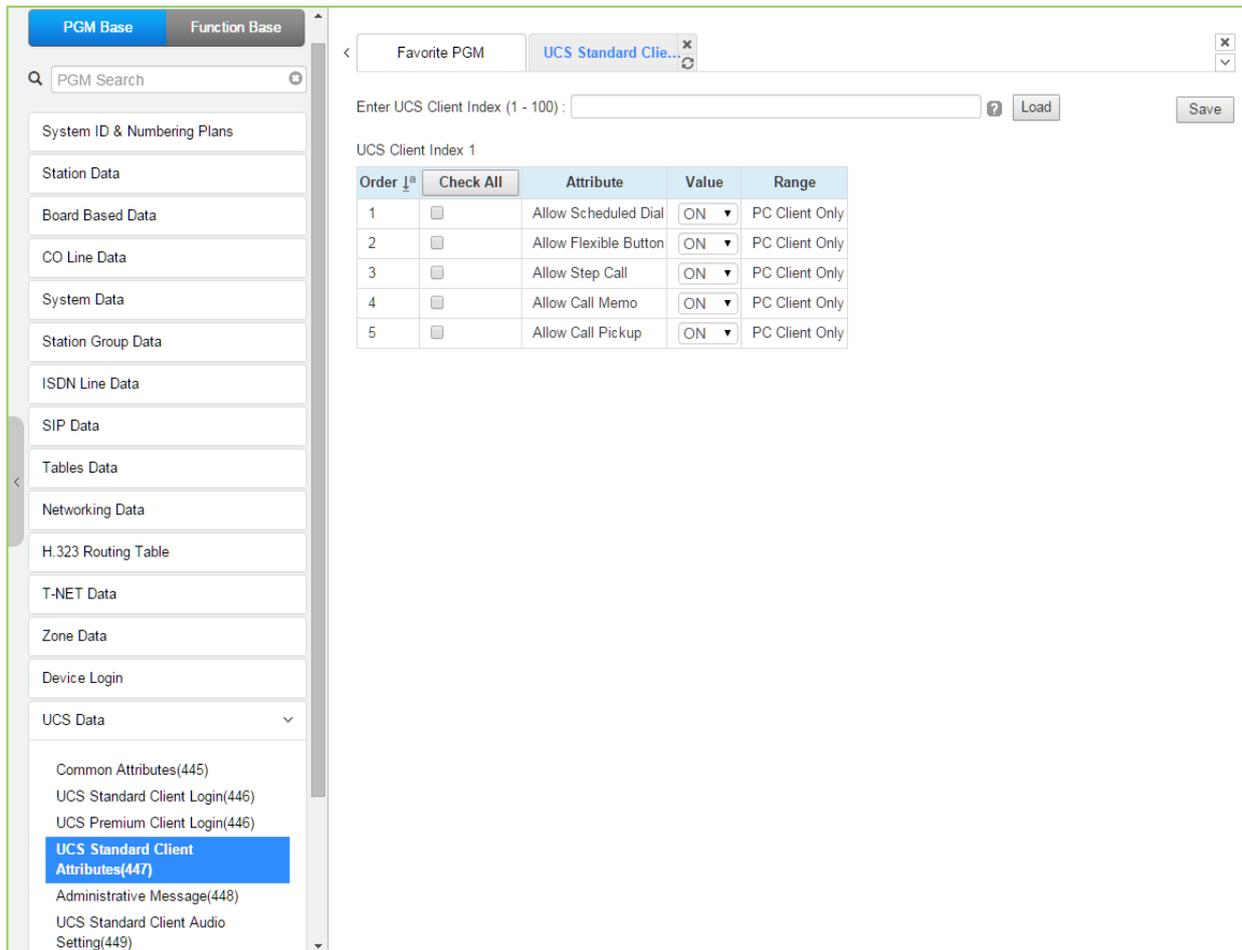

Figure 6.15.2-1 UCS Standard Client Login

The UCS Client information is configured for each client prior to registration. When the UCS Server is employed to provide Premium service, the iPECS UCP will transfer the information to the server after the client registers and, after registration, any changes to the client configuration are accomplished in the UCS Server.

Table 6.15.2-1 UCS Standard Client Login

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Station Number	The station number for the UCS Client can be assigned here and will update PGM 103 and 4433		
User ID	The UCS Client Id for the UCS Client can be assigned here and will update PGM 103 and 443.	Min. 4 characters	
User PWD	The UCS Client password, up to 32 characters must be assigned.	Max. 32 characters	
Name	The user's name, up to 32 characters should be entered.	Max. 32 characters	
Office Phone	Set Office phone number.	Max. 32 characters	
Mobile Phone	Set Mobile phone number	Max. 32 characters	
Home Phone	Set Home phone number	Max. 32 characters	
Office Name	Set Office name	Max. 32 characters	
Office Department	Set Office Department	Max. 32 characters	
Email address	Set Email address	Max. 40 characters	
Linked Pair	If the UCS Client is part of a Linked pair, it must be enabled here. By entering the Station number of the iPECS IP Phone as the UCS Client Station Number, the stations will employ MAC linking.	OFF ON	OFF

Table 6.15.2-1 UCS Standard Client Login

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Unique ID	When the UCS Server is employed, each client must have a unique Id of at least 40 characters, which identifies the clients to the UCS Server. Once the Client registers, iPECS UCP will send the information to the UCS Server. Note that the UCS Server Type in PGM 445 must be configured as "Premium".	Min. 40 characters	
Mutual presence permission	The presence status of the Client can be shared with other clients.	OFF ON	ON
License	The status of license will be displayed and select the license for usage among range.	STD-V STD-NV MOBILE	

6.15.3 UCS Premium Client Login – PGM 446

Selecting UCS Premium Client Login will display the UCS Premium client login input page. Enter the Client index and click **[Load]** to modify the Client data. Use the check boxes to indicate the attributes to modify. Click **[Save]** button after changing Value.
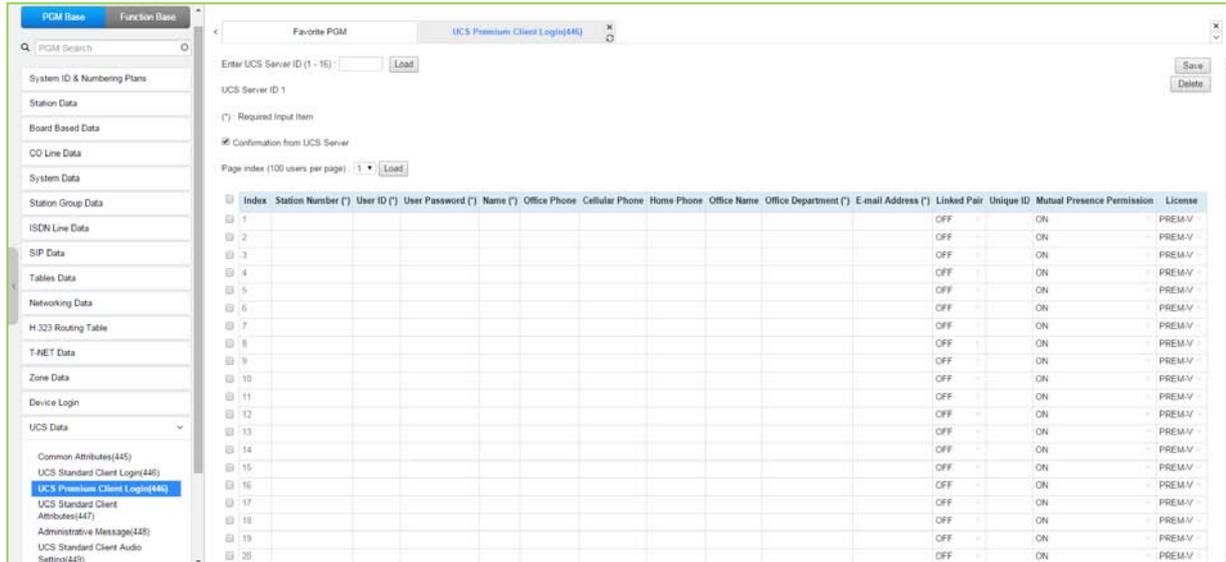
Figure 6.15.3-1 UCS Premium Client Login

The UCS Client information is configured for each client prior to registration. When the UCS Server is employed to provide Premium service, the iPECS UCP will transfer the information to the server after the client registers and, after registration, any changes to the client configuration are accomplished in the UCS Server.

Table 6.15.3-1 UCS Premium Client Login

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Station Number	The station number for the UCS Client can be assigned here and will update PGM 103 and 4433		
User ID	The UCS Client Id for the UCS Client can be assigned here and will update PGM 103 and 443.	Min. 4 characters	
User PWD	The UCS Client password, up to 32 characters must be assigned.	Max. 32 characters	
Name	The user's name, up to 32 characters should be entered.	Max. 32 characters	
Office Phone	Set Office phone number.	Max. 32 characters	
Cellular Phone	Set Cellular phone number	Max. 32 characters	
Home Phone	Set Home phone number	Max. 32 characters	
Office Name	Set Office name	Max. 32 characters	
Office Department	Set Office Department	Max. 32 characters	
Email address	Set Email address	Max. 40 characters	
Linked Pair	If the UCS Client is part of a Linked pair, it must be enabled here. By entering the Station number of the iPECS IP Phone as the UCS Client Station Number, the stations will employ MAC linking.	OFF ON	OFF

Table 6.15.3-1 UCS Premium Client Login

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Unique ID	When the UCS Server is employed, each client must have a unique Id of at least 40 characters, which identifies the clients to the UCS Server. Once the Client registers, iPECS UCP will send the information to the UCS Server. Note that the UCS Server Type in PGM 445 must be configured as "Premium".	Min. 40 characters	
Mutual presence permission	The presence status of the Client can be shared with other clients.	OFF ON	ON
License	The status of license will be displayed and select the license for usage among range.	PREM-V PREM-NV MOBILE	

6.15.4 UCS Standard Client Attributes – PGM 447

Selecting UCS Standard client attributes will display the UCS standard client attributes input page. Enter the UCS Client index then click **[Load]** to modify the client data. Use the check boxes to identify the parameters to modify. Click **[Save]** button after changing Value.

Figure 6.15.4-1 UCS Standard Client Attributes

The UCS Client can be allowed or denied access to several features as described in Table 6.15.4-1.

Table 6.15.4-1 UCS Standard Client Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Allow Scheduled Dial	Allows Scheduled Dialing from the client.	OFF/ON	ON
Allow Flexible Button	Allows the user to access Flex buttons on the UCS Client.	OFF/ON	ON
Allow Step Call	Allows access to the step-call feature by the UCS client.	OFF/ON	ON
Allow Call Memo	Allows the UCS Client access to the Call Memo feature.	OFF/ON	ON
Allow Call Pickup	Permits the UCS Client use of the Direct and Group Call Pick-up features.	OFF/ON	ON

6.15.5 Administrative Message – PGM 448

Selecting Administrative Message will display the administrative message input page. Click **[Send]** button after filling out the subject and Contents and checking the destination.
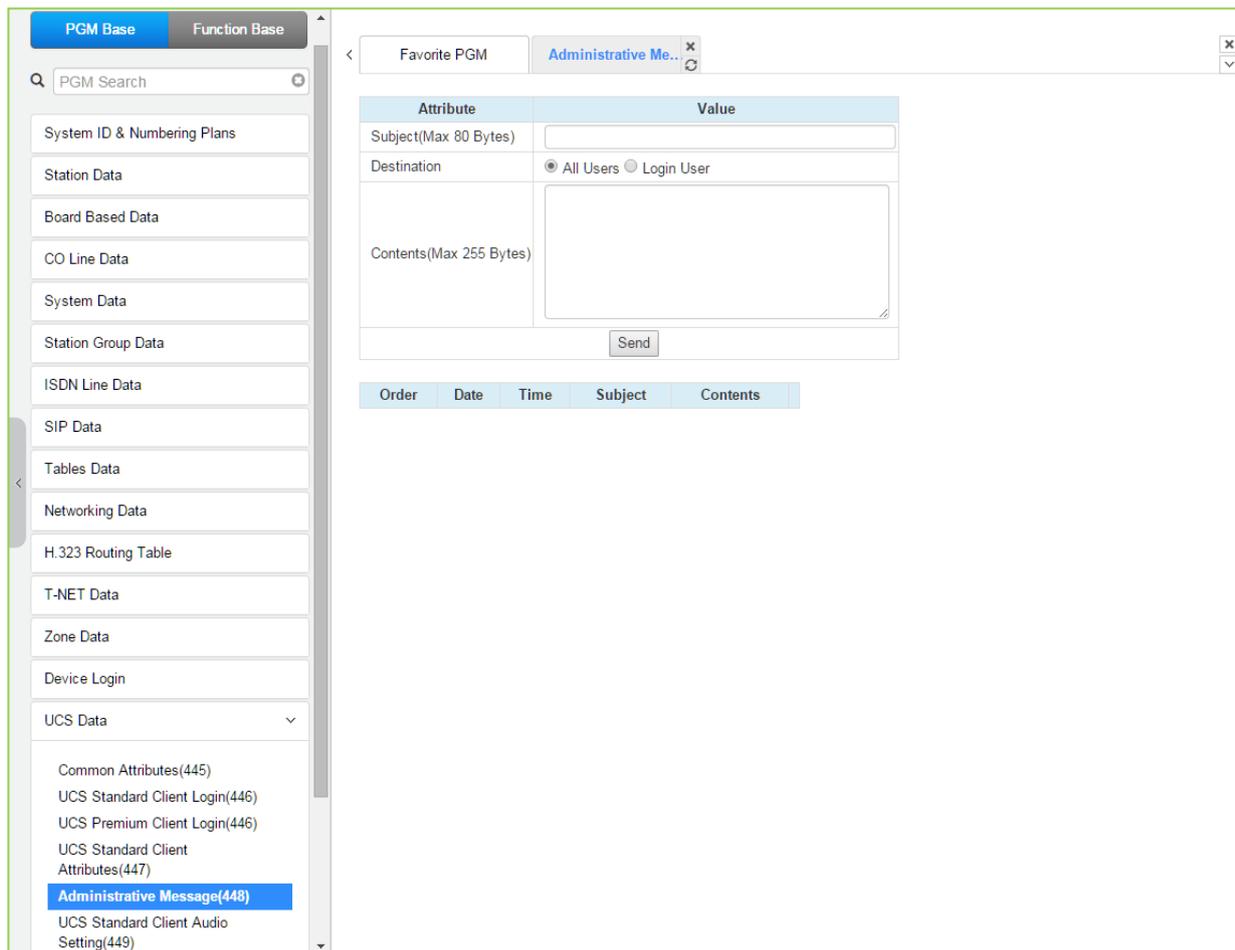
Figure 6.15.5-1 Administrative Message

An administrator can send a message to UCS Clients such as a Welcome message.

Table 6.15.5-1 Administrative Message

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Subject	Message subject	Max. 80 bytes	
Destination	The message can be sent to currently active UCS Clients or to All Clients. For the All Clients selection, the message is sent to all active Clients and as each Client logs in.	Login All	All
Contents	Message body	Max. 255 bytes	

6.15.6 UCS Standard Client Audio Setting – PGM 449

Selecting UCS Standard Client Audio setting will display the audio setting input page. Enter the Client index and click **[Load]** to modify the Client data. Use the check boxes to indicate the attributes to modify. Click **[Save]** button after changing Value.
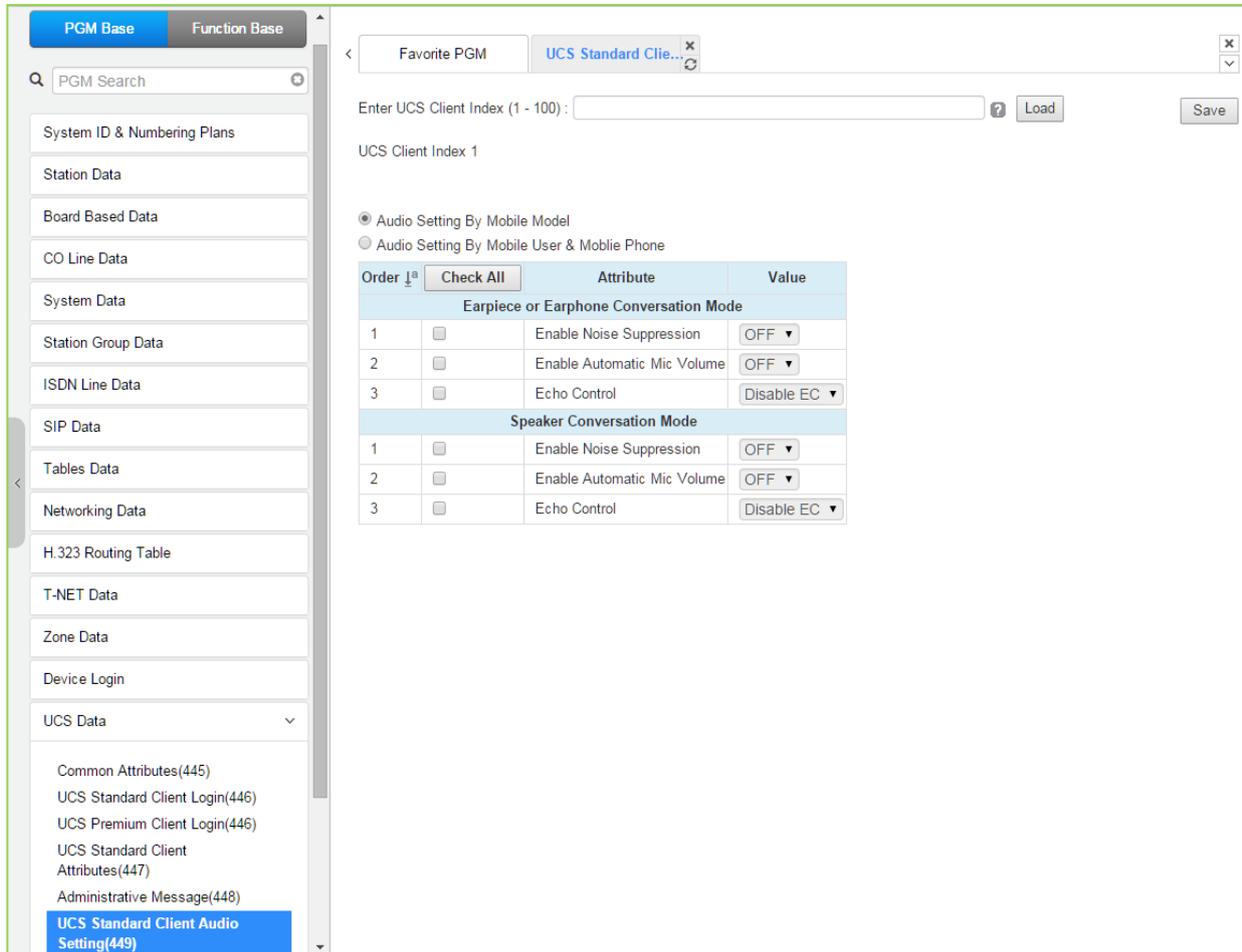
Figure 6.15.6-1 UCS Standard Client Audio setting

Audio characteristic for the Speaker and earphone mode can be established for best overall audio performance.

Table 6.15.6-1 Audio Setting by Mobile User & Mobile phone

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Earpiece or Earphone conversion mode			
Enable Noise Suppression	The system can implement various levels of "Noise suppression from "OFF" to "Very High" in six levels.	OFF ON	OFF
Enable Automatic Mic. Volume	The system can implement Auto gain control for the Earphone.	OFF ON	OFF
Echo Control	Echo cancellation can be enabled for the Earphone.	Disable EC Minimum Low Moderate	Disable EC

Table 6.15.6-1 Audio Setting by Mobile User & Mobile phone

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
		High Maximum	
Speaker Conversation Mode			
Enable Noise Suppression	The system can implement various levels of "Noise suppression from "OFF" to "Very High" in six levels.	OFF ON	OFF
Enable Automatic Mic. Volume	The system can implement Auto gain control for the Speakerphone.	OFF ON	OFF
Echo Control	Echo cancellation can be enabled for the Speakerphone.	Disable EC Minimum Low Moderate High Maximum	Disable EC

6.16 DECT Data - PGM 491

Selecting the DECT Data program group returns the sub-menu displayed in the left frame as shown in the following figure.
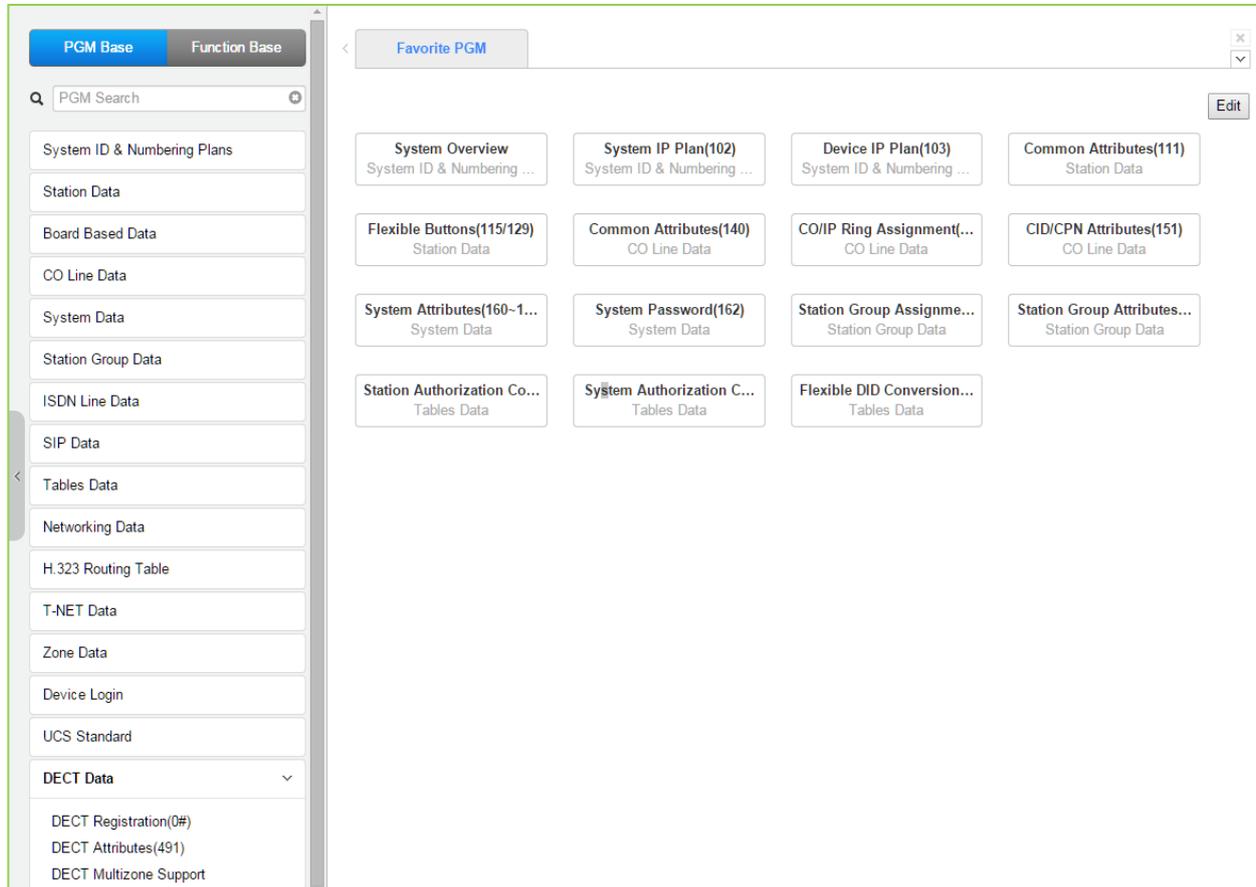
Figure 6.16-1 DECT Data Main Page

6.16.1 DECT Registration (0#)

Selecting DECT Registration will display the DECT Registration input page. Use the radial buttons to select the attributes to change. Click **[Save]** button after changing Value.
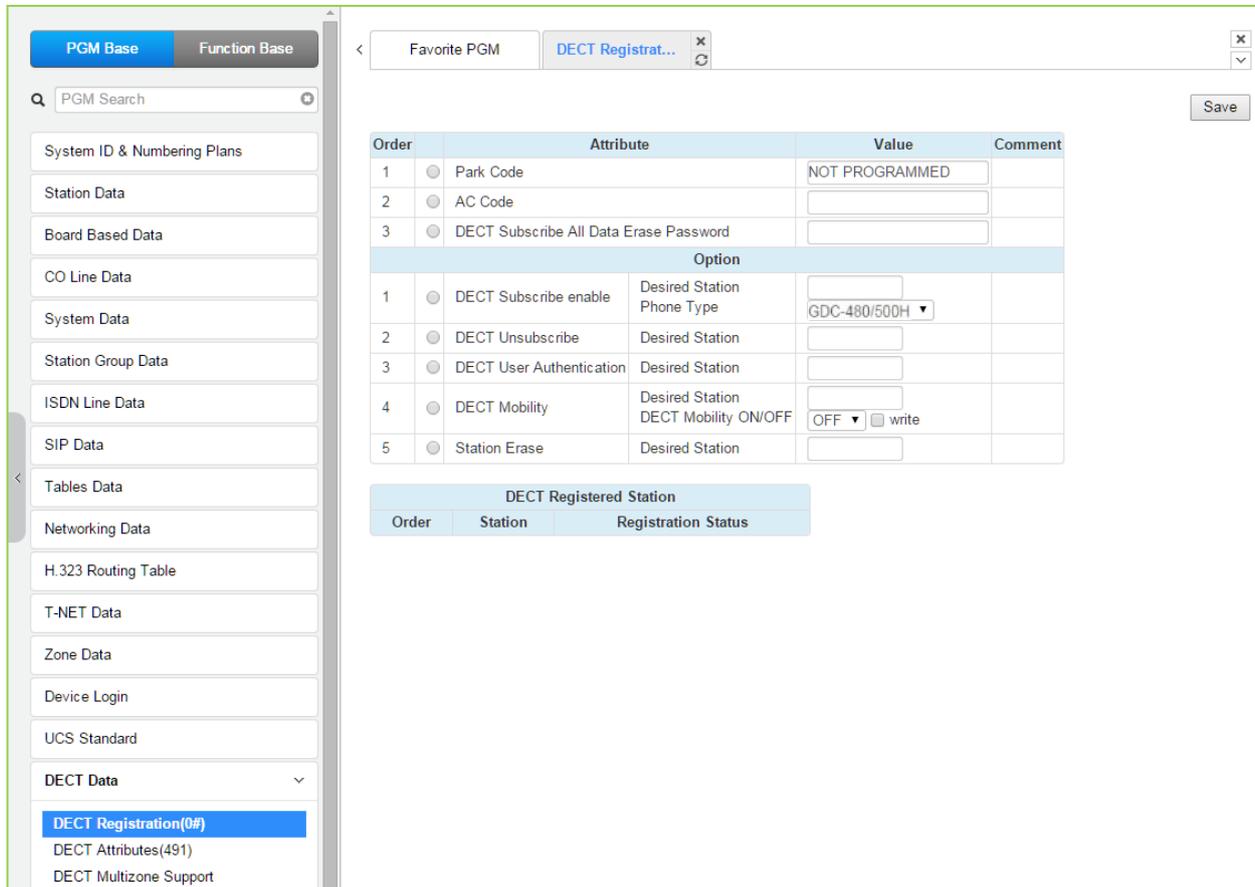
Figure 6.16.1-1 DECT Subscription Screen

On this page, the DECT PARK and authorization codes are defined. Several "Options" are available to enable subscription, unsubscribe a DECT station, enable mobility, etc.   A chart is included displaying the registered and subscribed DECT terminals.

Table 6.16.1-1 DECT Registration

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Park Code	The PARK (Portable Access Rights Key) Code is a unique system Id entered at the DECT handset during the subscription process.	14 digits	Not programmed
AC Code	The Authentication Code is entered at a DECT handset to verify subscription.	Up to 8 digits	
DECT Subscribe All Data Erase Password	To erase all data of DECT, enter the password.		
Option			
DECT Subscribe Enable	Enables the system to accept subscription from a DECT handset.		
Desired Station	Desired station number for the wireless DECT handset.	Station Number	

Table 6.16.1-1 DECT Registration

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Phone Type	Several types of handsets may be selected.	GDC-480/500H GDC-400/450H Standard GAP	GDC-480H/500H
	Press **[Send]** button after entering the number and type.		
DECT Unsubscribe	To terminate a subscription, enter the station number for the DECT handset. Note the Station must be connected, otherwise use Station Erase field below.	Station number	
DECT User Authentication	To subscribe successfully, the user will be required to enter the Station Authorization Code from PGM 227.		
DECT Mobility	When a DECT handset is registered to 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 **[write]** box then Save the page.	OFF ON	OFF
Station Erase	To terminate the registration for a DECT phone that is not connected, enter the registered station number and click **[Save]** button.	Station number	

6.16.2 DECT ATTRIBUTES - PGM 491

Selecting the DECT Attributes will display the DECT ATTRIBUTES input page. Click **[Save]** button after changing Value.

PGM Base	Function Base

Q PGM Search	
System ID & Numbering Plans	
Station Data	
Board Based Data	
CO Line Data	
System Data	
Station Group Data	
ISDN Line Data	
SIP Data	
Tables Data	
Networking Data	
T-NET Data	
Zone Data	
Device Login	
UCS Standard	
DECT Data	∨
DECT Registration(0#)	
DECT Attributes(491)	
DECT Multizone Support	

<	Favorite PGM	DECT Attrib... ✕

Save

Order	Attribute	Value	Remark
1	Auto Call Release	OFF ▾	
2	Base Fault Alarm	Disable ▾	
3	Chain Fault Alarm	Disable ▾	
4	Handover Sensitivity (GDC-500H only)	Normal ▾	If this value is changed, all WTIMs will restart.

Figure 6.16.2-1 DECT ATTRIBUTES

DECT Attributes define functions associated with the DECT equipment and operation as shown in Table 6.16.2-1.

Table 6.16.2-1 DECT Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Auto Call Release	If enabled, when the other party of an active internal call disconnects, DECT terminal returns to idle.	OFF ON	OFF
Base Fault Alarm	If enabled, DECT Base station alarms are sent to the Attendant.	Disable Enable	Disable
Chain Fault Alarm	If enabled, alarms are sent to the Attendant indicating a fault in the link between WTIM modules.	Disable Enable	Disable
Handover Sensitivity (GDC-500H only)	The user can control the handover sensitivity of GDC-500H. (Normal or Strong). If this value is changed, all WTIMs will restart.	Normal Strong	Normal

6.16.3 DECT Multizone Support

Selecting the DECT Multizone support will display the DECT Multizone support input page.

Figure 6.16.3-1 DECT Multizone Support

It is for roaming of DECT phones in large sites with more than 3 WTIMs.

6.17 Hotel Data

Selecting the hotel data returns the sub-menu displayed in the left frame as shown in the following figure.
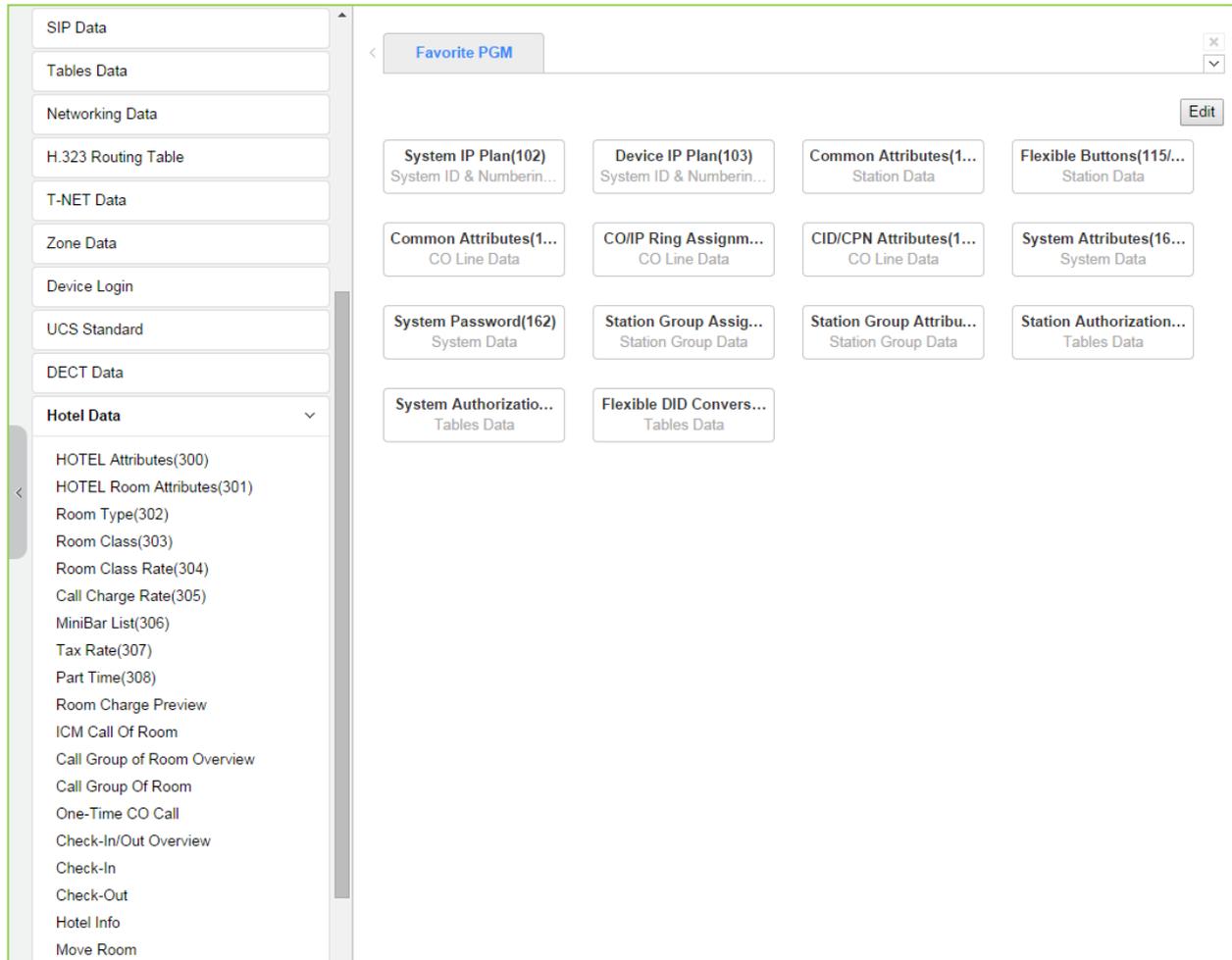
Figure 6.17-1 Hotel Data main page

With Hotel Data, we describe the function on another Hotel Feature and Programming manual. Refer to "Hotel Feature and Programming manual'.

6.18 Redundancy Data for UCP600 & 2400

Selecting the redundancy data returns the sub-menu displayed in the left frame as shown in the following figure.

PGM Base	Function Base		Favorite PGM			×

Q PGM Search ⊙

				Edit

System ID & Numbering Plans

Station Data

Board Based Data

CO Line Data

System Data

Station Group Data

ISDN Line Data

SIP Data

Tables Data

Networking Data

T-NET Data

Zone Data

Device Login

UCS Standard

DECT Data

Hotel Data

Redundancy Data ∨

Redundancy Attributes(502)
Redundancy IP Address(503)

System IP Plan(102)	Device IP Plan(103)	Common Attributes(...	Flexible Buttons(115...
System ID & Numberi...	System ID & Numberi...	Station Data	Station Data

Common Attributes(...	CO/IP Ring Assignm...	CID/CPN Attributes(1...	System Attributes(16...
CO Line Data	CO Line Data	CO Line Data	System Data

System Password(162)	Station Group Assig...	Station Group Attrib...	Station Authorizatio...
System Data	Station Group Data	Station Group Data	Tables Data

System Authorizatio...	Flexible DID Conver...
Tables Data	Tables Data

Figure 6.18-1 Redundancy Data main page

6.18.1 Redundancy Attributes – PGM 502

Selecting Redundancy attributes will display the page shown. Click **[Save]** button after changing Value.
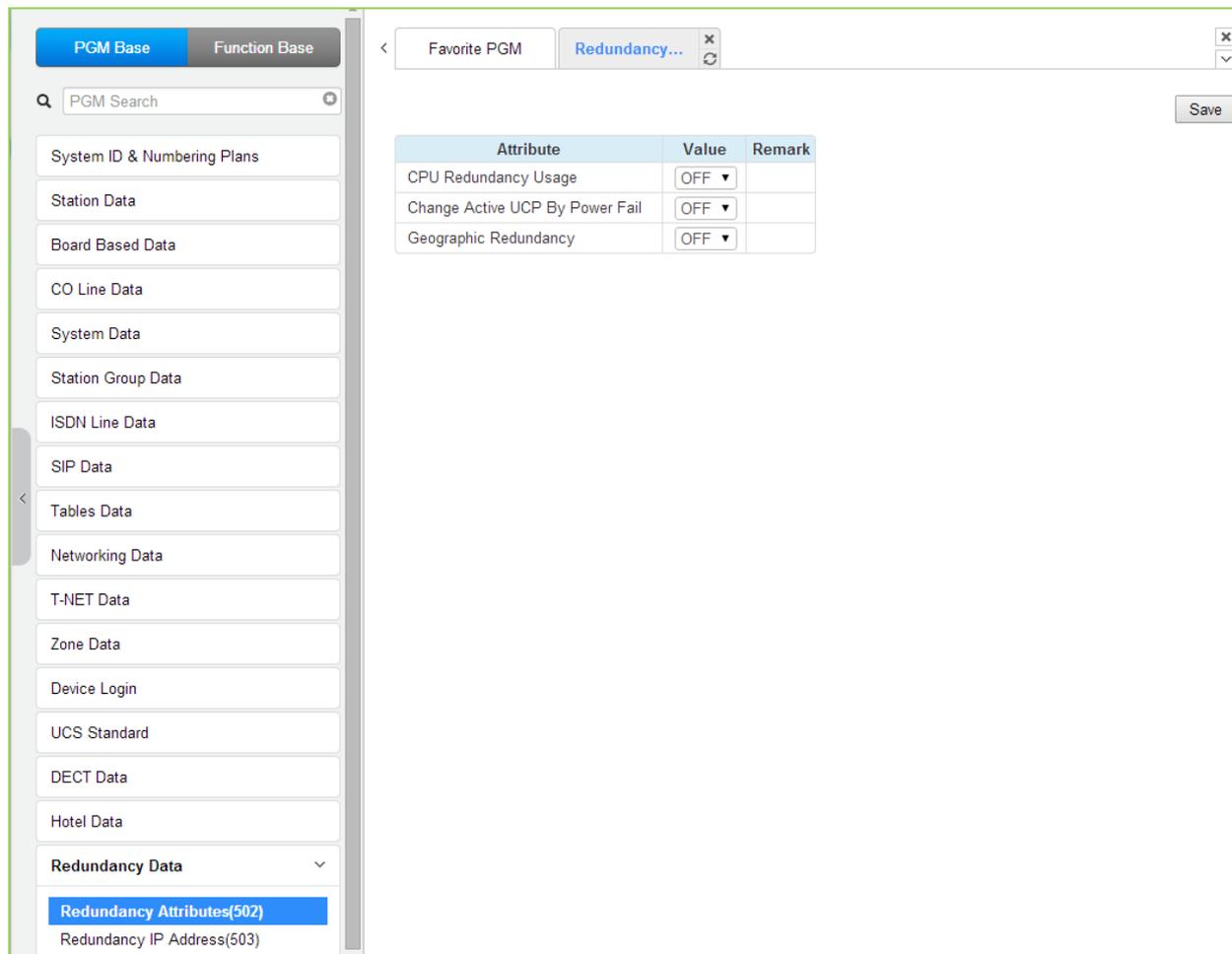
Figure 6.18.1-1 Redundancy Attributes

The Redundancy attributes enable redundancy and activate "Geographical" redundancy.

Table 6.18.1-1 Redundancy Attributes

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CPU Redundancy usage	When redundancy is employed, this field informs the master UCP that a redundant UCP is available.	OFF ON	OFF
Change Active UCP By Power Fail	When power fails, the active UCP is changed to the standby mode and the standby UCP becomes active.	OFF ON	OFF
Geographical Redundancy	Geographical redundancy permits remote location of the redundant UCP modules.	OFF ON	OFF

6.18.2 Redundancy IP Address - PGM 503

Selecting the redundancy IP Address will display the page shown. Click **[Save]** button after changing Value.

Figure 6.18.2-1 Redundancy IP address

For proper operation, the IP addressing parameters of the LAN1 ports of the redundant UCP modules must be defined for Redundancy.

6.19 Initialization - PGM 450

Selecting Initialization returns the sub-menu displayed in the left frame in as shown in the following figure.
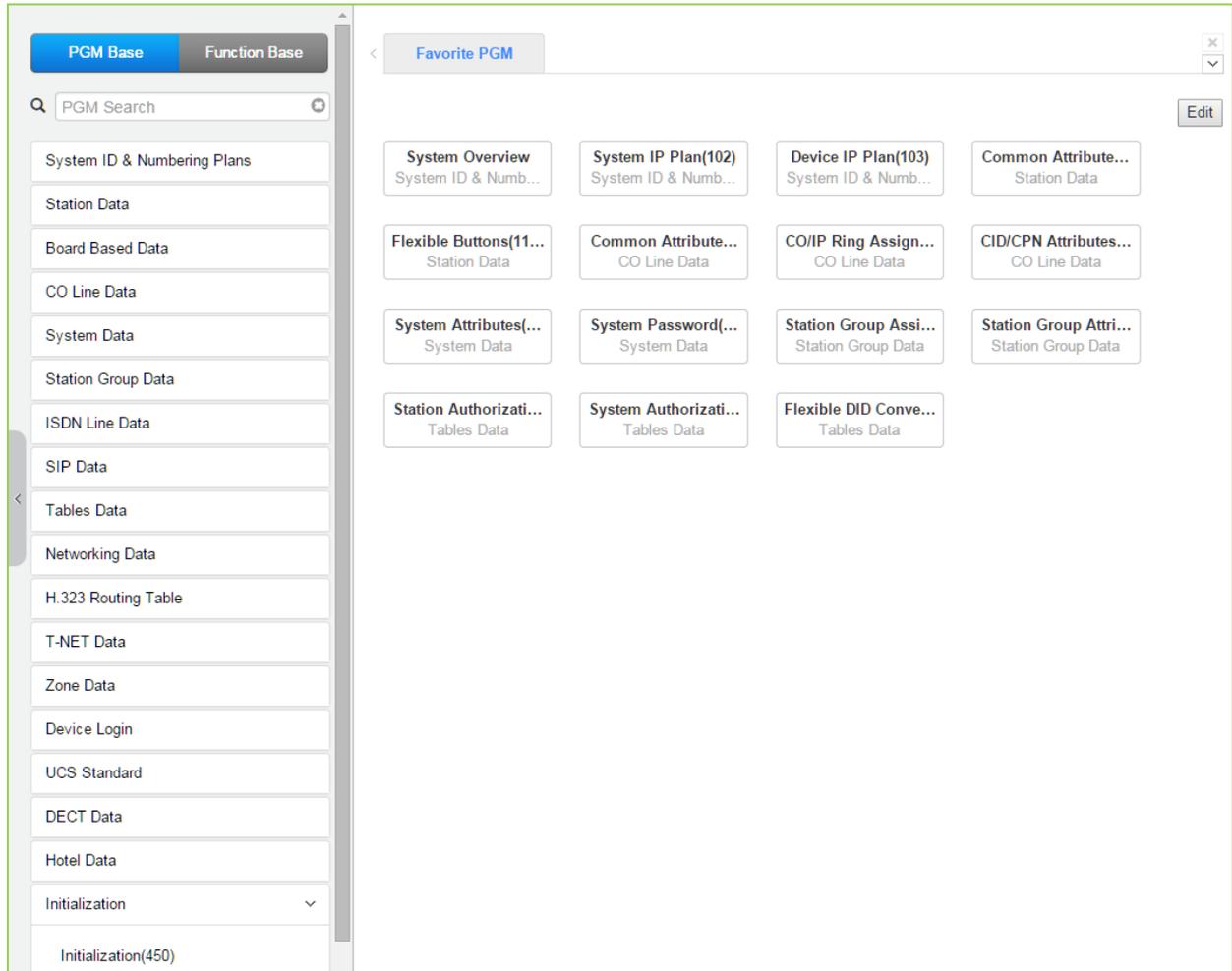
Figure 6.18-1 Initialization

6.19.1 Initialization Table - PGM 450

Selecting Initialization will display the Initialization Table data entry page. Use the check boxes to indicate the segment of the Database to initialize then click **[Initialize]** to reset the database to initial values. A range must be entered to initialize certain data such as Station Data. Click **[Reset System]** to restart the system.

Order	Select	Attribute	Value		Remark	Result
1	☐	All Database				
2	☐	Flexible Number Plan			PGM106~109	
3	☐	Station Data	[ ]	- [ ]	PGM111~114,115/129,116~127,Station Name Display,Station Speed Dial	
4	☐	CO Line Data	[ ]	- [ ]	PGM140~147,150~153,133,Name	
5	☐	System Data			PGM100,105,106,238,160~177,179,183,197,250,251,253,255,436~444,491,Custom Msg,PPTP Attr	
6	☐	Station Group Data			PGM190~192	
7	☐	ISDN Table			PGM201~203,206,231	
8	☐	System Timer			PGM180~182,186	
9	☐	Toll Table			PGM224	
10	☐	LCR Data			PGM220~223	
11	☐	Other Tables			PGM227~229,232~233,236,239,240	
12	☐	Flexible Button			PGM115,129	
13	☐	Networking Data			PGM320~325	
14	☐	SIP Data			PGM126,133,210~212,215,216	
15	☐	T-NET Data			PGM330~336	
16	☐	Zone Data			PGM436~441,444	
17	☐	Remote Device Registration	[ ]	- [ ]	PGM442	
18	☐	Station User Login	[ ]	- [ ]	PGM443	
19	☐	Personal Group Data			PGM260,261	
20	☐	UCS Standard Data			PGM445~449	
21	☐	Hotel Data			PGM300~308	
22	☐	Remove default password(*)			PGM227(Authorization Code Only)	

If you reset system when Dip-Switch 4 is ON status, all database will be initialized.(include PGM101~104)

Figure 6.19.1-1 Initialization Menu

The system has been pre-programmed with certain features using "default data". The default data are loaded into memory when the system is initialized. The system should always be initialized when first installed or the database appears corrupted. In addition, the system can be restarted from this page and, if redundant processors are equipped, the active and standby UCP modules can be modified.

The system can be initialized manually during installation. If the system is reset with the UCP module Mode switch pole 4 in the ON position, the entire database, including the System Id and Numbering Plan, will initialize.

7. MAINTENANCE

The Maintenance tab main screen displays information on the system software versions as well as information on modules and terminals connected to the system. From this page, the database upload/download, software upgrade, access control, license install and other management functions shown in Figure 7-1 are accessed. In addition, at the top of the central screen, detailed software version information is provided. Also, you can monitor the device in real time by clicking **[Real-time Device Monitoring]**.

Real time monitoring for Device

You can monitor the current status about Device in real time. The new information is highlighted in red.

Figure 7-1 Maintenance Main Page

Among Menus on Maintenance, the following functions are not available at UCP2400.

1. **VSF Prompt message**
2. **VSF System greeting**
3. **Voice Mail USB Backup on VSF Mail Management**

7.1 S/W Upgrade

The iPECS UCP employs a NAND based memory file system thus, html, UCP upgrade and iPECS appliance image files can be uploaded without impact to the current database. All of the iPECS series modules can be upgraded by remote access through the system. Selecting S/W Upgrade from the Maintenance page displays the Appliance Version page and sub-menus display in the left frame as shown in Figure 7.1-1.

Figure 7.1-1 S/W Upgrade

There are two types of upgrade images, the application and kernel image. If both are required, upgrade the Kernel image first and then the Application.

The version may be different according to upgrading SW. The web page for Maintenance is an example about how to show the page on each subject.

7.1.1 File Upload

From the File Upload page, Figure 7.1.1-1, click **[Select file]** button and then open the pop-up folder. Select the desired file to upload to the UCP system memory and click the **[Start]** button. The file is sent to the system's memory, saved, and automatically loaded upon a system reset or restart.

Prior to upload, verify sufficient memory is available in the UCP File System for the files to be uploaded. Refer to section 7.5.1 to view the free disk space and delete any unused files. Note names of files required for normal system operation will be grey out and cannot be deleted.

Html image files are extracted and previous HTML files are deleted at completion of the upload process.
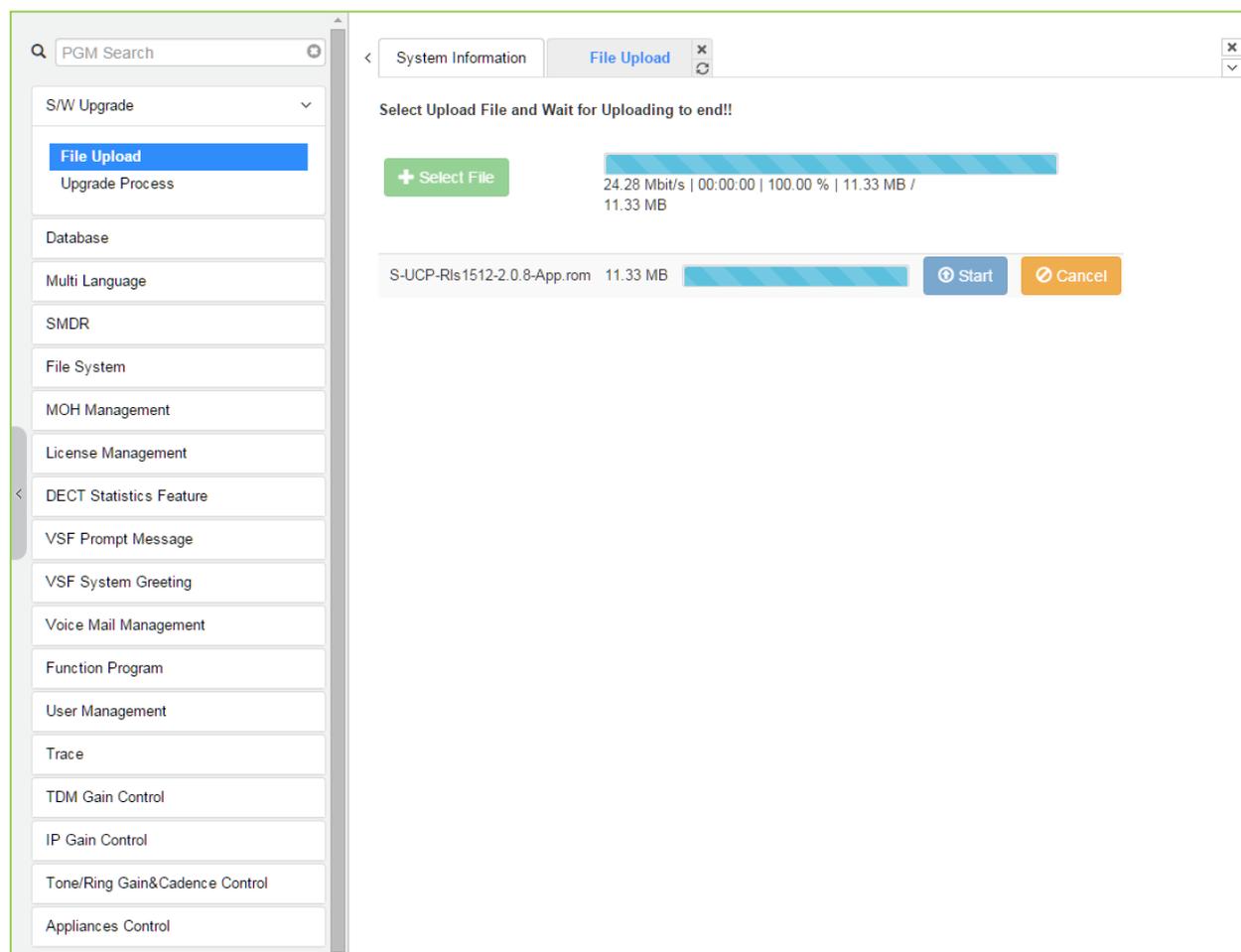
Figure 7.1.1-1 File Upload

* If file upload succeeds, a success page will be displayed.

7.1.2 Upgrade Process

When an iPECS Appliance image is uploaded, appliances of the same type as the uploaded image display and can be selected for upgrade, as shown in Figure 7.1.2-1. Select the desired appliance and click the **[Upgrade]** button, the upgrade process starts and a progress screen displays.
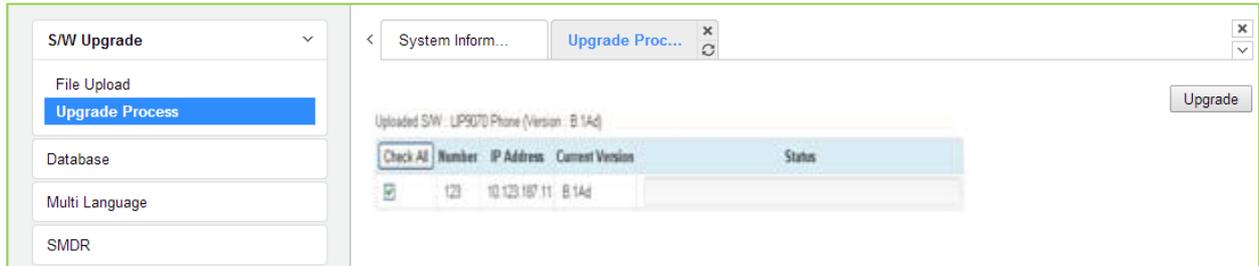
Figure 7.1.2-1 Upgrade Process

7.1.2.1 Upgrade Process View

The Upgrade Process View provides a status window; refer to Figure 7.1.2.1-1 Upgrade Process View, for Module and terminal upgrade activity in process.
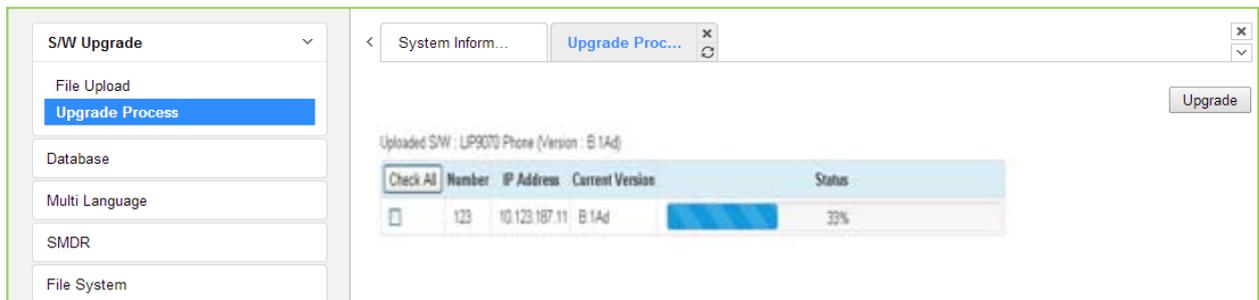
Figure 7.1.2.1-1 Upgrade Process View

7.1.2.2 iPECS UCP Upgrade Process

iPECS Software Full Upgrade Sequence

The following shows the order in which the upgrade process proceeds and firmware files for iPECS UCP and modules. Note the xxxx in the ROM file names indicates the version number of the file.

UCP Upgrade Process

1. Upload UCP application image.

iPECS UCP	UCP application image
	(Example, S-UCP-Rls1601-2.0.12-app.rom

2. Restart UCP.

Note:

If the new system database is not compatible with existing system database, it will be necessary to initialize the system database. This can be done manually using the Initialize Dip Switch located on the UCP module or via the Initialization Web page.

Upgrade of the UCP includes HTML files; a separate upload of the HTML files is not required. HTML files can be separately upgraded under the File System

Appliances Upgrade Process

1. Upload Appliance application image to UCP

LIP 8012/8024/8040E application image:	GS98Pxxxx.rom (xxxx indicates the version)
LIP 8002E application image:	GS99Pxxxx.rom (xxxx indicates the version)

2. Select the target appliances and click **[Upgrade]**.

3. Wait until upgrade process completes.

4. The device will restart automatically when upgraded successfully.

7.1.3 Upgrade HTML Files

The "File View" Menu is used to upload file and reload the system's html files. Upload time of the html files will take 5 ~ 10 minutes.

7.1.3.1 Appliances Upgrade (Device and iPECS Phone)

Upload appliance image, and click **[Upgrade process]** button to select upgrade appliances. If appliances are selected, click **[Upgrade]** button. The page shown in Figure 7.1.2.1-1 will be displayed indicating the Upload command has been sent and upgrade process is working. This page will display the Upload status. When the appliance upgrade process is successful, the status is updated to "Success". If the upgrade process fails, the process is attempted an additional three (3) times before abandoned.

7.1.3.2 Direct Appliances Upgrade

Should the above UCP system managed upgrade process fail, appliances (Devices and iPECS Phones) can be upgraded directly using the appliance IP address as the upgrade destination address. Note the later may require local access.

Figure 7.1.3.2-1 iPECS Phone Direct Connect Upgrade

7.2 Database

7.2.1 Database Upload

The Database Upload selection will display the Database Upload page as shown in Figure 7.2.1-1. Select **[Add Files]** then select the database file desired from the local PC and click **[Start]** to upload the database to the system.
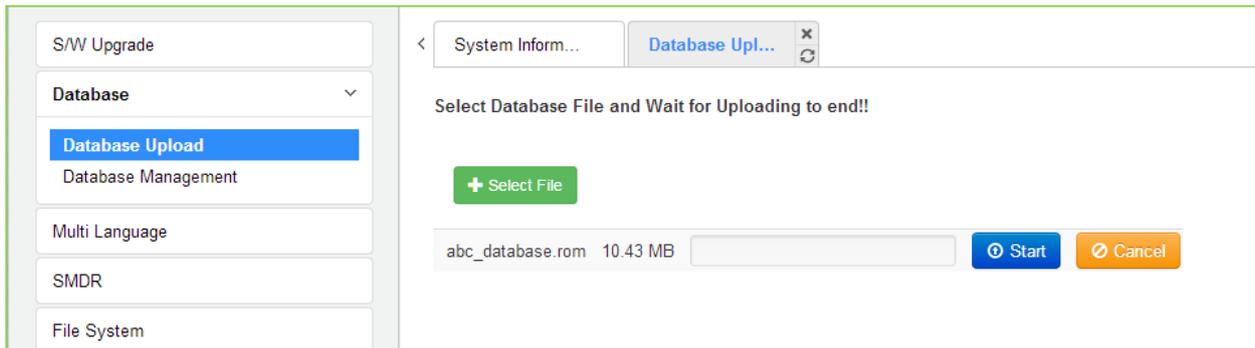
Figure 7.2.1-1 Database Upload

7.2.2 Database Management

Selecting Database Management will display the Database Download page Figure 7.2.2-1. Select the structure to determine the portion or structure of the download. Click **[Download]** then select the directory and file name in the pop-ups that follow to save the file to the PC. After building a database file, the download function will be available. A maximum of five downloadable files for various types of data and structure can be built; building additional files deletes the oldest.
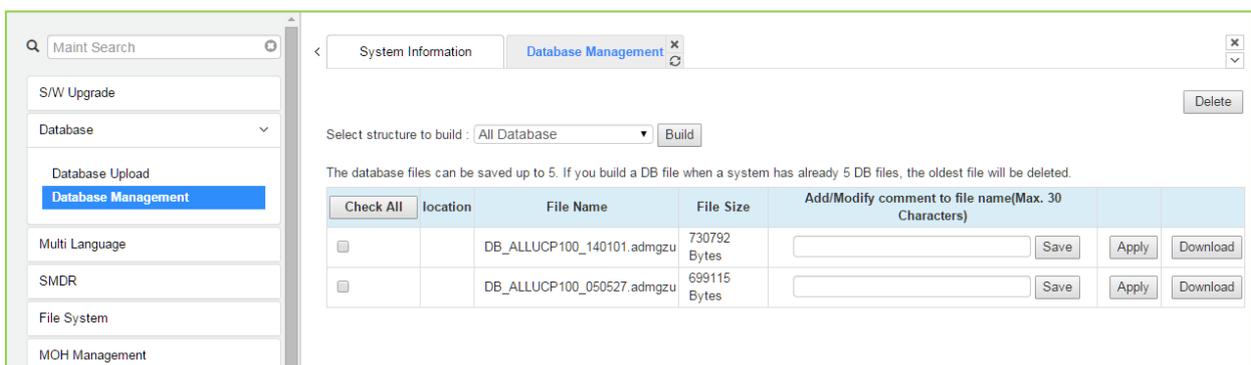
Figure 7.2.2-1 Database Download Management

You can add or modify the file name by entering the characters including special character (#, $, %, &, ', ( ), -, ., @, _). The comment will be created as [XXXXX] before '.admgzu'. It is useful to search DB after downloading. A user can apply DB file from USB memory. The DB file name is composed of system type, DB type, date & time and version information.

7.3 Multi Language

The system can employ either English or a "Local Language" for the Web page displays. The multi-language file, which is a csv (comma separated variable) formatted file, is downloaded to a PC and the English terms are translated to the local language. The modified file can then be uploaded to the iPECS UCP. Once uploaded, the user can select the "Change Language" button and select either English or the Local Language translation for Web page displays.

7.3.1 Multi Language File Upload

The Multi Language File Upload selection will display the Multi Language File Upload page as shown in Figure 7.3.1-1. By selecting the Multi Language File desired from the local PC, the desired Multi Language File can be uploaded to the iPECS UCP.
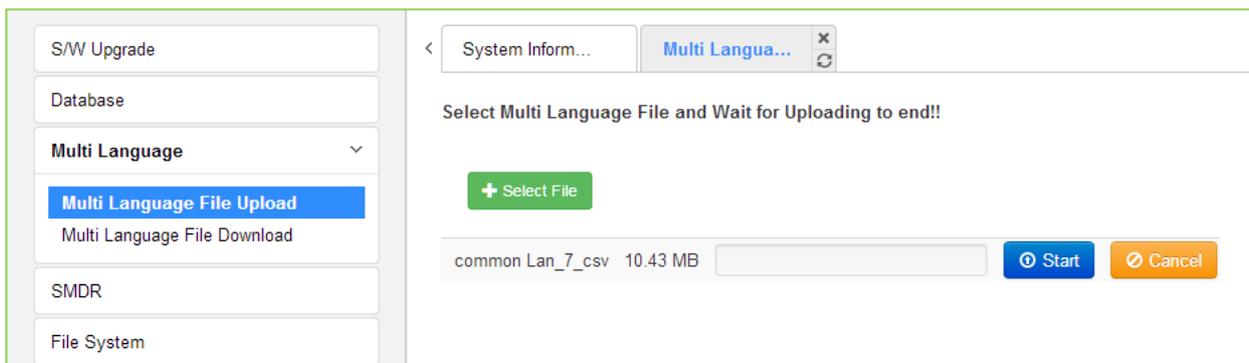
Figure 7.3.1-1 Multi Language File Upload

7.3.2 Multi Language File Download

Selecting Multi Language File Download will display the Multi Language File Download page Figure 7.3.2-1. Selecting this option will download the iPECS UCP Multi Language file to the local PC. The file can be stored in the PC, modified and uploaded to an iPECS UCP using the file upload procedures in section 7.3.1.
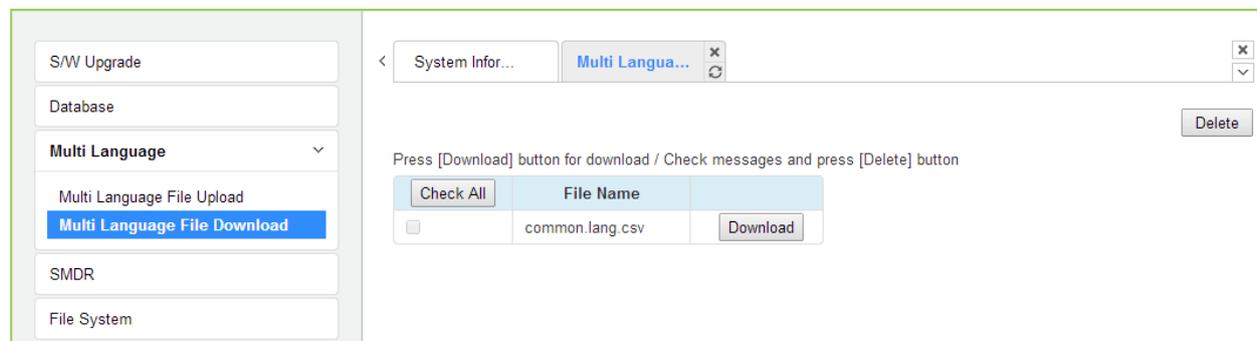
Figure 7.3.2-1 Multi Language File Download

7.4 SMDR

The iPECS UCP can download SMDR data in a SYLK format file (.slk). The file is compressed using the "gz" file format. After unzipping, the file can be opened under any common spreadsheet application. The system will provide a view of SMDR data for the station range entered in the Web page Figure 7.4-1. This page may also be employed to delete SMDR records for the station range entered.
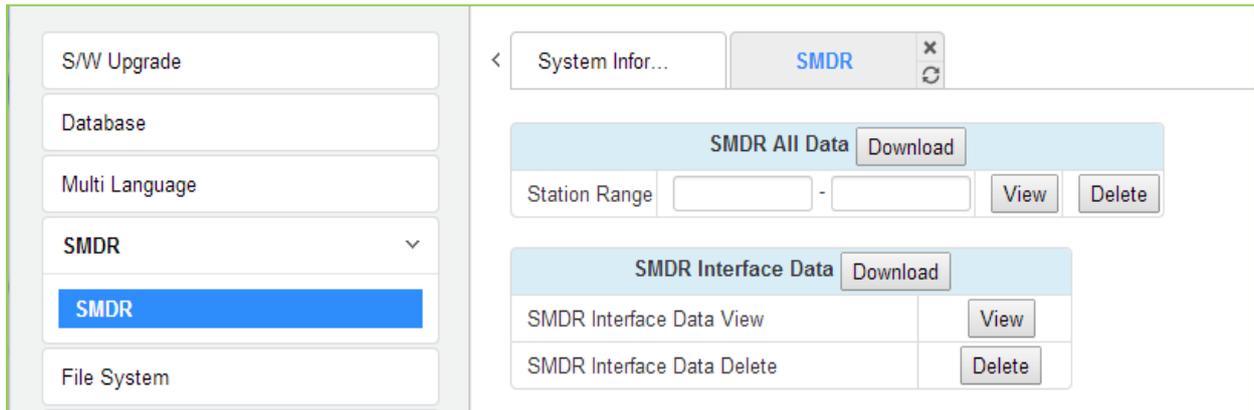
Figure 7.4-1 SMDR Access

7.5 File System

7.5.1 File View & Delete

The File View & Delete page displays a list of the various files stored in the UCP file system memory. To delete a file, check the box for the file and click **[Delete]**. Files can also be downloaded to the PC using the **[Download]** button to the right of the file name.

Note the system requires the files displayed in grey and the system will not allow deletion of these files.
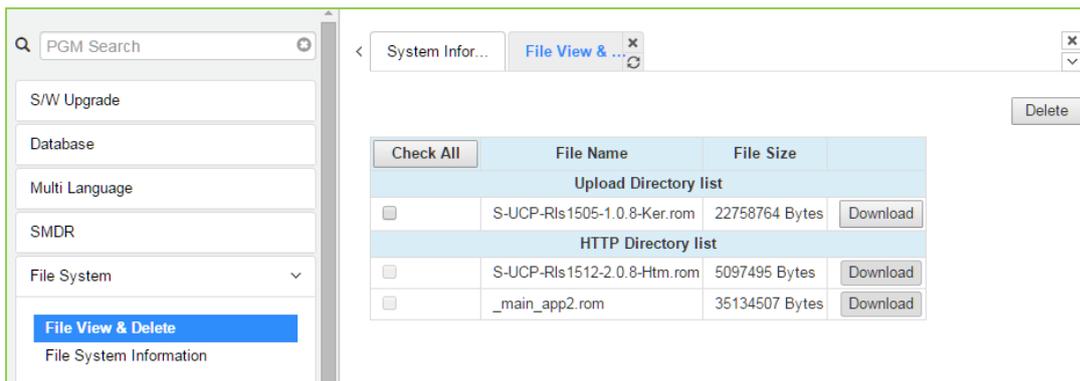
Figure 7.5.1-1 File View and Delete

7.5.2 File System Information

The File System Information page displays the disk status including the total and free disk space as shown in Figure 7.5.2-1. Use this page to determine if the free space on the disk is sufficient for uploading Upgrade files, section 7.1.1.
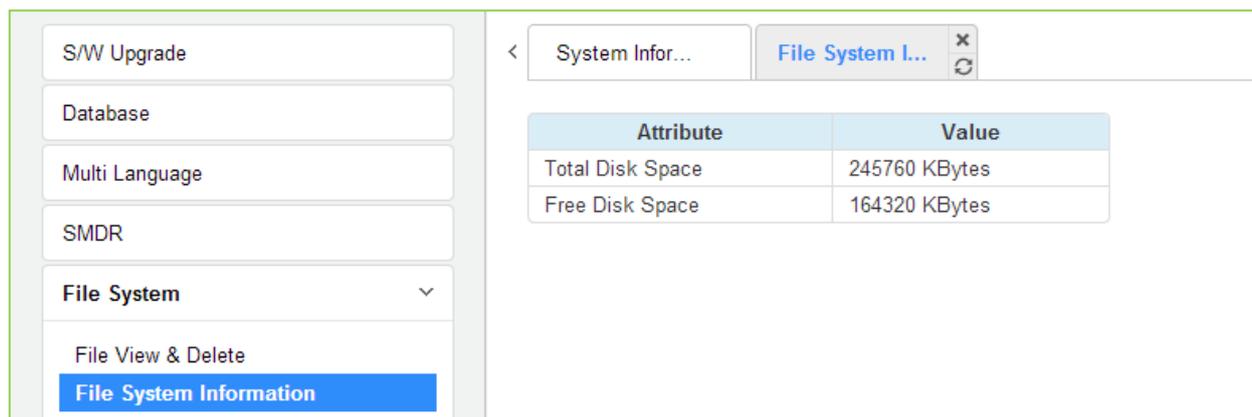
Figure 7.5.2-1 File System Information

7.6 MOH (Music On-Hold) Management

7.6.1 MOH Delete & Download

The MOH Delete & Download page displays a list of the thirteen files stored in the file system memory. To delete a file, check the box for the file and click **[Delete]**. Files can also be downloaded to the PC using the **[Download]** button to the right of the file name.
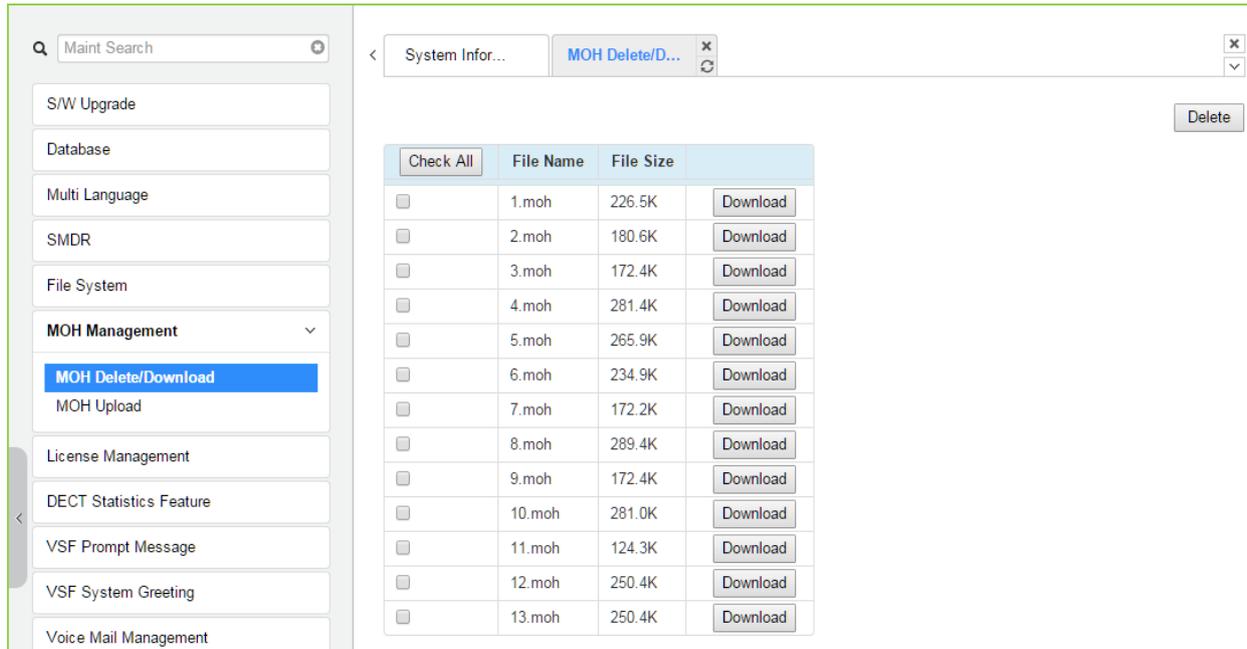
Figure 7.6-1 MOH Delete & Download

7.6.2 MOH Upload

Selecting "MOH Upload" will display the page shown in Figure 4.5.6.2-1. Here MOH can be uploaded to the system for each of up to thirteen (13) MOH. MOH files format must be followed as shown in the page 'G.711 a/u-Law wav (8000Hz, 16bit, mono)'.
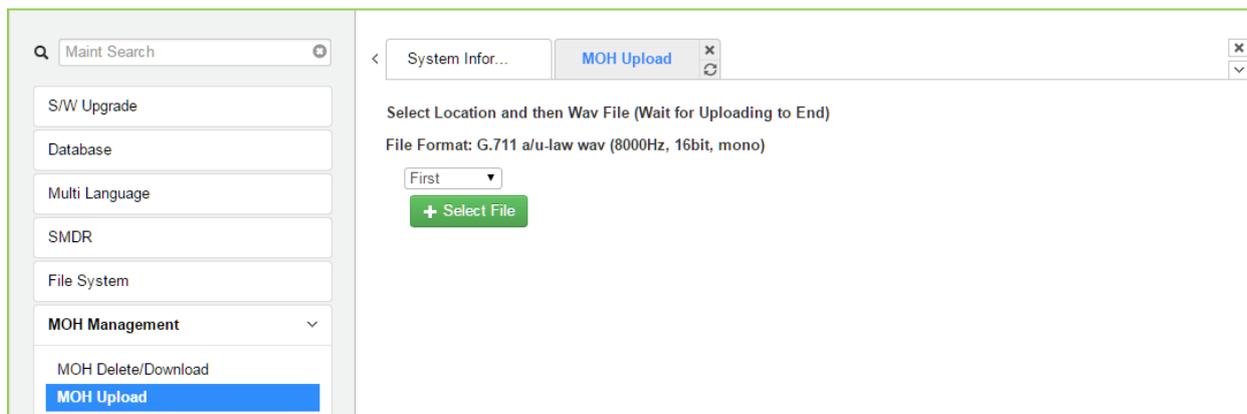
Figure 7.6-1 MOH Upload

7.7 License Install

Various licenses can be installed to expand capacity of the system and Auto Attendant/Voice Mail services available in the UCP system as well as soft clients and feature applications. After obtaining a license, the code is entered in the Value column as shown in Figure 7.7-1.

| PGM Base | Function Base | | Favorite PGM | System Overview | ✕ | | ✕ |

System License Overview

Serial No. : 000E56F40D137213

License	Status	Currently used	Purchased
Software Permissions	Activated		Not activated
Total System Port Expansion	199 copy(s)	50 (STN 1 + CO 49)	50 copy(s)
Total IP Extension	199 copy(s)	1 (LIP 1 + SIP1st 0)	30 copy(s)
Third Party SIP Extension	199 copy(s)	0	0 copy(s)
VOIP Virtual Switching Channel(8ch/copy)	2 copy(s)	0 ch	0 copy(s)
VMU Recording Time Add 10-Hour	Not activated		Not activated
Mobile Extension	199 copy(s)	0	30 copy(s)
IP Networking or QSIG	Activated		Not activated
Transparent Network(T-NET) or Local Survivability	Activated		Not activated
Hotel Feature	Activated		Not activated
FIDELIO Interface	Activated		Not activated
Third Party TAPI Interface	Activated	Disconnected	Not activated
Third Party SIP Application Server Interface	10 copy(s)	0	0 copy(s)
Third Party SIP Application Channel Interface	199 copy(s)	0	0 copy(s)
MS LYNC EV Channel	199 copy(s)	0	0 copy(s)
MS LYNC RCC Gateway	Activated		Not activated
MS LYNC RCC Client(2010)	199 copy(s)	RCC(max:398, set:0, using:0)	0 copy(s)
MS LYNC RCC Client(2013)	199 copy(s)	RCC(max:398, set:0, using:0)	0 copy(s)
MS LYNC RCC or Voice Client(2010)	199 copy(s)	Voice(0)	0 copy(s)
MS LYNC RCC or Voice Client(2013)	199 copy(s)	Voice(0)	0 copy(s)
ClickCall Application	199 copy(s)	(set:0, using:0)	2 copy(s)
UCS Client Desk Standard with Non Voice	100 copy(s)	0	0 copy(s)
UCS Client Desk Standard with Voice	100 copy(s)	0	2 copy(s)
UCS Client Desk Premium with Non Voice	199 copy(s)	0	0 copy(s)
UCS Client Desk Premium with Voice	199 copy(s)	0	0 copy(s)
UCS Client Mobile	199 copy(s)	0	2 copy(s)
IP Attendant for Office	50 copy(s)	0	0 copy(s)
IP Attendant for Hotel	50 copy(s)	0	0 copy(s)
IP Call Recording Server	10 copy(s)	0	0 copy(s)
IP Call Recording Agent	199 copy(s)	0	0 copy(s)
IP Call Recording Trunk	199 copy(s)	0	0 copy(s)

PGM Base sidebar:
- System ID & Numbering Plans
  - System ID(100)
  - **System Overview**
  - Device Port Num Change(101)
  - System IP Plan(102)
  - Device IP Plan(103)
  - CO Device Sequence Number(104)
  - Flexible Station Number(105)
  - Flexible Numbering Plan(106~109)
  - 8 Digit Extension Table(238)
- Station Data
- Board Based Data
- CO Line Data
- System Data
- Station Group Data
- ISDN Line Data
- SIP Data
- Tables Data
- Networking Data
- H.323 Routing Table
- T-NET Data
- Zone Data

Figure 7.7-1 License Install

7.7.1 License upload

Before License upload, make sure that the date of system is set correctly. Click the Add files button and then open the pop-up folder. Select a valid license file to upload to the UCP system and click the **[Start]** button. If the file which is sent to the system is "System License File", it will be saved and automatically applied without restart. The enabled features by uploading license file can be shown in "System Overview" page.
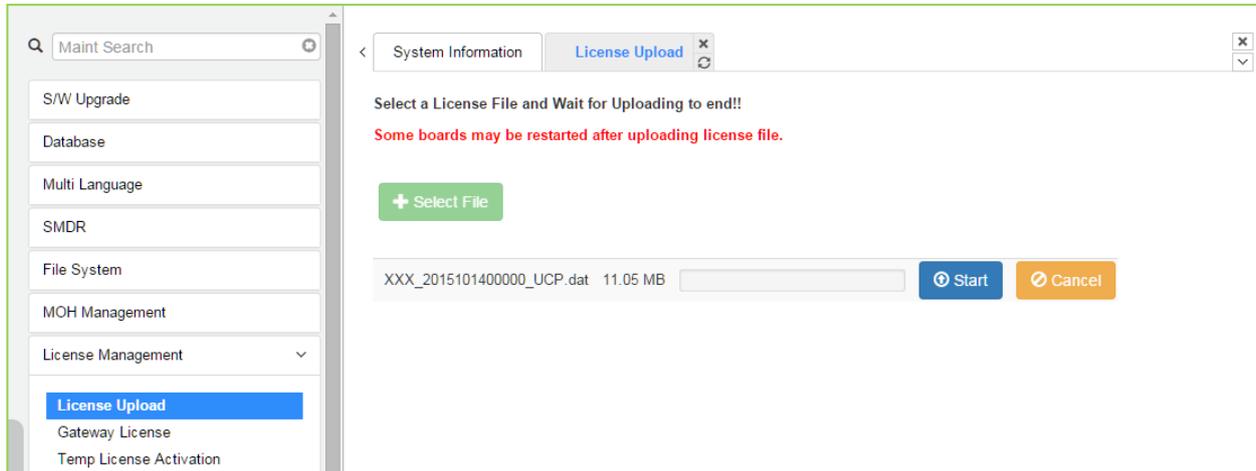
Figure 7.7.1-1 License Upload

7.7.2 Gateway License

All gateways which can have gateway license type are displayed in this web page. If a valid gateway license file was uploaded by "7.7.1 License Upload" and a gateway with matched serial number is registered in system, **[Apply]** button will be activated. This button is used to send the license file to the gateway. After the license file is sent, it will be deleted and the gateway will be restarted to apply features of the license file.

1) The following figure is before uploading the license.

2) After re fresh, the following figure is displayed and click the apply button to activate.
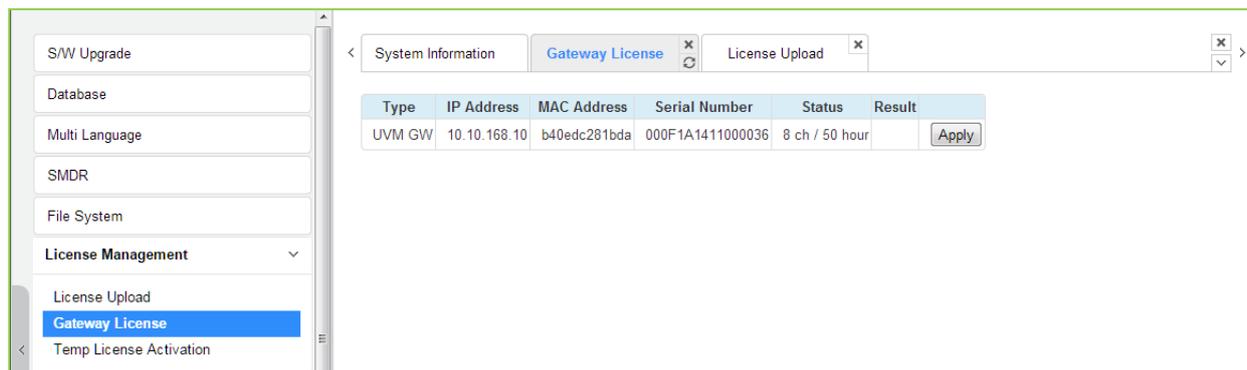
3) The final step is to reset the gateway license and then refresh. You can see the following figure for Gateway license.
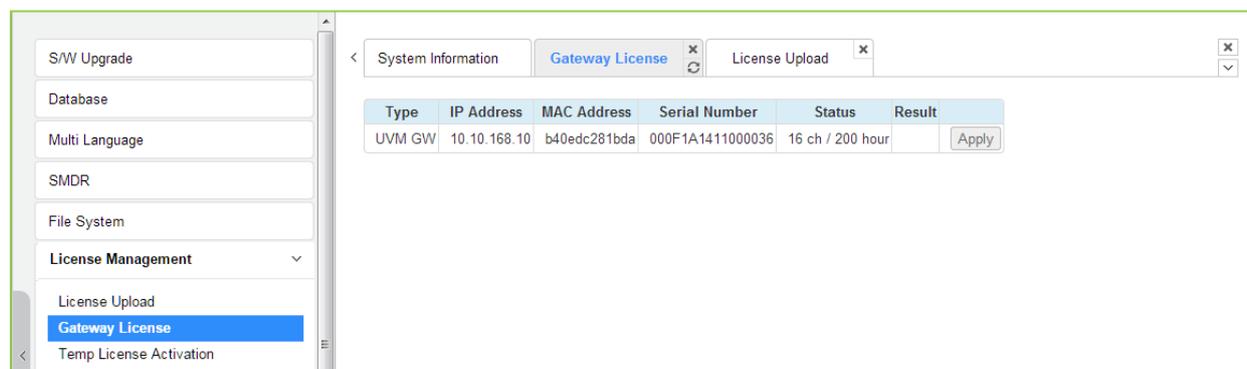
7.7.3 Temp License Activation

Before Temp License Activation, make sure that the date of system is set correctly. In this page, Temp License can be activated by clicking [**Activation**] button. Remaining days and activation times are displayed in State column. The Temp License can be activated totally 6 times. It is valid for 60 days at first activation and is valid 30 days from second activation. Therefore the license file which has correct port number and features must be uploaded. Otherwise, system may enter "Limited Service Mode" due to the mismatched license.
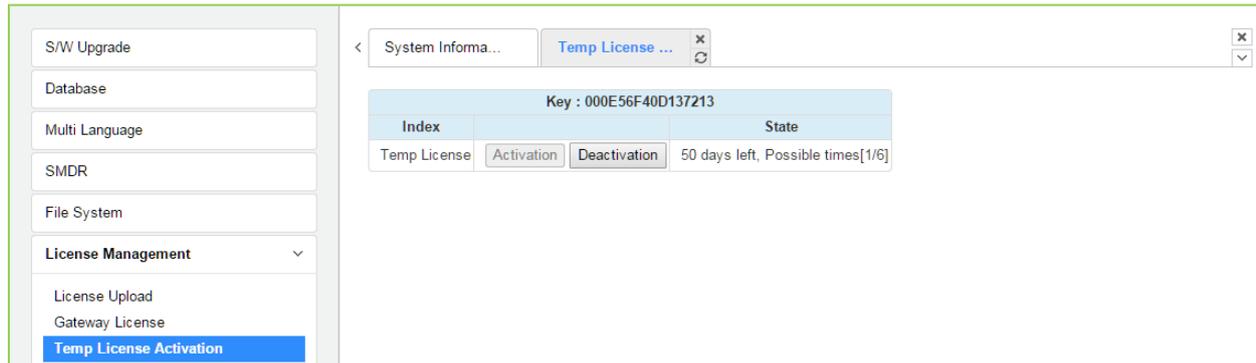
Figure 7.7.3-1 Temp License Activation

Notification

1) Please backup your DB before you activate a temporary license.

2) Must verify the present license before the temporary license expires.

3) If a temp license expires, upload a new license file or use within the scope of current license file. Otherwise, the system may go to [Limited Service Mode].

7.8 DECT Statistics Feature

Selecting DECT Statistics Feature displays the page shown in Figure 7.8-1. Select the desired radial button and, where needed, enter the WTIM sequence number then click **[Send]** to view the statistics.

Figure 7.8-1 DECT Statistics Feature

7.9 VSF Prompt Upload

iPECS UCP delivers system prompts in up to six (6) languages. The prompts for each language are stored in a separate file. Files for each of the languages supported are available from the local Ericsson-LG Enterprise representative. The entire prompt file or individual prompts may be uploaded to the system. Once uploaded to the system the file is employed to play prompts to the iPECS UCP user.

UCP2400 doesn't support VSF prompt message.

7.9.1 Prompt Selection

Selecting Prompt Selection displays the page shown in Figure 7.9.1-1. The System Voice Prompt language files are stored in the VSF memory and are enabled with the Prompt Selection page. A reference "Position" is assigned a language selected from the drop-down menu. Once the language for each position is selected, saving the page enables all the languages selected.
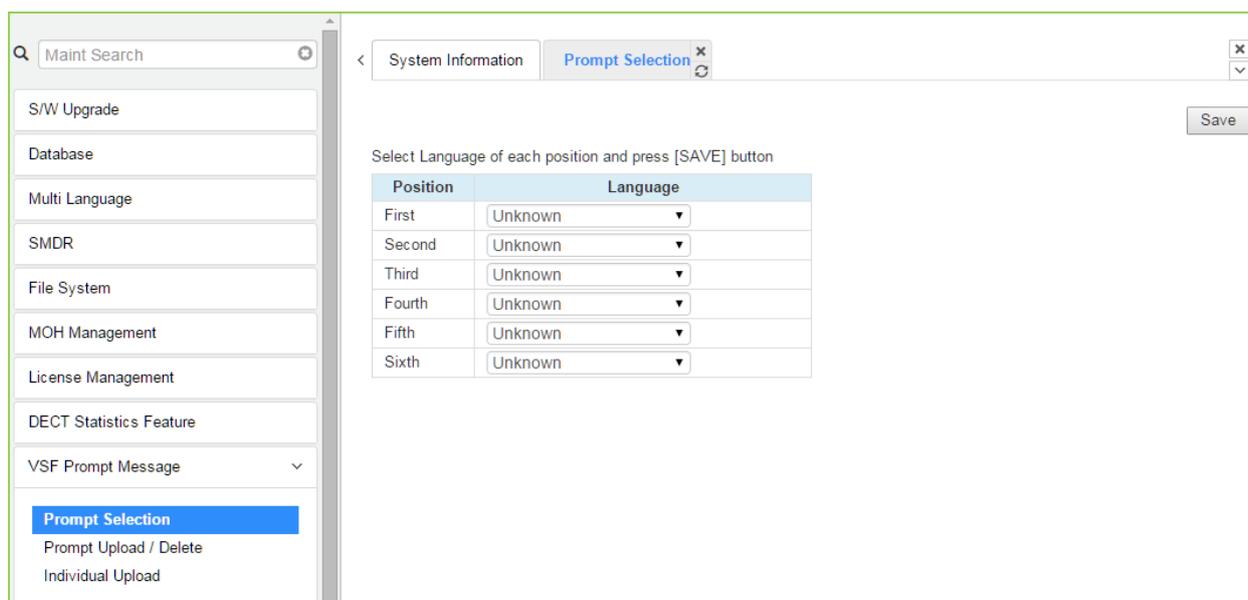
Figure 7.9.1-1 Prompt Selection

7.9.2 Prompt Upload/Delete

Selecting Prompt Upload/Delete displays the page shown in Figure 7.9.2-1. From this page, the entire prompt file for selected languages can be deleted and updated prompt files uploaded to the system. The Add file button is used to upload files. The check box and Delete button are used to delete old or unused language files.
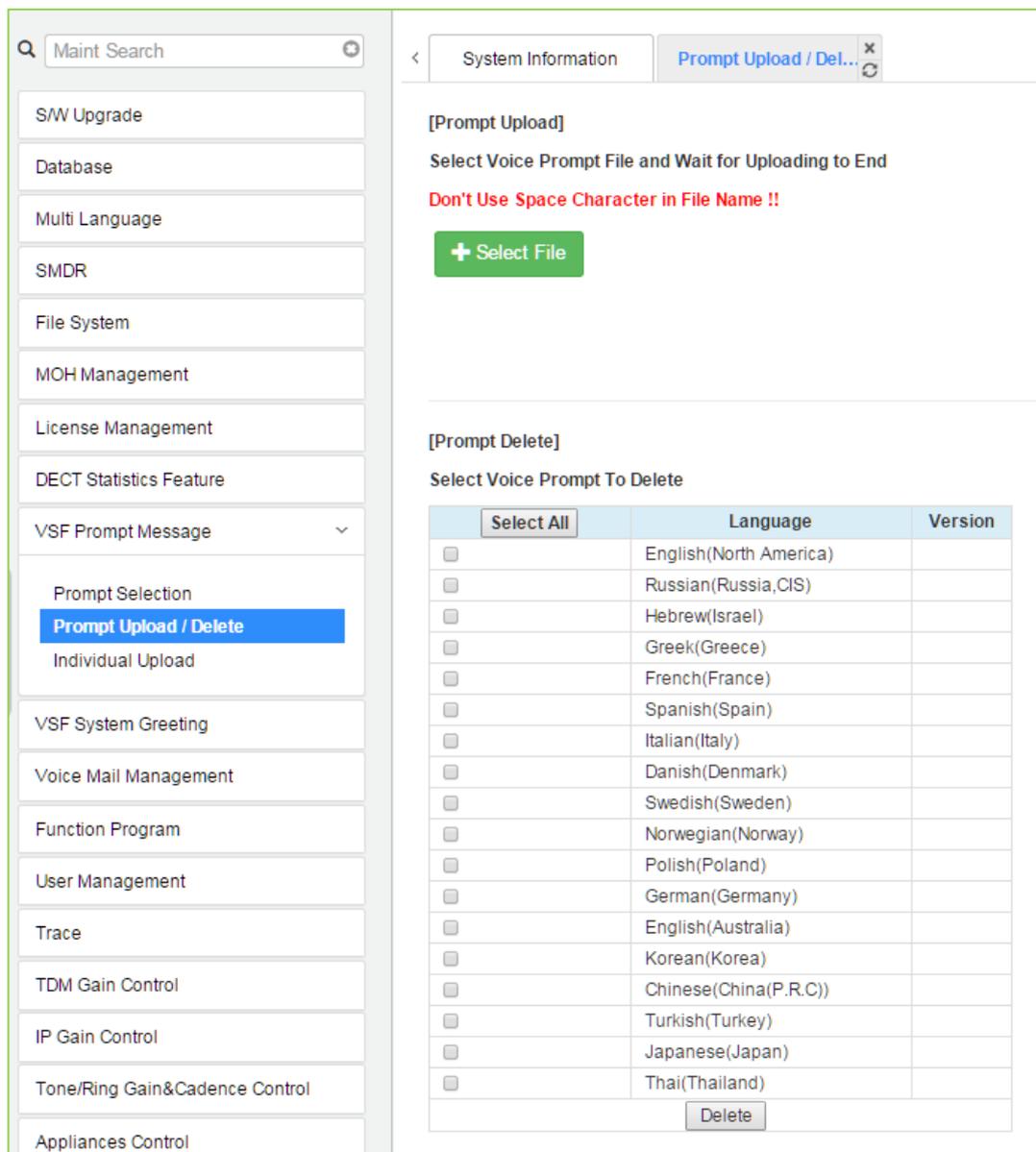
Figure 7.9.2-1 System Prompt Upload/Delete

7.9.3 Individual Upload

Selecting "Individual Upload" will display the page shown in Figure 7.9.3-1. Here individual system prompts can be uploaded to the system for each of up to six (6) languages. Prompt files must be appropriately named and must be in a G.711 a/u-Law format.
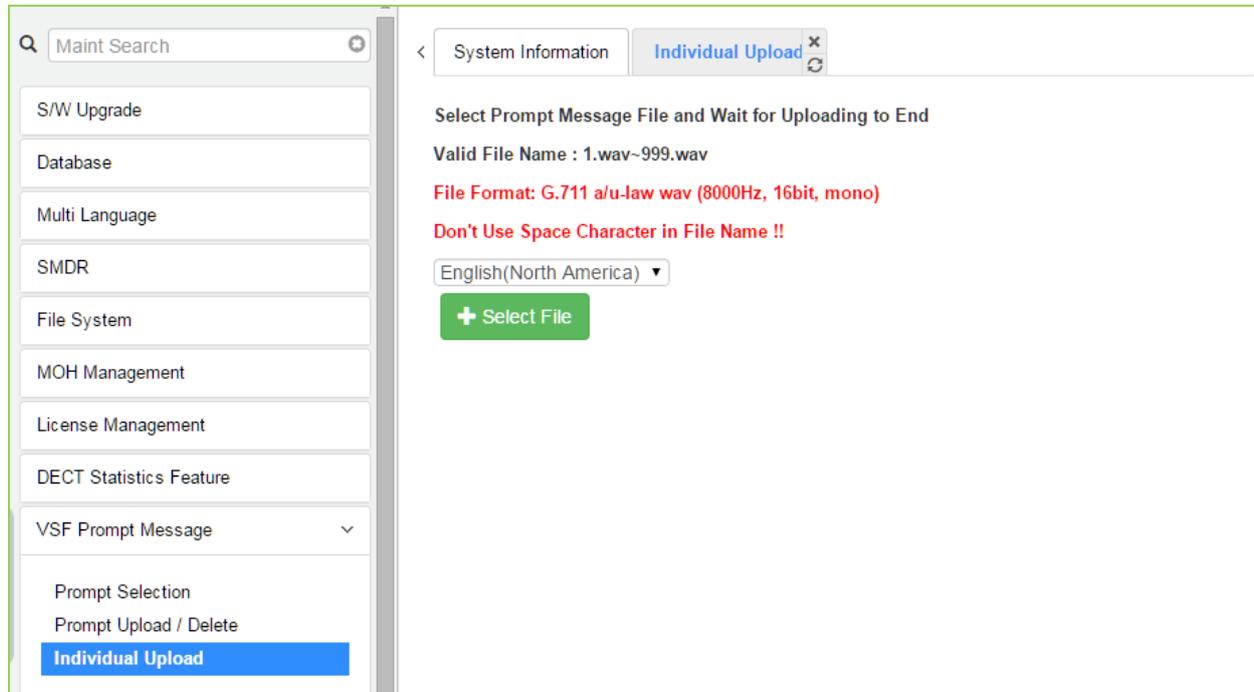
Figure 7.9.3-1 Individual Prompt Upload

7.10 VSF System Greetings

The System Greetings (Announcements) can be downloaded from the system or Uploaded to the system. Individual announcements can be recorded externally and then uploaded to the system. Like prompts, the individual announcement files must be in a .wav format using the g.711 codec. Individual greetings for each Language position, see section 7.8.1, can be uploaded as well as downloaded.   Clicking on the message will download the message to the browser, which will play the message.   To save the message, right click and select save as.   Locate a directory and change the file name, if desired and click save.

All announcements can be downloaded from the system as a back-up file and uploaded to the VSF memory as required.

UCP2400 doesn't support VSF system greeting.

7.10.1 Individual Upload

Selecting Individual Upload displays the page shown in Figure 7.10.1-1. From this page, individual pre-recorded announcements for each language position, see section 7.9.1, can be uploaded from the PC to the VSF gateway. Select the language position and assure the file name matches the appropriate System Announcement number. The file name must be in the format as shown on the page.
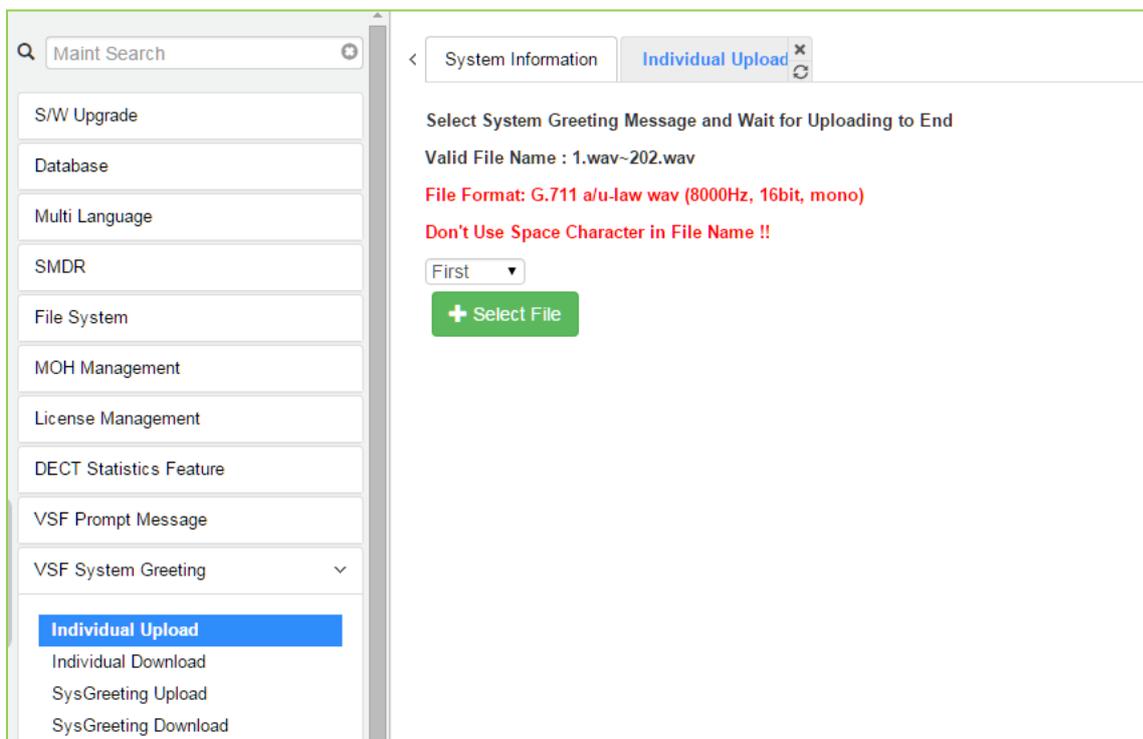
Figure 7.10.1-1 Individual Upload

7.10.2 Individual Download

Selecting Individual Download displays the page shown in Figure 7.10.2-1. The page will display a list of System announcement in the VSF memory. Individual files may be downloaded with the Download button next to the announcement file name or using the checkbox and the delete button the announcement can be deleted from the VSF memory.
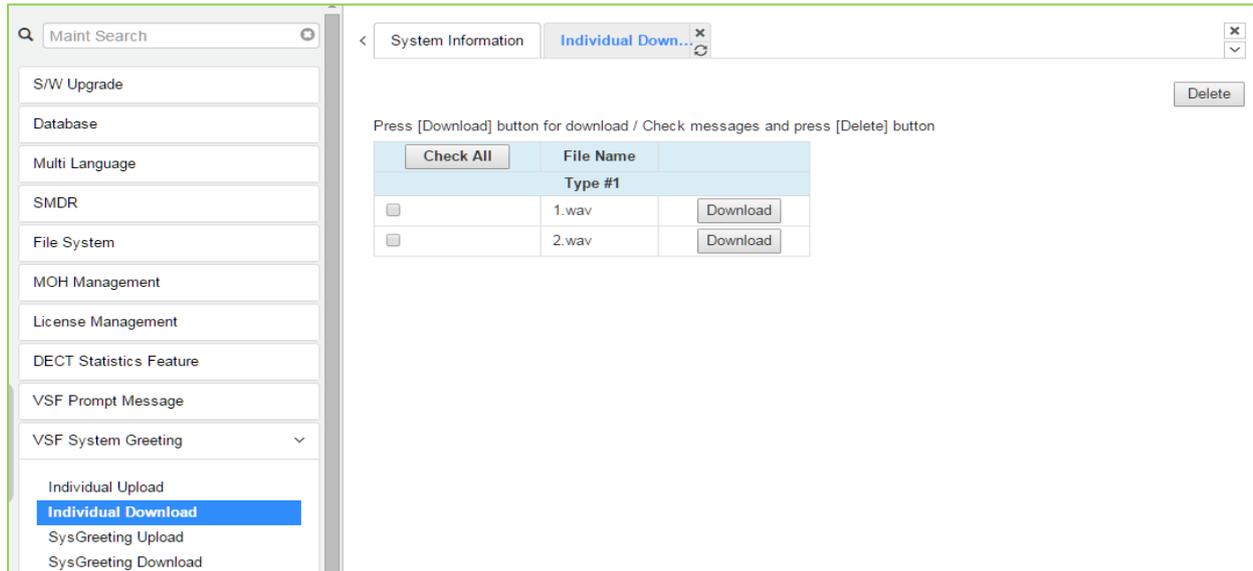
Figure 7.10.2-1 Individual System Announcement Download

7.10.3 System Greeting Upload

Selecting System Greeting Upload displays the page shown in Figure 7.10.3-1. From this page, announcements for each language "Position", see section 7.9.1, can be uploaded to the VSF memory. First, select the language position then click the Add Files button to select the file to upload from the PC.
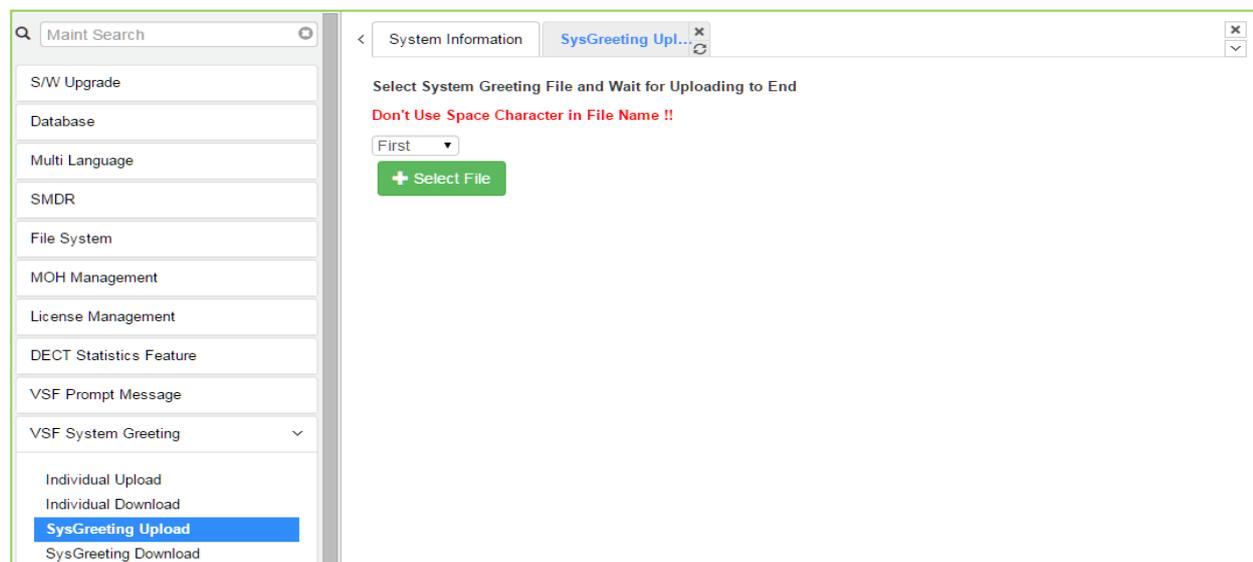
Figure 7.10.3-1 System Announcements Upload

7.10.4 System Greeting Download

Selecting System Greeting Download displays the page shown in Figure 7.10.4-1. From this page, announcements for each language "Position", see section 7.9.1, can be downloaded from the VSF memory as a back-up file. First, select the language position, click the Download button then follow the normal save file process to store the file to the PC.
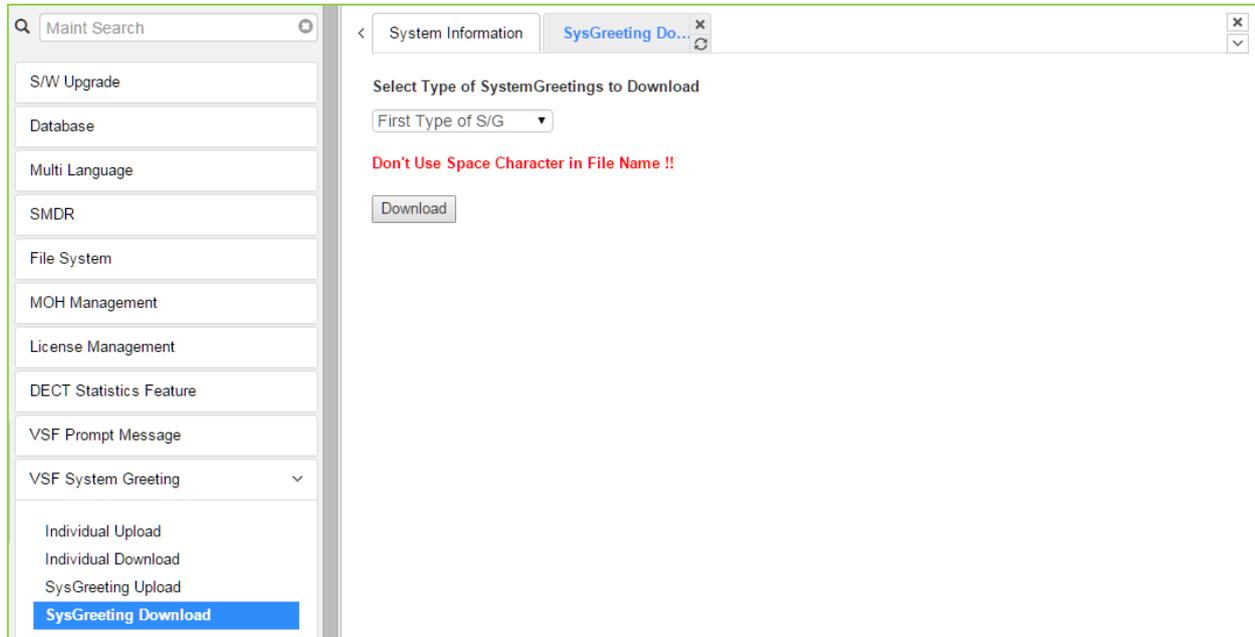
Figure 7.10.4-1 System Announcement Download

7.11 Voice Mail Management

The Voice Mail Management permits the administrator to view the status of the built-in Voice Mail boxes and delete the all messages, delete messages for specific stations, or back-up all the messages to the PC.

7.11.1 Voice Mail Delete

Selecting Voice Mail Delete displays the page as shown in Figure 7.11.1. From this page, messages stored in the VSF gateways can be deleted for all stations or a station range. In addition, using the "Display Station Voice Mail Status" button, the number of messages for each station is displayed.

Figure 7.11-1 Voice Mail Delete

A user can delete user greeting if you set "Including user greeting" option when a user delete voice mail.

7.11.2 Voice Mail USB Backup

Selecting Voice Mail Backup displays the page shown in Figure 7.11.2-1.   This page permits the Administrator to send all Voice messages in the VSF gateway memory to the USB port of the UCP module. Note the VSF gateway must be idle and a USB memory device installed in the UCP module.

UCP2400 doesn't support Voice mail USB Backup.
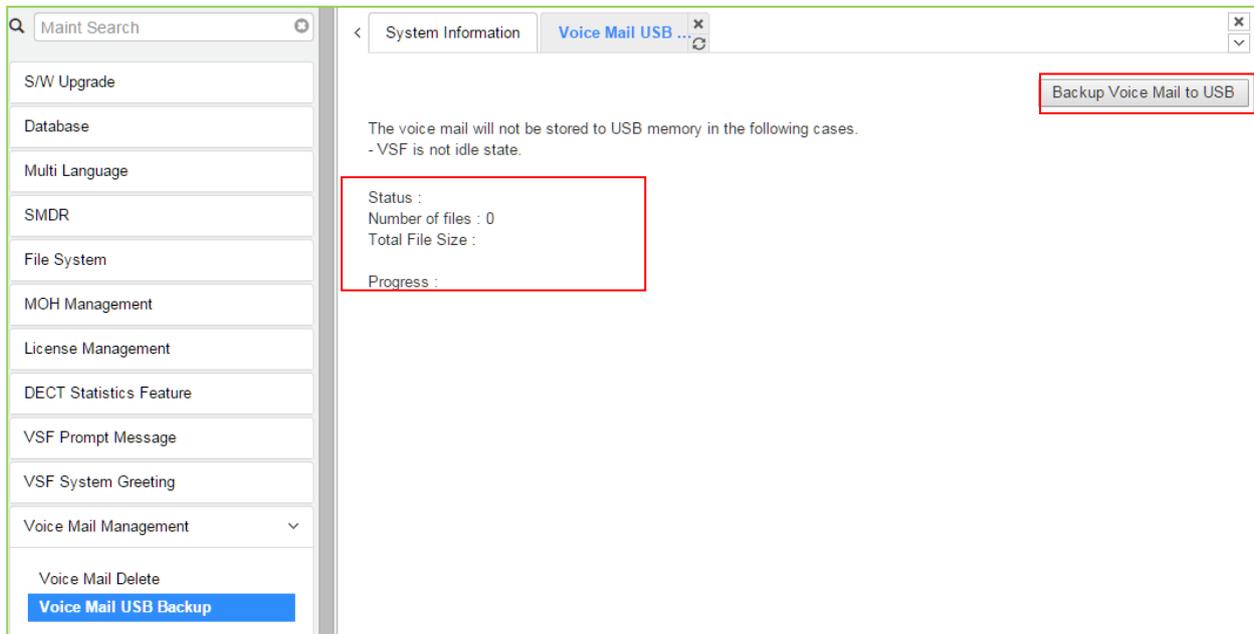
Figure 7.11.2-1 Voice Mail USB Backup

To back up Voice Mail, click the backup Voice Mail to USB button. User can find all information about the status, Number of files, Total file size, and the percentage of Progress by text.

7.12 Function program

The Function Program allows the administrator to build a set of programs with attributes in a structure customized to the administrator. The Function programs can then be accessed under the Administration pages using the "Function Base" button at the top of the left navigation page. Single level and multi-level programs can be structured. The User Function Management permits functions to be deleted or, with the "Add Function" button, new program functions can be added.

7.12.1 User Function Management Page

Selecting User Function Management displays the page shown in Figure 7.12.1-1. Using the check box and Delete button Function programs previously saved can be deleted.
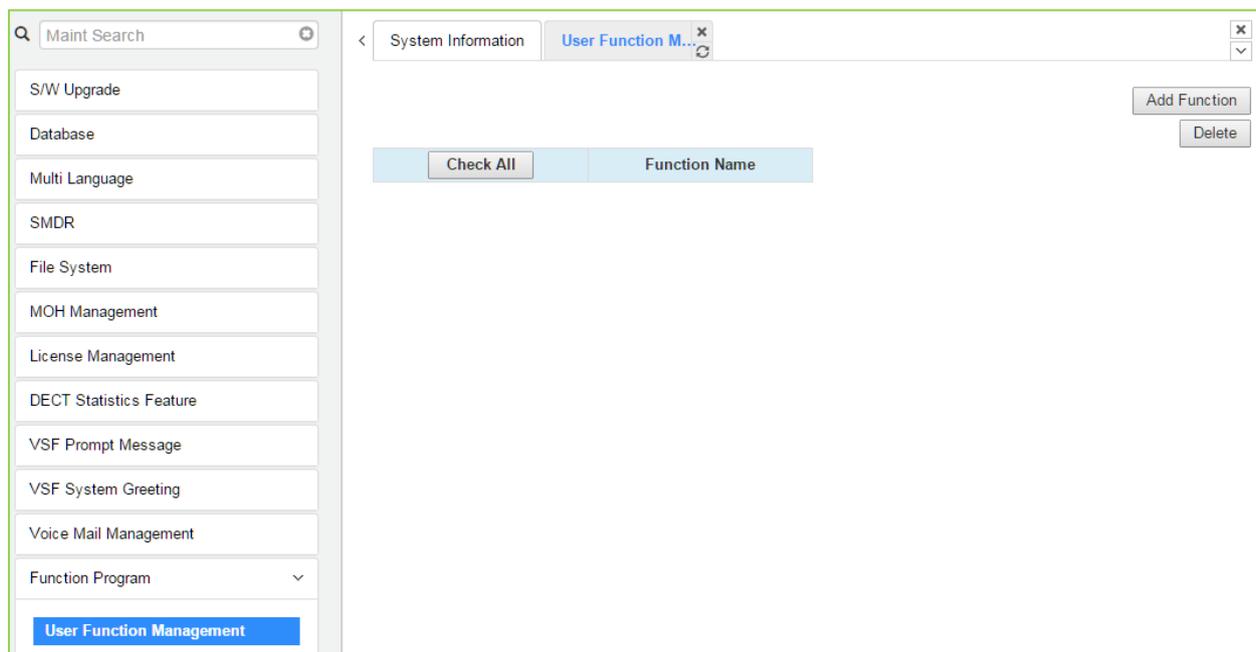
Figure 7.12-1 User Function Management

Adding Function

This step is divided into 5 steps and you can make the desired function each step. The following is the guide to make Name and add the function on each step:

1) To enter the function name, English, Numbering, Underscore (_) and Parentheses are available. Function name has to be filled out.

2) To enter the step name, English, Numbering, and Special letters except Double quotation marks are available.

3) The step name doesn't need to be filled out, but the function has to be configured each step.

4) To cancel or close this tab, click the close button (X) and pop up the blow;

1. Click the Add Function button.

2. On the below window, click the desired PGM in the left frame. First fill out the function name and step name. To configure the step 1, click the Make table to check the desired function and then click the Save button.

   ✓ Check All: check all functions

   ✓ Save: Save the checked functions

   ✓ Initialize: Initialize the checked functions

3. The selected functions are displayed and click the Save button after checking each functions. The rest steps are the same as the step 1.
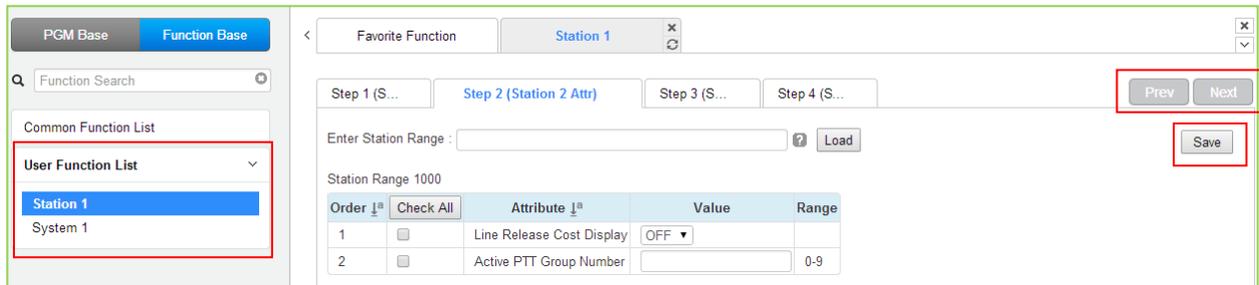

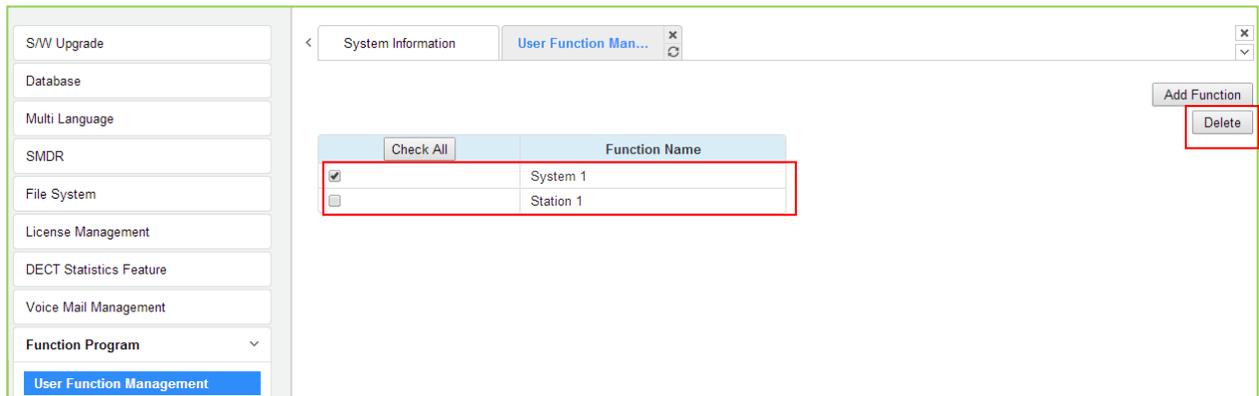4. Finally, click the Save Function to save and then click the OK button.

5. To check the function, click Function Base button in the left frame and User Function List and you will see the following figure.

You can enable or disable the function by checking each function and then click the save after setting Value. Also move to each step by clicking the next button or Previous button.

Deleting Function

To delete the user function, click Maintenance at the top of window and then click the Function Program -> User Function Management. Check the desired function to delete and click the delete button.

7.13 User Management

iPECS UCP supports up to 50 system accounts and up to ten (10) users may simultaneously access system Web services. The default ID is 'admin' and the password is '1234'. The Access privilege is determined based on the entered system account (ID/Password) and the privilege assigned for the user.

Note that access to the Station Web portal employs the Station number and Authorization code managed in PGM 227.

Please follow up the below instructions to make User ID:

1) Maximum 16 characters & digits

2) In English only

3) First letter must be Alphabet. The rest are available: Alphabet, number, underscore (_)

4) Don't use 'blank'.

It is strongly recommended that a unique User ID and strong password be entered to minimize the risk of admin and maintenance access by unauthorized personnel.

User should register more than a maintenance ID.

Figure 7.13-1 User Management

7.14 Trace

The UCP system software incorporates routines to monitor and output detailed call and feature processing information, and event logs. Information is provided on a system or device level as requested. User can find the information on this page.
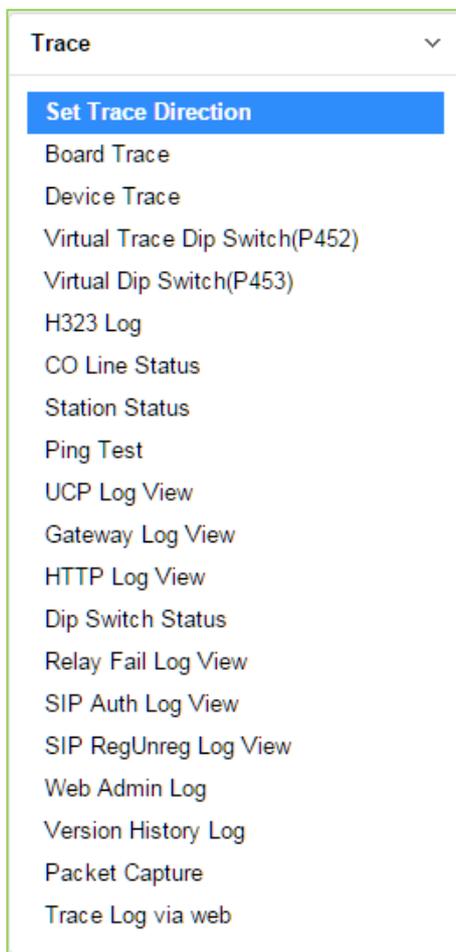
Figure 7.14-1 Trace Main Page

We explain [Packet Capture] and [Trace Log via web] among them because it is very useful to use Web admin.

Packet Capture

A user can start and stop capturing packets and download packet capture result. The dump file will be replaced when new capturing is started.

Default option is included. The file name is packetdump.cap. -i eth0 -w packetdump.cap -c 10000

- Option explanation

protocol filter: tcp or udp or icmp

port filter: both source and destination port -> port #number, source port -> src port #number, destination port -> dst port #number

IP filter: both source and destination host IP -> host #IP_ADDR, source host IP -> src host #IP_ADDR, destination host IP -> dst host #IP_ADDR

Example

UDP and Port 5588 and Source IP address 10.1.1.100: udp and port 5588 and src host 10.1.1.100

Trace Log via web

A user can start and stop logging and download trace log result. The log file will be replaced when new logging is started.
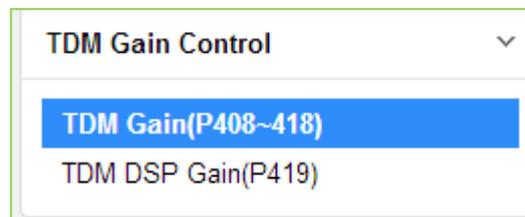
1. Set 'Trace Attribute' to 'COM1(UART1)' from 'Set Trace Direction' menu.

2. Set 'Board Trace', 'Device Trace' and 'Virtual Trace Dip Switch(P452)' menu.

3. Click 'Start' button.

7.15 TDM Gain Control

Control voice gain of TDM device for each direction.

Figure 7.15-1 TDM Gain Control Main Page

7.16 IP Gain Control

Audio gain for signals from and to each type of device is adjustable through the IP Gain Control page. The default gain values for the system are set to match the Nation Code. For more information to change Value, it is recommended that you ask your dealer or an authorized Ericsson-LG Enterprise representative. We can't guarantee the damage according to changing Value arbitrarily.
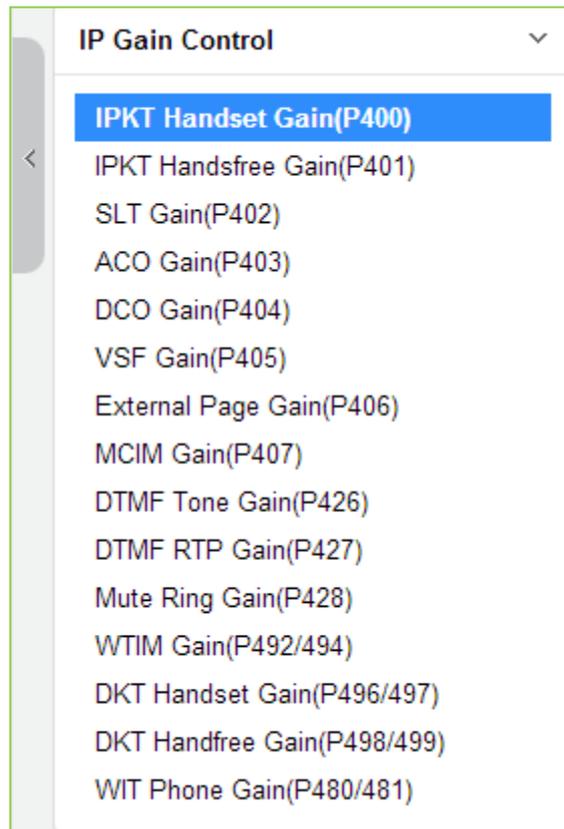
Figure 7.16-1 IP Gain Control Main Page

7.17 Tone/Ring Gain & Cadence Control

Selecting Tone Table will display the page. Here the tones, cadence and gain used with features can be managed. For more information to change Value, it is recommended that you ask your dealer or an authorized Ericsson-LG Enterprise representative. We can't guarantee the damage according to changing Value arbitrarily.
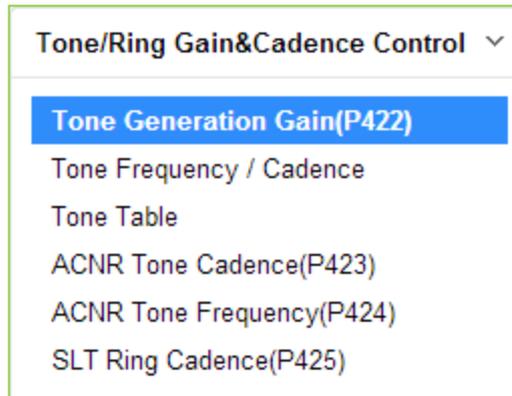
Figure 7.17-1 Tone/Ring Gain & Cadence Control Main Page

7.18 Appliances Control

The Appliances Control page permits control of several characteristics of IP phones and analog CO lines such as Comfort Noise level and side tone. For more information to change Value, it is recommended that you ask your dealer or an authorized Ericsson-LG Enterprise representative. We can't guarantee the damage according to changing Value arbitrarily.


Figure 7.18-1 Appliances Control Main Page

8. STATION PROGRAM (USER PORTAL)

This Station Program helps our customer program each station so that each station can be easily set the value and find which features are used. For information, refer to the above section.

To access Station program:

First, set the station authorization code table (PGM 227) for each station as shown in the following figure and then click **[Log Out]**.

Figure 8.1 Station Authorization Code Table – PGM 227

Second, enter the desired station number in the user ID and enter the authorization code in the password as shown in Figure 8.2.

Figure 8.2 Login Page

You will see the below Web page and find Station program sub-menu in the left frame.
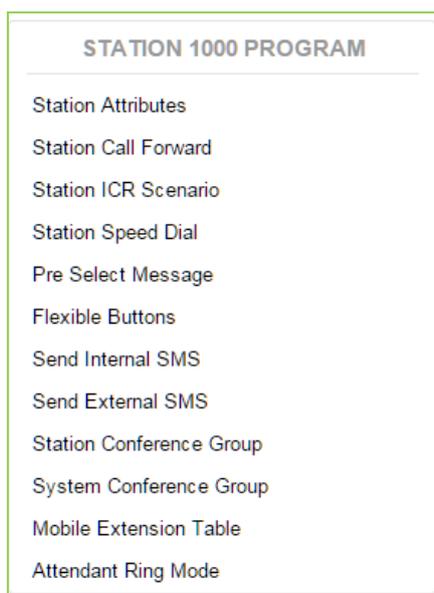
Figure 8.3 Station Portal Main Page

In the portal, users can modify various station attributes, set-up call forwarding, create Station ICR scenarios, assign Flexible buttons, program Station Speed dial numbers, send Short Messages and create conference groups. The following sections provide details on each of the available Station Program Web pages.

The above Admin. feature may be different according to 'Station Web Level' (Station Data > Common Attribute (111)).

The Portal main page has three sections,

- Station selected – Upper left frame
- Web site directory & navigation section – Left frame
- Info and Entry section – Central frame

8.1 Station Attributes

Selecting Station Attributes will display the input entry page.

Figure 8.1-1 Station Portal Station Attributes

Station Attributes define features and functions available to the station. Refer to Table 8.1-1 for a description of the features and the input required.

Table 8.1-1 STATION ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
DND	DND (Do-Not-Disturb) can be activated from the User Portal.	OFF ON	OFF
ICM Signaling Mode	The user may select Hands-free (H), Privacy (P) or Tone Ring (T) for the ICM Signaling mode.	H, T, P	T
Call Coverage Mode	The Call Coverage feature permits an iPECS Phone user to receive ring and answer calls for other stations.	OFF ON	OFF
Delay Ring Cycle	When a covered station rings, the **{CALL COVERAGE}** button LED will flash at the covering station and the station will receive ring (immediate or delayed, 0 to 15 ring cycles).	0~15	0
Headset Usage	The user may enable the iPECS IP or LDP-7000 series phone for use with a Headset.	OFF ON	OFF
Authorization Code	Authorization codes are employed to control access to the system resources and facilities. Walking COS, CO/IP Group access DISA callers and certain Call Forward types may require the input of a valid Authorization code. Codes up to 12 digits may be entered.	Max. 12 digit (Include *)	N/A

Table 8.1-1 STATION ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
Call Wait	When a busy station receives a call, the call may queue to the station instead of receiving busy tone. With Call Wait, the caller hears Ring-back and the user sees the CO line button LED flash.	OFF, For External/Internal, For External, For Internal	For External/Internal
Choice Exec/Sec Message	When a call forwards to the Secretary of an Executive/Secretary pair, messages can be left for the Executive (ON) or Secretary (OFF).	OFF ON	ON
Wake up Time	The user can register a Wake-up time and set the Wake-up alarm to repeat on a daily basis. Available only system Attendant New 5 Wake up usage is Off.	hhmm (Must be 4 digits)	
Attendant Wake up time	The user can register Attendant Wake-up time and set the Wake-up alarm to repeat on a daily basis. Available only system Attendant New 5 Wake up usage is Off.	hhmm (Must be 4 digits)	
Headset or Speaker Mode	The device, Speakerphone or Headset, to receive audio for a call can be selected.	Speaker Headset	Speaker
Headset Ring Mode	Ring for an incoming call can be delivered to the Headset, Speaker or both the Headset and Speaker.	Speaker Headset	Headset
User Name Registration	A name can be entered to display in the LCD of iPECS IP and LDP Phones calling the station.	Max. 20 characters	
BGM	From the portal, the user can select the source for Background Music that is played over the iPECS IP or LDP phone speaker while the station is idle.	No BGM, Internal/External Music 1, External Music 2, VSF MOH, SLT MOH1~5, VSF MOH2~3	No BGM
Station Ring Type	The user may select one of eight ring signals for Intercom calls.	1-8	1
CO Ring Type	The user may select one of eight ring signals for incoming outside calls.	1-8	1
SMTP Server Address	This field defines the address or url of the SMTP mail server for the notification.	IP v4 addressor Mail server name	
VSF MSG – User Mail Address	The system can send an E-mail to the user as notification of a new Voice message in the user built-in Voice Mailbox. This field defines the E-mail address to notify when a new message is received.	E-mail address	
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 System No-answer timer.	000-600 seconds	0
CLIR Service	CLIR (Calling Line Identification Restriction), an ISDN service, removes calling party Id sent from the ISDN to the called party with a RESTRICT instruction in the SETUP message. If enabled here, the system will send the RESTRICT instruction to the PSTN when an outgoing ISDN call is placed.	OFF ON	OFF
Log in/out Agent	An Agent may change their Station group duty status	No Action,	No Action

Table 8.1-1 STATION ATTRIBUTES

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
from Station Group	(ON or OFF duty) through the User Portal.	Log in Agent-ON Duty, Log out Agent-OFF Duty	
LCD Back Light Usage	The backlight of iPECS IP and LDP Phones is assigned to stay off, light only when the station is busy, or light constantly.	ALWAYS OFF, BUSY ONLY, ALWAYS ON	BUSY ONLY
VM Message No	This field displays the message status of the built-in Voice Mail. The number of New, Saved and Urgent messages is shown.		
Company Directory - First Name	The user's First Name for the Company Directory is assigned.		
Company Directory - Last Name	The user's Last Name for the Company Directory is assigned.		
Message Rewind/Fast-Forward Time	When the user selects to Rewind or Fast-forward while listening to a message in the built-in Voice Mailbox, the message will rewind or Fast-forward by the time entered.	3 ~ 99 seconds	4

8.2 Station Call Forward

Selecting Station Call Forward displays the input entry page.

Figure 8.2-1 Station Portal Call Forward

Stations can be programmed so that incoming calls re-route to another station (local or networked), a station group, the built-in Voice Mailbox, or over a system CO/IP line (Off Net). Call Forward can be separately assigned to forward calls unconditionally, or when the call encounters a Busy, No-Answer or Busy/No Answer condition.

8.3 Station ICR Scenario

Selecting Station ICR Scenario displays the input entry page.

Figure 8.3-1 Station Portal ICR Scenario

Station ICR is an extension of call forward where the user enters scenarios to define the call forward feature. Each station has ten (10) routing scenarios that define conditions for routing a user's incoming calls. Each scenario may define time of day, day of week, date, caller ID and destination for incoming calls. In addition, the scenarios may be prioritized; calls are routed to the destination with the highest priority-matching scenario.

8.4 Station Speed Dial

Selecting Station Speed Dial will display the input entry page. Enter the Speed Dial Bin range then click **[Load]** to enter Speed Dial information.

Figure 8.4-1 Station Speed Dial Portal

Each station can store commonly dialed numbers for easy access using Station Speed Dial bins. Each Speed Dial number can be up to 25 digits in length and may include special instruction codes for analog and ISDN lines. The CO Line used with the Speed Dial number must be entered and a name can be assigned for use with Dial-by-Name dialing.

8.5 Pre-selected Message

Selecting Pre-selected Message will display the input entry page.

STATION 1000 PROGRAM	‹	Station Information	Pre Select Message ✕	✕

Station Attributes
Station Call Forward
Station ICR Scenario
Station Speed Dial
Pre Select Message
Flexible Buttons
Send Internal SMS
Send External SMS
Station Conference Group
System Conference Group
Mobile Extension Table
Attendant Ring Mode

Save

Index	Messages	Attribute	Range
#	MESSAGE DEACTIVATED		
0	STATION CUSTOM MESSAGE		Max 24 Characters
1	LUNCH RETURN TIME		hhmm (Must be 4 digits)
2	ON VACATION RETURN AT DATE		MMDD (Must be 4 Digits)
3	OUT OF OFFICE RETURN TIME		hhmm (Must be 4 digits)
4	OUT OF OFFICE RETURN DATE		MMDD (Must be 4 Digits)
5	OUT OF OFFICE RETURN UNKNOWN		
6	CALL TO STATION		Phone Number
7	IN OFFICE STATION		Station Number
8	IN A MEETING TIME		hhmm (Must be 4 digits)
9	AT HOME		
10	AT BRANCH OFFICE		
11	SYSTEM CUSTOM MESSAGE		
12	SYSTEM CUSTOM MESSAGE		
13	SYSTEM CUSTOM MESSAGE		
14	SYSTEM CUSTOM MESSAGE		
15	SYSTEM CUSTOM MESSAGE		
16	SYSTEM CUSTOM MESSAGE		
17	SYSTEM CUSTOM MESSAGE		
18	SYSTEM CUSTOM MESSAGE		
19	SYSTEM CUSTOM MESSAGE		
20	SYSTEM CUSTOM MESSAGE		
VSF MESSAGE PLAY TO CO INCOMING CALLER	OFF ▾		

Figure 8.5-1 Pre-selected Message Portal

Users can select a message to be displayed in the LCD of a calling iPECS IP or LDP Phones. There are ten pre-defined messages (index 1-10) and several messages allow for auxiliary information such as a time, date or number.

A user may activate Custom Display Messaging to send a custom text message to the LCD of a calling iPECS IP and LDP Phones. Up to 11 Custom Messages (ten system level and one for each user) may be entered in the system database. System level Custom Messages (index 11-20) may be entered from the Attendant or Administrator phone or via the Web Admin & Maintenance page. The Station Custom Message (index 0) may also be assigned from the station or via the Station portal Web page.

8.6 Flexible Buttons

Selecting Flexible Buttons will display the input entry page. Select the Station Type from the drop-down menu then select **[Load]** to modify the Flexible Button data.

Button	Type	Value	Label
Flex Button 1	CO Line	1	
Flex Button 2	CO Line	2	
Flex Button 3	CO Line	3	
Flex Button 4	CO Line	4	
Flex Button 5	CO Line	5	
Flex Button 6	CO Line	6	
Flex Button 7	CO Line	7	
Flex Button 8	CO Line	8	
Flex Button 9	CO Line	9	
Flex Button 10	CO Line	10	
Flex Button 11	Loop		
Flex Button 12	Loop		
Flex Button 13	Station Number	1000	
Flex Button 14	Station Number	1001	
Flex Button 15	Station Number	1002	
Flex Button 16	Station Number	1003	
Flex Button 17	Station Number	1004	
Flex Button 18	Station Number	1005	
Flex Button 19	Station Number	1006	
Flex Button 20	Station Number	1007	
Flex Button 21	Station Number	1008	
Flex Button 22	Station Number	1009	
Flex Button 23	Station Number	1010	
Flex Button 24	Station Number	1011	

STATION 1000 PROGRAM

Station Attributes
Station Call Forward
Station ICR Scenario
Station Speed Dial
Pre Select Message
Flexible Buttons
Send Internal SMS
Send External SMS
Station Conference Group
System Conference Group
Mobile Extension Table
Attendant Ring Mode

Station Information   Flexible Buttons

Select Station Type : Master Station ▼ Load   Save

Station Type : Master Station

Figure 8.6-1 Flexible Buttons

Each Flex button for each iPECS IP and LDP Phone, and DSS Console can be assigned a function (Type) from the drop-down menu. After selecting the Type for a button, enter the value, if required.

If the station employs the LCD to display the button function, the LSS Label field can be used to assign a label that displays in the LCD.

8.7 Send Internal and External SMS

Selecting Send Internal SMS displays the input entry page.

Figure 8.7-1 Send Internal and External SMS

A short message can be sent to the LCD of other iPECS IP and LDP Phones; enter the station range to receive the message and the message body then click **[Send]**.

8.8 Station Conference Group

Selecting Station Conference Group displays the input entry page. Enter the Conference Group Number then click **[Load]** to enter the group attributes and identify members of the group as shown in Figure 8.8-2.

STATION 1000 PROGRAM		Station Information	Station Conferen... ✕ ↻
Station Attributes			
Station Call Forward		Enter Station Conference Group Number (0 - 99) : [ ] [Load] [Overview]	
Station ICR Scenario			
Station Speed Dial			
Pre Select Message			
Flexible Buttons			
Send Internal SMS			
Send External SMS			
Station Conference Group			
System Conference Group			
Mobile Extension Table			
Attendant Ring Mode			

Figure 8.8-1 Station Conference Group Portal

Figure 8.8-2 Station Conference Group Attributes & Members

Enter a Group Name and Group members, which can be internal and external parties. In addition, the various attributes such as password for group entry can be modified.

8.9 System Conference Group

Selecting System Conference Group displays the input entry page. Enter the Conference Group Number then click [Load] to enter the Conference Group attributes and members as shown in Figure 8.9-2.
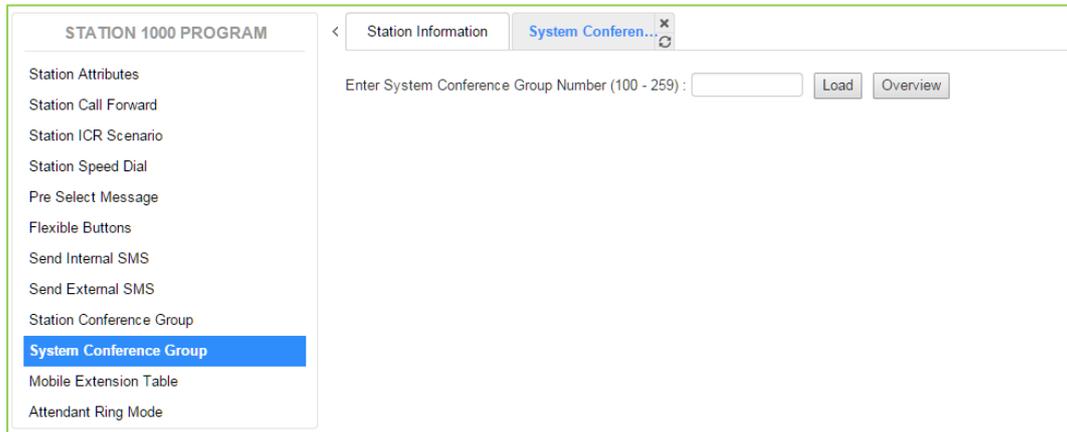
Figure 8.9-1 System Conference Group
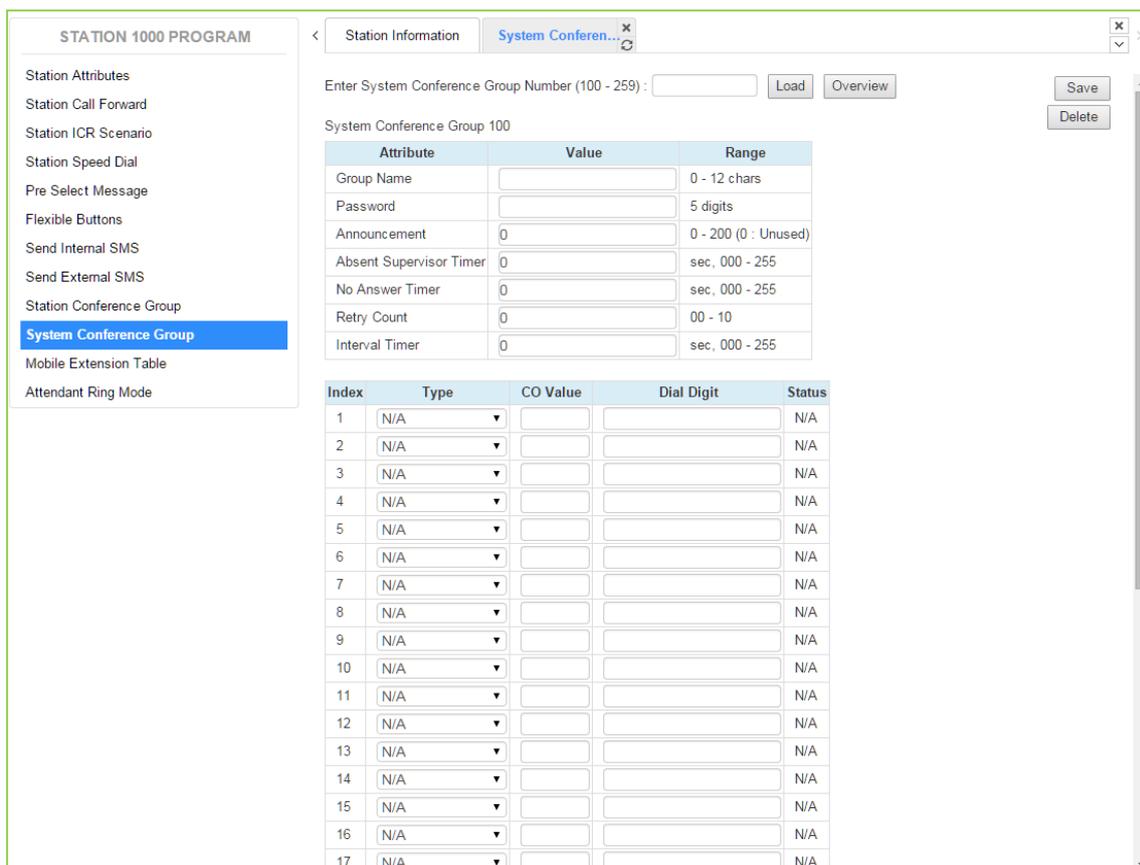
Figure 8.9-2 System Conference Group Attributes & Members

Enter a Group Name and Group members, which can be internal and external parties. In addition, the various attributes such as password for Group access can be modified.

8.10 Mobile extension Table

Selecting Mobile Extension displays the attributes entry page.

Figure 8.10-1 Mobile extension

Mobile Extension attributes enable Mobile Extension and control the feature use. The user can activate Mobile Extension to receive business calls on their mobile and request Mobile notification of new Voice messages.

8.11 Attendant Ring Mode

Selecting Attendant Ring Mode displays the input page.

Figure 8.11-1 Attendant Ring Mode

An Attendant can change the active system Ring mode from the Auto mode and select a specific mode including a specified scenario group for incoming call routing.

9. ON-LINE WEB USER GUIDE

We provide On-line web user guide about the frequent use of features to a user.

You can access it in the login page as below without entering ID and Password.

We didn't explain the user guide because you can easily get the information on the following features on the web.

Appendix A *USER PROGRAM CODES*

User Program Codes, refer to Table A-1 and A-2, are digit sequences users and the Attendant may dial while in the User Program Mode to assign Flex buttons and affect the status of a feature or setting. For more information on the User Program Mode, refer to the **iPECS UCP Features and Operation Manual**.

Many of these codes may be assigned to a button of an iPECS IP or LDP Phone by pressing the **[PGM]** button as the first entry of a Flexible button assignment, see Flex button Programming in the **iPECS UCP Features and Operation Manual**.

Table A-1 Station User Program codes

To use these codes, press the Trans/PGM key then enter the below code.

Code	Description	Code	Description
11X	Intercom Differential Ring	82	{COLR} Button Assignment
12X	CO Line Differential Ring	83	{ATD DND} button assignment
13	Intercom Answer Mode (1: HF/2: TONE/3: PV)	84	{Account Code} Button Assignment
14X	Call Coverage Attribute Setting	85	{LOOP} Button Assignment
15X	Station Ring Download	86	{ATD Intrusion} Button Assignment
21	Knock Down Station COS	87	{INTERCOM} Button Assignment
22	Restore Station COS	88	{Camp-on} Button Assignment
23	Walking COS	89	{Send Keypad Facility IE} Button Assignment
24	ICR Scenario	8#	{OHVO} Button Assignment
25	LIP Keyset Stat	8*1	DID Restriction Button Assignment
26	Call Profile	8*2	DISA Restiction Button Assignment
30	VM Mobile Notify	8*3	Bomb Threat History Button Assignment
31	Message Retrieve Method	8*5	Headset Button Assignment
32	Message Retrieve Example	8*6XYZA	Toggle Ring Mode Button Assignment X,Y,Z,A : 1(Day), 2(Night), 3(Timed), 4(Auto)
33	User Authorization Code Registration	90	{SPEED} Button Assignment
34	DID Call Wait	91	{CONFERENCE} button assignment
35	Message Wait in Executive/Secretary pair	92	{CALLBACK} button assignment
36	Send SMS Message	93	{DND} button assignment
37	Register Mobile Extension	94	{FLASH} button assignment
38	Make Mobile Extension active	95	{MUTE} button assignment
39	Register Mobile Extension CLI	96	{MONITOR} button assignment
41	Set Wake-Up Time	97	{REDIAL} button assignment
42	Wake-Up Time Disable	98	{CALL FORWARD} button assignment
51XX	Custom/Pre-select Message Display (XX = 00-20)	99	{PTT} button assignment
52	Register Custom Message (Message 00)	9*XX	{In-Room Indication} button assignment (XX = 01-10)
53	Active Conference Room	*6XXX	Set Forced Fwd Dest (XXX : hunt group no.)
54	Deactive Conference Room	*7XXX	Forced FWD To Dest (XXX : hunt group no.)
55	Monitor Conference Group	*8	Register Bluetooth
57	{Call Log Display} Button Assignment	*9	Bluetooth Usage
61	Headset/Speakerphone Mode	*0	Hotdesk Login
62	Change Ring Mode	**	Hotdesk Logout
71	LCD Display Mode		
72	Version Display		
73	Background Music		[1] STA RING TYPE / [2] CO RING TYPE
74	Station User Name Registration		
75	Display Phone IP Address		[3] ANSWER MODE / [4] CALL COVER ATTR
76	Change Phone IP Address		
77	Display Phone MAC Address		[5] STA RING DOWNLOAD
78	Network Config		[1] RING/NEWKEYSET
79	Display Phone Version		[2] COS
7#	Forced Release Button Assignment		[1] COS DOWN / [2] COS RESTORE
7*	Display Serial number/Package for SMEMU		
80	{Record} Button Assignment - With Voice Mail		[3] WALKING COS
81	{CLIR} Button Assignment		

Table A-2 ATTENDANT USER PROGRAM FIXED FUNCTION CODES

CODE	FUNCTION
0111	Print SMDR, by Station
0112	Delete SMDR, by Station
0113	Display Call Charge
0114	Abort Print
0115	Print Lost Call Report
0116	Delete Lost Call Report
0121	Print Traffic Analysis (All summary)
0122	Print Traffic Analysis (All summary periodically)
0123	Abort "Print Traffic Analysis (All summary periodically)"
0124	Print Traffic Analysis (Attendant)
0125	Print Traffic Analysis (Call summary)
0126	Print Traffic Analysis (Call Hourly)
0127	Print Traffic Analysis (H/W Usage)
0128	Print Traffic Analysis (CO summary)
0129	Print Traffic Analysis (CO Hourly)
021	Knock Down Station COS
022	Restore Station COS
031	Authorization Code Registration
032	Erase Authorization
041	System Date/Time Mode
042	LCD Date Mode
043	LCD Time Mode
044	Set Wake Up Time from Attendant
045	Wake Up Disable from Attendant
046	PX Clock Set through ISDN message
051	Custom/Pre-select Message
052	DND/Call Forward/Pre-selected MSG Cancel
053	Custom Display Message (11-20)
054	Delete Conference Room
055	Monitor Conference Room
06	VSF – Record System Greeting
071	Dial By Name
072	Isolate Fault CO Line
073	Automatic Day/Night/Timed Ring Table
074	External Page Music -1 Assignment/Cancel
075	External Page Music -2 Assignment/Cancel
076	LCD Display Language
077	PTT Login with station range
078	Display cpu redundancy state
08	Emergency Log
*#	Admin Programming Code

Appendix B *FLEXIBLE NUMBERING PLAN*

The System Numbering Plan can be selected from 1 of 9 basic Numbering Plans. Table B-1 provides a brief description of the plans and Table B-2 and Table B-3 provide the Numbering Plan codes for each of the eight basic plans. The Number Plan is selected in Program 100 and individual codes in the plan can be changed using the Flexible Numbering Plan Programs 106 to 109.

Table B-1 Numbering Plan Description

Plan Number	Description	Range
1	Basic Numbering Plan	1000 – 3399
2	Korea Default	1000 – 3399
3	Australia Default	1000 – 3399
4	New Zealand Default	1000 – 3399
5	Italy Default	2000– 4399
6	Finland Default	2100 –4499
7	Sweden Default	1000 – 3399
8	Norway Default	1000 – 3399
9	Israel Default	1000 – 3399

Table B-2 Basic Numbering Plan 1-4, Default Values

Feature	Basic Numbering Plan				Remark
	1	2	3	4	
Intercom Call	1000 ~3399	1000 ~3399	1000 ~3399	1000 ~3399	
Internal Page Zone	*301~*400	*301~*400	*101~*200	*301~*400	
Internal All Call Page	543	*543	#3	#43	
Meet Me Page	544	*544	##	##	
External Page Zone 1-2	545-546	*545-*546	#41-#42	#41-#42	
External All Call Page	548	*548	#5	#5	
All Call Page	549	*549	#00	#00	
SMDR Account Code Enter	550	*550	550	#9	SLT
Flash Command to CO Line	551	*551	551	551	SLT
SLT Last Number Redial	552	*552	552	552	SLT
DND (Toggle On/Off)	553	*553	553	553	SLT
Call Forward	554	*554	554	554	
Speed Dial Programming	555	*555	555	555	SLT
Activate Message Wait/Callback	556	*556	556	*66	SLT
Message Wait/Callback Answer	557	* 557	557	*67	SLT
SLT Speed Dial Access	558	* 558	558	#8	SLT
DND/FWD cancel	559	* 559	559	559	SLT
SLT CO System Hold	560	* 560	560	560	SLT
SLT Program Mode Access	561	* 561	561	561	SLT
Attendant Unavailable	562	* 562	562	562	
AME Feature	564	* 564	564	564	
Alarm Reset	565	* 565	565	* 565	

Table B-2 Basic Numbering Plan 1-4, Default Values

Feature	Basic Numbering Plan				Remark
	1	**2**	**3**	**4**	
Group Call Pickup	566	* 566	**	*1	
Universal Night Answer	567	* 567	567	2	
Account Code with bin	568	* 568	568	568	
Walking COS	569	* 569	569	569	
ACD Agent On/Off Duty	571	* 571	571	571	
ACD Supervisor Login	572	* 572	572	572	
ACD Supervisor Logout	573	* 573	573	573	
ACD Help Code	574	* 574	574	574	
ACD Calls In Queue Display	575	* 575	575	575	
ACD Supervisor Status	576	* 576	576	576	
ACD Supervisor Monitor	577	* 577	577	577	
ACD Reroute Queued Call Answer	578	* 578	578	578	
ACD Reroute Queued Call No Answer	579	* 579	579	579	
Camp-On Answer	621	* 621	621	621	SLT
Call Parking Locations	#601~#800	#601~#800	#601~#800	#101~#300	
Station Group Pilot Number	401 – 500	*401 – *500	*401 – *500	#620 – #719	
Station User VSF Features Access	66x	66x	*66x	69x	x: 1 ~ 3
Call Coverage Ring	76	*76	*76	67	
Direct Call Pickup	77	*77	*77	*77	
Access CO Group	89xx	89xx	89xx	89xx	xxx: 000~201,
Access Individual CO/IP Line	88xxx	88xxx	88xxx	48xxx	xxx: 001~999
Access Held CO/IP	8*	8*	8*	4*	
Access Held Individual CO/IP	8#xx	8#xx	8#xx	4#xx	xx: 01~74
Access CO in First CO Group	9	9	9	1	
Attendant Call	0	0	0	0	
VM Message Wait Enabled	*8	*8	*8	*8	
VM Message Wait Cancel	*9	*9	*9	*9	
Door Open (1st Door)	#*1	#*1	#*1	#*1	
Door Open (2nd Door)	#*2	#*2	#*2	#*2	
Door Open (3rd Door)	#*3	#*3	#*3	#*3	
Door Open (4th Door)	#*4	#*4	#*4	#*4	
MCID Request	*0	*0	*0	*0	
Unsupervised conference time Extension code	##	##	* ##	*22	
PTT Group Login/Logout	#0	#0	#*0	*21	
ACD Agent primary login	581	* 581	581	581	
ACD Agent primary logout	582	* 582	582	582	
ACD Agent secondary login	583	* 583	583	583	
ACD Agent secondary logout	584	* 584	584	584	
Wrap-up end	585	* 585	585	585	
T-NET CM Login/out	586	* 586	586	586	
Enter Into Conf-Room	59	* 59	* 59	59	
Enter Into Conf-Group	68	* 68	* 68	68	
Station ICR	587	* 587	587	587	
Pick up Group Pick-Up	588	* 588	588	588	
Emergency Page	589	* 589	589	589	

Table B-2 Basic Numbering Plan 1-4, Default Values

Feature	Basic Numbering Plan				
	1	**2**	**3**	**4**	**Remark**
Remote Mobile Extension Control	580	* 580	580	580	
ACD Agent ON/OFF Duty-All group	58*	* 58*	58*	58*	
SLT ACNR	58#	* 58#	58#	58#	
ACD Supervisor Ring Mode	570	* 570	570	570	
Company Directory Name	563	* 563	563	563	
ISDN Supplementary Hold	57*	*57*	57*	57*	
ISDN Supplementary Conf	57#	*57#	57#	57#	
Forced Channel Seize	56*	*56*	56*	56*	
Override DND/Forward	56#	*56#	56#	56#	
Cancel Call Back					
Transfer to VSF Number	55*	*55*	55*	55*	
CCR	#2	#2	#2	#2	
Room type Conf Group join	5*0	5*0	5*0	5*0	
Last Number Redial (LNR)	[REDIAL]	[REDIAL]	[REDIAL]	[REDIAL]	Keyset
Save Number Redial	[Save]	[Save]	[Save]	[Save]	Keyset
Station Speed Dial Access	[SPEED] + XXX	[SPEED] + XXX	[SPEED] + XXX	[SPEED] + XXX	XXX: 000~099
System Speed Dial Access	[SPEED] + XXXXX	[SPEED] + XXXXX	[SPEED] + XXXXX	[SPEED] + XXXXX	XXXXX: 20000~31999

Table B-3 Basic Numbering Plan 5-8, Default Values

Feature		Basic Numbering Plan				
	5	**6**	**7**	**8**	**9**	**Remark**
Intercom Call	2000-4399	2100-4439	1000-3339	1000-3339	1000-3339	
Internal Page Zone	*101-*200	*301-*400	301- 400	*301- *400	*301- *400	
Internal All Call Page	#3	*543	43	*543	543	
Meet Me Page	##	*544	44	*544	544	
External Page Zone 1-2	#41- #42	*545 - *546	45- 46	*545 - *546	545-546	
External All Call Page	#5	*548	48	548	548	
All Call Page	#00	*549	49	*549	549	
SMDR Account Code Enter	50	*550	0	*550	550	SLT
Flash Command to CO Line	51	*551	51	*551	551	SLT
SLT Last Number Redial	52	*552	52	*552	552	SLT
DND (Toggle On/Off)	53	*553	53	*553	553	SLT
Call Forward	54	*554	54	*554	554	
Speed Dial Programming	55	*555	55	*555	555	SLT
Activate Message Wait/Callback	56	*556	56	*556	556	SLT
Message Wait/Callback Answer	57	*557	57	*557	557	SLT
SLT Speed Dial Access	58	*558	58	*558	558	SLT
DND/FWD cancel	59	*559	59	*559	559	SLT
SLT CO System Hold	690	*560	*10	*560	560	SLT
SLT Program Mode Access	691	*561	50	*561	561	SLT
Attendant Unavailable	692	*562	*12	*562	562	

Table B-3 Basic Numbering Plan 5-8, Default Values

Feature	Basic Numbering Plan					Remark
	5	6	7	8	9	
AME Feature	694	*564	*36	*564	564	
Alarm Reset	695	*565	*13	*565	565	
Group Call Pickup	**	*566	*14	*566	**	
Universal Night Answer	697	*567	*15	*567	567	
Account Code with bin	698	*568	*16	*568	568	
Walking COS	699	*569	*17	*569	569	
ACD Agent On/Off Duty	671	*571	*20	*571	571	
ACD Supervisor Login	672	*572	*21	*572	572	
ACD Supervisor Logout	673	*573	*22	*573	573	
ACD Help Code	674	*574	*23	*574	574	
ACD Calls In Queue Display	675	*575	*24	*575	575	
ACD Supervisor Status	676	*576	*25	*576	576	
ACD Supervisor Monitor	677	*577	*26	*577	577	
ACD Reroute Queued Call Answer	678	*578	*27	*578	578	
ACD Reroute Queued Call No Answer	679	*579	*28	*579	579	
Camp-On Answer	621	*621	*629	*621	*621	SLT
Call Parking Locations	#601 – #800	#601 – #800	#601 – #800	#601 – #800	#601 – #800	
Station Group Pilot Number	720 – 819	*401 - *500	#401 - #500	*401 - *500	620-719	
Station User VSF Features Access	*66x	66x	67x	66x	*66	x: 1 ~ 3
Call Coverage Ring	*76	*76	*76	*76	76	
Direct Call Pickup	*77	*77	*77	*77	*77	
Access CO Group	89xxx	89xxx	89xxx	#89xxx	89	xxx: 000-201
Access Individual CO/IP Line	88xxx	88xxx	88xxx	#88xxx	88	xxx: 001~999
Access Held CO/IP	8*	8*	8*	#8*	8*	
Access Held Individual CO/IP	8#xxx	8#xxx	8#xxx	#8#xxx	8#	xxx: 001~999
Access CO in First CO Group	0	9	9	0	9	
Attendant Call	9	0	0	#9	0	
VM Message Wait Enabled	*8	*8	*8	*8	*8	
VM Message Wait Cancel	*9	*9	*9	*9	*9	
Door Open (1st Door)	#*1	#*1	*31	#*1	#*1	
Door Open (2nd Door)	#*2	#*2	*32	#*2	#*2	
Door Open (3rd Door)	#*3	#*3	*33	#*3		
Door Open (4th Door)	#*4	#*4	*34	#*4		
MCID Request	*0	*0	*35	*0	*0	(Except USA version)
Unsupervised conference time Extension code	*##	##	*37	##	##	
PTT Group Login/Logout	#*0	#0	*38	#0	#0	
ACD Agent primary login	681	*581	*40	*581	581	

Table B-3 Basic Numbering Plan 5-8, Default Values

Feature		Basic Numbering Plan					
		5	6	7	8	9	Remark
ACD Agent primary logout		682	*582	*41	*582	582	
ACD Agent secondary login		683	*583	*42	*583	583	
ACD Agent secondary logout		684	*584	*43	*584	584	
Wrap-up end		685	*585	*44	*585	585	
T-NET CM Login/out		686	*586	*45	*586	586	
Enter Into Conf-Room		*59	*59	*46	*59	59	
Enter Into Conf-Group		*68	*68	*47	*68	*68	
Station ICR		687	*587	*48	*587	587	
Pick up Group Pick-Up		688	588	*49	*588	588	
Emergency Page		689	*589	*50	*589	589	
Remote Mobile Extension Control		680	*580	**	#1	580	
ACD Agent ON/OFF Duty-All group		68*	*58*	*5#	*58*	58*	
SLT ACNR		68#	*58#	*51	*58#	58#	
ACD Supervisor Ring Mode		67*	*570	*52	*570	570	
Company Directory Name		*21	*563	*53	*563	563	
ISDN Supplementary Hold		*22	*57*	*54	*57*	57*	
ISDN Supplementary Conf		*23	*57#	*55	*57#	57#	
Forced Channel Seize		*24	*56*	*56	*56*	56*	
Override DND/Forward		*25	*56#	*57	*56#	56#	
Cancel Call Back							
Transfer to VSF Number		55*	*55*	*59	*55*	55*	
CCR		#2	#2*	#2	#2	#2	
Room type Conf Group join		5*0	5*0	5*0	5*0		
Last Number Redial (LNR)		[REDIAL]	[REDIAL]	[REDIAL]	[REDIAL]	[REDIAL]	Keyset
Save Number Redial		[Save]	[Save]	[Save]	[Save]	[Save]	Keyset
Station Speed Dial Access		[SPEED] + XXX	[SPEED] + XXX	[SPEED] + XXX	[SPEED] + XXX	[SPEED] + XXX	XXX: 000~099
System Speed Dial Access		[SPEED] + XXXXX	[SPEED] + XXXXX	[SPEED] + XXXXX	[SPEED] + XXXXX	[SPEED] + XXXXX	XXXXX: 20000~3 1999

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