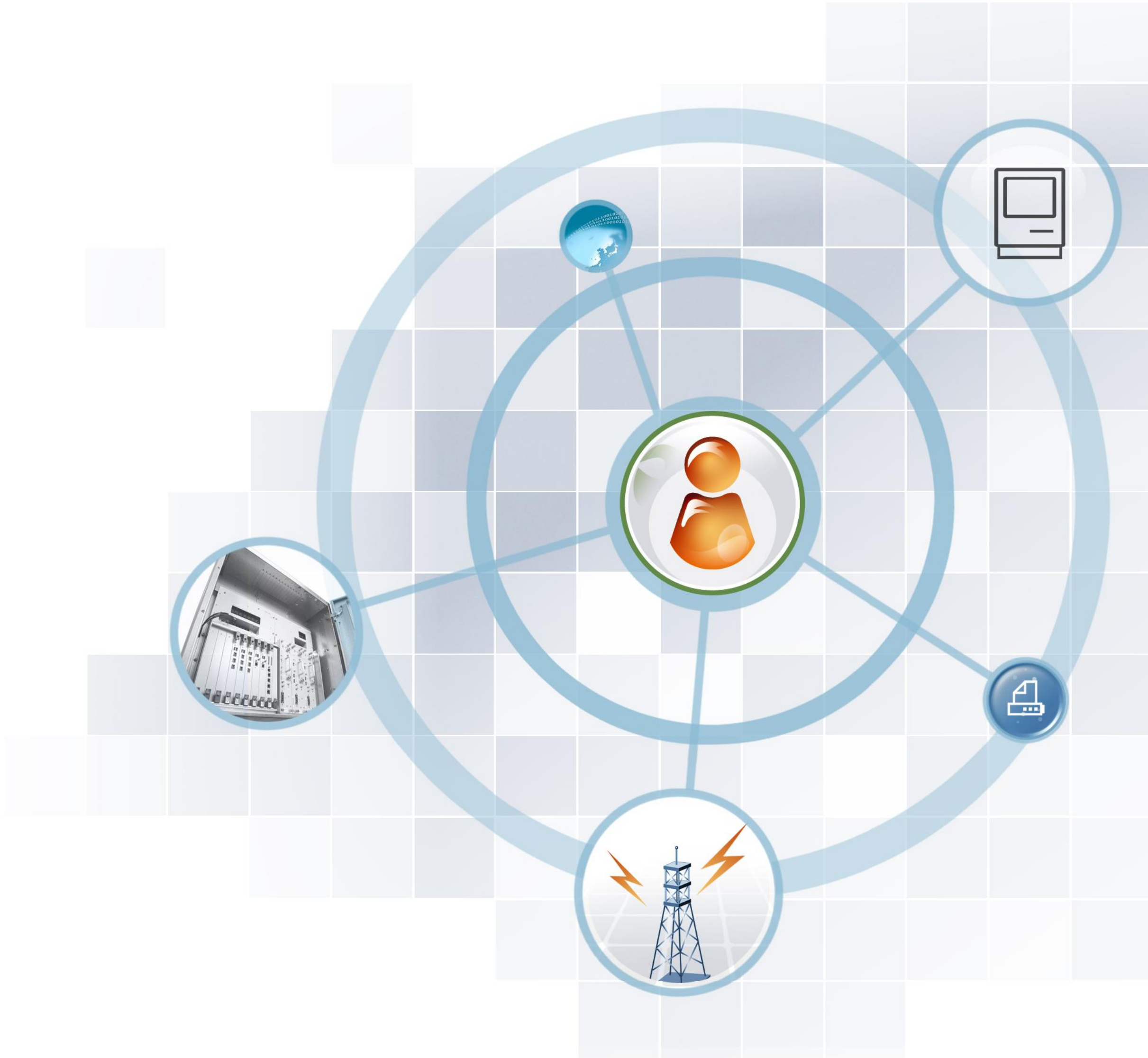
2600-00ILU9GA4 Ver. 3.0

SCM Compact (IPX-S300B)

Operation Manual



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TRADEMARKS

This manual should be read and used as a guideline for properly installing and operating the product.

All reasonable care has been made to ensure that this document is accurate. If you have any comments on this manual, please contact our documentation centre at the following homepage:

Homepage: http://www.samsungdocs.com

INTRODUCTION

Purpose

This Samsung Communication Manager (SCM) Compact Operation Manual describes how to manage, configure and troubleshoot the IPX-S300B.
This manual is written based on the version 1.0.0 of the IPX-S300B software. The IPX-S300B is the product model name of the SCM Compact system.

Audience

This manual is intended for the administrators and operators to understand about the IPX-S300B and help administer the systems.

Document Content and Organization

This manual consists of 8 Chapters, and a list of Abbreviations. Summaries of each chapter are provided below.

CHAPTER 1. IPX-S300B Introduction

This chapter describes the IPX-S300B overview and network environments.

CHAPTER 2. Configuring IPX-S300B Server

This chapter describes how to configure IPX-S300B system for working.

CHAPTER 3. Cabinet/Slot Configuration

This chapter describes how to configure IPX-S300B system for working.

CHAPTER 4. Configuring Phone and Gateway

This chapter describes how to configure phones and gateways.

CHAPTER 5. Call Service

This chapter describes how to use call service features.

CHAPTER 6. Application Features

This chapter describes how to use the application features.

CHAPTER 7. System Management

This chapter describes how to use the management features.

CHAPTER 8. Troubleshooting Guide

This chapter describes how to solve troubles.

ABBREVIATION

Describes the acronyms used in this manual.

Conventions

The following types of paragraphs contain special information that must be carefully read and thoroughly understood. Such information may or may not be enclosed in a rectangular box, separating it from the main text, but is always preceded by an icon and/or a bold title.

WARNING

WARNING

Provides information or instructions that the reader should follow in order to avoid personal injury or fatality.

CAUTION

CAUTION

Provides information or instructions that the reader should follow in order to avoid a service failure or damage to the system.

CHECK

CHECKPOINT

Provides the operator with checkpoints for stable system operation.

NOTE

NOTE

Indicates additional information as a reference.

Console Screen Output

- The lined box with '`Courier New`' font will be used to distinguish between the main content and console output screen text.
- '**`Bold Courier New`**' font will indicate the value entered by the operator on the console screen.

Revision History

VERSION	DATE OF ISSUE	REMARKS
3.0	10. 2016.	SCM Compact V2.0 Software Added Features - Internal recording system - Multicast paging - System information display on device - Nursing home equipment interworking - Emergency call service - VM/AA DB, Data backup - Ring deflect - Ringing preference - LAN Redundancy feature etc.
2.0	10. 2015.	Added a warning phrase in Safety concerns
1.0	10. 2015.	First Version

SAFETY CONCERNS

The purpose of the Safety Concerns section is to ensure the safety of users and prevent property damage. Please read this document carefully for proper use.

Symbols

Caution

Indication of a general caution

Restriction

Indication for prohibiting an action for a product

Instruction

Indication for commanding a specifically required action

WARNING

When the operator is running a delete command on a certain user data, the registration information of the terminal is deleted if the user terminal registration is done at the time. Also, when registration is done for the user terminal to be deleted and the line is busy or in progress of handling a call, the registration information of the terminal is deleted. Therefore, the operator must check the registration status of the user terminal and also the call status in relation to the terminal before running the delete command.

User Interaction service operates according to the contents described in Scenario file. Incorrect scenario technology may cause a problem in the system, so caution should be taken when changing the service scenario.

If the time is changed during a IPX-S300B operation, errors may occur. Therefore, the time change should be carried out after stopping the operation.

Running Database Restore during a IPX-S300B operation can cause a serious error. Thus the Database Restoring must be done after stopping all operations.

Deletion of the Feature Code is applied to all users of the User Group. To cancel only a certain user’s service, only the service should be deleted from the Class of Service instead of deleting the Feature Code.

Deletion of the application server is applied to all users referring to the application server. To delete only a certain user’s service, the service should be deleted only in Class of Service instead of deleting the application server.

Registration and Deletion of Service in Class of Service are affected to all users referring to the Class of Service. To register/clear the service of a certain user, the Class of Service that only that user refers to must be created.

TABLE OF CONTENTS

LIST OF FIGURES

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

CHAPTER 1. IPX-S300B Introduction

The IPX-S300B (model name: IPX-S300B) provides an IP-based service platform for using a wide range of applications for communications within an enterprise. It is aimed at the SMB market in existing small-scale hybrid PBXs.
The IPX-S300B provides VM/AA and conference functionality as well as trunk/extension and general PBX function in a single box. The highly scalable structure allows easy expansion of VM/AA and conference channel cards when necessary. The product also provides built-in NAT and SBC Lite functions.
The call processing software is installed in the main cabinet, and analog extension/trunk, PRI and BRI circuits can be expanded using optional card slots.

1.1 IPX-S300B System Architecture

The diagram below illustrates how IPX-S300B (IPX-S300B) can be implemented in a voice-and-data network.

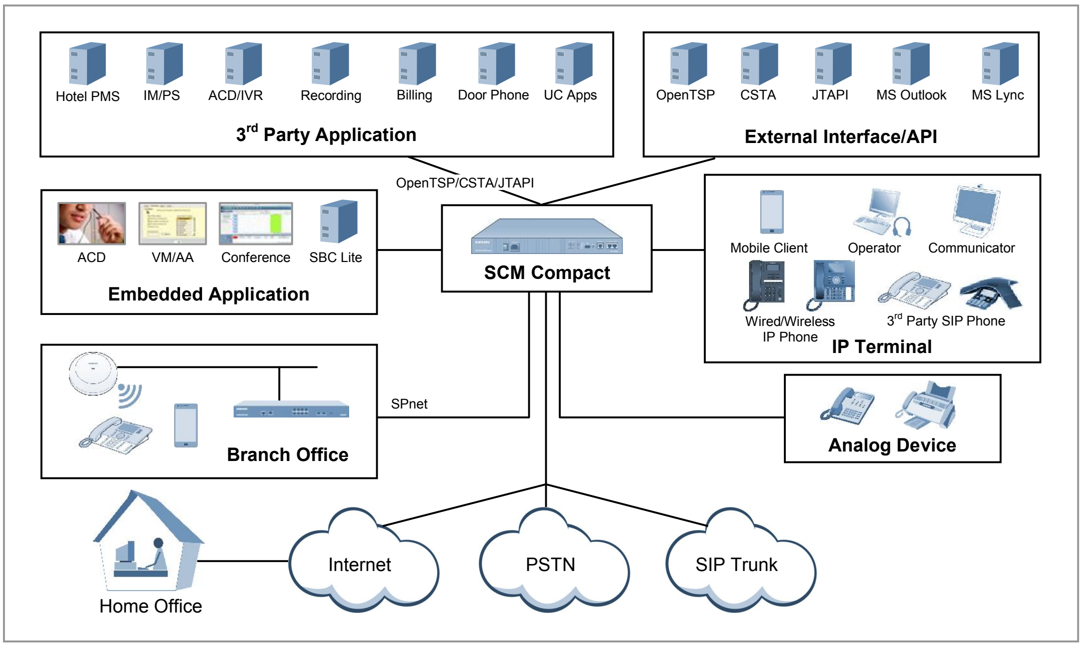

Figure 1. IPX-S300B System Architecture

1.1.1 IPX-S300B

The IPX-S300B uses the SIP protocol to provide IP-PBX call processing functionality and uses analog extension (FXS), analog trunk (FXO), ISDN-PRI, ISDN-BRI interfaces, etc. to communicate with the existing voice networks. The product also provides built-in NAT, DHCP and SBC Lite functions.

1.1.2 Voice Gateway (Media Gateway)

The voice gateway is responsible for connecting the existing telephone and networks.
It communicates with the existing telephone network (PBX or PSTN) through the PRI interfaces as well as analog interface like FXO.
The voice gateway acts as a media gateway, which performs the conversion between PCM data and packet data. It communicates with IPX-S300B over the standard SIP protocol.
The IPX-G5x0 product series can be used as voice gateway.

1.1.3 IP Phones

IP Phone provides general telephone calls on the IP network. For providing basic calling and supplementary services, IP Phone communicates with IPX-S300B (IPX-S300B) using the SIP protocol (Samsung SIP Extension) that is partially extended from the standard.
The following types of IP phones are supported by IPX-S300B.

- SMT-i6000 Series
- SMT-i5343, SMT-i5243, SMT-i5220S, SMT-i5220, SMT-i5210S, SMT-i5210, SMT-i5264
- SMT-i3105, SMT-i2205G, SMT-i2205S, SMT-i2205

IP Video Phone

Video Phone allows voice and video calls over an IP network.
For provision of basic call and additional services, IP Video Phones communicate with the IPX-S300B using an expanded version of the SIP protocol (Samsung SIP Extension).

Soft Phone

Soft Phone runs on a PC (Personal Computer) as a software program. It allows voice-only calls or voice and video calls over an IP network.

Wi-Fi Phone

Wi-Fi Phone connects to an IP network using the Wi-Fi protocol and allows voice calls.

PDA Phone

PDA Phone connects to an IP network over the Wi-Fi protocol. It is provided as a software program that runs on existing PDAs and allows voice-only calls or voice and video calls.

Analog Phone

The system provides analog voice call service using the FXS port which is an optional card.

1.1.4 SCM Administrator

SCM Administrator is web-based service with a graphical user interface for managing IPX-S300B. You can use a web browser on a normal PC to access IPX-S300B and execute various IPX-S300B services like Fault Management, Performance Management, Security Management, Accounting Management and Subscriber Management.

1.1.5 Application Server

The IPX-S300B provides the following built-in services. VM/AA and conference channels can be expanded by adding optional cards.

Music On Hold (MOH) Server

MOH provides the music on hold when a call is put on hold and the voice announcement in the case of errors. It also provides an announcement when a call is queued in the embedded ACD server. In addition, the MOH server collects DTMFs from the caller while executing the user interaction services such as DISA.

SBC Lite

Normally, an SBC server is required for signaling and media (voice and video) connections when establishing calls between phones in the public IP side and those in the private IP side. IPX-S300B includes the NAT traversal function which is one of the services of SBC.

Conference Server

Responsible for combining all the individual voice data in calls involving three or more users into one data set. In a conference call, each phone is connected for a 1:1 call with the conference server, but the conference server combines the data from all the different phones into one data set so the parties can hear each other.
The conference server included in IPX-S300B not only provides the normal conference feature with which the caller pages all the parties to include in the conference, but also provides an advanced conference feature with which a conference room can be set up and the parties can voluntarily call to enter the conference room and participate in the conference. The product provides a default conference capacity and services add-on conference. By adding VPU card, IPX-S300B can be expanded channels and can be added conference service (For example, Meet-me conference, Paging on Answer). The Meet-me conference service requires a license.

ACD (Automatic Call Distribution) Server

The ACD server distributes the incoming calls to agents according to the status of the agents. It also collects real-time call statistics on groups and agents and aggregates the information.

IVR (Interactive Voice Response) Server

The IVR server provides the interactive voice response service that collects DTMFs from the caller and provides proper responses according the DTMF.

VM/AA (Voicemail/Auto attendant) Server

The voice mail service, in particular, constitutes a key component of the enterprise communication system by allowing the calling party to be connected to the VM/AA server and leave a voice message in the called party's mailbox. When there is a new voice mail, the user is notified of by an indicator light on the user's phone or in a notification email sent to the user's Outlook account.

The user can then call the VM/AA server to listen to, reply to, send, or delete the voice mail. The product provides a default VM/AA service, which can be expanded with more channels by adding VPU cards. The VM/AA function requires a license.

Mobility Server

The product provides in-house extension phone call service using smartphones.
When using both the cellular network and Wi-Fi network, the integrated dialer allows switching between the extension network and cellular network. The Mobile Extension function also allows the smartphone to be used in the same way as an extension phone outside the office.
This service requires a separate license.

1.2 Internet Telephony Network

1.2.1 Network Configuration

IPX-S300B is a system that provides Internet telephony on an IP network. Its installation requires special attention compared to installation of ordinary data systems.

Incorrect installation of the Internet telephony system on an IP network may result in IP packet loss, packet delay, jitter, and other problems that can seriously affect the quality of voice calls. Therefore, when implementing an Internet telephony service, your network should be configured with all available QoS technologies, providing redundancy and rapid response in the case of network disruptions. In other words, your IP network devices must be configured for:

- 802.1P (CoS on LAN)
- 802.1Q (VLAN)
- Traffic classification
- Traffic shaping
- 802.1D (STP), 802.1w (RSTP), 802.1s (MSTP)

The following diagram illustrates the IP network structure for Internet telephony between HQ and its branch.

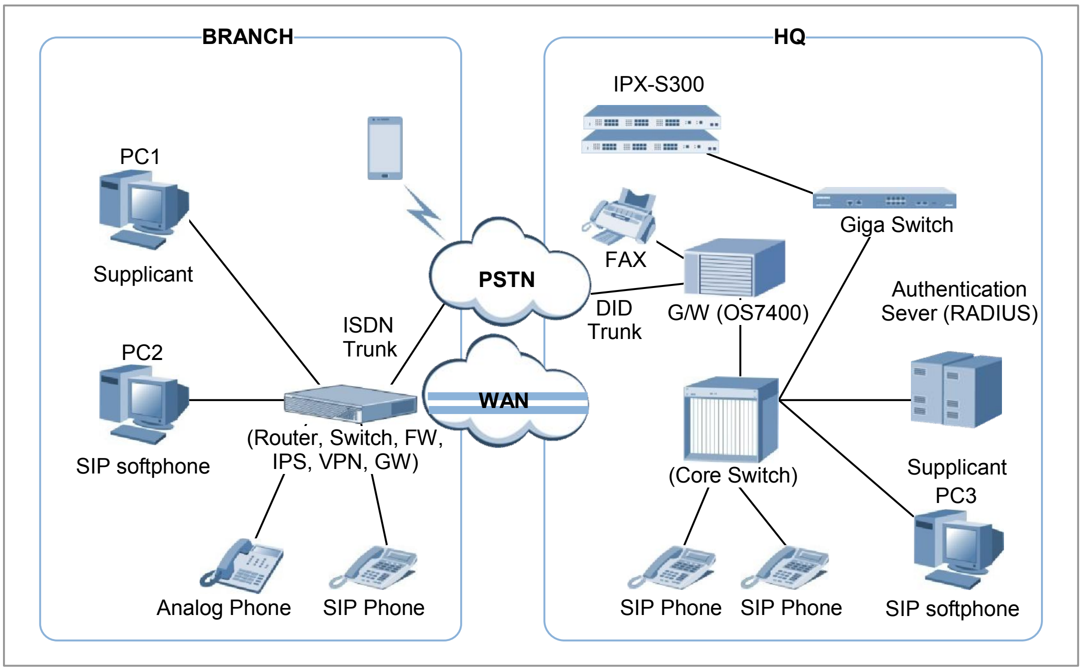

Figure 2. Internet Telephony Architecture

1.2.2 Network Requirements

For a reliable Internet telephony service, your IP network must meet the following requirements.

Quality of Service (QoS)

The following requirements must be met for QoS.

- Packet loss rate: 0.1 % or less
- Packet delay: Average 100 ms or less, maximum 200 ms or less
- Jitter: 40 ms or less

The measurements should be performed as follows.

- Your measurements must include the time period with the highest traffic volume on the IP network.
- You must take the measurements at least eight times a day for one week or longer.
- Your measurements must include the longest path on which the Internet telephony service is provided.

Virtual Local Access Network (VLAN)

VLAN is a method of dividing physical switching ports into logical broadcast domains. In other words, the switching ports on the LAN are configured into one domain. Within this single domain, multicast and broadcast data is transmitted without limits. But there are many applications that generate multicast and broadcast data on an IP data network, including ARP, RARP, VRRP, RIP, and OSPF. VLAN is a technology that divides this multicast and broadcast domain to reduce unnecessary traffic.

When implementing an Internet telephony service on a VLAN, the network must be designed to meet the following requirements.

- Each VLAN must be unique in the entire network.
- A single VLAN must be assigned to a single IP subnet.
- A single VLAN must not have 250 or more devices. The recommended number is 100.
- Each VLAN must run its own STP. (For more information, see the section on STP.)
- Each VLAN must be assigned to a single IP address subnet. (For more information, see the section on DHCP.)

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Spanning Tree Protocol (STP)

When implementing an Internet telephony service on a large-scale network, each VLAN usually has many devices (switches, IP phones, PCs, etc.), generating a lot of broadcast data that can potentially lead to many problems. To reduce potential problems and ensure reliable Internet telephony service, it is necessary to have each VLAN run its own STP so that topological looping can be prevented and only the ports causing broadcast storms can quickly be closed.

Dynamic Host Configuration Protocol (DHCP) & Domain Name System (DNS)

DHCP is the protocol that configures initial information, such as IP address, subnet mask, and default gateway, for the various hosts on the network. The DNS is a server that performs IP address mapping for the hosts on the network. An IP network runs DHCP and DNS so that there is no need to manually configure and manage the initial configuration information, including the IP addresses of all the hosts on the network. It also automatically processes the initial configuration information, including IP address, whenever a host changes its location on the network.

For effective operation of an Internet telephony service, a DHCP pool is recommended so that each VLAN used for the Internet telephony service has a unique IP address.
This allows easy identification and management of the devices on the Internet telephony service network.

It is recommended that you run the DHCP lease time flexibly according to the characteristics of the site running the Internet telephony service. In most cases, IP phones are rarely relocated and therefore unnecessary data generation can be eliminated by keeping the DHCP lease time at one week or longer.

Network address translation (NAT) Traversal

When recognizing that there is a network-aware application program behind the NAT device, the NAT traversal feature obtains the external IP address and performs port mapping in order to transfer the data from the external port of the NAT device to the internal port used by the application program. All this is done automatically. There is no need for the user to manually perform port mapping or use any other mechanism.

To facilitate use of this technology in Internet telephony, IPX-S300B provides the media proxy feature for securing communication paths between the IP phones on the NAT network and the systems and phones on the public IP network.

As such, when it is difficult for IPX-S300B to obtain a public IP address, it can use the port mapping configuration of the existing NAT system to obtain a private IP address for IPX-S300B itself and provide a reliable Internet telephony service with IP phones on the public IP network or IP phones on another NAT network.

To implement this technology, the port information below must be set to open for the NAT system, and port mapping must be configured for the NAT system.

Following is a list of ports must be open when the IPX-S300B is located under NAT.

Service	TCP Port	UDP Port	Description
General	20, 21	-	FTP Server
	22	-	Secure Shell
	23	-	Telnet
	80, 443	-	HTTP Web Server
	389	-	LDAP Server
	123	123	NTP
Provisioning	-	69	TFTP Server
	8088	-	Gateway Provisioning
	-	6000	Phone upgrade from Proprietary to SIP
NMS	-	161	SNMP Agent
Personal Management	8080, 9500	-	Personal Assistant for Call Service
	4002, 4004, 4102	-	Single Sign-On, PWP for UMS/Conference
System Management	20001, 20002, 20003, 20005, 20006, 20007	-	SCM Administrator
	5432	-	PostGRE DBMS connection
Call	5060, 5061	5060	SIP signaling
UMS	5080, 8624	5080	Call signaling for UMS
	-	14002~14130	RTP path for UMS
	25, 143, 993, 587	-	Signaling for E-mail Server
	3681, 3683, 3684, 22001	-	Signaling for Outlook client
	22001	-	UMS File Server
Conference	3333	5090, 5098	Call signaling for Conference
	-	40200~40399	RTP path for VPU Card
MOH	-	35000~35999	RTP path for MOH/Announcement
Others	6000~6127	-	CSTA link for each user group
	9050, 9052	-	PMS link
	9090, 9092, 9094	-	Proprietary Application server link
		-	-
	9010, 9011	-	MVS client link

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Service	TCP Port	UDP Port	Description
	18124, 18126	-	mySingle link
	18202	-	Wi-Fi Phone service (Presence, Location Service)
	2195, 443		WE VoIP Push Service: iOS (2195), Android (443)
	10306, 2300	-	CDR (Call Data Record)

LAN Redundancy

When the IPX-S300B system is operated in the LAN only mode, LAN redundancy is supported.

In the IPX-S300B system, two Ethernet ports can be connected to the same switch or another switch simultaneously. (However, the switch must support STP.)

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

CHAPTER 2. Configuring IPX-S300B Server

This chapter describes the basic information configuration for using the IPX-S300B system. After configuring the basic information, the system can handle calls for extension lines and trunks, which is a basic feature of the PBX.

2.1 Connecting to SCM Administrator

SCM Administrator is the IPX-S300B management program running on the Java Web Start platform. This section describes the connection environment, login method, and page layout of SCM Administrator.

2.1.1 Environment

To use the SCM Administrator normally, the following PC environment must be prepared. Item Software

Item	Software
CPU/Memory	Pentium D or higher, 2 GB DRAM or higher
Operating System	Windows XP or later
JRE	v6.0 update13 or later
Web Browser	Microsoft Internet Explorer version 10.0, Firefox 3.5, Chrome 5.0 or later

You can download the JRE from:
http://www.oracle.com/technetwork/java/index.html

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

2.1.2 Login

When you enter <http://*{System IP Address}*> or <http://*{System IP Address}/admin*> into the address bar of your web browser, the login page of SCM Administrator is displayed.

If the value of 'Root URL Owner' in **[Configuration > Miscellaneous > System Options]** was changed to 'pwp', you should enter <http://*{System IP Address}/admin*> for connecting to SCM Administrator. <http://*{System IP Address}*> url will open Personal Assistant. The default IP address of the system is 192.168.0.1. Therefore, for the initial configuration, you need to enter <http://192.168.0.1> or <http://192.168.0.1/admin>.

The login page of SCM Administrator is displayed as illustrated below: Enter the user ID and password to log in. The default administrator account is the same as below.
(User ID: admin, Password: samsung*#)

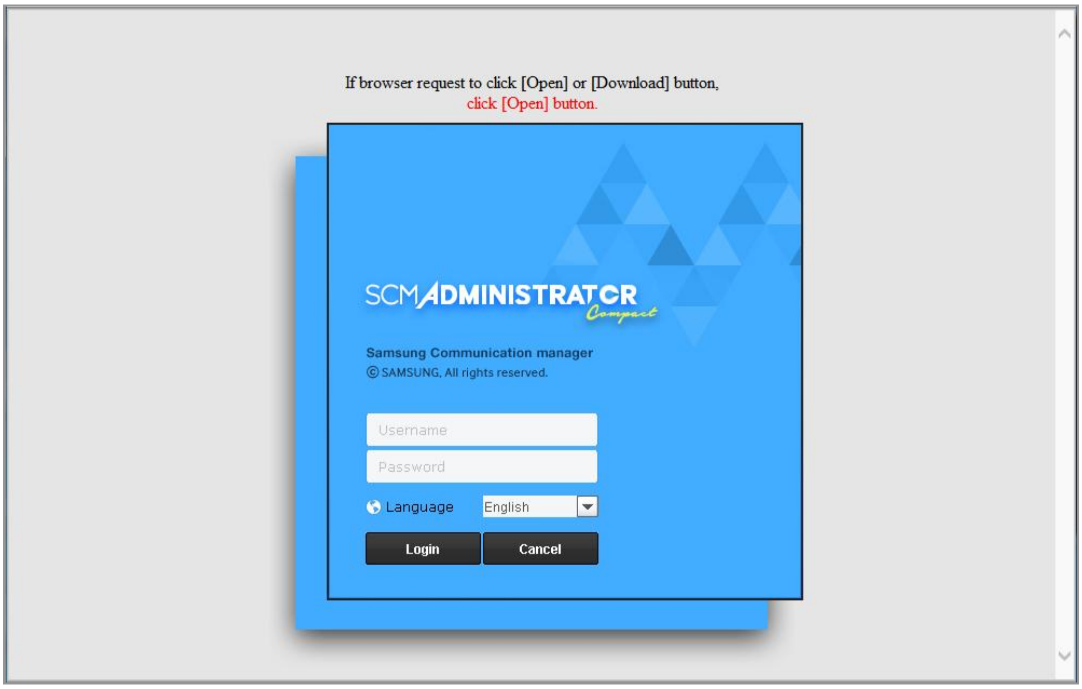

Figure 3. Login window

2.1.3 Connecting to SCM Administrator from remote place

You can remotely log in to SCM Administrator for programming purposes. If IPX-S300B is using a public IP address on a public IP network, you can log in using the same method.

If you want to access SCM Administrator remotely, you must first connect to the private IP network and set ‘System Public IP Address’ to the public IP address of the NAT network in the **[CONFIGURATION > Miscellaneous > System Options]** menu.

Enter <http://(WAN IP Address)> or <http://(WAN IP Address)/admin> in the address bar of your web browser. The SCM Administrator login screen appears.

> NOTE
>
> **If you have changed the Root URL Owner field’s value from ‘admin’ to ‘pwp’ in the [CONFIGURATION > Miscellaneous > System Options] menu**, you must connect to <http://(IPX-S300B IP address)/admin>. Connecting to <http://(IPX-S300B IP address)> opens the Personal Assistant login page.

> NOTE
>
> If the login page does not appear on the screen, check the pop-up blocker and security settings in your web browser’s Settings.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

2.1.4 Page Layout

The SCM Administrator page layout is as follows.

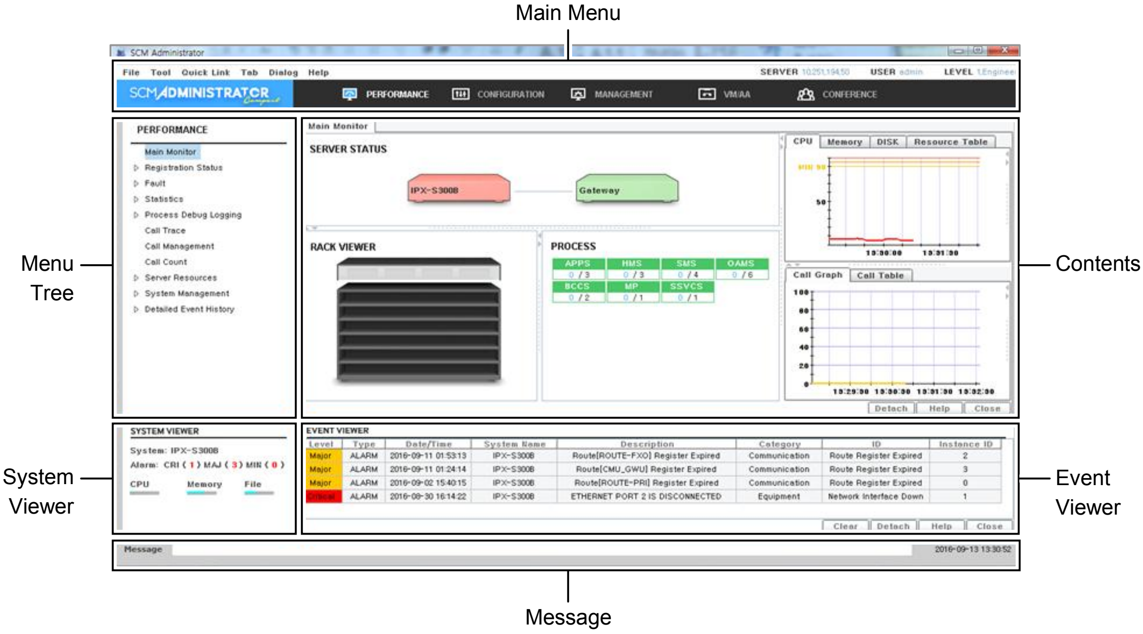

Figure 4. SCM Administrator window

- Main Monitor: Provides the menus of SCM Administrator grouped by feature.
- Menu Tree: Provides the submenus.
- Main Page: Displays the setting page for the functions supported by each menu.
- System Viewer: Displays the current status of the system (system name, active/standby, number of alarms generated, and CPU, memory, and file system usage status).
- Event Viewer: Displays the events that occurred in the system.
- Message Window: Displays the results of the features executed by the user.

2.1.5 Displaying on the CONFIGURATION Menu

The options on the CONFIGURATION menu are displayed according to the following rules:

- Display in blue: The fields that must be filled in in a given menu.
- Display in black: The optional fields that are not necessary to input.
- Display in gray: The fields that do not need to be filled in depending on what options are selected for other fields.

2.2 IPX-S300B Configuration Wizard

When first time login to the IPX-S300B system with an administrator account, the Configuration Wizard windows will be starting.

When click **[generate]** button after enter the all required data for each of the steps, all data are completed setting for incoming and outgoing call.

> **NOTE**
>
> In each step, you must fill in the fields that are written in blue text with an asterisk (*) added in the end. Otherwise, you cannot proceed to the next step. Clicking Default Setting] deletes all settings that you have modified and returns to the default settings.

2.2.1 Step 1. Configuring System

This window runs one time only when first time login to the IPX-S300B system with an administrator account.

This step can configure basic system information and license key.

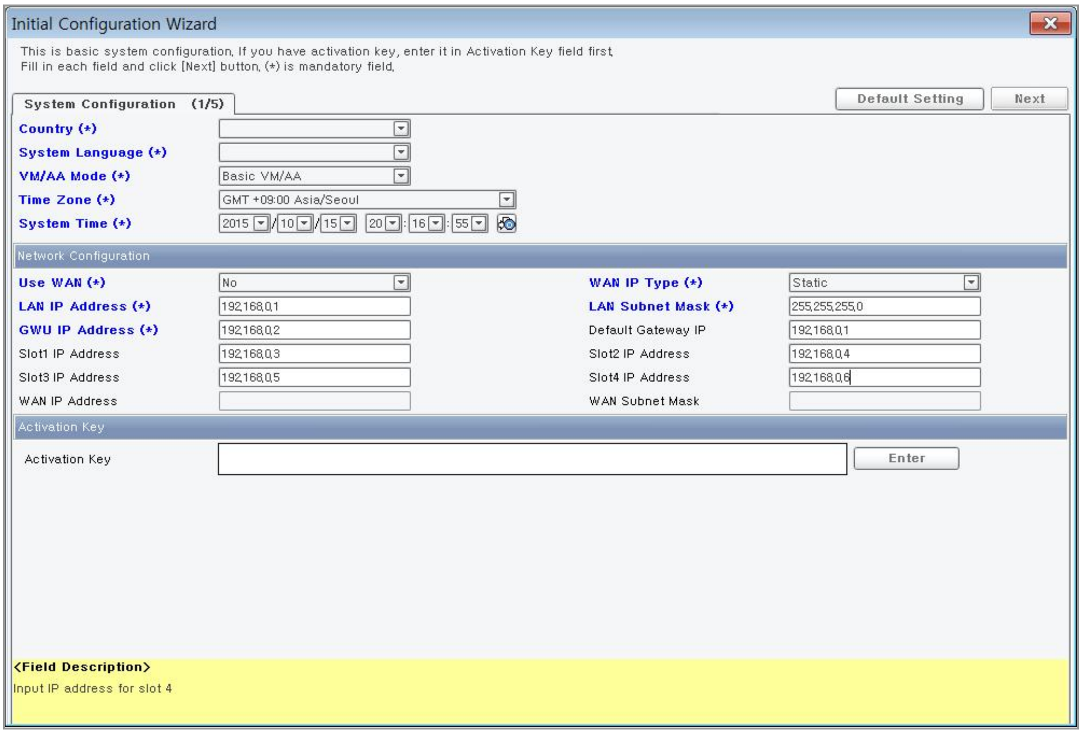

Figure 5. Initial Configuration Screen

Configuring IPX-S300B system Basic

This step can select country and language for IPX-S300B.

Item	Description
Country	Please choose the country where the IPX-S300B installed
System Language	Please choose the language of the system
VM/AA Mode	Please choose the mode of the setting window of the VM/AA - Basic VM/AA: This enables only menus for basic auto attendant and voicemail features. This option simplifies menu tree of VMAA, so you can set up voicemail with ease. - Extended VM/AA: This enables all menus of voicemail systems. Select this option when you want to use complex yet diverse voicemail features.
Time Zone	Select a time zone for your system.
System Time	Set the time for your system.
Use WAN	Select whether to uses WAN connection. - No: Uses your system in a LAN environment without using the WAN feature. - Yes: Uses the WAN feature.
WAN IP type	If you have chosen to use WAN connection, a static IP address is used for your WAN IP address.
LAN IP Address	The LAN IP address of your system.
LAN Subnet Mask	The LAN subnet mask of your system.
Default Gateway IP	The gateway IP address of your system. - When WAN is enabled: The default gateway IP address for your WAN IP address. - When WAN is disabled: The default gateway IP address for your LAN IP address.
GWU IP Address	The IP address of the GWU board (must be in the same range with the LAN IP address).
Slot1 IP Address	If a PRI or VPU board has been fitted in slot 1, an IP address needs to be entered for the board. (Must be in the same range with the LAN IP address.)
Slot 2 IP Address	If a PRI or VPU board has been fitted in slot 2, an IP address needs to be entered for the board. (Must be in the same range with the LAN IP address.)
Slot 3 IP Address	If a PRI or VPU board has been fitted in slot 3, an IP address needs to be entered for the board. (Must be in the same range with the LAN IP address.)
Slot 4 IP Address	If a PRI or VPU board has been fitted in slot 4, an IP address needs to be entered for the board. (Must be in the same range with the LAN IP address.)

Item	Description
WAN IP Address	The WAN IP address of your system.
WAN Subnet Mask	The WAN subnet mask of your system.

Entering Activation Key

This step can enter the Activation (License) key for IPX-S300Bworking.
After input the Activation Key, should click the **[Submit]** button for applying the system.
If you do not enter the activation key, the system runs with a evaluation key or a sample key.

Item	Description
Activation Key	Please input the Activation Key for IPX-S300B working

2.2.2 Step 2. Number Configuratoin

This step can configure the information minimally required such for each user group. For example, the first extention number and the number of all extentions belong to each telephone type, the common access number for the operator and trunk, and an operator group information related with it. The initial value which allows to the number of users for each group will depend on the number of activation keys. The total number of users for each group can not exceed 512.

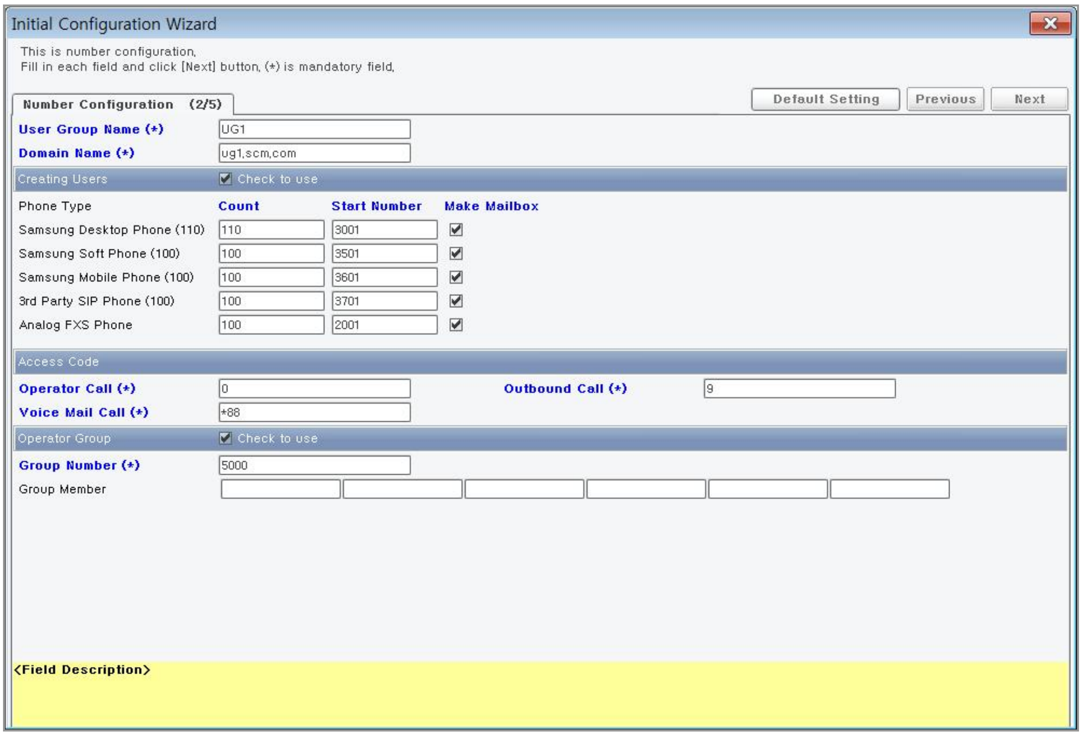

Figure 6. Number Configuration for each User Group

Making User Group

This step can make name and domain of user group, location and service group.

Item	Description
User Group Name	It is tenant group and a similar concept. Root of all data is a user group with independent number system.
Domain Name	Please input the name of the domain of SIP URI

Making User Information

You can create phone numbers and voicemail for each phone type. The number of phone numbers that can be created is limited to the number defined in the activation key.

Item	Description
Samsung Desktop Phone	SIP phones made by Samsung (To support a basic call service and other supplementary services, a SIP protocol that has been extended from the standard SIP protocol are used.)
Samsung Soft Phone	Soft phones made by Samsung. (This type of phone is provided as a software program that can be used on common PCs.)
Samsung FMC Phone	Software phones made by Samsung for mobile phones. (This type of phone connects to an IP network via a Wi-Fi protocol.)
3rd Party SIP Phone	SIP phones made by manufacturers other than Samsung.
Analog FXS Phone	The number of the analog phone connected to the gateway is designated.

Making Common Number

This step can make the common access numbers like operator call number and trunk access number.

Item	Description
Operator Call	It is access number for operator calling.
Outbound Call	It is access number for outbound call via default trunk group.
Voice Mall Call	It is access number for voice mail system calling.

Making Operator Group

The operator group uses hunt group of each ring plan. Therefore minimum a hunt group should be made. In this step can make a hunt group.

Item	Description
Group Number	It is hunt group number for assigned operator group.

Item	Description
	(Cannot duplicated with other extension numbers.)
Member Number	It is user number of hunt group member. (Cannot duplicated with other extension numbers.)

2.2.3 Step 3. Configuring SIP Trunks and DID Routing

You can configure SIP trunks and DID routing table.

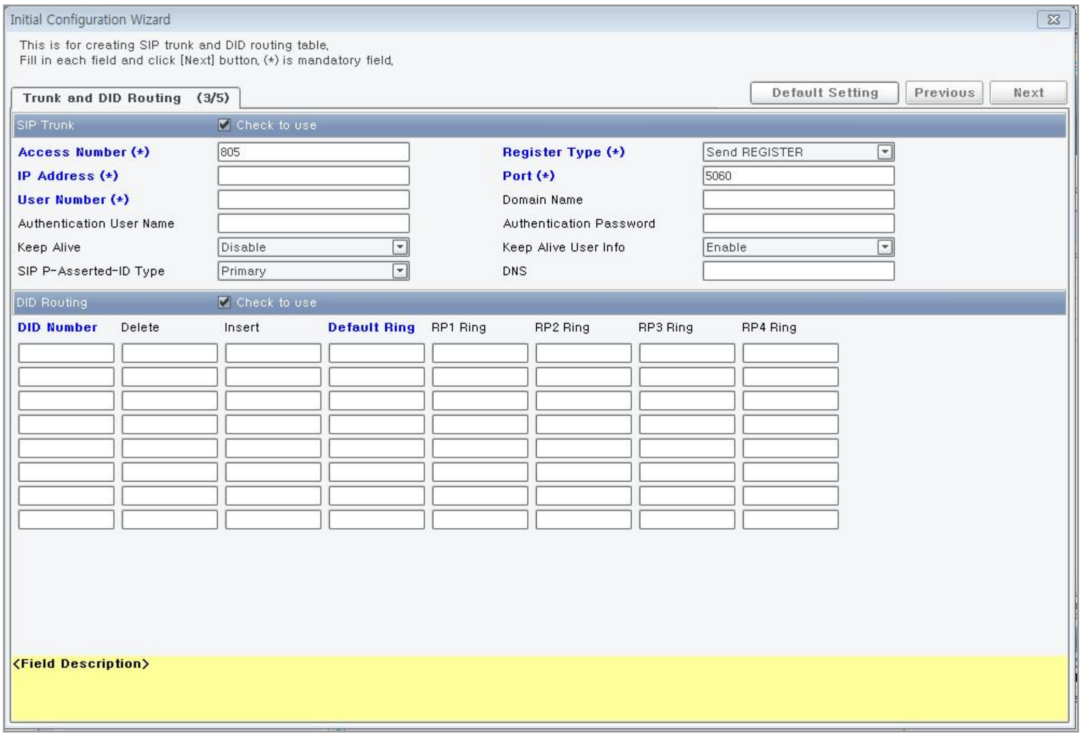

Figure 7. Configuring SIP Trunks and DID

Configuring SIP Trunks

This step configures SIP trunks.

Item	Description
Access Number	The trunk access code used when making an outbound call.
Register Type	Input a register type for SIP trunk. - Receive REGISTER: Accepts the registration of the trunk by receiving REGISTER from the trunk route. - Send REGISTER: Registers the trunk by sending REGISTER from the SCM to the trunk. - None: Does not perform registration between the trunk and the IPX-S300B.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
IP Address	Input the IP address of the SIP trunk.
Port	Input the port number of the trunk link.
User Number	Input the user number for a User Info part of SIP URI.
Domain Name	Input the domain name that a part of host of SIP URI contains.
Authentication User Name	Input a name used at the time of a registration.
Authentication Password	Input a password used at the time of a registration.
Keep Alive	Setting Keep Alive to Enable periodically sends OPTIONS messages to check the status of the trunk route.
Keep Alive Userinfo	Whether to use a Userinfo field in OPTIONS messages.
SIP P-Asserted-ID Type	Select the SIP connect interoperation type. It is how to use P-Asserted-ID header to deliver the representative number. - None: P-Asserted-Id is not used and the representative number is not delivered. - Primary: The representative number is delivered with P-Asserted-ID header. - Secondary: The representative number is delivered in the From header and P-Asserted-ID contains an individual number.
DNS	Input an IP address of main DNS server.

Configuring DID Routing

This step can make the ring destination number table for DID inbound call from SIP or PSTN trunk.

Item	Description
DID Number	Input DID number for inbound call
Delete	When modifying DID number, it is the digit count to delete the part of the start of the DID number.
Insert	When modifying DID number, it is the digit count to add the part of the start of the DID number.
Default Ring	Input the default ring destination number.
RP1~RP4 Ring	Input the ring plan 1~4 destination number.

2.2.4 Step 4. Summary and Confirmation

This step can show the values input in each previous step. To create a database as is, click the **[Generate]** button. To change a setting, click the **[Previous]** button and modify the setting.

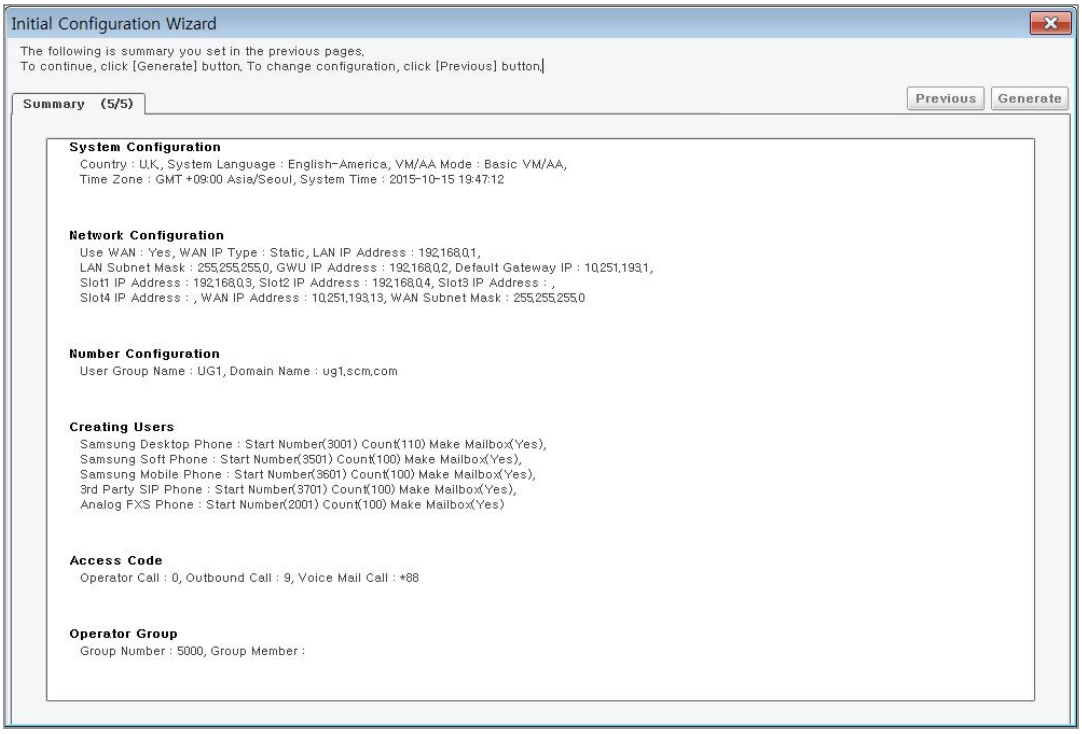

Figure 8. Summary and Confirmation

2.2.5 Step 5. Configuration in Progress

This shows the events generated in the system and their results. If an error has occurred in the system, you can click the Retry button, modify the relevant part, and then run the process again.

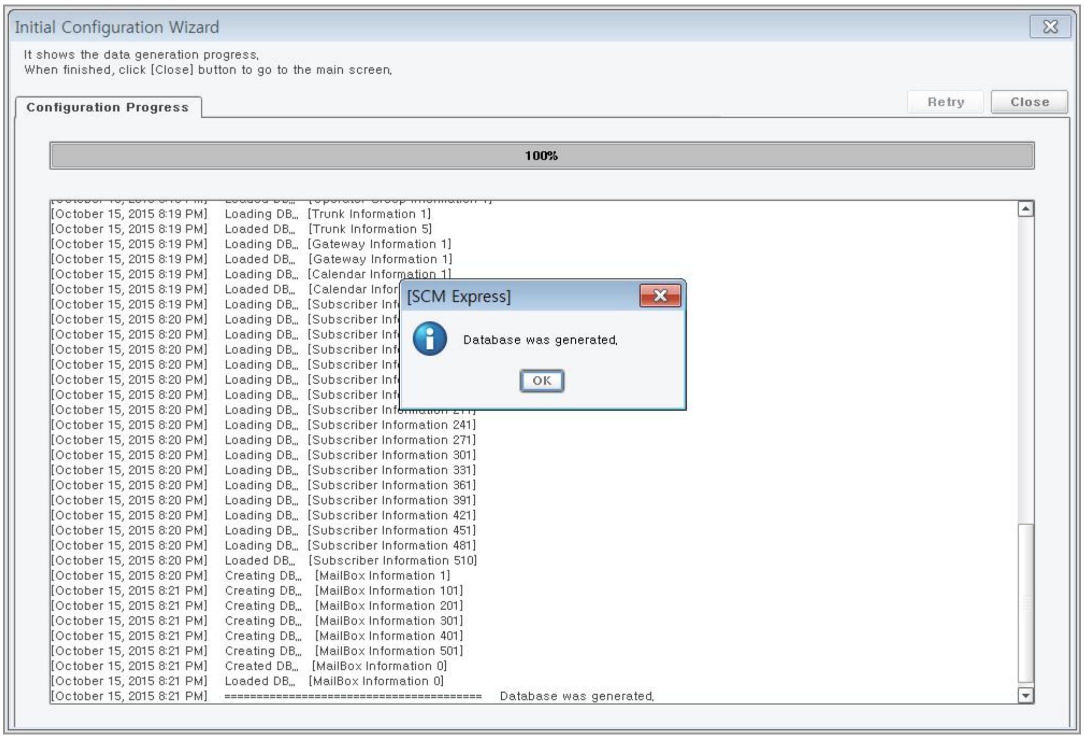

Figure 9. Configuration in Progress

2.3 Changing All Users Data

It is often necessary to change information for many subscribers simultaneously.
SCM Administrator allows changing all user's data at one time.
When modifying an Excel file offline, you can modify each field's information as well as add or delete lists. All changes made to an Excel file can be reflected in the IPX-S300B. You can change all users' data at once in the **[Tool > Customer Data Import/Export]** menu in the upper left-hand corner of the SCM Administrator, executing the following steps.

1) **Click menu in the [Tool > Customer Data Import/Export]**. The current settings will be shown.
2) Click the 'Export' button and specify a filename to save as.
3) Modify the saved file offline.
4) Click the 'Import' button and select the excel file you worked on offline.
5) Click the 'Apply' button to apply the new setting to the system.

You can change these user data in Customer Data Import/Export.

- When adding new data, execute the following steps. When delete data, execute the steps in reverse order. Department: **This is the type of information you create in the [CONFIGURATION > User Group > Department]** menu.
- Single Phone User: This is the type of information you create in the **[CONFIGURATION > User > Single Phone User]** menu.
- User Profile: This is the type of information you create in the **[CONFIGURATION > User > User Profile]** menu.

2.3.1 Customer Data Import/Export

SCM Administrator's data file export/import feature allows exporting information from some of the IPX-S300B through SCM Administrator menu items into an Excel spreadsheet, which can then be edited offline and be imported back.

When editing the Excel spreadsheet offline, you can edit the information for each field and also add lists. IPX-S300B can be updated with any changes made in the Excel spreadsheet.

The data file export/import feature can be executed by using the **[Tool > Customer Data Import/Export]** menu in the top left corner of SCM Administrator. The following features are available.

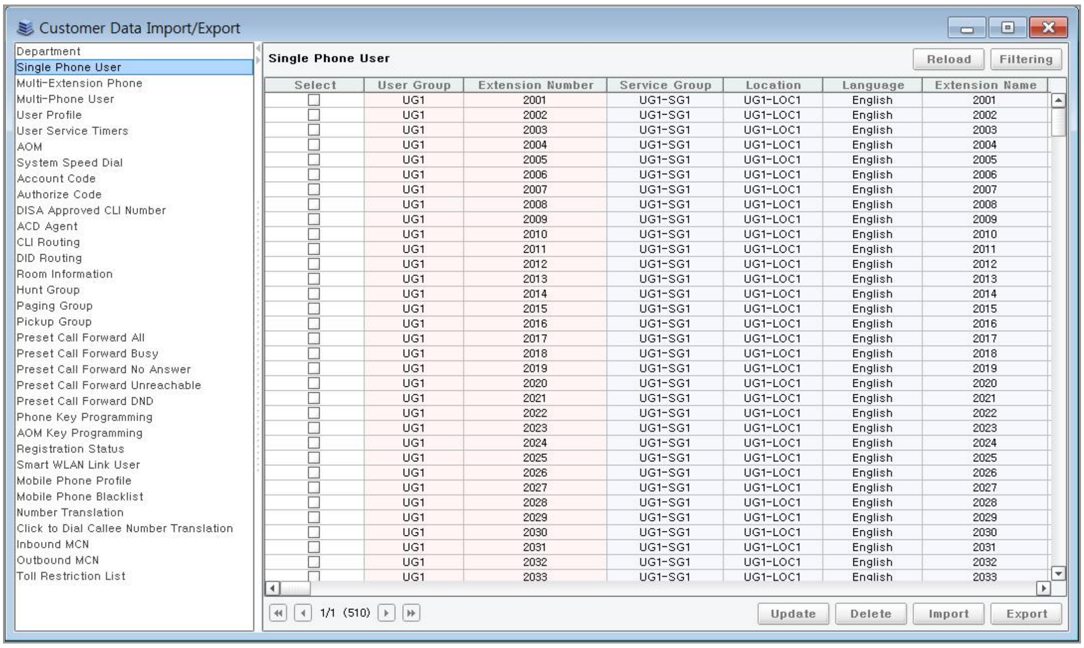

Figure 10. File Export/Import

Refresh

This feature re-reads data from the system and displays it on the screen. All editing data, such as data you imported previously and data you changed, disappear from the screen.

Update

You use this feature to apply data displayed on the screen, and you execute it in the following order:

1) Select the **[Tool > Customer Data Import/Export]** menu, and then the currentsettings will be displayed.
2) To add data, import your desired Excel file's data to the screen by clicking the 'Import' button.
3) If you want to modify the current data without adding new data, change the values of your desired fields.
   Changing the value of a field automatically checks the checkbox for the 'select' field matching the information.
4) Click the 'Update' button to apply the information lists on the screen to the system.
5) If no error has occurred, the screen is automatically refreshed. If there has been an error, an error dialog appears. Error dialog shows the information lists that cannot be applied to the system due to an occurrence of an error as well as the cause of the error.

Import

You can import a list of information which had been exported previously as an Excel file to the system by using the **[Import]** feature, which must be executed in the following order:

1) In the **[Tool > Customer Data Import/Export]** menu, click a menu you wish to use. A list of currently registered information will be displayed.

2) Click the **[Import]** button and specify the name of the file to import.

3) If duplicate data exists in the system, the list of duplicate data will be displayed in a dialog box.

4) The first column indicated as **[DB]** shows the information currently registered in the system. The second column indicated as 'Excel' shows the information in the Excel file. You can select fields for which to save the information.

5) The following buttons are shown in the duplicate dialog box. If there is no duplicate data, the duplicate dialog box will not be shown and the next step will be performed automatically.

Button	Description
Update	The information selected in the dialog box will be updated on the data file import/export screen.
Update All	The data in the Excel file will be selected for all the following items with duplicate information for updating the data file import/export screen.
Skip	The current item with duplicate information will not be updated.
Skip All	The duplicate data for all the following items will not be updated.

6) The information from the Excel file will be shown on the screen.
The **[Update]** field on the information list indicates whether the selected information will be added or changed.

Update field	Description
Indicated as **[Add]**	This information does not exist in the system and will be added when **[Update]** is clicked.
Checkbox is checked	This information exists in the system and will be changed when **[Update]** is clicked.
Checkbox is not checked	This information exists in the system and nothing will happen when **[Update]** is clicked.

After performing the import feature, if you do not see the information on the screen, check the following:

- Check that the names of fields at the top of the Excel spreadsheet are the same as the data file import/export screen.
- If a field's property is combo box, check that the string in the Excel file is within the selection range.

Export

You can save a list of information currently registered in the system into an Excel file by using the **[Export]** feature, which must be executed in the following order:

1) In the **[Tool > Customer Data Import/Export]** menu, click a menu you wish to use. A list of currently registered information will be displayed.
2) Click the **[Export]** button and specify a name of the file to save.
3) An Excel file will be created with the specified file name. Open the Excel file to check whether the list of information has been exported correctly.

Filtering

Data on the screen can be viewed selectively using the following options.

Option	Description
Update	Only the information available for updating is shown.
Add	Only the information available for adding is shown.
Option	You can filter information by the field used as the index of each menu. - A~: Only the information with the selected field value 'A' is shown. - A~B (string): Only the information for which the first character of the selected field is within the specified range is shown. - A~B (number): Only the information for which the value of the selected field is within the specified range is shown.

2.3.2 Customer Data Import/Export

The types of information for which you can use the customer data import/export feature are as follows:

- Department: This is the department information created in the **[CONFIGURATION > User Group > Department]** menu.
- Single Phone User: This is the subscriber information created in the **[CONFIGURATION > User > Single Phone User]** menu.
- Multi-Extension Phone: This is the type of user information you create in the **[CONFIGURATION > User > Multi-Extension Phone]** menu.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

- Multi-Phone User: This is the type of user information you create in the **[CONFIGURATION > User > Multi-Phone User]** menu.
- User Profile: This is the user profile information created in the **[CONFIGURATION > User > User Profile]** menu.
- User Service Timer: This is the timer option's information set in the **[CONFIGURATION > User > User Service Timer]** menu.
- AOM: This is the add-on module (AOM) information created in the **[CONFIGURATION > User > AOM]** menu.
- System Speed Dial: This is the system speed dial information created in the **[CONFIGURATION > Service > Speed Dial > System Speed Dial]** menu.
- Account Code: This is the account code information created in the **[CONFIGURATION > Service > DTMF Detection Service > Account Code]** menu.
- Authorize Code: This is the authorize code information created in the **[CONFIGURATION > Service > DTMF Detection Service > Authorize Code]** menu.
- DISA Approved CLI Number: This is the DISA approval CLI number created in the **[CONFIGURATION > Service > DTMF Detection Service > DISA Approved CLI Number]** menu.
- ACD Agent: This is the ACD agent information created in the **[CONFIGURATION > Application > ACD > ACD Agent]** menu.
- CLI Routing: This is the CLI routing information created in the **[CONFIGURATION > Trunk Routing > CLI Routing]** menu.
- DID Routing: This is the DID routing information created in the **[CONFIGURATION > Trunk Routing > DID Routing]** menu.
- Room Information: This is the Room Information created in the **[CONFIGURATION > Service > Hotel Service > Room Information]** menu.
- Hunt Group: This is the Hunt Group Information created in the **[CONFIGURATION > Service > Group Service > Hunt Group]** menu.
- Paging Group: This is the Paging Group Information created in the **[CONFIGURATION > Service > Group Service > Paging Group]** menu.
- Pickup Group: This is the Pickup Group Information created in the **[CONFIGURATION > Service > Group Service > Pickup Group]** menu.
- Preset Call Forward All: This is the Preset Call Forward All Information activated in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu.
- Preset Call Forward Busy: This is the Preset Call Forward Busy Information activated in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu.
- Preset Call Forward No Answer: This is the Preset Call Forward No Answer Information activated in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu.

- Preset Call Forward Unreachable: This is the Preset Call Forward Unreachable Information activated in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu.
- Preset Call Forward DND: This is the Preset Call Forward DND Information activated in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu.
- Phone Key Programming: This is the Phone Key Programming Information created in the **[CONFIGURATION > User > Phone Key Programming]** menu.
- AOM Key Programming: This is the AOM Key Programming Information created in the **[CONFIGURATION > User > AOM Key Programming]** menu.
- Registration Status: This is the Registration Status Information created in the **[PERFORMANCE > Registration Status > Registration Status]** menu.
- Smart WLAN Link User Profile: This is the Smart WLAN Link User Profile Information created in the **[CONFIGURATION > Wireless Enterprise > Smart WLAN Link Configuration > Smart WLAN Link User Profile]** menu.
- Mobile Phone Profile: This is the Mobile Phone Profile Information created in the **[CONFIGURATION > Wireless Enterprise > Mobile Phone Profile]** menu.
- Mobile Phone Blacklist: This is the information specified in the **[CONFIGURATION > Wireless Enterprise > Mobile Phone Blacklist]** menu.
- Number Translation This is the information specified in the **[CONFIGURATION > User Group > Change User Group > Number Translation]** menu.
- Click to Dial Callee Number: This is the information specified in the **[CONFIGURATION > User Group > Change User Group > Click to Dial Callee Number]** menu.
- Inbound MCN Table: This is the information specified in the **[CONFIGURATION > Trunk Routing > Inbound MCN]** menu.
- Outbound MCN Table: This is the information specified in the **[CONFIGURATION > Trunk Routing > Outbound MCN Table]** menu.
- Toll Restriction List: This is the information specified in the **[CONFIGURATION > Trunk Routing > Toll Restriction List]** menu.

2.3.3 Phone Software Upgrade

The phone software upgrade feature can be executed by using the **[Tool > Phone Software Upgrade]** menu in the top left corner of SCM Administrator. The following figure shows the phone software upgrade screen.

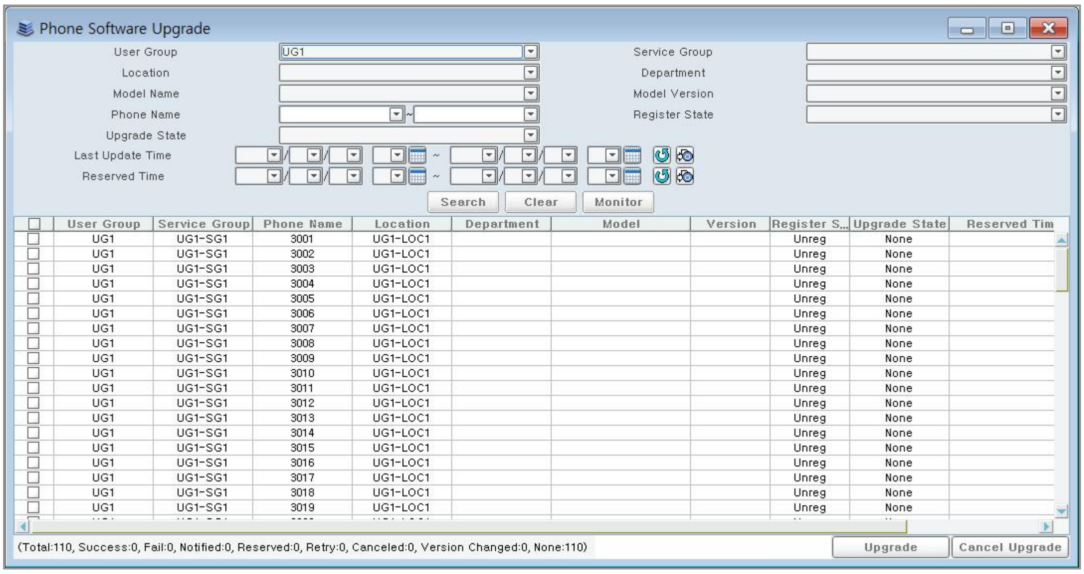

Figure 11. Phone Software Upgrade

Filtering area

Data on the screen can be viewed selectively using the filtering options.

Table area

Table displays the phone list which is Samsung-Desktop-Phone type.
[Upgrade State] column means the upgrade result of the last time. The upgrade state has the following types.

Option	Description
Success	The phone was re-registered after upgrading.
Fail	No response has been made from the phone for a certain period of time, or the phone is not re-registered.
Reserved	The phone has an upgrading reservation at reserved time.
Retry	System sent Notification to phone but there is no response message, so it will retry to send Notification after **[Phone Upgrade Retry Delay Time]**.
Notified	System sent Notification to phone and received response message, but the phone has not re-registered yet.
Canceled	The upgrading reservation was canceled.
Version Changed	Version of the phone was changed for another reason.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Option	Description
	Ex) The phone can be restarted.
None	The phone has no upgrade history.

Upgrade

Check checkboxes of the phones for upgrading and click **[Upgrade]** button. Only registered phones (which registered state is 'Reg') can be upgraded. The phones registered in slave system can be upgraded through SCM Administrator of slave system.

Upgrading is processed with the following steps:

1) Confirm versions: Input the upgrade version for phone models. The initial values are read from **[CONFIGURATION > Phone Setting > Software Upgrade Configuration]**.
2) Confirm upgrade: It displays the upgrade phone list and non-upgradable phones are marked with red color. You can set upgrade start time and interval.
3) Confirm Result: SCM Administrator send upgrade message to system and it displays the result. You can see the estimated start time, last time and time-out. If some errors are occurred, it will be displayed.

Cancel Upgrade

Check checkboxes of the phones for canceling and click **[Cancel Upgrade]** button. Only phones which state is 'Reserved', 'Retry' or 'Notified' can be canceled. The phones registered in slave system can be canceled through SCM Administrator of slave system.

Canceling upgrade is processed with the following steps:

1) Confirm cancel upgrade: It displays the canceled phone list and non-cancelable phones are marked with red color.
2) Confirm Result: SCM Administrator send cancel message to system and it displays the result. If some errors are occurred, it will be displayed.

2.3.4 Multibatch Key Programming

SCM Administrator's multibatch key programming feature allows setting programmable keys of several phones at one time.

The multibatch key programming feature can be executed by using the **[Tool > Multibatch Key Programming]** menu in the top left corner of SCM Administrator.
The following features are available.

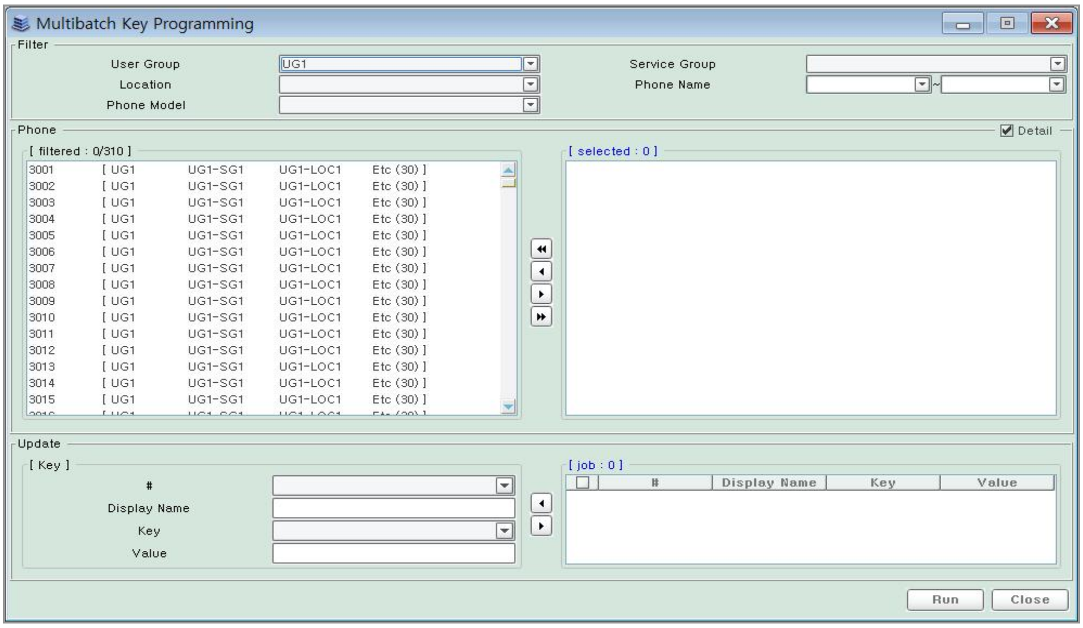

Figure 12. Multibatch Key Programming

Filter

You can specify the group of phones. The result of selected filter is displayed in the **[Phone]** frame.

Phone

You can select the filtered phones for updating key values. Clicking the **[>>]** button will move all of filtered phones to right panel. Clicking the **[>]** button will move the selected phones in left panel to right panel. **[Detail]** checkbox is used for toggling display about phone's detail information like User Group, Service Group, Location and Phone Model.

Update

After inputting the values (Key Number; #, Display Name, Key, Value), click **[>]** button and then the job will be added. And you can add the other job again.
If the number of programmable key on the selected phones are different, you can set for the number of the smallest key.
After adding all jobs, press **[Run]** button for updating the key setting of the phones.

2.4 Creating information for Individual Users

This section describes the procedure for creating information for individual users.

The user can be defined single phone user and multi-line user.

- Single Phone User: The phone has only one extension number. Also this extension number can be used only in that phone.
- Multi-Extension Phone: User information based on physical elements, such as the profile of the phone.
- Multi-Phone User: User information based on logical elements, such as the extension number of the phone.

When using Multi-line, phone and user can also be assigned as M:N. In other words, one phone can have multiple extension numbers by using the multiple appearance features, and one extension number can be assigned to multiple phones by using the multi-device feature.

2.4.1 Preparing User Creation

The followings are mandatory items for creating subscriber.

- Location: Indicates the location of the user. This allows selecting different trunk call paths using the same number within the same user group. You can also specify the priority codec for each location.
- User Group: Similar to the tenant group. A user group has its own numbering system and forms the basic unit of all data. You can assign all the users within the system to different groups and allow each group to use different sets of IPX-S300B supplementary services.
- Service Group: Lower-level group of the user group. Each user group can have many subgroups, and each subgroup can be allowed to use different sets of supplementary services.

2.4.1.1 User Group

User groups are the most basic units in IPX-S300B, and each user group is classified by its host.
User group enables users to use the extension number, even though it is already used in same IPX-S300B.

Making User Group

Administrators can create a user group in the **[CONFIGURATION > User Group > Creating User Group]** menu and multiple locations and service groups can be created. Entering the menu gets you started from ‘Step 2. Configuring User Group’ of the IPX-S300B Configuration Wizard, and in each step, you create necessary information by entering the information.

Changing User Group

When you created user group in the **[CONFIGURATION > User Group > Creating User Group]** menu, you didn't input data that saved default value.

You can change user group using the **[CONFIGURATION > User Group > Change User Group > Information]** menu.

The details are described below:

Items	Description
Name	Specify a name for the user group. Pay special attention to choosing a name, as it is used as an identifier when selecting the user group in other menus and cannot be changed.
Host	Enter a domain which is used as the host of SIP URI. Pay special attention to entering a domain, as the host cannot be changed.
Company Name	Enter a name of company which the user group belongs to. This is used for Premium CID.
CDR Storage Options	Specify how to create CDR data. - None: Does not create CDR data. - Local: Saves CDR data to the IPX-S300B hard disk. - FTP: Saves CDR data to a file in IPX-S300B and periodically transfers the file to a specified FTP server. - RADIUS: Transfers CDR data to the RADIUS server by using the RADIUS protocol whenever CDR data is created. - TCP: Transfers CDR data to the CDR server that is connected through TCP by using a unique method in IPX-S300B whenever CDR data is created. - TCP_SMDR: The Method to interoperate with the Accounting Server that is used for IPX-S300B system. (IOT mandatory) - TCP_ACK: When it is disconnect from CDR server, CDR data Will be stored inside SCM. When it is reconnect with CDR server, CDR data will be transfer to CDR server in order of occurrence
Authentication Method	Authentication method used when registering a user to IPX-S300B. - Internal: Uses internal authentication in the IPX-S300B. - LDAP: Performs authentication by interoperating with an external LDAP server. - RADIUS_S1: Performs authentication by using scenario 1 of an external RADIUS server. - RADIUS_S2: Performs authentication by using scenario 2 of an external RADIUS server.
MOH ID	Specify the ID of a sound source played when a call is on hold.
MOH Enable	Specify whether to play the system's sound source when a call is on hold.

Items	Description
	When the MOH is not in use, the phone plays its own on-hold tone or remains silent when a call is on hold.
Transfer Ring-back Tone	An MOH is played when the caller is put on hold for a transfer. When you page the number to which you are transferring the caller and hang up the phone, the MOH being played is changed to the ringtone. You can specify this transfer ringtone.
User Group Code	When a user make a call to the users belongs to different user group, a user should dial this User Group Code and Extension Number.
CLI Number	When the users of this User Group make a call, this CLI Number if configured is used as CLI for all users.
QOP (Quality of Protection)	Specify the QOP information when digest authentication is used.
Realm	Specify the Realm information when digest authentication is used.
Algorithm	Specify the algorithm when digest authentication is used.
LDAP Root Directory	You can enter the base directory path of the LDAP server when LDAP is selected as the authentication method.
Restriction Policy	Specify a restriction policy to apply to the users belonging to the user group. A restriction policy only applies to trunk calls. You can select one of the existing restriction policies. You can create restriction policies in the **[CONFIGURATION > Routing > Restriction Policy List]** menu.
Default Access Code	You can set default access code for trunk call routing.
Min. Digit (Default Access Code)	You can set minimum digit length to use Default Access Code.
Service Call Access Code	You can set service call access code for trunk call routing. This access code is used for default access code relating mobile service.
Media Option	You can set the allowed media to the call. - RTP only: IPX-S300B don’t accept sRTP media. - sRTP only: IPX-S300B doesn’t accept RTP media to the call.
sRTP Mine offer Option	This is the function that contains the 2 mine (AVP/SAVP) in the INVITE message of sRTP calls.
External Cooperation ID	You can specify the IPX-S300B ID, If you want to work with the external device for special purposes, IPX-S300B is, the ID of the IPX-S300B.
Restriction User Group	Specify call restrictions between user groups. The current user group is restricted from making calls to the selected user groups.
Transfer Caller ID for VM/AA	Select the Caller ID Number which is provided to VM/AA. When VM/AA receives the Semi-Blind Transfer Call, Specify the Caller ID Number of transferer or transferee.

2.4.1.2 Location Information

Location refers to the actual location of the user. You can set a different location for each user and endpoint to apply call restrictions between different locations according to the bandwidth, assign preferred Codecs within a location or between specific locations, or select different trunk call paths for locations. Your users may actually be in different locations, but if there is no need to apply separate call restrictions, preferred codecs, or trunk call paths by location, we recommend that you only create one location.

Making Location Information

When you created user group in the **[CONFIGURATION > User Group > Creating User Group]** menu, one or more locations can be created.

Changing Location Information

When you created user group in the **[CONFIGURATION > User Group > Creating User Group]** menu, you didn't input data that saved default value.

You can change location using the **[CONFIGURATION > Location > Location]** menu. And you can create new location. When you create new location, the following items are mandatory.

Item	Description
Name	Specify a name to identify the location. Pay special attention to choosing a name, as it is used as an identifier when selecting the location in other menus and cannot be changed.
Bandwidth	Specify the maximum available bandwidth for the location. You can enter up to 1000000.
Intra-Location Video Codec	Select a preferred video codec for the location.
Inter-Location Video Codec	Select a preferred video codec for the other location.
Intra-Location Audio Codec	Select a preferred audio codec for the location.
Inter-Location Audio Codec	Select a preferred audio codec for the other location.
Intra-Location Forced Codec	Select an allowed single audio codec for the location.
Inter-Location Forced Codec	Select an allowed single audio codec for the other location.
Announcement Codec	Select a preferred announcement codec for the location.

2.4.1.3 Service Group

A service group is a small unit in a user group and is used for applying different service policies. Related policies include external call restriction policy, per-group class of service settings, and routing plan for service groups.

Making Service Group

When you created user group in the **[CONFIGURATION > User Group > Creating User Group]** menu, one or more service groups can be created.

Changing Service Group

When you created user group in the **[CONFIGURATION > User Group > Creating User Group]** menu, you didn't input data that saved default value.

You can change service group using the **[CONFIGURATION > User Group > Service Group]** menu. And you can create service group.

Item	Description
User Group	Specify a user group to which the service group belongs. You can select one of the existing user groups.
Name	Specify a name for the service group. Pay special attention to choosing the name, as it is used as an identifier when selecting the service group in other menus and cannot be changed.
Service Group Code	Specify code of service group. It identifies service group.
CLI Number	Specify CLI number.
Class of Service	Specify a class of service to apply to the users belonging to the service group. You can select one of the existing Class of Service. You can create Class of Service in the **[CONFIGURATION > Service > Feature Service > Class of service]** menu.
Restriction Policy	Specify a restriction policy to apply to the users belonging to the service group. A restriction policy only applies to trunk calls. You can select one of the existing restriction policies. You can create restriction policies in the **[CONFIGURATION > Routing > Restriction Policy List]** menu.
Dial Tone	Specify a digit to create a virtual dial tone. The profile information is downloaded to the phone, allowing the phone to play a dial tone when a specified digit is entered.
Dial Plan	Specify a dial plan to apply to the users belonging to the service group. You can select one of the existing service group based dial plan. You can create Service Group based dial tone in the **[CONFIGURATION > Phone Setting > Service Group based**

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
	Dial Plan] menu.
Application Server Service Group	Specify a node0's application server service group to apply to the users belonging to the service group. You can select one of the existing Application Server Service Group. You can create Application Server Service Group in the **[CONFIGURATION > Application > Application Server Group]** menu.
Node1's App Server Service Group	Specify a node1's application server service group to apply to the users belonging to the service group. You can select one of the existing Application Server Service Group. You can create Application Server Service Group in the **[CONFIGURATION > Application > Application Server Group]** menu.
Call Recording Method	You can select call recording method Conference Recording or Phone Recording.
Auto Attendant Ring Plan Schedule	Specify a dial plan to apply to the users belonging to the service group for auto attendant service
CFUR Service Schedule	Specify a dial plan to apply to the users belonging to the service group for CFUR service
Send Extension Number Usage	Select how to use **[Send Extension Number]** when a call is setup between different service groups. This options is worked when **[Send Extension Number]** is entered. - With Service Group Code Service Group Code is prefixed to **[Send Extension Number]** as a calling number when a call is setup between different service groups. - Without Service Group Code For a call between different service groups **[Send Extension Number]** is used as a calling number without prefix.
Phone Recording Start Time	In case of auto record using a Samsung phone, the call for a recording server can be made when the call for recording service is ringing or answered. The 'Answer' option is recommended when the recording server is internal VM for smooth support of better instruction for recording start. - Ringing The call for a recording server can be made when the call for recording service is ringing. - Answer The call for a recording server can be made when the call for recording service is answered.

2.4.2 Making Single Phone User

Single Phone User means general phone user. Phone and User are paired on a 1:1 basis. You can specify the extension number of the phone, the authentication method of the phone, and so on.

You can create devices using the **[CONFIGURATION > User > Single Phone User]** menu.

Item	Description
Basic Configuration	
User Group	Select a user group.
Service Group	Specifies a subscriber's service group. You can select one out of pre-defined service groups.
Location	Specifies the location where a subscriber is located. You can select one out of pre-defined locations.
Language	Specifies a language.
Extension Number	Specifies the extension number of a subscriber. Used as a separator when specifying a subscriber in another menu and it cannot be changed. Thus extra caution is required.
Extension Name	Specifies a subscriber's name.
Mobile Phone Number	Specify the Mobile phone number to be used to Move to Mobile service.
Use Mobile Phone Number	Specify the usage of Mobile Phone Number - None: do not use. - Ring Only: MOBEX behavior. Your phone and Mobile Phone are ringing simultaneously. - Dial Only: Mobile Remote Call behavior. Make a connection to the Mobile Voice Server; execute the Click To Dial service. - Both: Ring Only, Dial Only both services are provided.
Application User ID	Input an ID of an application user.
Application Password	Input a password of an application user.
Authentication User ID	Enter a user ID which is used for authentication during subscriber registration.
Authentication Password	Enter a password which is used for authentication during subscriber registration.
Phone Login ID	Specifies a profile ID. A Samsung phone can do auto environment configuration by getting the information of a phone in IPX-S300B by using a profile ID and a password.
Phone Login Password	Specifies a profile password. A Samsung phone can do auto environment configuration by getting the information of a phone in IPX-S300B by using a profile ID and a password. (mandatory if there is a profile ID)

Item	Description
PIN Number	Specifies a password which is used when using a service such as DISA or CRU.
Make Mailbox	Specifies if using mailbox - Yes: It will make a mailbox number same with extension number.
Phone Type	Select a phone type. - Samsung-SIP-Phone: Select this when a subscriber's number is specified to a Samsung SIP phone. - Samsung-Soft-Phone: Select this when a subscriber's number is specified to a Samsung Softphone. - Samsung-Mobile-Phone: Select this when a subscriber's number is specified to a Samsung mobile Softphone. - Samsung-PC-operator: Select this when a subscriber's number is specified to an IP operator Softphone. - 3rd-Party-SIP-Phone: Select this when a subscriber's number is specified to a 3rd party phone other than Samsung. - FXS-Phone: Select this when a subscriber's number is specified to an analog phone port which is connected to a gateway.
Phone Verification	Specifies device authentication method. Device authentication is required to use multiple devices. - None: No device authentication. - IP Address: Performs authentication by using an IP address device. This is applicable only when a fixed IP address is used. - MAC Address: Performs device authentication by using the MAC address of a device. Because a Samsung phone has a MAC address in its REGISTER, you can use the information for registration. - IP Address (NAT): Performs authentication by using an IP address device for a device which is under NAT of a remote location.
MAC Address	MAC Address of a device (mandatory if authentication method is MAC_ADDRESS)
IP Address	IP Address of a device (mandatory if authentication method is either IP_ADDRESS or NAT_IP_ADDRESS)
Private IP Address	Private IP address of a device in the NAT environment (mandatory if authentication method is NAT_IP_ADDRESS)
Use mVoIP	Select the use of mVoIP
VMS Extension Number	Specify extension number for VMS
Protocol	
Protocol	Specifies a protocol. You can select one out of UDP, TCP, or TLS.
Check Registration Protocol	Specify whether to check the protocols (UDP, TCP, TLS) when registration.
Media	Select a media type that a device is going to use. Specifies the priority of an encryption algorithm for media encryption.

Item	Description
	- RTP: Does not encrypt a media. - sRTP (AES/ARIA): Encrypts media into the ARIA or AES protocol, and uses AES first. - sRTP (ARIA/AES): Encrypts media into the ARIA or AES protocol, and uses ARIA first. - sRTP (AES): Encrypts media into the AES protocol. - sRTP (ARIA): Encrypts media into the ARIA protocol.
URI Type	Specifies SIP in general as URI type. - SIP - SIPS
TLS Connection	Specify the TLS connection type: - Normal: use TLS Full handshaking method. - Reuse: reuses the existing TLS connection established by initial Message. - Resume: use simplified handshaking method using TLS Session ID.
TLS Key Decryption Password Type	Specify TLS Key Decryption Password Type.
TLS Key Decryption Direct Password	Specify TLS Key Decryption Direct Password.
TLS Key Decryption Device ID	Specify TLS Key Decryption Device ID
TLS Key Decryption Salt	Specify TLS Key Decryption Salt
TLS Key Decryption IC	Specify TLS Key Decryption IC
TLS Key Decryption DX	Specify TLS Key Decryption DX
MOH SIP Media Mode	Specify the SDP media mode which will be send to the held side when hold.
DTMF	Specifies the DTMF signaling system of a device. - RFC2833: Sends a DTMF signal in the RFC2833 protocol. - Invoice: Sends a DTMF signal together with a general voice signal. - Outband: Sends a DTMF signal in the RFC2976 protocol.
RFC2833 DTMF Payload	Specifies a payload when DTMF is RFC2833.
SIP-PRACK option	Select the use of SIP PRACK
Send RTCP on Hold	Select the use of Send RTCP
Use InterProxy	When use only TLS protocol, this options is activated. If ENABLE, it can operate with inter PROXTB.
PROXYB Index	Inform the PROXYB index that is assigned
Number Translation	
Service Group Local Number	Notify information of station name to Phone.

Item	Description
Service Group Local CLI Number	Short name of phone user.
Send Extension Number	Specify value of Display number field in internal call.
Phone Display Name	Specify Phone Display name - Extension Number - Send Extension Number
Send CLI Number	When making trunk call, specify value of userinfo field.
Send CLI Name	Specify value of Display name field in trunk call.
CLI for Forwarded Call	Select the use of CLI for Forwarded Call - Originator - Forwarder - Forwarder with Originator Name
Display Option	Specify the priority order of CLI information display on the terminal. - Normal: (line 1) Peer number (line 2) Reason > Divert number > Peer name - Peerinfo: (line 1) Peer number (line 2) Peer name - Reason: (line 1) Blank (line 2) Reason - Diversion: (line 1) Divert number (line 2) Diverted info string * Reason/Diversion type is valid only when a reason/diversion field exists.
Service	
MOH Announcement ID	Specify the sound source ID of an on-hold tone played when the user is put on hold.
Service Schedule	Select the Service Schedule to provide Leaving Office Class of Service. When Service Schedule status is unset, Leaving Office Class of Service activates.
Class of Service	Specifies a service level which will be applied to a subscriber. You can select a service level from the service levels created in **[Configuration > Service > Function Service > Service Level]** menu.
Extension Lock	Can restrict the use of a subscriber’s number. - None: No restriction. - Dialing-Only: Uses a subscriber’s number only for an outgoing call. - Answering-Only: Uses a subscriber’s number only for an incoming call.

Item	Description
	- Both: Prevent a subscriber's number from outgoing/incoming.
Restriction Policy	Specifies call restriction policy which will be applied to a subscriber. The call restriction policy is only applied to a trunk call, and you can select one out of pre-defined call restriction policies which are created in **[Configuration > Trunk Configuration > Call Restriction Policy Configuration]** menu.
Allow Selective Call	Select selective call allow list.
Mobile Number Auto Update	Select the use of auto update mobile number when mobile phone is registered
Desk Phone Simultaneous Ring Delay	Select the use of Desk Phone Simultaneous Ring Delay.
Ping Ring Type	Specify notification type. - Audio + Visual: You can be notified by phone pop-up display and ring - Audio: You can be notified by ring only - Visual: You can be notified by phone pop-up display. - None: There no any ping ring notification.
Caller Ring Type	If you specify a certain value of this and make a call to an extension, the extension will hear this ring type.
Use Virtual Ringback	Specify whether to use ring back tone stored in the Phones.
External Ringback Tone Use	Specify whether to use External Ring Back Tone. - NONE: do not use. - Internal: only used to internal call. - External: only used to external call - Both: used to both internal and external calls.
Error Announcement	Select the use of announcement when call has failed.
Account Code Use	Can specify whether to enter an account code. - Force: It is mandatory to enter an account code when sending an external call. - Voluntary: It is not necessary to enter an account code when sending an external call. You may enter an account code after pressing the Account Code button if you are using a Samsung phone.
Call Monitoring	Specify whether to monitor this user or not.
Call Recording Method	You can select call recording method Conference Recording or Phone Recording.
Nurse Call Service	Select the use of Nurse Call Service. If Enable is chosen, IPX-S300B analyzes a dialed number to get a bed and room number which is used for calling number. The pattern of dialed number is 'Bed & Room Number + Nurse Call Delimiter + Called Number'.
Nurse Call Prefix	Nurse Call Prefix is inserted in front of the bed and room number. Except Nurse Call Prefix, other number translation is not applied to

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Item	Description
	the calling number of Nurse Call.
Change to Multi Type	Change Single Phone User to Multi-type
Call Appearance	Select Multi-Device Number Type. - SCA (Single Call Appearance) - MCA (Multi Call Appearance)
User Type	Select Multi-Device User Type. - Normal - Secretary - Manager
Interworking	
Department	Specifies a subscriber's department. You can select a department out of departments which are created in **[Configuration > User Group > Department]** menu. Department information is very useful when searching for a subscriber in an application.
Position	Specifies a subscriber's position. You can select a position out of positions which are created in **[Configuration > User Group > Position]** menu. Position information is very useful when searching for a subscriber in an application.
Directory Service	Configure whether Directory Service provides or not for a user.
LDAP DN Number	Enter a user directory number which is used when interoperating with the LDAP server.
Application Server Service Group	Specify an application server service group to apply to the user.
Gateway Name	Specifies a gateway which handles a subscriber's call in survival mode (a mode when a basic telephone service is available from a gateway which is located in the same area because communication between a subscriber's phone and IPX-S300B is lost).
mySingle Click To Dial CLI	Select CLI of desk or mobile phone number when click-to-dial application of smart phone makes a call - Origin: Desk phone number - Remote: Mobile phone number
Auto Answer by Click to Dial	If a remote device (PC) sends a calling command to IPX-S300B when remote calling (Click to Dial) service is used, IPX-S300B sends a ring to a phone. When the phone answers, it also sends a call to the phone according to a command. At this time, it specifies whether to send a call after auto speaker phone response instead of sending a ring to a subscriber. There may be no auto answer if the phone is not a Samsung phone.
User Account Number	Specify a number of Account Code.
CMS Monitoring	Specify whether to use CMS Monitoring

Item	Description
FMS Zone Name	Specify a FMS Zone. You can select one of FMS zone created in the **[CONFIGURATION]** > Wireless Enterprise > FMS Zone
Phone	
Phone Call History	Specify save call history at phone or not
Resume after Transfer-Cancel	Select auto resume or hold when transfer cancel
Time Zone	Specifies the time zone of phone. - Same as System: It will be set same with time zone of system.
Accept Login Override	Specify whether to allow login override.
Push Notification	Select the use of Push Service
Use Smart WLAN Link	Select the use of Smart WLAN Link Service at Mobile Phone (need to get License for Smart WLAN Link)
Input Number Display	Specify display input digit
Off Hook Alarm	Specify whether to use alarm in case of off-hook state for a long time.
Web Access Protocol	Select the use of http/https or https for Web Access Protocol
Telnet-Access	Select the use of Telnet-Access at phone
Use Wi-Fi	Select the use of WiFi-Access at phone
EHS Headset	Select the use of EHS Headset
Phone TX Gain	Specify a value of Phone TX Gain.
Comfort Noise	Select the use of Comfort Noise at Samsung-Desktop-Phone
Use Bluetooth	Select the use of Bluetooth at Phone
Use USB	Select the use of USB at Phone
Use NFC	Select the use of NFC at Phone
NFC Mobile Phone Name	Specify NFC Mobile Phone Name
NFC Auto Login	Specify use NFC Auto Login
Display Forward Number on Idle Screen	Specify whether phone will display the forward number on screen when Forward function is activated.
Hotspot Channel (Country)	Specify Hotspot Channel for WiFi configuration phone use.
CallLog for XML P-CID	Specify whether to log the call of Premium CID by XML Server.
Use Syslog	Specify whether to use a syslog function.
Syslog Type	select the type of syslog.
Syslog Level	select the log level of syslog.
Roaming Trigger	Roaming Trigger Parameter for IP Phone WIFI module.
Roaming Delta	Roaming Delta Parameter for IP Phone WIFI module.
Roaming Scan Period	Roaming Scan Period Parameter for IP Phone WIFI module.

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2.4.3 Making Multi-Extension Phone

Multi-Extension Phone information refers to the user's physical elements. You can specify the device user type, the extension number of the phone, the authentication method of the phone, and so on. Phone authentication is required when using multi-device, when multiple phones use one extension number.

You can create devices using the **[CONFIGURATION > User > Multi-Extension Phone]** menu.

Item	Description
User Group	Specify a user group to which the device belongs. You can select one of the existing user groups.
Extension Name	Specify a name for the phone. Pay special attention to choosing the name, as it is used as an identifier when selecting the device in other menus and cannot be changed.
Phone Authentication type	Specify the device authentication method. Device authentication is required to use multi-device. - None: No device authentication. - IP Address: Authenticates the device by using the IP address. This option is available only when using fixed IP addresses. - MAC Address: Authenticates the device by using the MAC address of the device. Every Samsung phone has a unique MAC address that can be used for this purpose. - IP Address (NAT): Authenticates the device by using the IP address when the device is on a remote NAT network.
Profile login ID	Specify the profile ID. Samsung phones can perform automatic configuration by fetching the phone information from IPX-S300B using the profile ID and password.
Profile login Password	Specify the profile password. Samsung phones can perform automatic configuration by fetching the phone information from IPX-S300B using the profile ID and password.
User Type	Select the phone user type. - Normal: The type of phone used by a normal user. - Secretary: The type of phone used by a secretary when using the Manager/Secretary feature. - Manager: The type of phone used by a manger when using the Manager/Secretary feature.
Phone Type	Select a phone type. - Samsung-Desk-Phone: Select this when assigning the user's number to a Samsung SIP phone. - Samsung-Soft-Phone: Select this when assigning the user's number to a Samsung soft phone. - Samsung-Mobile-Phone: Select this when assigning the user's

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Item	Description
	number to a Samsung mobile soft phone. - FMS-Phone: A virtual user created to replace a trunk call with a virtual user call so that Zone service for a particular trunk can be enabled.
Language	Specify the language.
URI Type	The URI type can be set SIP or SIPS, the SIP is used normally.
Protocol	Specify the protocol (select from among UDP, TCP, and TLS).
DTMF Method	Specify the DTMF signaling method that will be used by the device. - RFC2833: Sends a DTMF signal via the RFC2833 protocol. - Invoice: Sends a DTMF signal together with an audio signal. - Outband: Sends a DTMF signal via the RFC2976 protocol.
Media Type	Select the media type that will be used by the device. (Specify the priority order of encryption algorithms as well if media is to be encrypted.) - RTP: Does not encrypt media. - sRTP (AES/ARIA): Encrypts media using the ARIA or AES protocol. (AES is used preferentially.) - sRTP (ARIA/AES): Encrypts media using the ARIA or AES protocol. (ARIA is used preferentially.) - sRTP (AES): Encrypts media using the AES protocol. - sRTP (ARIA): Encrypts media using the ARIA protocol.
Allow Duplicate Login	Specify whether to allow duplicate login.
Phone Display Information	Specify where the terminal will get CLI information though below a specified header. - Normal: Specifies the received SIP message's from header as CLI information. - Peer Info: Specifies the received SIP message's P-Asserted-Identity header as CLI information. - Reason: Specifies the received SIP message's Reason header as CLI information. - Diversion: Specifies the received SIP message's Diversion header as CLI information.
TLS Connection	Specify the TLS connection type: - Normal: use TLS Full handshaking method. - Reuse: reuses the existing TLS connection established by initial Message. - Resume: use simplified handshaking method using TLS Session ID.
MOH SIP Media Mode	Specify the SDP media mode which will be send to the held side when hold.

When creating a multi-line phone, you can choose whether or not to enter the following information as needed.

Item	Description
MAC Address	Enter a MAC address if you have specified the authentication method as MAC Address.
IP Address	Enter an IP address if you have specified the authentication method as IP Address.
Private IP Address	Enter a private IP address if you have specified the authentication method as IP Address (NAT).
License Phone Type	Set the license type of the terminal created. Normally, a created terminal's phone type is set to the same as its license phone type. However, in the case of Samsung desktop phones, you can create a terminal using another terminal's license.
RFC2833 DTMF Payload	Specify the payload value for when using RFC2833 DTMF.
Standard time	Specify the difference between the SCM's time, which is the standard time, and each device's time. (If a phone is located in a different time zone from the SCM, you can change the phone's displayed time to be different from the system's.)
Originating Number	Specify an originating name when a call has originated from a trunk.
Off-the-hook Alarm	Set whether to trigger an alarm when off-the-hook status has continued for an extended period of time.
Check Registration Protocol	Specify whether to examine the protocol used for registration, such as UDP, TCP, or TLS.
Phone Tx Volume	Specify the Tx volume of the phone (0-5). If you do not specify a value, the TX volume that has been specified for the phone model in **[CONFIGURATION > Phone Settings > Volume]** will be applied.
TLS Key Decryption Password Type	Specify TLS Key Decryption Password Type.
TLS Key Decryption Direct Password	Specify TLS Key Decryption Direct Password.
TLS Key Decryption Device ID	Specify TLS Key Decryption Device ID
TLS Key Decryption Salt	Specify TLS Key Decryption Salt
TLS Key Decryption IC	Specify TLS Key Decryption IC
TLS Key Decryption DX	Specify TLS Key Decryption DX
NFC Mobile Phone Name	Specify NFC Mobile Phone Name
Phone Call History	Specify save call history at phone or not
Main Ext.	Specify the extension number that will be used as the multi-line

Item	Description
	phone's main number, which is the extension number of the line that gets picked in idle or off-the-hook status.
Ext. Select Method When Main Ext is in Use.	Specify how to select an extension number when the main ext. is off-the-hook because it is being used by another phone. - First: Select the first available extension number in a specified user sequence. - Next: Select the available extension number that is next to the main ext. in a specified user sequence.
NFC Auto Login	Specify use NFC Auto Login
NFC Auto Login Phone Name	Specify the name of the multi device used for NFC auto login service.
Use InterProxy	When use only TLS protocol, this options is activated. If ENABLE, it can operate with inter PROXTB.
PROXYB Index	Inform the PROXYB index that is assigned
Input Number Display	Specify whether to display input digit or not in conversation
Use mVoIP	Select the use of mVoIP
Resume after Transfer-Cancel	Select auto resume or hold when transfer cancel
Telnet-Access	Select the use of Telnet-Access at phone
Use Wi-Fi	Select the use of Wifi-Access at phone
SIP-PRACK option	Select the use of SIP PRACK for MS Lync
Comfort Noise	Select the use of Comfort Noise at Samsung-Desktop-Phone
Send RTCP on Hold	Select the use of Send RTCP for MS Lync
Web Access Protocol	Select the use of http/https or https for Web Access Protocol
EHS Headset	Select the use of EHS Headset
Push Notification	Select the use of Push Service
Use Bluetooth	Select the use of Bluetooth at Phone
Use USB	Select the use of USB at Phone
Use NFC	Select the use of NFC at Phone
Use Smart WLAN Link	Select the use of Smart WLAN Link Service at Mobile Phone(need to get License for Smart WLAN Link)
Display Forward Number on Idle Screen	Specify whether phone will display the forward number on screen when Forward function is activated.
Hotspot Channel (Country)	Specify Hotspot Channel for Wifi configuration phone use.
CallLog for XML P-CID	Specify whether to log the call of Premium CID by XML Server.
Use Syslog	Specify whether to use a syslog function.
Syslog Type	select the type of syslog.
Syslog Level	select the log level of syslog.
Roaming Trigger	Roaming Trigger Parameter for IP Phone WIFI module.

Item	Description
Roaming Delta	Roaming Delta Parameter for IP Phone WIFI module.
Roaming Scan Period	Roaming Scan Period Parameter for IP Phone WIFI module.

2.4.4 Making Multi-Phone User

Multi-Phone User information refers the user's logical elements. You can access the basic information, such as the user's phone number, and various options, including the methods for using the services. IPX-S300B user services are provided based on the settings included in the user information.

You can create user information using the **[CONFIGURATION > User > Multi-Phone User]** menu.

Item	Description
User Group	Specify a user group to which the device belongs. You can select one of the existing user groups.
Service Group	Specify a service group for the user. You can select one of the service groups already defined.
Location	Specify the location of the user. You can select one of the existing locations.
Extension Number	Specify the user's extension number. Pay special attention to choosing the number, as it is used as an identifier when selecting the user in other menus and cannot be changed.
Application User ID	Specify a universal user ID used with applications.
Application Password	Specify a universal user Password used with applications.
Name	Specify the name of the user.
Service Password	Specify a password that will be used when using services.
Multi Type	Specify the multi type - Multi Line: Select this option if the extension number is to be allocated to one phone as a multi-line number. - Multi Line & Multi Device: Select this option if the extension number is to be allocated to multiple phones.
Use Restrictions	Set up restrictions for using the user number. - None: No restriction on the use of the user number. - Dialing-Only: Uses the user number for originating purposes only. - Answering-Only: Uses the user number for terminating purposes only. - Both: Prohibits the use of the user number for both originating and terminating purposes.
User Authentication ID	Enter a user ID that will be used for authentication when registering the user.

Item	Description
User Authentication Password	Enter a password that will be used for authentication when registering the user.

When creating a multi-line user, you can choose whether or not to enter the following information as occasion demands.

Item	Description
Mobile Phone Number	Specify the Mobile phone number to be used to Move to Mobile service.
Mobile Phone Number Use	Specify the usage of Mobile Phone Number - None: do not use. - Ring Only: MOBEX behavior. Your phone and Mobile Phone are ringing simultaneously. - Dial Only: Mobile Remote Call behavior. Make a connection to the Mobile Voice Server; execute the Click To Dial service. - Both: Ring Only, Dial Only both services are provided.
Department	Specify the user's department. You can select one of the departments created in the **[CONFIGURATION > User Group > Department]** menu. Department information is useful when searching for a user in an application.
Position	Specify the user's position. You can select one of the positions created in the **[CONFIGURATION > User Group > Position]** menu. Position information is useful when searching for a user in an application.
Send CLI Number	When making trunk call, specify value of userinfo field.
Send CLI Name	Specify value of Display name field in trunk call.
Service Group Local Number	Short name of phone user.
Service Group Local CLI Number	Notify information of station name to Phone.
Multi-line Type	Specify the call processing option for when you have specified User Type as Multi Device. - SCA: Allows only one call at a time for the user number. - MCA: Allows multiple calls at a time for the user number. Even if the user number is engaged, it is possible to make calls from other phones that have the same user number.
Class of Service	Specify a class of service to apply to the user. You can select one of the class of service created in the **[CONFIGURATION > Service > Feature Service > Service Class]** menu.
Restriction Policy	Specify a restriction policy to apply to the user. The restriction

Item	Description
	policy only applies to trunk calls. You can select one of restriction policies created in the **[CONFIGURATION > Routing > Restriction Policy List]** menu.
Gateway Name	Specify a gateway that handles the user's calls in survival mode (when the phone is disconnected from IPX-S300B, it connects to a gateway in the same location to provide the basic telephony services.).
MOH ID	Specify the sound source ID of an on-hold tone played when the user is put on hold.
Account Code Use	You can specify whether to enter the account code. - Force: Entering the account code is mandatory when making external calls. - Voluntary: Entering the account code is not mandatory. You can voluntarily enter the account code during the call by pressing the Account Code button.
LDAP DN Number	Enter a user directory number used when interoperating with the LDAP server.
Auto Click To Dial	When using the click-to-dial service, if you send the call command from the remote device (PC) to IPX-S300B, IPX-S300B calls your phone. When you answer the call, an outgoing call is made from the phone according to your command. If you use this option, your phone does not ring but automatically responds to the call from IPX-S300B through the speakerphone as the call is made. Automatic answering may not work for non-Samsung phones.
External Ring-back Tone Use	Specify whether to use External Ring Back Tone. - NONE: do not use. - Internal: only used to internal call. - External: only used to external call. - Both: used to both internal and external calls.
Call Monitoring	Specify whether to monitor this user or not.
Send Extension Number	Specify the caller id when make a call to extension.
Use Virtual Ring back	Specify whether to use ring back tone stored in the Phones.
Multi-Device Conference Join	Specify whether for multi-devices to join to same conference
Caller Ring Type	If you specify a certain value of this and make a call to an extension, the extension will hear this ring type.
Application Server Service Group	Specify an application server service group to apply to the user.
Ping Ring Type	Specify notification type. - Audio + Visual: You can be notified by phone pop-up display and ring - Audio: You can be notified by ring only

Item	Description
	- Visual: You can be notified by phone pop-up display. - None: There no any ping ring notification.
CMS Monitoring	Specify whether to use CMS Monitoring
A-A Primary Node	Specify Primary node when Active-Active mode
A-A Dual Registration	Specify dual registration enable or not when Active-Active mode. If select dual registration enable, Phone register to both node.
VMS Extension Number	Specify extension number for VMS
Call Recording Method	You can select call recording method Conference Recording or Phone Recording.
Allow Selective Call	Select selective call allow list.
Phone Display Name	Specify Phone Display name - Extension Number - Send Extension Number
Error Announcement	Select the use of announcement when call has failed.
Desk Phone Simultaneous Ring Delay	Select the use of Desk Phone Simultaneous Ring Delay.
CLI for Forwarded Call	Select the use of CLI for Forwarded Call - Originator - Forwarder - Forwarder with Originator Name
Mobile Number Auto Update	When this value is 'Yes', Mobile Phone Number is automatically updated when the phone registers
mySingle Click To Dial CLI	Select CLI of desk or mobile phone number when click-to-dial application of smart phone makes a call - Origin: Desk phone number - Remote: Mobile phone number
User Account Number	Specify a number of Account Code.
Service Schedule	Select the Service Schedule to provide Leaving Office Class of Service. When Service Schedule status is unset, Leaving Office Class of Service activates.
DirectoryService	Specify whether to provide an directory service to the subscriber.
Enable Mailbox	Specify whether to enable the voicemail feature. (Enabling the feature creates a voicemail number for the extension number.)
Phone Number	Specify the phone to which you have allocated the user number. (Only phones that match both the user type and the phone type can be specified.)

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2.5 Making Individual Trunk information

This section describes a process of creating information for trunks. The information shown below are the mandatory requirements for trunk calls, listed in the required order of creation.

1) Route: This is the trunk port information for external connection to ITSP SIP servers, gateways, and other entities that interoperate with IPX-S300B.
2) Priority Routing: Specifying the preferred routes that are connected to the endpoint allows automatic selection of alternative routes and other factors.
3) Location Based Routing: You can specify different call routes according to the caller's location.
4) Access code: This is the access code used for selecting trunk call routes.

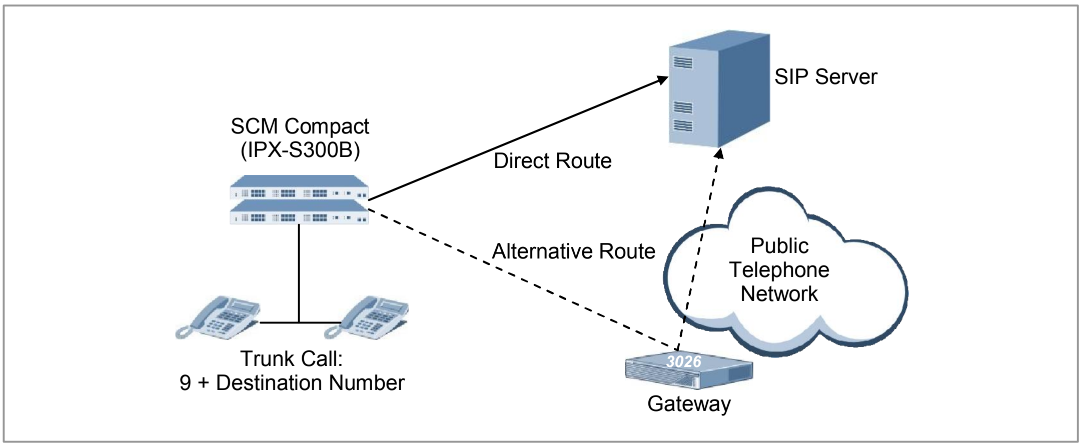


Figure 13. Trunk Call Connection with IPX-S300B

The diagram above illustrates the process of making a trunk call from IPX-S300B, in which the user dials '9 + destination number' to call an external phone. Upon receiving this number, IPX-S300B removes 9, which is the access code for making external calls, and only sends the destination number to the SIP server or the gateway. In general, IPX-S300B is configured with a direct link for calling the SIP server. But if the link with the SIP server is severed, the call is made through the gateway, which is an alternative route.

2.5.1 Making Routes

A route means a conceptual path connected to an SIP server, a gateway, and other entities interoperating with IPX-S300B. The route includes information on the handling method for outgoing/incoming calls from each external connection endpoint as well as the number translation policy information.

You can create routes using the **[CONFIGURATION > Trunk Routing > Route]** menu. Route options are categorized to 6 group as follows:

- Basic Configuration
- SIP Registration
- Additional SIP
- Trunk Service
- Interworking
- Number Translation and display

<Route-Basic Configuration>

Item	Description
Route Type	Specify that the trunk route is used for a certain user group only.
User Group	Select a user group to which the route belongs. If route type is set to Common, the user group also must be set to Common.
Route Name	Input a name of a route
Location	Specify a location to which the endpoint belongs.
Register Type	Specify a registration method for the endpoint. - Receive REGISTER: REGISTER is received from the endpoint for registering the endpoint. - Send REGISTER: IPX-S300B sends REGISTER to the endpoint for registration. - None: No registration is performed between the endpoint and IPX-S300B.
Port	Specify a port number for the endpoint.
User Name	Specify the user name to use in user info of SIP URI.
Domain Name	Enter a domain to use as the host of SIP URI.
Multiple Registration	Enable this feature if you want to register multiple accounts for the same ISP. - You cannot make a route with setting Multiple Registration to Enable if there is another route which Multiple Registration is Disble. For the same ISP. - You can set up Multiple Registration only when Register Type is set to Send Register.
User Number Range	Enter the called number range which is used to find receiving Rout.

Item	Description
	- 'Number', '**[-]**' and 'X' can be entered for 'User Number Range'. - 'User Number Range' has to be started with number. Range can be expressed with '**[-]**'. 'X' means any number. - User Number Range is supported 3 digits and the range is possible within 100. ex) Example ・0312791000: In case a called number is same to 0312791000 ・0312791XXX: In case a called number is starting with 0312791 and its length is 7. ・031279[1-4]XXX: In case a called number is starting with 0312791, 0312792, 0312793 and 0312794 and its length is 7.
Proxy Server	Specify the primary proxy server address for the endpoint.
Secondary Proxy Server	Enter the IP Address of secondary Proxy Server. This server will be used when primary server is no response. * It works only when Failover is enabled.
Authentication User Name	Enter the user authentication name used for registration.
Authentication Password	Enter the authentication user password used for registration.
DNS	Enter the IP address of DNS server.
DNS2	Enter the IP address of second DNS server.
Tie Trunk	Decide whether receiving calls from trunk to SCM Compact is relayed to the same gateway or to another turnk or not. - TIE: A trunk that allows tandem calls. - Normal: A trunk that does not allow tandem calls.
Use Internal SBC	Select whether to use internal SBC when creating a trunk route. Select Yes when the trunk route is located on a WAN.
Access Number	If you specifys an Access number when creating a route, basic information such as 'Priority Routing-Location Based Routing-Access Code' will be configured automatically. And the Number Type of Access Code will be set to Normal.

<Route-SIP Registration>

Item	Description
Check Registration Protocol	Specify whether to check the protocols when register.
Register Address	If using a separate registration server, enter the address of the registration server.
Register Expires (sec)	This is the expiration period for registration. IPX-S300B must retry registration within this period.
Maximum Register	Specify the number of times to retry sending the REGISTER message.

Item	Description
Retry	If registration fails even after retrying to send the registration message the specified number of times, delete the registration information for the endpoint, then try again after some time.
Register Retry Interval (sec)	Enter the interval for resending the REGISTER message.
Keep Alive	It is used to verify the connection using SIP OPTION message.
Keep Alive Interval (sec)	Specify an interval (seconds) for exchanging Keep Alive Messages
Maximum Keep Alive Retry	Specify an maximum retry count to sending Keep Alive Messages
Keep Alive Retry Interval (sec)	Specify an retry interval to sending Keep Alive Messages
Keep Alive User Info	When sending OPTION message, User Info will be used depends on this option.
Black List Expires (sec)	Specify the expiration time in second from 0 to 86400. In case of no response when it send ‘Blacklist Check Message’ to ISP, it blocking the IP address of the ISP during this expiration time.
Backlist Check Message	Select a type of messages. REGISTER, OPTIONS, and INVITE messages are used. Only if there is no response, the IP address will be added to black list
DNS SRV Query	Select whether to use of DNS SRV.
DNS SRV Version	Specify the version of DNS SRV.
Failover	When ISP server has no static IP address but domain name or has a secondary proxy server, IPX-S300B provides failover service between IP lists
Failover Type	Select the interworking ISP. The specific failover flow will be applied to the trunk.
Failover Domain Name	Specify failover domain name that ISP provide.
Failover response	Specify what kind of response message will be used to failover behavior. For example, if you input 503, When IPX-S300B get response 503 message from the ISP upon request, The failover is start.
Failover Timeout (sec)	When IPX-S300B send message to ISP server, there is no response after this ‘Failover Timeout’ value, IPX-S300B send message again to the available IP address within the list of ISP Server. So, It is valid when the IP address of ISP server is more than one.
Retry Pause Time (sec)	When it fails both sent and retry of REGISTER or OPTIONS messages, it wait as long as this retry pause time and try again.
Recovery Method	When there is no response to attempts to call, select the recovery methods. - Registration: Re-register to the server that is responded. - Blacklist: IP list of no responding server * It works when Failover is enabled.

Item	Description
Register Timeout (sec)	Specify the timeout value for expiration of REGISTER message.
Register in Expire (%)	Specify the point of re-registration after registration. It specifies the percentage of ‘Register Timeout’ value.

<Route-Additional SIP>

Item	Description
Protocol Type	Select UDP, TCP, or TLS as the protocol to use.
URI TYPE	Select SIPS if the protocol is TLS. Select SIP in other cases.
TLS Connection	Specify the TLS connection type: - Normal: use TLS Full handshaking method. - Reuse: reuses the existing TLS connection established by initial Message. - Resume: use simplified handshaking method using TLS Session ID.
TLS Connection Reconnection	If you want to reconnect TLS connection which is configured reuse, set TLS Connection Reconnection to Enable. Then, TLS connection is refreshed once a day.
NAT Traversal	If enabled, it provides Media Proxy function to this ROUTE.
Call Authentication	Specify whether to allow user authentication for INVITE.
SIP P-Asserted-ID Type	Select a type of representative number. - Primary: The P-Asserted-Identity header contains the Primary number and the From header contains residential number. - Secondary: The P-Asserted-Identity header contains the residential number and the From header contains Primary number.
Use Request URI User Info	Specify whether to use the SIP Request URI when selecting the Route for outgoing call.
Contact Header Format	Select a type of Contact header in SIP message. - IP Address - Domain Name
PRACK Support	Specify whether to support PRACK message.
Keep To-Tag in Response	If a target is changed by call forward or call transfer, To-Tag can be kept with this option.
Multi 183 Message Block	When IPX-S300B receives multiple 183 messages, the messages are blocked except the first-received one.
Reliable 18x Response	Select whether to send reliable 18x response.
MOH SIP Media Mode	Specify the media mode (Send/Receive, Send Only, and Inactive) when make a call hold to trunk side.
Refer Relay	Specify whether to relay a REFER message to this trunk when a user transfer the call.
302 Response	Specify whether to deliver 302 response messages.

<Route-Trunk Service>

Item	Description
Trunk Access Code	Select access code which is used for a outbound call from call log at the phone.
Default Access Code for Tandem	Select whether to use default access code or not for tandem call.
Available Route	Select whether to use this route or not.
Route Lock	You can specify whether to use a route for external calls coming through the endpoint connected to the route. To restrict the use of a route, set - None: All incoming and outgoing calls through the route are allowed. - Outbound Locked: Outgoing calls are restricted and only incoming calls through the route are allowed. - Inbound Locked: Incoming calls are restricted and only outgoing calls through the route are allowed. - All Locked: All incoming and outgoing calls through the route are restricted.
Trunk Restriction Policy	Select to use restriction policy within tandem call. This option is applied in inbound case.
Call Forward Block	Specify whether to allow or block the calls being forwarded to this route.
Anonymous Call Reject	Specify whether to reject an incoming call with below: - Anonymous: if there is anonymous string in calling URI, reject the call. - No Number: if there is no number in calling URI, reject the call - Both: reject the call which has either anonymous string or no number in calling URI.
Class of Service	“Class of Service” can be applied to Route for Call Restriction. If any “Class of Service” is not assigned, a default “Class of Service” of User Group is applied to Route.
Maximum Call	Set the maximum number of simultaneous call inbound and outbound direction.
Maximum Inbound Call	Set the maximum number of simultaneous call with inbound direction.
Maximum Outbound Call	Set the maximum number of simultaneous call with outbound direction.
Allow Reroute Reason Code	Specify the Reason Code that it will be used to reroute the call when the reason code was received.
Outbound Error Announcement	Select the use of announcement when outbound call has failed.
Inbound Error Announcement	Select the use of announcement when incoming call has failed.
Call Forward Announcement Iteration	Specify the iteration count of announcement that it was played when a call is being forwarded.

Item	Description
Caller Ring Type	Specify the ring type for inbound call to this route.
Use Virtual Ringback	Enable this feature to ignore 'External Ringback Tone' of a called user when a call is received from the trunk. - Enable: Makes to connect the build-in ringback tone. - Disable: Connects the external ringback tone to the trunk
Use Real Ringback	Enable this feature to ignore the 180 message received after the first 183 message but process the 183 message when a call has been made from the trunk. - Disable/Enable
Virtual Ringback for Outbound Call	Enable this feature to ignore real ringback tone. SCM Compact coverts the 183 message from trunk to the 180 message to send virtual ringback tone.
Privacy Header Value	Set up the Privacy Header so that the information within the office can be protected from the outside. - Available values: header, session, user, none, critical, id, history (Default value: id;critical.) Caution: If there are the value 'none' and another value at the same time, only 'none' is added to the header. In other words, for example, in the case of 'id;none,' only 'none' is added to the header.
Ignore Auto Answer	Enable this feature to ignore the auto answer when a truck call is received to a subscriber for which auto answer has been set up.

<Route-Interworking>

Item	Description
DTS Mode	Specify the trunk option to use DTS Service
Send Paging On Answer Info	Select whether to send Paging On Answer Info or not. It is used for between IPX-S300B.
Gateway Channel Display	Specify channel information in the REGISTER message from gateway.
Outbound Proxy Server	If Trunk use dual network by indirectly using SBC, specify the SBC's IP address. It used for specific carrier purpose.
Outbound Proxy Server Type	If SBC is used with Name List, select the option as Name List.
Statistic Mode	Specify the statistic mode for specific carrier.
Local Domain	You can separately specify the SCM's domain. Normally, when a message is sent to a trunk, domain or IP address of the route is used as the SCM's domain value. However, there are cases where a separate domain is requested when linking to certain equipment. This feature is used in such cases. So, this feature is not used when the equipment you are linking has not made a request for a separate domain.

Item	Description
Option Port Update	This option concerns, if the port where the SCM has received messages differs from the port set up in the route when OPTION messages are transmitted both ways, whether to make the SCM send messages to the port set up for the trunk route. - Disable/Enable
Transfer INVITE Direction	Specify the target to which no-SDP INVITE will be sent when an extension line connects to the trunk because of a transfer.

<Route-Number Translation>

Item	Description
Forced Send CLI Number	Select the CLI Number for the trunk calls instead of according to the system's internal priority order. If None is selected, the CLI number is decided by the internal priority order.
Outbound CLI Prefix	If there is no configuration of 'Send CLI Number' in a user and there is only extension number, when the user make a call through this route and there is prefix, add this prefix to the extension number and send it as calling number.
Send CLI Name for User	When make a call to outbound, select the caller Name below: - User Name: send 'User Name' as caller name. - Send CLI Name: Send 'Send CLI Name' as caller name.
Send CLI Name for Inbound Call	If there is no caller name for inbound call, use below options. - None: not used. - Receive CLI Number: use caller number as caller name.
Send CLI Name for Internal Call	Specify a display name for subscriber. Tandem call does not support this option.
CLI for Forward Call	Specify the type of caller ID for the call being forwarded to new destination. - Originator: Original caller's number and name. - Forwarder: Forwarder's caller number and name: - Forwarder with Originator Name: Forwarder's number and Originator's name.
Transfer Caller ID	Specify the type of caller ID for the call being transferred to new destination (Transfer Target). - Held Party Number (Transferee) - Transfer Party Number (Transferor)
Tandem Blind Transfer CLI	Within Tandem call, specify the calling number. This option is activated in Blind-Transfer case.
Outbound Diversion Number	Select whether to use CLI Number or CLI Name in Diversion header when a user makes an outbound call. It used for individual billing purpose.
Tandem Diversion	Specify a number for Diversion header when a incoming call direct to

Item	Description
Number	outbound.
Use Anonymous Call Diversion Header	Select whether to use anonymous to Diversion header.
Anonymous URI	Specify a URI type when sending an anonymous call.
Inbound DID Delete Length	Specifies the length of digits to delete from the first position of the DID number for inbound call.
Inbound CLI Delete Length	Specifies the length of digits to delete from the first position of the calling number for inbound call.
Inbound DID Insert Digits	Specifies the digits to insert from the first position of the DID number for inbound call.
Inbound CLI Insert Digits	Specifies the digits to insert from the first position of the calling number for inbound call.
Modify E.164 Format	Specify whether to use E.164 format for calling number or called number for outgoing call through this route.
Inbound MCN	Specify multiple rules to modify an inbound calling number. MCN is processed before simple delete/insert number modification. Only one rule can be applied to each call.

2.5.2 Making LCR

A Least Cost Route (LCR) is a method of selecting a route when processing outgoing trunk calls. There are three different types of LCR, as shown below.

Priority Routing

A priority routing allows automatic selection of alternative routes when the default outgoing path becomes unavailable. Priority is assigned to the direct route and alternative routes. When calls going out through the high-priority route fail, they can be retried through the low-priority routes.

You can create route sequences in the **[CONFIGURATION > Trunk Routing > Priority Routing]** menu. This menu is used for creating route sequences.

Item	Description
User Group	Specify a user group to which the route sequence belongs.
Priority Routing Name	Specify a name for the route sequence. Pay special attention to choosing the name, as it is used as an identifier when selecting the route sequence in other menus and cannot be changed.
Route Priority	Assign priority to the route. - Direct Route: Specify the top priority route. - Alternative Route1 to Alternative Route8: Select the routes according to their priority levels.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Route Name	Select a route for the route priority level.

When creating a Priority Routing Table, you can choose to enter or not to enter the following information as occasion demands.

Item	Description
Outbound DOD Delete Length	Specifies the length of digits to delete from the first position of the called number for outbound call.
Outbound DOD Insert Digits	Specifies the digits to insert from the first position of the called number for outbound call.
Outbound CLI Delete Length	Specifies the length of digits to delete from the first position of the calling number for outbound call.
Outbound CLI Insert Digits	Specifies the digits to insert from the first position of the calling number for outbound call.
Outbound MCN	Specify MCN rules for outbound call.

2.5.3 Making Location Based Routing

A location based routing allows each location to use its own LCR. Since each location is set with its own LCR, if you have created multiple locations, you must set an LCR for each of the locations. If no LCR is set for a particular location, the location is not allowed to make trunk calls.

You can create route partitions in the **[CONFIGURATION > Routing > Location Based Routing]** menu. This menu is used for creating Location Based Routing.

Item	Description
User Group	Select a user group to which the Location Based Routing belongs.
Location Based Routing Name	Enter a name for the Location Based Routing. Pay special attention to choosing the name, as it is used as an identifier when selecting the route partition in other menus and cannot be changed.
Location select	Select whether to use the 'Location'. If Location selects use 'disable', all location can use this Location Based Routing.
Location	Select a location to set the routes.
Routing Type	Select a type of LCR for the location. - Time-Based Routing: Select to use a Time-Based Routing. - Priority Routing: Select to use a Priority Routing. - Load Balance Routing: Select to use a Load Balance Routing.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

When you input data into a Location Based Routing, some of following items are selected depending on the type of the LCR selected.

Item	Description
Time-Based Routing	If the LCR type is set to Time-Based Routing, select a Time-Based Routing to use.
Priority Routing	If the LCR type is set to Priority Routing, select a Priority Routing to use.
Load Balance Routing	If the LCR type is set to Load Balance Routing, select a Load Balance Routing to use.

Time-based Routing

A time-based routing sequence contains time conditions so that each service group can use different route sequences according to the conditions.

Load-balanced Routing

A load-balanced routing allows use of the selected routes in a specified ratio.
Calls are distributed between the routes identified as available for calls, and therefore there is no need for configuring alternative routes.

Among the three types of LCRs, the route sequence type is used by default. For more information on LCRs, including the setup method and route selection, see the 'Least Cost Route (LCR) Policy' section of '4.1 System Features'.

2.5.4 Configuring Access Codes

Access codes are number to be used for dialing a directly outgoing call. They are also used for analyzing the destination numbers to determine which location-based routes to be used for outgoing trunk calls.

You can create access codes using the **[CONFIGURATION > Routing > Access Code]** menu. The following items are mandatory.

Item	Description
User Group	Select a user group to which the access code belongs.
Access Code	Enter an access code to use when making calls to trunks instead of extension numbers. Pay special attention to choosing the code, as it is used as an identifier when selecting the access code in other menus and cannot be changed.
Number Type	Select a type of access code. The access code can be the beginning portion of the external destination number, or an internal code is used within the boundary of the SIP servers or gateways. - Normal: When the calling number for an outgoing trunk call is analyzed, the digit corresponding to the access code is deleted from

Item	Description
	the number dialed by the user, and then the call is made to the trunk. - Internal: When the calling number for an outgoing trunk call is analyzed, the digit corresponding to the access code is not deleted from the number dialed by the user and the call is made to the trunk as is. - Emergency: When the calling number for an outgoing trunk call is analyzed, only the digit corresponding to the access code from the number dialed by the user is used as the destination number, and then the call is made to the trunk. - Pattern: When analyzing the calling numbers for outgoing trunk calls, a wild card (expressed as X) is used to denote the length. The call is made to the trunk without deleting the digit corresponding to the access code from the number dialed by the user. - DTS: When the calling number for an outgoing trunk call is analyzed, the digit corresponding to the access code is not deleted from the number dialed by the users and the call is made to the trunk as is. DTS Access code can use DTS Trunk, other Trunk is not allowed.
Location Based Routing Name	Select a Location Based Routing to use with this access code.

When creating an access code, you can enter the following items optionally as needed.

Item	Description
Minimum Digit Length	This is the minimum length of the digit used for analyzing the number dialed by the user.
Maximum Digit Length	This is the maximum length of the digit used for analyzing the number dialed by the user.

2.5.5 Configuring DID Routing

Destination of incoming trunk calls depend on the DID number. When entering a DID number, you can use wild cards (entered by *) to enter multiple numbers at a time. If the called number is set to 'B', a translated DID number is used as the called number.

You can also specify different called numbers for different times of the day. Time periods are defined by ring plans.

For more information on DID number translation and assigning called numbers by ring plans, see the 'DID Routing' section of '4.1 System Features'. Also, for more information on ring plans, see the 'Ring Plan' section of '4.1 System Features'.

You can assign routes by DID number using the **[CONFIGURATION > Routing > DID Routing]** menu. The following items are mandatory.

Item	Description
User Group	Select a user group to which the calls are directed.
DID Number	Enter a DID number for incoming trunk calls.
Default Destination	Specify a called number to which the incoming calls with the selected DID number are directed. The default called number is used if the current ring plan is not RP1 through RP15.

When configuring DID routing, you can enter the following items optionally as needed.

Item	Description
DID Name	Enter this if you want to have a DID name displayed on the called party's phone.
Delete Length	Enter the length of the digit deleted from the called number.
Insert Digit	Enter the digit to insert into the called number.
MOH ID	Select a MOH sound source to play when the Music on Hold service is performed for incoming trunk calls with the selected DID number.
ACD Queuing Level	Specify a level of priority for this DID pattern queues in ACD.
Ring Plan Schedule	Specify a Ring Plan Schedule which is used for the DID.
RP1-RP15 Destination	Enter a called number to which the incoming calls with the selected DID is directed when the current ring plan is RingPlan1 to RingPlan15.
RP1-RP15 Maximum Call	Specify a maximum number of allowed terminating calls for each ring plan. As many as a specified number of calls are received, and the excess calls are rejected.

2.6 Step of Call Processing

When new call arrives (receive INVITE), IPX-S300B handles the call with below procedures.

Distinction of internal or external call

1) When IPX-S300B receives new INVITE, it checks whether this call is coming from internal or external IPX-S300B gets the IP address of caller and compare it with 'Proxy Server' or 'Domain Name' (if Proxy Server or Domain name is configured with IP address) in **[CONFIGURATION > Trunk Routing > Route]** menu.
   If it matches then IPX-S300B process it as external call.
2) IPX-S300B gets the domain and user info of P-Asserted-ID header in INVITE message and compare these with 'Domain Name' and 'User Name' in **[CONFIGURATION > Trunk Routing > Route]** menu.
   If it matches then IPX-S300B process it as external call.
3) IPX-S300B gets the caller's number and compare it with extension numbers in **[CONFIGURATION > User > Single Phone User]** menu.
   If it matches then IPX-S300B process it as internal call.
4) If a call is not matched one of above three cases, IPX-S300B decide it as unauthorized access and declines the call.

Processing of Internal Call from a subscriber

When the IPX-S300B receive a call from a subscriber (internal call), it handles the call as follows.

1) It rejects the call in case that calling restriction service was set for the originating subscriber.
2) It connects a call to the dialed number.

Processing of External Call from a trunk

When the IPX-S300B receive a call from a trunk (external call), it handles the call as follows.

1) First, check the IP address and/or domain name of caller. If there is no matched route within Route List, it rejects the call.
2) If there is a matched Route and there is number translation configuration, it performs translation of caller/called number. After this process, the translated called number (DID) and calling number (CLI) will be used for call processing.
3) Search the caller number with the lists of 'CLI Number' at the **[CONFIGURATION > Trunk Routing > CLI Routing]** menu, If there is matched number, process it as follows:
   - If 'Call Reject' option is enabled in the matched CLI number, IPX-S300B reject the call.
   - If there is a value in 'Name Translation' field in the matched CLI number, the caller name will be replaced with it.

- If there is a number in 'RPx Destination' fields in the matched CLI number, the call will be destined to the number regarding on each Ring Plan.
- If there is a number in 'Default Destination' field, the call will be routed to the default destination number.

4) Search the called number with the lists of 'DID Number' at the **[CONFIGURATION > Trunk Routing > DID Routing]** menu, If there is matched number, process it as follows:
   - If there is a number in 'RPx Destination' fields in the matched DID number, the call will be destined to the number regarding on each Ring Plan.
   - When a maximum call number has been specified for each ring plan, calls received after the specified call number is reached are rejected.
   - If there is a number in 'Default Destination' field, the call will be routed to the default destination number.
   - If the destination value is 'B', the prefix with 'Delete Length' of DID number will be delete and 'Insert Digit' value will be added instead. The call will be route with the translated number.
   - If the destination value is 'E', the prefix with 'Delete Length' of DID number will be delete and 'Insert Digit' value will be added instead. The call will be route with the translated number. The difference with 'B' is that this call cannot be routed to trunk.
   - The configurable types of destination number in DID routing table are User's Extension Number, User Group Code with User's Extension Number, Hunt Group Number, ACD Group Number, VM/AA Access Number, Route Access Code with External number and UMS access number.

Incoming call processing

Following steps explans how to calls are routed to the terminating number determined in the internal call processing and external call processing described above.

1) Incoming to extension
   - It searches the called number with the lists of extensions. If there is a matched extension number, it route the call to the extension.
   - It searches the called number with the lists of combination of User Group Code and User Group's Extension Number. If there is a matched combined number, it route the call to the number.
   - It rejects a call when Do Not Disturb service was activated at the extension.
2) If calling number is matched with restriction policy, it rejects the call.
3) Sending to trunk
   - It searches the called number with the lists of Access Codes, if there is a matched access code, it route the call to external through the Route.
4) Decision of CLI when a call sends to trunk:
   - For the calling number for originating calls, use 'Send CLI Number' in the **[CONFIGURATION > User > Normal User]** menu.

- If Send CLI Number has not been set up, use CLI Number' in the **[CONFIGURATION > User Group > Service Group]** menu as the calling number.
- If Service Group's CLI Number has not been set up, use CLI Number' in the **[CONFIGURATION > User Group > Change User Group > Information]** menu.
- If User Group's CLI Number has not been set up, you can add the **[CONFIGURATION > Trunk Routing > Route]** menu's 'Outbound CLI Prefix' in front of the extension number and use it as the calling number.
- Use the extension number as the calling number if there is no matching rule mentioned above

5) CLI/DID Number conversion to Trunk side:
   - If there is pattern with calling number type in Outbound MCN and the Calling number of a call is matched with one of calling number patterns, the Calling number of the call will be translated with the matched pattern.
     The converted calling number will be deleted with the length of **[Outbound CID Delete Length]** of **[CONFIGURATION > Trunk Routing > Priority Routing]** or **[CONFIGURATION > Trunk Routing > Load Balance Routing]**
   - And insert the 'Outbound CLI Insert Digits' and send it as CLI.
     If there is pattern with called number type in Outbound MCN and the Called number of a call is matched with one of called number patterns, the Called number of the call will be translated with the matched pattern.
     The converted called number will be deleted with the length of 'Outbound DOD Delete Length' of **[CONFIGURATION > Trunk Routing > Priority Routing]** or **[CONFIGURATION > Trunk Routing > Load Balance Routing]**
     And insert the 'Outbound DOD Insert Digits' and send it.
6) An outgoing call may be dropped if called number was matched with restriction policy. The call will be rejected with reason of call restriction.
7) If there is no matched route for a call, add default access code if exist as prefix to the called number and process the steps above 3), 4), and 5).
8) The call that route was not specified will be dropped with reason of invalid number.

CHAPTER 3. Cabinet/Slot Configuration

This chapter describes how to configure system configuration information necessary for installing and running the IPX-S300B on a cabinet and slot basis.

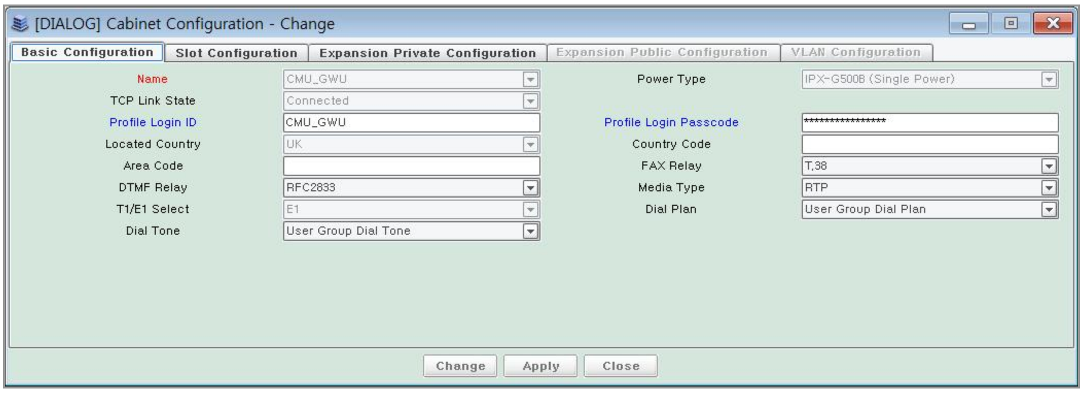
3.1 Cabinet/Slot Configuration

Configure cabinet information. The key configuration items are as follows:

- Basic Configuration
- Slot Configuration
- Expansion Slot Configuration

3.1.1 Basic Configuration

In Basic Configuration, enter the IPX-S300B's basic cabinet configuration information.



<Read Only Fields>
These fields show the cabinet's basic information and cannot be modified.

Item	Description
Name	Shows the name that will be set by default.
Power Type	Shows the power type. - IPX-G500B (Single Power)

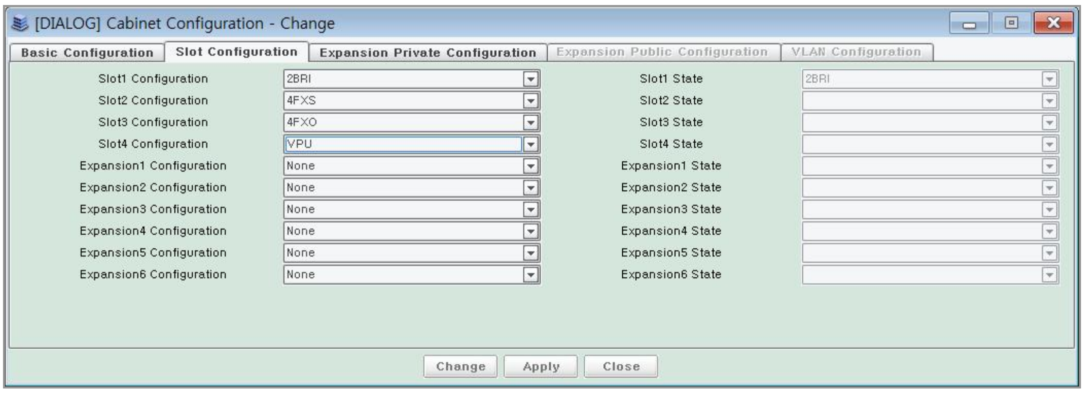
Item	Description
	- IPX-G500BP (Dual Power)
TCP Link State	Shows the status of the TCP connection between the CMU and the GWU.
Located Country	Enter the system's country information.
T1/E1 Select	Shows the PRI type. (T1 or E1.)

<Others>

Item	Description
Profile Login ID	The ID used to request the provision of profile information. This field must be filled in. (Default: CMU_GWU.)
Profile Login Passcode	The passcode to request the provision of profile information. This field must be filed in.
Country Code.	Enter the country code.
Area Code	Enter the area code of your area.
Fax Relay	Specify the Fax relay type. (T.38 or Pass Through.)
DTMF Relay	Specify the DTMF relay type. (Outband, RFC2833, or In Voice.)
Media Type	Specify the media type. RTP, sRTP (AES/ARIA128), sRTP (ARIA128/AES), sRTP (AES/ARIA192), sRTP (ARIA192/AES), sRTP (AES), sRTP (ARIA128), or sRTP (ARIA192).
Dial Plan	Shows the dial plan used for the option card. (Default: User Group Dial Plan).
Dial Tone	Shows the dial tone used for the option card. (Default: User Group Dial Tone.)

3.1.2 Slot Configuration

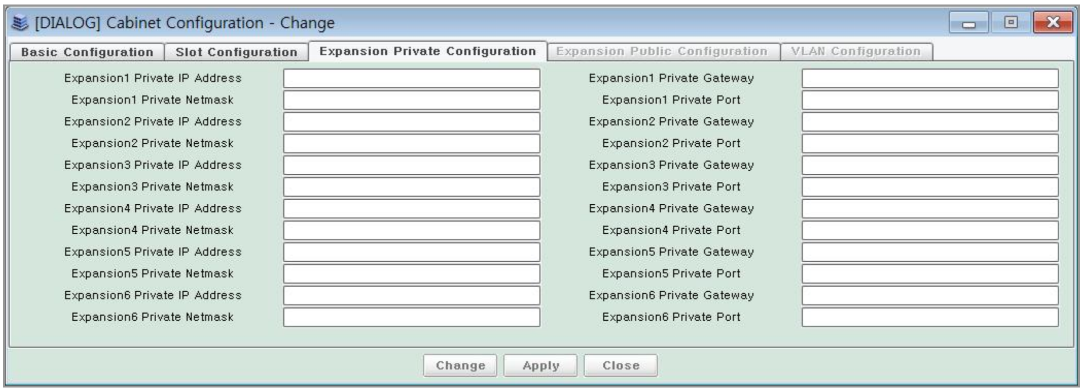
In Slot Configuration, enter slot configuration information that matches the cabinet's actual configuration.



Item	Description
Slot1-Slot4 Configuration	1) Enter information for slot 1, slot 2, slot 3, and slot 4. (4FXO, 4FXS, 2BRI, 2PRI, 1PRI, or VPU.) 2) The slot information you enter must match the type of the card actually inserted into the slot. If the type of the slot card is set to none, when a card is inserted into the slot, the slot is automatically set to the type that matches the card's type.
Slot1-Slot4 State (Read only fields)	Shows the types of cards that are actually installed in the cabinet.
Expansion1-6 Configuration	1) Enter the type of each ECU, from ECH 1 to 6. => IPX-G520S (20FXS) or IPX-G540S (40FXS). 2) The ECU types you enter must match the types of the ECUs that are actually connected to the cabinet. If the type of the expansion rack is set to none, when an ECU is connected to the cabinet, the expansion rack is automatically set to the type that matches the ECU's type.
ECU1-6 State (Read only fields)	Shows the types of the ECU that are actually connected to the cabinet.

3.1.3 Expansion Private Configuration

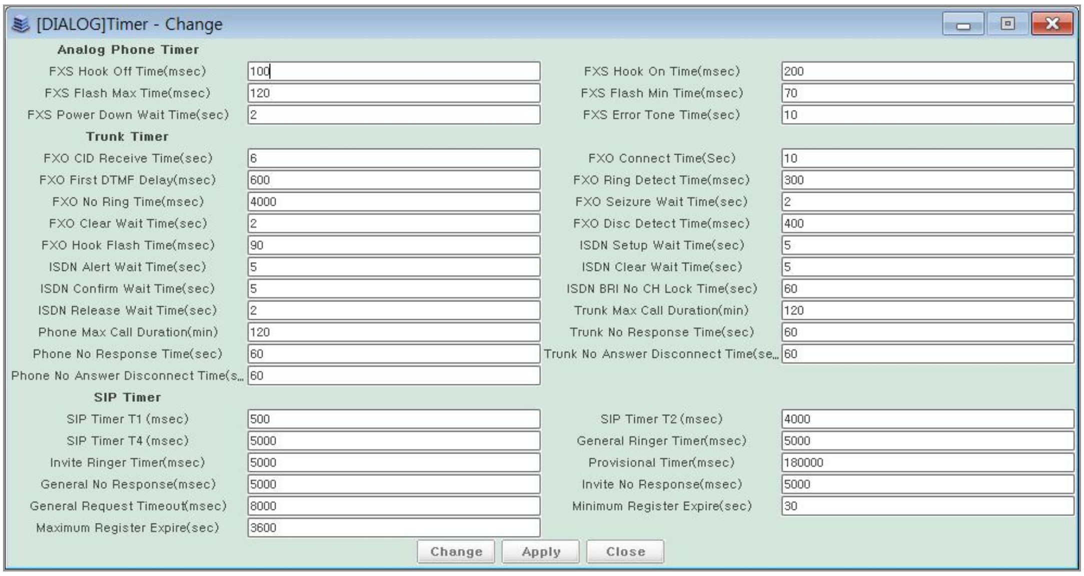
In Expansion Private Configuration, enter private IP information for each ECU.



Item	Description
Expansion1-6 Private IP Address	Enter the private IP address of each ECU, from ECU1 to 6.
Expansion1-6 Private IP Address Gateway	Enter the private gateway of each ECU, from ECU1 to 6.
Expansion1-6 Private Netmask	Enter the private netmask of each ECU, from ECU1 to 6.
Expansion1-6 Private Port	Enter the private port of each ECU, from ECU1 to 6.

3.2 Timer

In Timer, you configure information about timers used for the provision of analog phones, analog trunks, PRI trunks, and BRI trunks.



3.2.1 Analog Phone (FXS)

Item	Description
FXS Hook Off Time (ms)	The time based on which whether the analog phone is in off-hook status is determined.
FXS Hook On Time (ms)	The time based on which whether the analog phone is in on-hook status is determined.
FXS Flash Max Time (ms)	The maximum time for determining whether the analog phone is in hook flash status.
FXS Flash Min Time (ms)	The minimum time for determining whether the analog phone is in hook flash status.
FXS Power Down Wait Time (sec)	The length of time for which Loop Open (Power Down) will be sustained when the FXS Loop Open is used.
FXS Error Tone Time (sec)	The maximum length of time for which the error tone will play.

3.2.2 Trunk (FXO, ISDN)

Item	Description
FXO CID Receive Time (sec)	The timeout time for CID receipt after a 1st ring signal is detected from the FXO port.
FXO Connect Time (sec)	When making a FXO trunk call without PRS, the point in time at which the trunk connect time starts being calculated for

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Item	Description
	billing purposes.
FXO First DTMF Delay (ms)	The time delayed until the first DTMF is transmitted when dialing though an analog trunk.
FXO Ring Detect Time (ms)	The minimum length of time for which the ring signal that comes in the FXO port is sustained for determination of the ring signal's validity.
FXO No Ring Time (ms)	The point in time at which the ring signal that has been coming in the FXO port is judged not to be coming in any longer.
FXO Seizure Wait Time (sec)	When making an FXO trunk call, the wait time until the receipt of a response after a request for seizure is made to the port.
FXO Clear Wait Time (sec)	When terminating an FXO trunk call, the wait time that it takes to get back to idle status after a request for release is made to the port.
FXO Disc Detect Time (ms)	When the feeding voltage of the CO line connected to the FXO port has been cut off, the minimum length of time for deciding on Disconnect.
FXO Hook Flash Time (ms)	The time that it takes to transmit a hookflash signal to the analog trunk.
ISDN Setup Wait Time (sec)	The wait time after sending a Setup message.
ISDN Alert Wait Time (sec)	The wait time after sending an Alerting message.
ISDN Clear Wait Time (sec)	The wait time after sending a Disconnect message.
ISDN Confirm Wait Time (sec)	The wait time until receiving the next message after sending a Connect and Release message.
ISDN BRI No CH Lock Time (sec)	Not used.
ISDN Release Wait Time (sec)	The wait time until becoming available again after terminating a call.
Trunk Max Call Duration (min)	The length of time after which a trunk call is terminated.
Phone Max Call Duration (min)	The length of time after which an inter-extension call is terminated.
Trunk No Response Time (sec)	The time after which the call is terminated if a response to a request for making a trunk call has not been received from the IPX-S300B in Normal mode before the time has passed.
Phone No Response Time (sec)	The time after which the call is terminated if a response to a request for making an inter-extension call has not been received from the IPX-S300B in Normal mode before the time has passed.
Trunk No Answer Disconnect Time (sec)	The time after which the call is terminated if a response has not been made from the other end before the time has passed, even though the phone at the other end has been ringing after a trunk call was made.

Item	Description
Phone No Answer Disconnect Time (sec)	The time after which the call is terminated if a response has not been made from the other end before the time has passed, even though the phone at the other end has been ringing after an inter-extension call was made.

3.2.3 SIP Timer

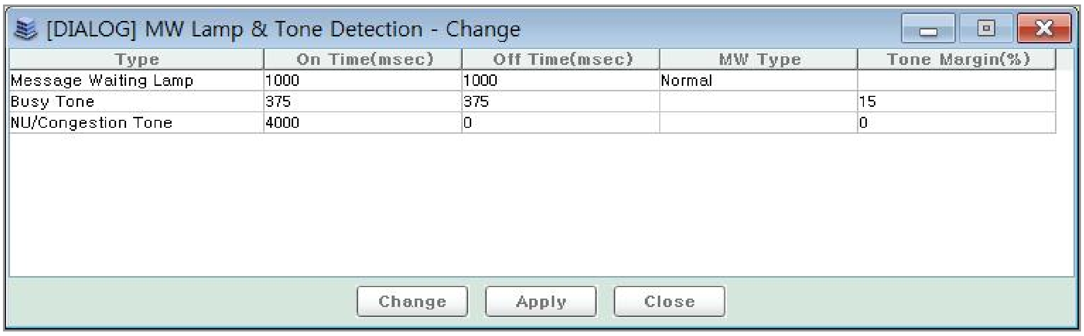
Item	Description
SIP Timer T1 (ms)	SIP T1 Timer
SIP Timer T2 (ms)	SIP T2 Timer
SIP Timer T4 (ms)	SIP T4 Timer
General Ringer Timer (ms)	General Ringer Timer
Invite Ringer Timer (ms)	Invite Ringer Timer
Provisional Timer (ms)	Provisional Timer
General No Response (ms)	General No Response Time
Invite No Response (ms)	Invite No Response Time
General Request Timeout (ms)	General Request Timeout Time
Minimum Register Expire (sec)	The minimum length of time for the expiration of the SIP phone's registration.
Maximum Register Expire (sec)	The maximum length of time for the expiration of the SIP phone's registration.

3.3 Ring Cadence

Ring Type	1st On (msec)	1st Off (msec)	2nd On (msec)	2nd Off (msec)	3rd On (msec)	3rd Off (msec)
Gateway1 Ring	400	200	400	3000	0	0
Gateway2 Ring	1000	3000	0	0	0	0
Gateway3 Ring	400	100	400	2000	0	0
Gateway4 Ring	200	200	200	2000	0	0
Gateway5 Ring	200	200	200	4000	0	0
Gateway6 Ring	400	300	400	1000	0	0

Item	Description
Ring Type	The list of rings used for the option card.
1st on (ms)	The length of time for which the first ringing continues.
1st off (ms)	The duration of silence after the first ringing stops.
2nd on (ms)	The length of time for which the second ringing continues.
2nd off (ms)	The duration of silence after the second ringing stops.
3rd on (ms)	The length of time for which the third ringing continues.
3rd off (ms)	The duration of silence after the third ringing stops.

3.4 MW Lamp & Tone Detection-Change



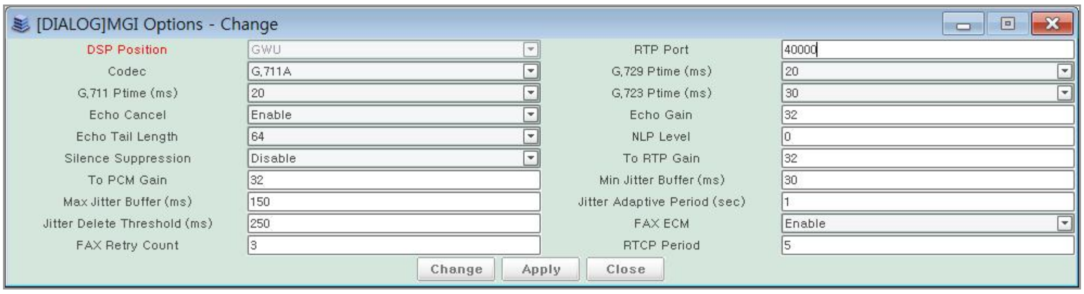
Item	Description
On time (ms)	A period of time during which the message waiting lamp is on/a period of time during which a tone to detect is on
Off time (ms)	A period of time during which the message waiting lamp is off/a period of time during which a tone to detect is off
Message waiting type	Normal (normal mode using a ring signal)/PRS (mode using polarity reversal)
Tone Margin (%)	Additional margin value for on/off time which is set when Busy Tone or NU/Congestion Tone is detected

3.5 [SIP Option Configuration]

Item	Description
UDP Port	Port number to be used in signaling UDP used in a GWU module
IPX-S300B Register Expire (sec)	Registration expiration time to be used in registering a trunk or FXS separately
IPX-S300B Representative Register Expire (sec)	Registration expiration time to be used in registering a key number
Use Session Timer	Enables/disables Session Timer
Session Expire Time (sec)	Expiration period to be applied when Session Timer is used
Unauthorized SIP ACL Status	Enable/disenables ACL blocking function for unauthorized SIP messages
SIP Common Msg Block Timer (sec)	Time to be applied when being blocked by unauthorized SIP message (excluding REGISTER)
SIP REGISTER Block Time (sec)	Time to be applied when being blocked by unauthorized REGISTER message
SIP REGISTER Retry Limit	Allowed number of REGISTER messages

3.6 [MGI Option Configuration]

MGI (Media Gateway Interface) converts PCM voice into IP voice packets, and IP voice packets into PCM voice data. In MGI options, options related to such conversion can be set. In this case, the communications for IP voice packets use RTP (Real-Time Transport Protocol).



Item	Description
RTP port	Start port number for RTP to transmit
Codec	Voice codec information for RTP to transmit (G.729, G.729a, G.711a, G.711u, G.723)
G.729 P-time (ms)	Transmission interval for G.729/G.729a RTP packets
G.711 P-time (ms)	Transmission interval for G.711a/G.711u RTP packets
G.723 P-time (ms)	Transmission interval for G.723 RTP packets
Echo Cancel	Can remove packet delays, and echoes generated by voice reflection on PSTN, during a call.
Echo Gain	Can set the call sensitivity for echo cancel. (18-38 (-14-+6 dB))
Echo Tail Length	Can set the tail length used during echo cancel (8-128)
NLP Level	Can specify the NLP (Non-Linear Processor) control value, and is valid when echo cancel is enabled (0/1/2). 0. Normal level of NLP engagement (Default) 1. NLP Tune Option 2 (Reduced level of NLP engagement) 2. NLP Tune Option 1 (Increased level of NLP engagement) NLP Tune Option 2 (Reduced level of NLP engagement): can choose NLP Tune Option 2 in order to decrease the NLP level, when beeping disconnecting sound is generated in a start section of voice due to double talk phenomenon. NLP Tune Option 1 (Increased level of NLP engagement): when 2-to-4 wire hybrid/line conditions are generated, can choose NLP Tune Option 2 in order to increase NLP Level
Silence Suppression	Can set a function not to transmit voice packets by sensing mute sounds during a call.

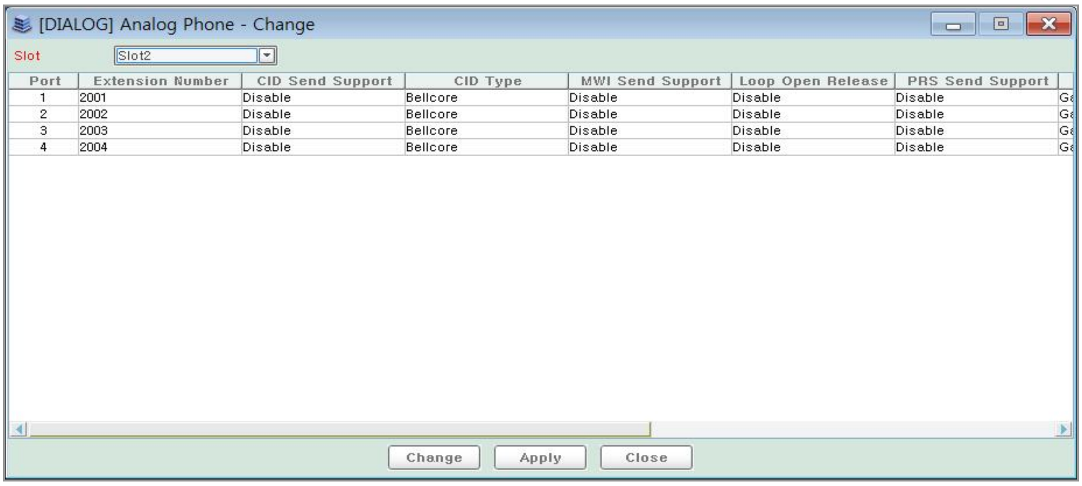
Item	Description
To RTP Gain	Can set the call sensitivity for PCM voice inputted by a DSP. (18~38 (-14~+6 dB))
To PCM Gain	Can set the call sensitivity when voice packets are converted into PCM voice. (18~38 (-14~+6 dB))
Min Jitter Buffer (ms)	Selects minimum Jitter Delay value (0~150 ms).
Max Jitter Buffer (ms)	Selects maximum Jitter Delay value (30~200 ms).
Jitter Adaptive Period (sec)	Can select a Jitter Adaptive Period (1~10 s).
Jitter Delete Threshold (ms)	Can select a Jitter Delete Threshold (150~500 ms).
Fax ECM	Sets an additional recovery feature for fax transmission errors when a T.38 fax is used.
FAX Retry Count	Specifies the number of retry attempts when errors occur using a FAX (0~3).
RTCP Period	Selects an interval time value for RTCP period (2~10).

3.7 Analog Phone Configuration

Generates information on an analog phone to be connected to FXS port in the **[CONFIGURATION > User > Single Phone User]** menu.
The phone type should be Analog-FXS-Phone.

3.7.1 Analog Phone Number Specification

Specifies generated information on a subscriber at a port by slot in Environment configuration-Cabinet slot-Analog phone configuration.
If FXS is used as manual or automatic in cabinet configuration, FXS extension number is assigned automatically when available FXS phone numbers exist.

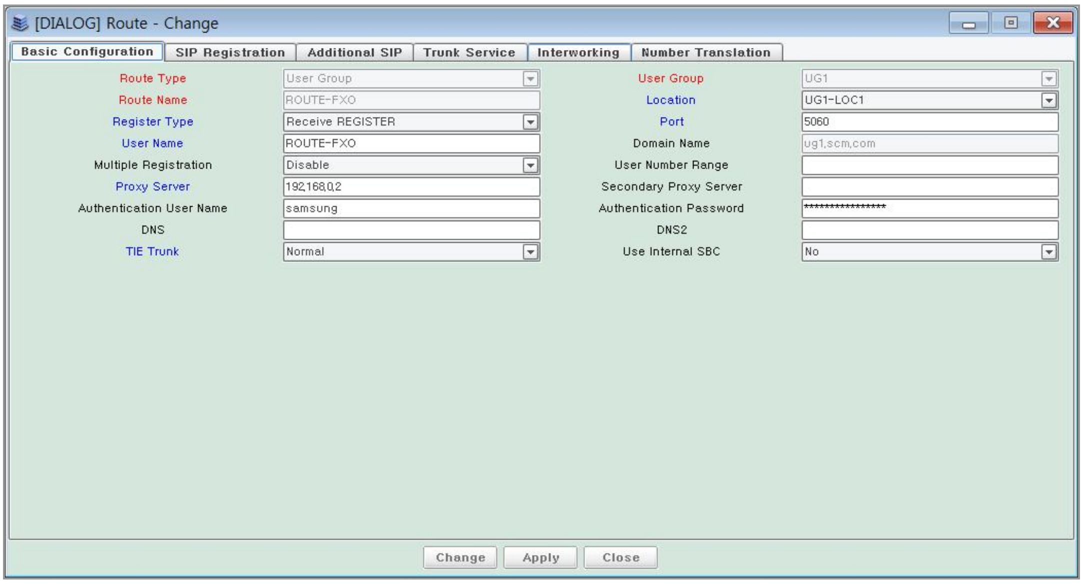


Item	Description
Extension number	Extension number to set in a given port
CID Send Support	Sets whether CID will be transmitted to a target analog phone.
CID Type	Sets either Bellcore or ITU-T.
MWI Send Support	Sets whether to use Message Waiting Indication.
Loop Open Release	Sets whether to perform loop open to a given port when a call with the other side terminates.
PRS Send Support	Sets whether to generate PRS signals when the other party responds during a outgoing call.
Ring type	Sets which to use among rings 1-6 in **[Ring configuration]**
Tx Gain	Sets the gain for the voice coming out from a given analog phone (from -4 dB to +4 dB).
Rx Gain	Sets the gain for the voice coming into a given analog phone (from -4 dB to +4 dB).
Feature Tone detection	Sets whether to play out Feature Dial Tone when Call Forward All), DND or Extension Lock features are set for a given analog phone.

3.8 FXO/PRI/BRI Trunk Configuration

From the menu of **[CONFIGURATION > Trunk Routing > Route]** Sets the routing information for the FXO/PRI/BRI trunk connected to the IPX-S300B system.

- Set registration mode to a 'Receive REGISTER'.
- Set proxy server to the IP address of GWU. The IP address of GWU means 'GWU IP address' in the menu **[CONFIGURATION > Network > Network Configuration]**.
- Set authentication user as 'Disable' in SIP additional configuration tab.
- Enter a calling number to generate trunk calling code.
- Authentication user ID and password are can be added arbitrarily.
- When setting for the first time via Configuration Wizard, trunk routes called 'ROUTE-FXO', 'ROUTE-PRI' and 'ROUTE-BRI' are automatically generated.
- Trunk routes other than those generated in Default can be generated as follows.



3.8.1 Analog Trunk Route Configuration

Specifies the name of a generated FXO trunk route to a port by slot in Environment configuration-Cabinet/Slot-Analog trunk route configuration.

When Slot card type is None in Cabinet configuration-Slot configuration, and when FXO card is set, 'ROUTE-FXO' is specified as trunk route name for a given port if 'ROUTE-FXO' is among trunk routes.
Specifies 'ROUTE-FXO' to trunk route name.

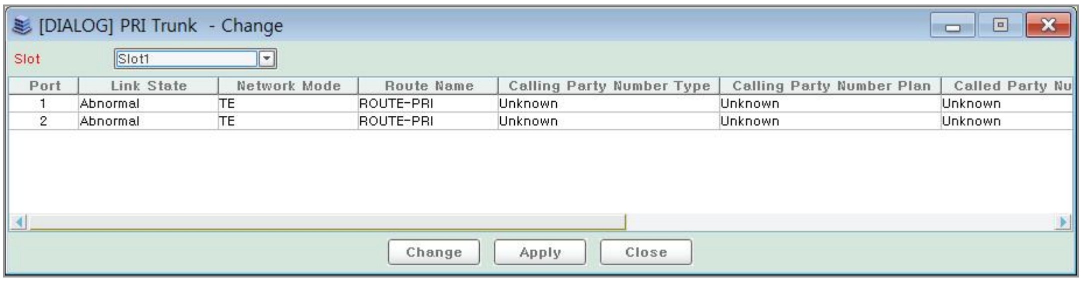
For an incoming number ring, attendant service extension group numbers generated during the execution of Configuration Wizard are automatically assigned.

Item	Description
Route name	Sets the trunk route to which a given analog trunk belongs.
Link status	Shows the current connection status.
CID Receive Support	Sets whether to receive CID coming from an analog trunk.
Ring Destination number	Sets a number at which a ring sounds when incoming through a given analog trunk.
PRS Receive Support	Disable (not receive PRS)/Answer-Disconnect (When once received, respond to a call from the other party; when twice received, terminate a call from the other party)/Disconnect (When once received, terminate a call from the other party) If not set to Disable, an analog trunk used will be released automatically.
Tone detection feature	Sets whether to detect the tone received at an analog trunk Disable (do not detects tone)/Busy Tone (detects Busy Tone)/NU/Congestion Tone (detects NU/Congestion Tone)/Both (detects both Busy Tone and NU/Congestion Tone). When tone is detected, an analog trunk being used is released automatically.
Loop length	Sets the distance to a telephone station
Outgoing sensitivity	Sets the gain of voice going out into an analog trunk (from -4 dB to +4 dB)
Incoming sensitivity	Sets the gain of voice coming into an analog trunk (from -4 dB to +4 dB)

3.8.2 PRI Trunk Configuration

From the menu [CONFIGURATION > Cabinet/Slot > BRI Trunk/Station], Specifies the trunk route name of a generated PRI to a port by slot and specifies network mode of it.

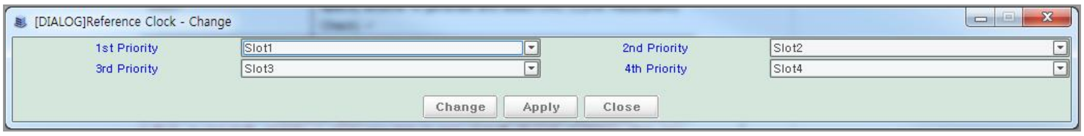
When the slot card type is 'none' on the Slot Configuration Tab in the **[CONFIGURATION > Cabinet/Slot > Cabinet Configuration]** menu, and when PRI card is mounted to the slot, 'ROUTE-PRI' is automatically specified as trunk route name for a given port if 'ROUTE-PRI' is among trunk routes.



Item	Description
Link status	Shows the current connection status.
Network Mode	Sets network mode to connect to PRI (TE: Terminal/NT: Network)
Route Name	Sets a trunk route to which a PRI trunk belongs.
Calling Party Number Type	Sets a calling party number type - Unknown - International - National - Network - Subscriber - Abbreviate - Extension
Calling Party Number Plan	Sets a calling party number plan - Unknown - ISDN - Data - Telex - Nation - Private - Extension
Incoming number type	Sets an incoming number type - Unknown - International - National - Network

Item	Description
	- Subscriber - Abbreviate - Extension
Incoming number plan	Sets an incoming number plan - Unknown - ISDN - Nation
CRC	Specify whether to generate and detect CRC (Cyclic Redundancy Check)
Outgoing number display/hiding	Sets whether to display or hide an outgoing number.

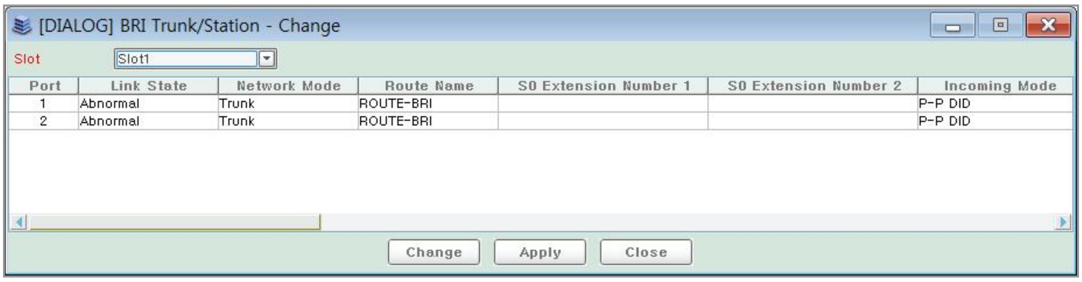
In the **[CONFIGURATION > Cabinet/Slot > Reference Clock]** menu, the slot to be used as the system reference clock can be prioritized. When a PRI card and a BRI card are mounted simultaneously, the PRI card is given priority.



Item	Description
1st Priority	#1 priority slot
2nd Priority	#2 priority slot
3rd Priority	#3 priority slot
4th Priority	#4 priority slot

3.8.3 BRI Trunk/Extension Configuration

Specifies the name of a generated BRI trunk to a port by BRI slot and specifies network mode according to whether it is a trunk or extension in Environment configuration-Cabinet/Slot-BRI trunk configuration For an extension, enters an extension number.



Item	Description
Link status	Shows the current connection status.
Network Mode	Sets network mode to connect to a given BRI port (Trunk: connects to trunk/Station: connects to BRI Station)
Route Name	Sets a route name to which a BRI trunk belongs
S0 extension number 1	First S0 extension number
S0 extension number 2	Second S0 extension number
Incoming mode	Sets incoming mode. - Point to Point Normal - Point to Point DID - Point to Multi Point
Normal ring incoming number	After setting incoming mode to Point to Point Normal, specifies a number at which a ring sounds when a call is received to a given BRI trunk.
Calling Party Number Type	Sets a calling party number type - Unknown - International - National - Network - Subscriber - Abbreviate - Extension
Calling Party Number Plan	Sets a calling party number plan - Unknown - ISDN - Data - Telex - Nation - Private - Extension

Item	Description
Incoming number type	Sets an incoming number type - Unknown - International - National - Network - Subscriber - Abbreviate - Extension
Incoming number plan	Sets an incoming number plan - Unknown - ISDN - Nation
Outgoing number display/hiding	Sets whether to display or hide an outgoing number.
MSN number 1-8	Sets up to eight MSN numbers.

In Configuration-Cabinet/Slot-Reference Clock Setting, the slot to be used as the system reference clock can be prioritized. When a PRI card and a BRI card are mounted simultaneously, the PRI card is given priority.

Item	Description
1st Priority	#1 priority slot
2nd Priority	#2 priority slot
3rd Priority	#3 priority slot
4th Priority	#4 priority slot

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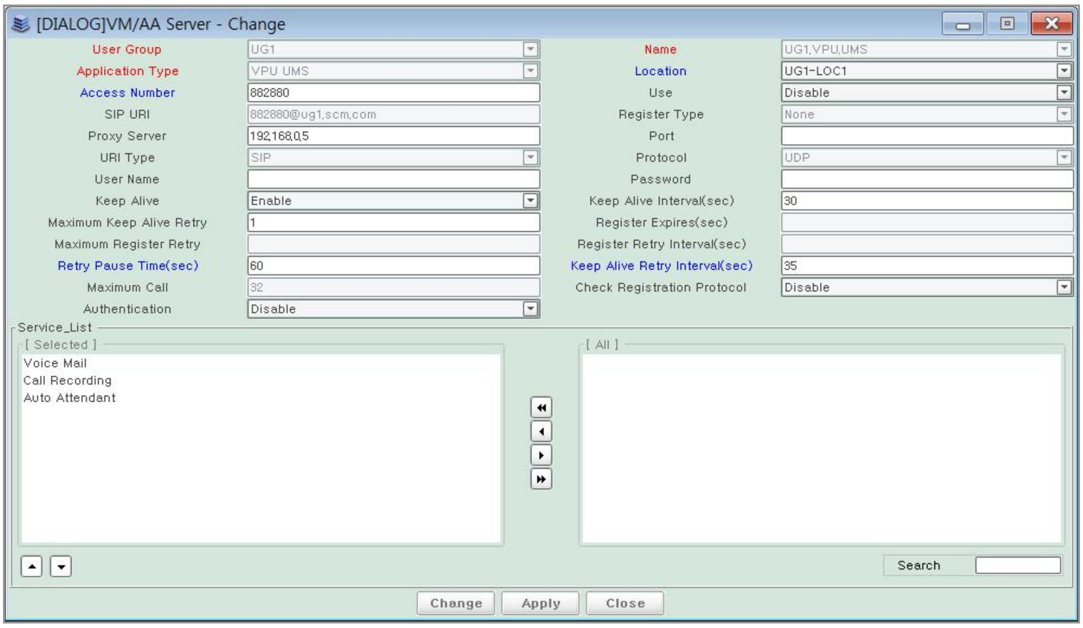
3.8.4 VPU Configuration

When Slot card type is None in Cabinet configuration-Slot configuration, and when VPU card is set, slot card type is automatically specified as VPU, and the proxy server IPs of VPU.UMS in Environment configuration-Application-Voice mail/Automatic attendant service server, and of VPU. Conference in Environment configuration-Application-Video conference, are specified as IPs assigned to a VPU slot.



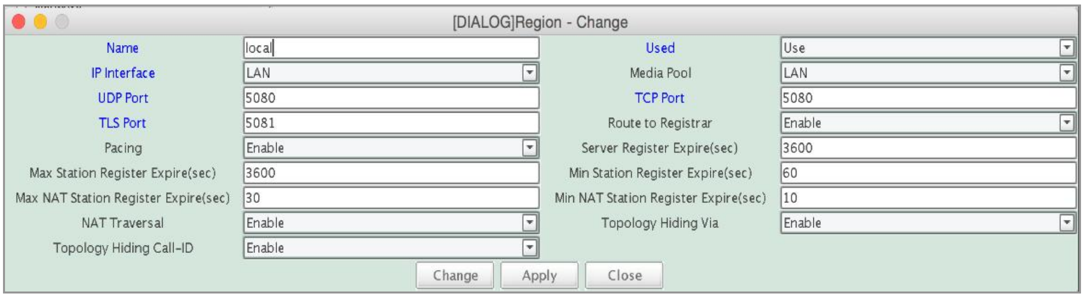
3.9 SBC Enabling

3.9.1 Media Pool configuration

Set Port Pool configuration in order to send RTP for LAN and WAN interface.
Selects the Media pool in the menu **[CONFIGURATION > Network > SBC > Media Pool]**.

Item	Description
Name	Name of Media Pool. Generally, spool for WAN and LAN interface are called WAN and LAN, respectively.
Used	Sets whether to use or not.
IP Address	IP address of WAN or LAN interface
Minimum RTP Port	Start value of RTP port range
Maximum RTP Port	End value of RTP port range

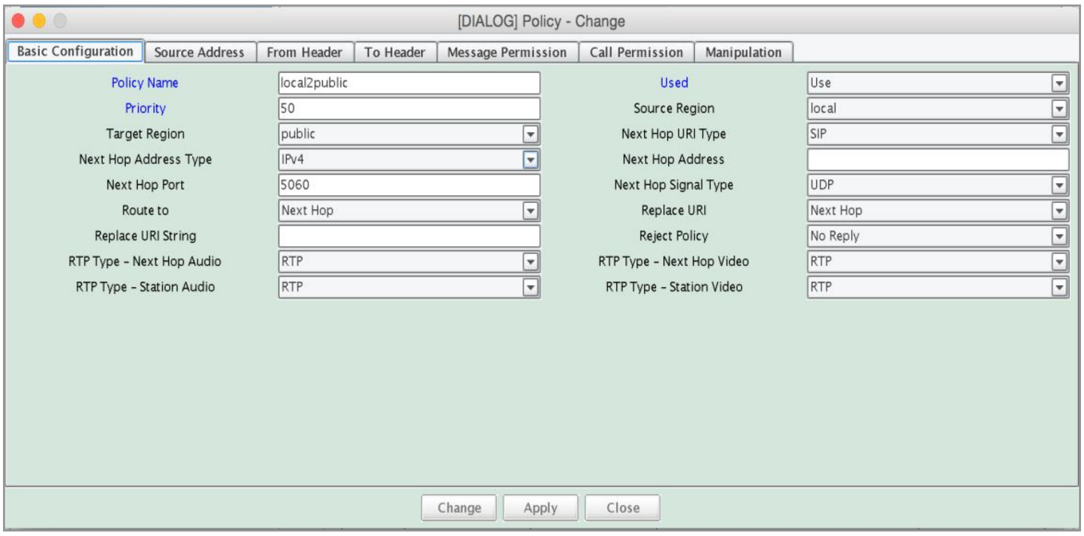
3.9.2 Region Configuration



Item	Description
Name	Name of a region to generate
Used	Sets whether to use or not.
IP Interface	Interface configuration to use in a region (WAN/LAN)
Media Pool	Media Pool configuration to use in a region.
UDP Port	UDP port number for SIP signaling to listen to in a region to generate.
TCP Port	TCP port number for SIP signaling to listen to in a region to generate.
TLS Port	TLS port number for SIP signaling to listen to in a region to generate.
Router to Registrar	Sets whether to record subscriber information by REGISTER passing

Item	Description
	through SBC.
Pacing	Whether to use a feature to adjust the rate of REGISTER messages transmitted from terminals.
Server Registrar Expire (sec)	Expire time to use when SBC sends REGISTER messages
Max Station Registrar Expire (sec)	Allowed maximum expire time for register messages received from an object. If larger than a predetermined value, respond with a maximum value.
Min Station Registrar Expire (sec)	Allowed minimum expire time for register messages received from an object. If smaller than a predetermined value, respond with 423 Interval too brief.
Max NAT Station Registrar Expire (sec)	Allowed maximum expire time for register messages received from an object below NAT. If larger than a predetermined value, respond with a maximum value.
Min NAT Station Registrar Expire (sec)	Allowed maximum expire time for register messages received from an object below NAT. If smaller than a predetermined value, respond with 423 Interval too brief.
NAT Traversal	NAT recognition mode configuration 1) none: Not processed as NAT (all objects are assumed to be in public) 2) rport: Processed as NAT when ‘rport’ parameter is included in SIP messaged 3) always: Processed as NAT when the actual outgoing address of messages is not identical with the IP of a header.
Topology Hiding Via	Sets whether to delete previous Via headers when sending messages.
Topology Hiding Call-Id	Sets whether to newly generate Call-ID objects when sending messages.

3.9.3 Policy Configuration



Item	Description
Policy Name	Policy name
Used	Sets whether to use or not.
Priority	Sets the priority of policy. Finds policy with higher priorities in the first place; and when the priorities are equal, they are determined arbitrarily. Predetermined values can be 0-99, with smaller ones having higher priorities.
Source Region	Sets the source-region for policy When messages are received, and if they are not incoming from a given region, given policy will not be selected.
Target Region	Sets the target region for policy. Calls and messages routed by given policy will be transmitted to a given region.
Next Hop URI Type	Sets URL type for a destination for policy.
Next Hop Address Type	Sets address type for a destination for policy.
Next Hop Address	Sets address for address type for a destination for policy. Calls and messages routed by given policy will be transmitted to a given address. However, this is the case only when route-to property is Next-Hop. In other cases, they are routed according to each type.
Next Hop Port	Sets s port for a destination for policy.
Next Hop Signal Type	Sets signal type for a destination for policy.
Route to	Sets routing mode. - Next Hop: Routed to the address set by Next Hop. - To Header: Routed to the host name of the To header of messages. - Request URI: Routed to the host address of the Requested-URI of a start line.
Replace a URI	Sets the modification rules of Request-URI values on SIP messages. - None: Not changed.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
	- Next Hop: Changed to Next Hop value. - All: Changed Replace to URI String.
Replace URI String	Value used when Replace a URI All is performed.
Reject Policy	Sets response policy for messages not conforming to the permission of policy. If set to be No-reply, error messages will not be responded.

CHAPTER 4. Configuring Phone and Gateway

This chapter describes how to configure phones and gateways connected to IPX-S300B.

4.1 Configuring Phone

4.1.1 Phone Installation

4.1.1.1 Installation Type

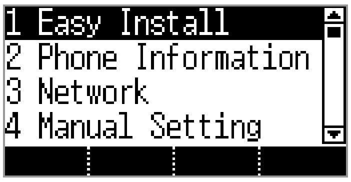
How to register a phone in the system is as follows.

Configuration Type	Description
Standard	User can configure all the information to register with system manually. This mode is including following steps. - SIP server setting - SIP authentication setting - NTP server setting
Server	All the information to register with system is downloaded from a Configuration server. - If the system using MAC Address authentication type, ID/Password is not mandatory. - Please contact the system administrator detail information about the phone authentication type.
PnP (Plug & Play)	This feature allows a phone to be registered with the system when powered on so that it becomes available for service. - To use the 'PnP' mode, the PnP environment must be configured by the system administrator. - If you choose the 'PnP' mode, the network mode of the phone is changed to DHCP and the network setting step is skipped. - Please contact the system administrator about detail information about the 'PnP' mode.

4.1.1.2 Easy Installation of SMT-i3100/3105/5210/5210S/5220/5230

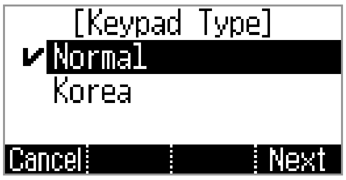
1) To get to the SETUP MODE unplug the power cord from the phone. Press and hold the **[*]** button while you plug power back into the phone. Release the **[*]** when you see Samsung in the display.
   When the phone reboot is complete, the Language Menu will display. Select the language to use and press the **[Next]** soft button to advance to the Configuration Menu.
   - The system administrator can change the language of the phone after registered with the system.

2) Select the **[1 Easy Install]** Menu.



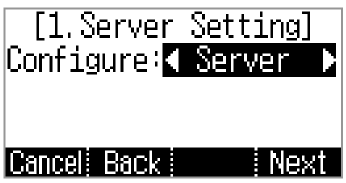
3) Select the Keypad Type.
   - In case of use Korean language, select Korea, in other case select Normal.

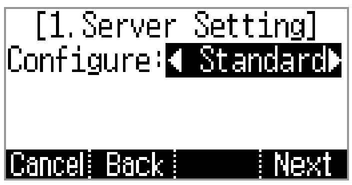
   Press the **[Next]** Button.



4) Select the configuration mode. The steps of the easy install menu might be changed depending on type of the configuration mode.

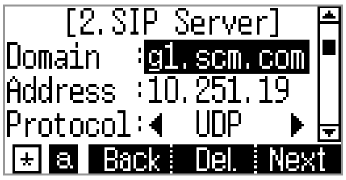
   Press the **[Next]** Button.

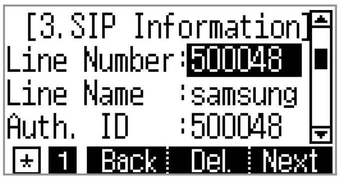


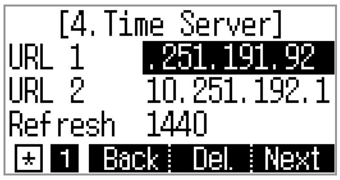


5) If the **[Standard]** configuration mode is selected, The following steps are added.
   - SIP Server: Enter the SIP server information.
   - SIP Register Information: Enter the sip register information.
   - Time Server: Enter the time server (NTP) URL and update interval.

   Press the **[Next]** Button.

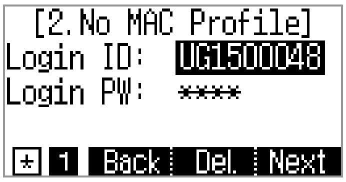






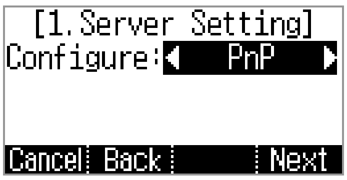
6) In case the **[Server]** configuration mode is selected, You can enter ID/Password.
   - If the system using MAC Address authentication type, ID/Password is not mandatory.
   - Please contact the system administrator detail information about the configuration server address and authentication type.

   Press the **[Next]** Button.



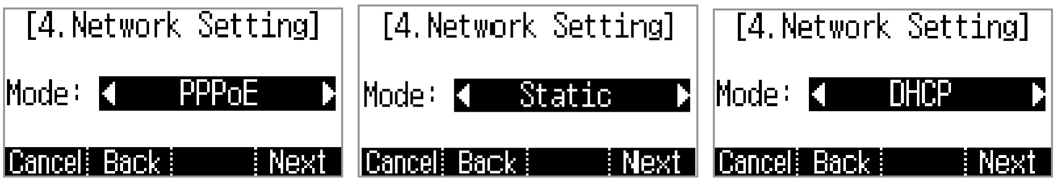
7) In case the **[PnP]** configuration mode is selected, There is nothing to do in this step.
   - Please contact the system administrator about detail information about the PnP mode.

   Press the **[Next]** Button.

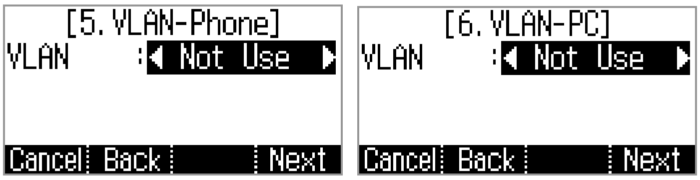


8) Select the network mode and enter detail network information.
   - If you choose the PnP configuration mode, the Network mode of the phone is changed to DHCP and the Network Setting step is skipped in the easy Install feature.
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.

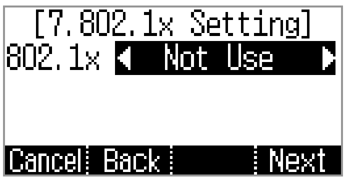
   Press the **[Next]** Button.



9) Enter VLAN information.
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.



10) Enter 802.1x information.
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.



11) LLDP Setting

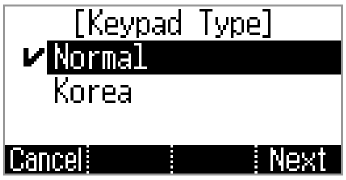
12) All steps of the easy install is ended, press the **[Yes]** button to finish the easy install. Then the phone will be restart.
   - You can change the information already entered. Press the **[Back]** button till the step you want, and edit the information

4.1.1.3 Easy Installation of SMT-i6010/6011

1) In case the phone is not registered with the system, **[Easy]** button will be displayed. Press **[Easy]** button, then the easy install menu will be shown.
   - The system administrator can change the language of the phone after registered with the system.

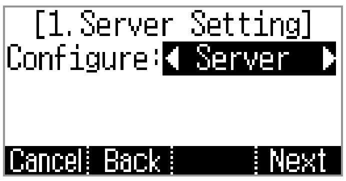
   Select the Keypad Type.
   - In case of use Korean language, select Korea, in other case select Normal.

   Press the **[Next]** Button.



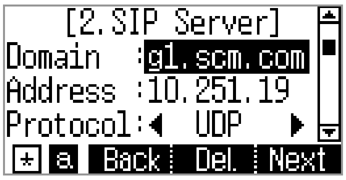
2) Select the configuration mode. The steps of the easy install menu might be changed depending on type of the configuration mode.

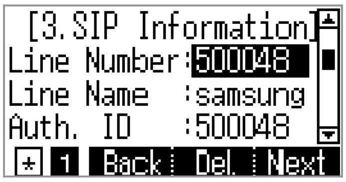
   Press the **[Next]** Button.

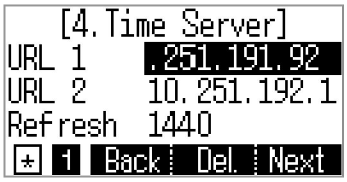


3) If the **[Standard]** configuration mode is selected, The following steps are added.
   - SIP Server: Enter the SIP server information.
   - SIP Register Information: Enter the sip register information.
   - Time Server: Enter the time server (NTP) URL and update interval.

Press the **[Next]** Button.

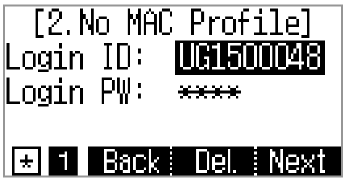






4) In case the **[Server]** configuration mode is selected, You can enter ID/Password.
   - If the system using MAC Address authentication type, ID/Password is not mandatory.
   - Please contact the system administrator detail information about the configuration server address and authentication type.

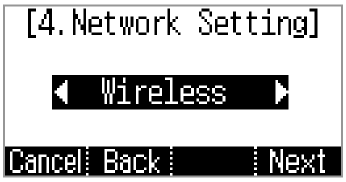
   Press the **[Next]** Button.

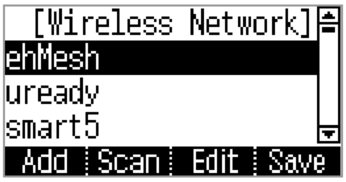


5) In case the **[PnP]** configuration mode is selected, there is nothing to do in this step. Press the **[Next]** Button.
   - Please contact the system administrator about detail information about the PnP mode.

6) Enter network information.
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.
   - If the **[Wired]** network is selected, you can choose **DHCP/Static/** or **PPPoE** mode.

7) In case the **[Wireless]** network is selected,
   * Wireless mode is not supported for 6010.
   - The phone will start to search AP automatically.
   - Wi-Fi is only running under AC Adaptor connection.
   - After select AP in the searched list, connect to Static or DHCP mode.



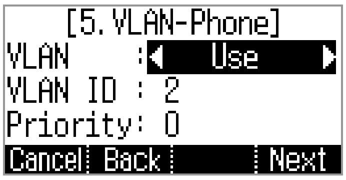


NOTE

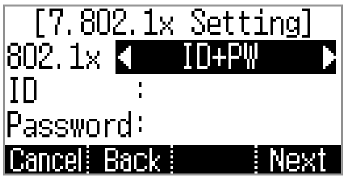
SMT-i6010 's WiFi availability

WiFi is not supported by SMT-i6010.

8) Enter VLAN information.
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.



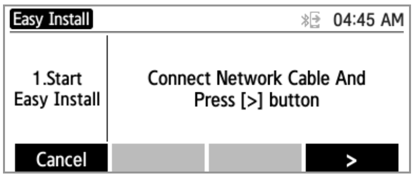
9) Enter 802.1x information
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.



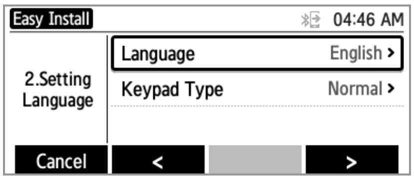
10) After Setting LLDP service, press the **[Finish]** button to finish the easy install. Then the phone will be restart.
   - You can change the information already entered. Press the **[Previous]** button till the step you want, and edit the information.

4.1.1.4 Easy Installation of SMT-i6020/6021

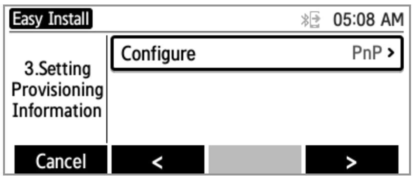
1) In case the phone is not registered with the system, **[Easy Install]** button will be displayed. Press **[Easy Install]** button, then the easy install menu will be shown. Check a network cable and press **[Next]** button.

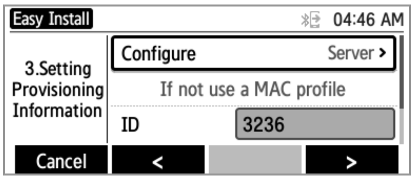


2) Select the language to use.
   - The system administrator can change the language of the phone after registered with the system.

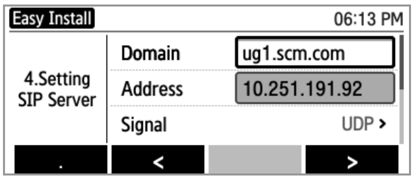


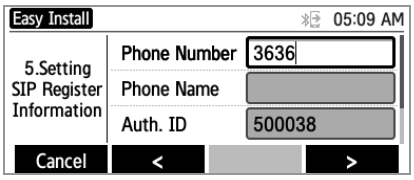
3) Select the configuration mode. The steps of the easy install menu might be changed depending on type of the configuration mode.



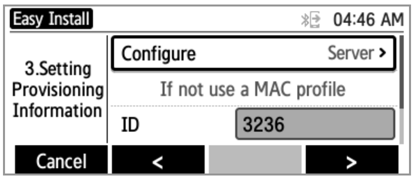


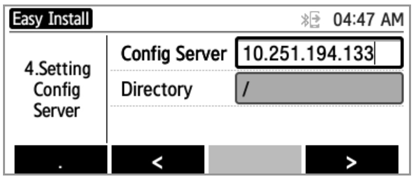
4) If the **[Standard]** configuration mode is selected, The following steps are added.
   - SIP Server: Enter the SIP server information.
   - SIP Register Information: Enter the sip register information.
   - Time Server: Enter the time server (NTP) URL and update interval.





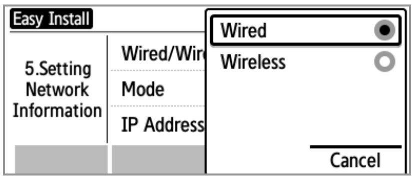
5) In case the **[Server]** configuration mode is selected, You can enter ID/Password.
   - If the system using MAC Address authentication type, ID/Password is not mandatory.
   - Please contact the system administrator detail information about the configuration server address and authentication type.



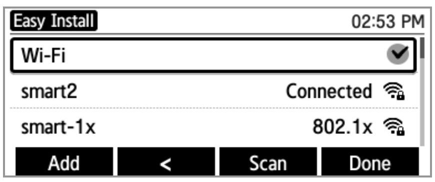


6) In case the **[PnP]** configuration mode is selected, there is nothing to do in this step. Press the **[Next]** Button.
   - Please contact the system administrator about detail information on the PnP mode.

7) Enter network information.
   - If the **[Wired]** network is selected, you can choose **DHCP/Static/** or **PPPoE** mode.
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.



8) In case the **[Wireless]** network is selected,
   * Wireless mode is not supported for 6020
   - The phone will start to search AP automatically.
   - Wi-Fi is only running under AC Adaptor connection.
   - After select AP in the searched list, connect to Static or DHCP mode.

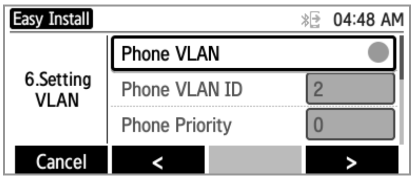


NOTE

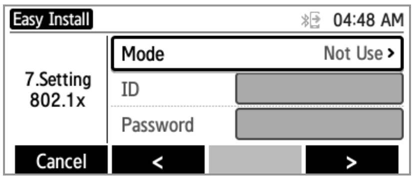
SMT-i6020 's WiFi availability

WiFi is not supported by SMT-i6020.

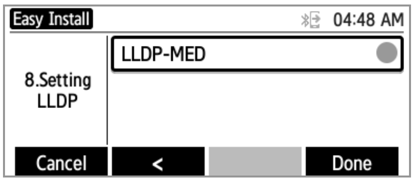
9) Enter VLAN information.
  - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.



10) Enter 802.1x information
  - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.



11) After Setting LLDP service, press the **[Finish]** button to finish the easy install. Then the phone will be restart.
  - You can change the information already entered. Press the **[Previous]** button till the step you want, and edit the information.



4.1.1.5 Easy Installation of SMT-i5243

1) In case the phone is not registered with the system, **[Easy Install]** button will be displayed. Press **[Easy Install]** button, then the easy install menu will be shown.

   Check a network cable and press **[Next]** button.

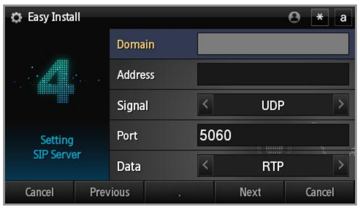
2) Select the language to use.
   - The system administrator can change the language of the phone after registered with the system.

3) Select the configuration mode. The steps of the easy install menu might be changed depending on type of the configuration mode.

   Press the **[Next]** Button.

4) If the **[Standard]** configuration mode is selected, the following steps are added.
   - SIP Server: Enter the SIP server information.
   - SIP Register Information: Enter the sip register information.
   - Time Server: Enter the time server (NTP) URL and update interval.

   Press the **[Next]** Button.



5) In case the **[Server]** configuration mode is selected, you can enter ID/Password.
- If the system using MAC Address authentication type, ID/Password is not mandatory.
- Please contact the system administrator detail information about the configuration server address and authentication type.

Press the **[Next]** Button.

6) In case the **[PnP]** configuration mode is selected, there is nothing to do in this step. Press the **[Next]** Button.
- Please contact the system administrator about detail information on the PnP mode.

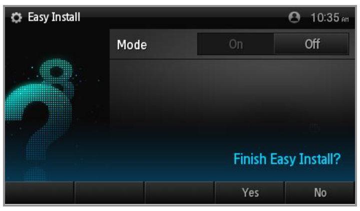
7) Select the network mode and enter detail network information.
- If you choose the PnP configuration mode, the Network mode of the phone is changed to DHCP and the Network Setting step is skipped in the easy Install feature.
- If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.

8) Enter VLAN information.
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.

9) Enter 802.1x information
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.

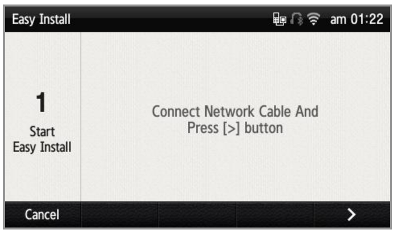
10) LLDP Setting

11) All steps of the easy install is ended, press the **[Done]** button to finish the easy install. Then the phone will be restart.
   - You can change the information already entered. Press the **[Previous]** button till the step you want, and edit the information.



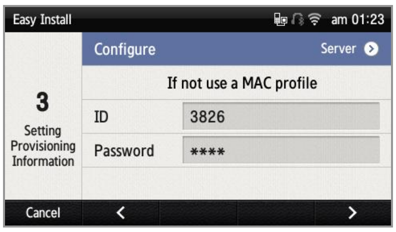
4.1.1.6 Easy Installation of SMT-i5343

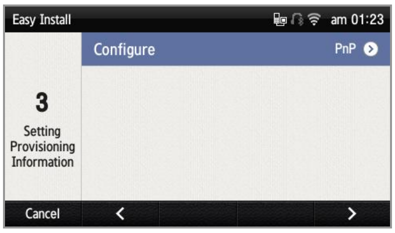
1) In case the phone is not registered with the system, **[Easy Install]** button will be displayed. Press **[Easy Install]** button, then the easy install menu will be shown. Check a network cable and press **[Next]** button.



2) Select the language to use press **[>]** button.
   - The system administrator can change the language of the phone after registered with the system.

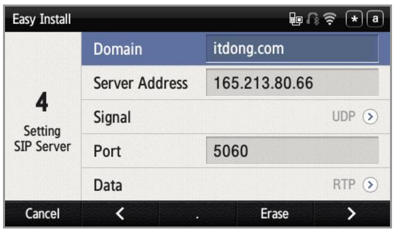
3) Select the configuration mode. The steps of the easy install menu might be changed depending on type of the configuration mode.
   Press the **[>]** Button.





4) If the **[Standard]** configuration mode is selected, The following steps are added.
   - SIP Server: Enter the SIP server information.
   - SIP Register Information: Enter the sip register information.
   - Time Server: Enter the time server (NTP) URL and update interval.

   Press the **[>]** Button.



5) In case the **[Server]** configuration mode is selected, You can enter ID/Password.
   - If the system using MAC Address authentication type, ID/Password is not mandatory.
   - Please contact the system administrator detail information about the configuration server address and authentication type.

   Press the **[>]** Button.

6) In case the **[PnP]** configuration mode is selected, there is nothing to do in this step. Press the **[>]** Button.
   - Please contact the system administrator about detail information about the PnP mode.

7) Enter network information.
   - If the **[Wired]** network is selected, you can choose **DHCP/Static/** or **PPPoE** mode.
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

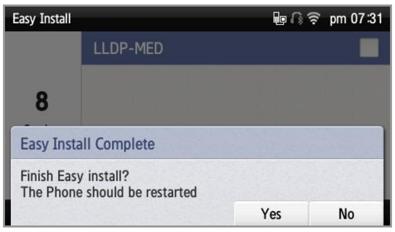
8) In case the **[Wireless]** network is selected,
   - The phone will start to search AP automatically.
   - Wi-Fi is only running under AC Adaptor connection.
   - After select AP in the searched list, connect to Static or DHCP mode.

9) Enter VLAN information.
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.

10) Enter 802.1x information
   - If you enter the wrong information, the phone may not be registered with a system or cannot connect to the network.

11) Setting LLDP servoce

12) After Setting LLDP service, press the **[Complete]** button to finish the easy install. Then the phone will be restart.
   - You can change the information already entered. Press the **[<]** button till the step you want, and edit the information.



4.1.2 Upgrading Software

This service allows the administrator to update profiles. Service profiles, user profiles and line profiles can be updated.
If necessary, phones can be upgraded. Multiple phones can be set for upgrades at a specified time.

This can be configured in the **[CONFIGURATION > Phone Setting > Upgrade Software]** menu.

Name	Description
Type	Specify an update method. The update methods are described in the Manual Phone Update section below.
User Group	Specify a user group to update.
Class of Service	Specify the service grade in which the phone you want to upgrade belongs.
Extension Number	If updating only one phone, specify the extension number of the phone to update.
Phone Name	If updating only one phone, specify the device name of the phone to update.
Model Name	Specify the model name of the phone to update.
Update Time	If updating multiple phones, specify the start option. - Now: Updated immediately. - Reserve: Updated at an appointed time.
Start Time	If updating multiple phones at an appointed time, specify the start time.
Start Device Name	If updating multiple phones, specify the device name of the starting phone.
Update Number	If updating multiple phones, specify the number of phones to update at a time. You can set up to 40.
Interval (min)	If updating multiple phones, specify the interval between updates.

4.1.2.1 Manual Phone Update

Phones can be updated manually in the following ways:

Service profile

This method sends an SIP Notify message which notifies new service settings to a phone in a user group. Although this message is sent automatically when changing feature codes, etc. you can have force it to be sent when necessary.

Line profile

This method forces the profile by extension number to be sent. Although this message is sent automatically when changing automatic call forward, etc., you can have force it to be sent when necessary.

User profile

This method forces the profile by phone name to be sent Although this message is sent automatically mainly when changing the phone button assignment of the phone, etc. you can have force it to be sent when necessary.

User profile User Group

This method updates all user profile of a specified model in a user group. When this method is used, phones of the selected model are updated starting from the start device, a specified number of phones at a time and at a specified time interval.

Phone profile

This method forcedly transmits profiles about items that are configured throughout the system. A message is automatically transmitted when there is a change to an option, but it is also possible to forcedly send a message if necessary.

Phone profile User Group

This method upgrades the phone profile of every phone in the user group.

Model update

This method updates a phones of a specified model. When this method is used, the phone of the selected model is updated.

Model update user group

This method updates all phones of a specified model in a user group. When this method is used, phones of the selected model are updated starting from the start device, a specified number of phones at a time and at a specified time interval.

Reboot

This method is used simply for restarting the phone.

Reboot User Group

This method restarts all the phones in a user group at once. This method processes 40 phones each second. If you use this method to update phones, the system may slow down due to overload. Always use the model update user group feature when updating phones.

4.1.2.2 Automatic Phone Update

When a phone is powered up for operation, it loads the phone profile and performs the following procedure:

1) Information such as the phone's main version, the VLAN ID, and the upgrade server is included in sec_boot.xml, which is first checked when the phone starts.
   The phone main version information is used for determining the need for automatic upgrade.
2) After the phone loads sec_boot.xml, it checks sec_{macaddress}.xml. If not found, the phone uses the profile ID and the password entered on the phone to load sec_user_{profildID}.xml and sec_phone.xml.
3) Profiles other than sec_phone.xml contain information on the services provided by the phone.
4) The profiles to be loaded on the phone are defined in sec_user_{profildID}.xml.
   The phone loads the specified profiles in order.
5) The phone downloads profiles in the order of sec_boot.xml > sec_phone.xml > sec_user_{profildID}.xml.

4.1.2.3 Automatic Upgrade Procedure

IPX-S300B can automatically upgrade the phone software.
Load the package on a selected profile server and reboot the phone to have the phone upgraded automatically.

The TFTP server and the upgrade server are included in IPX-S300B by default.
If necessary, you can have separate servers for these functions.

In the IPX-S300B, it is necessary to install profile servers in different places in order to reduce load and upgrade time. (Including sec_boot.xml.)

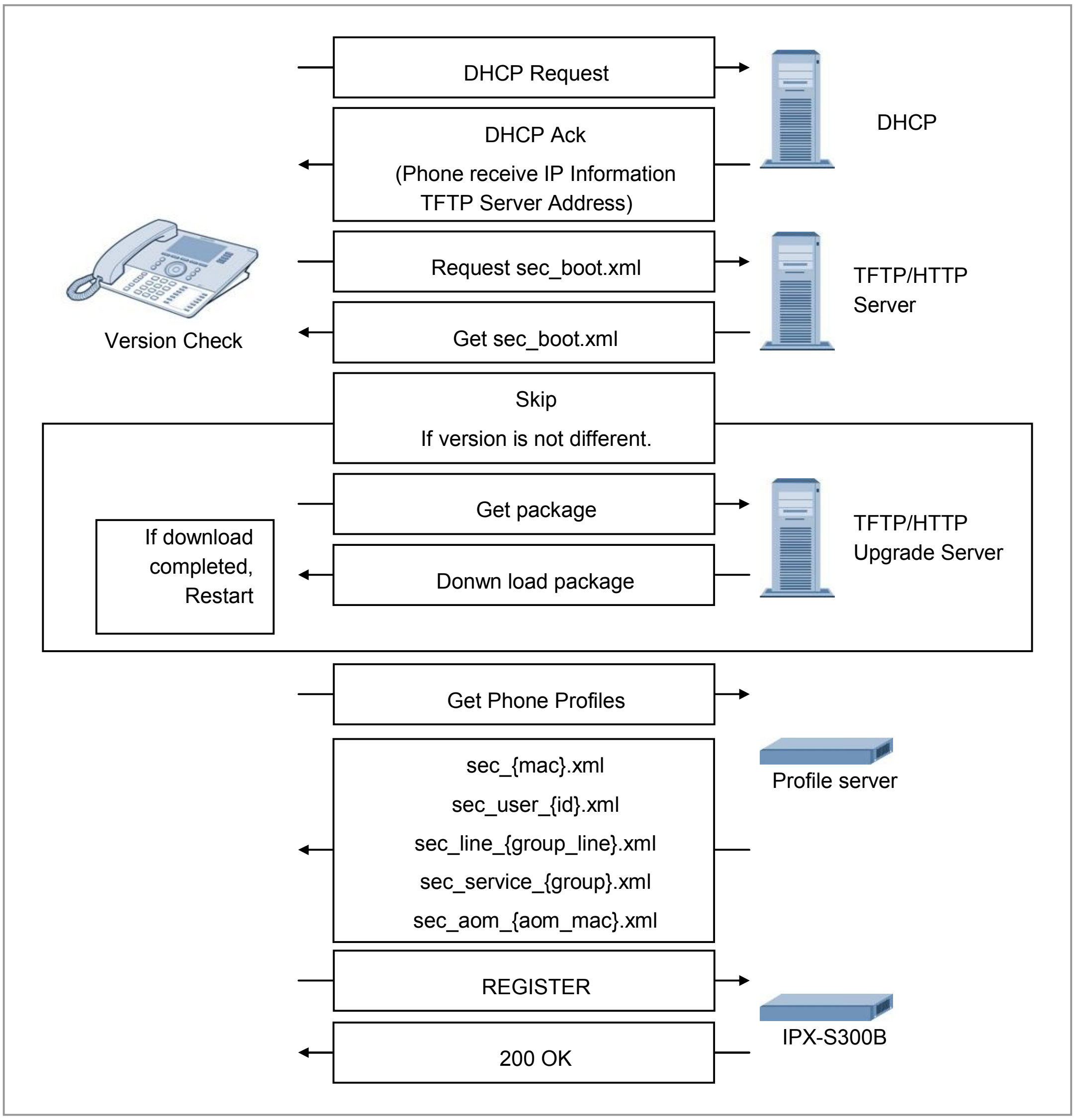


Figure 14. Automatic Upgrade Procedure

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

4.1.2.4 Installing Software Package in Profile Server

Software package should be installed on the profile server in the following order:

1) Save the phone software package in the root directory of a HTTP or TFTP server to use and decompress it.
2) Open the sec_boot.tpl file included in the phone software package, edit the phone main version, the profile server IP address, etc. and copy them to sec_boot.xml.
3) Use the update command in the **[CONFIGURATION > Phone > Phone Update]** menu to reboot the phone(s).

To use the IPX-S300B as an update server, upload your phone model's software package to theIPX-S300B by using the **[CONFIGURATION > Phone Setting > File Upload]** menu. Then, restart your phone by using the update command in the **[CONFIGURATION > Phone Setting > Upgrade Software]** menu.

4.1.3 Managing Phone Settings

4.1.3.1 Phone Profile Information

A phone's profile information can be viewed in the **[CONFIGURATION > Phone > Phone Profile Info]** menu.

Name	Description
Profile Type	Shows the type of the profiles viewed. There are service profiles and line profiles.
Profile Path	Shows the location within the profile server where the files are stored.
Profile Count	Shows the number of profiles created.

A service profile is applied to all phones in a user group. It includes definitions of available services. It includes information on feature codes, etc.

A line profile is applied to each phone number. It includes definitions of features available for each number. It includes information on call forward settings, etc.

4.1.3.2 Phone Display

You can set or change the items displayed on the phone by a preferred language.
This can be configured in the **[CONFIGURATION > Phone > Phone Display]** menu.

Name	Description
Key	Select a feature for which the information displayed will be changed.
Language	Select a language for which the information displayed will be changed.

Name	Description
Description	Enter the information displayed by language.

4.1.3.3 Software Version

You can configure the phone version. When the settings are changed, sec_boot.xml is changed.
This can be configured in the **[CONFIGURATION** > **Phone** > **Software Upgrade Configuration]** menu.

Name	Description
Phone Version (SMT-i2205)	The version of the phone software for the package update server
Phone Version (SMT-i2205S/D/G)	The version of the phone software for the package update server
Phone Version (SMT-i3100/3105)	The version of the phone software for the package update server
Phone Version (SMT-i5210)	The version of the phone software for the package update server
Phone Version (SMT-i5220)	The version of the phone software for the package update server
Phone Version (SMT-i5220S/D/S2)	The version of the phone software for the package update server
Phone Version (SMT-i5230)	The version of the phone software for the package update server
Phone Version (SMT-i5243)	The version of the phone software for the package update server
Phone Version (SMT-i5264)	The version of the phone software for the package update server
Phone Version (SMT-i5343)	The version of the phone software for the package update server
Phone Version (SMT-i5210S)	The version of the phone software for the package update server
Phone Version (SMT-i6010/6011)	The version of the phone software for the package update server
Phone Version (SMT-i6020/6021)	The version of the phone software for the package update server
Samsung Deskphone Manager Version	The version of the phone software for the package update server
Samsung Deskphone Manager Upgrade Server	Choose the upgrade server location - Internal: IPX-S300B - External: other Upgrade Server

4.1.3.4 NTP Option

You can configure the secondary NTP server and the refresh time. When using NAT, you can configure the secondary NTP used on the public IP network. When the settings are changed, sec_user_XXX.xml is changed, and a Notify message is sent to all phones.

This can be configured in the **[CONFIGURATION > Phone Setting > NTP Options]** menu.

Name	Description
NTP Second Server	If using a secondary NTP server, this sets the IP address of the NTP server.
NTP Refresh (minute)	If using a secondary NTP server, this sets the time interval for updating time for the NTP server.
NTP Public Second Server	If using a secondary NTP server on the public IP network, this sets the IP address of the NTP server.
NTP Public Refresh (minute)	If using a secondary NTP server on the public IP network, this sets the time interval for updating time for the NTP server.

4.1.3.5 Dial Plan

When dialing a phone number on the phone, if the number entered matches a specified rule, the phone can be set to send a sent message immediately to IPX-S300B without waiting for the next number or timeout. IPX-S300B can configure dial plans to be used by phones and send them to all phones.

Dial plans can be configured in the **[CONFIGURATION > Phone Setting > Dial Plan]** menu.

Name	Description
Start Timer	Specify the time to wait for the first dial since the handset is lifted. In case of timeout, an error will occur.
Long Timer	Specify the maximum waiting time between dials. In case of timeout, a sent message will be sent to IPX-S300B.
Short Timer	Specify the minimum waiting time between dials.
End of digit	Specify a digit to indicate that the last dial has been entered. When this digit is entered, a sent message is sent to IPX-S300B immediately.
DIGIT MAP 1~150	You can set up to 96 dial plans for phones. (Example: If a dial plan is set with 5xxx, the phone will make a call when a four-digit number starting with 5 is entered.)

4.1.3.6 Dial Tone

When dialing a phone number on the phone, if the number entered matches a specified rule, the phone play dial tone immediately. IPX-S300B can configure dial tone to be used by phones and send them to all phones.

Dial Tone can be configured in the **[CONFIGURATION > Phone Setting > Dial Tone]** menu.

Name	Description
DIGIT TONE 1~150	Enter dial rules which are used in a phone. e.g.) If you enter '5xxx', it means there will be an immediate playing dial tone when 4-digit with 5 as a leading figure becomes full.

4.1.3.7 SIP Options

You can configure the phone options. When the settings are changed, sec_phone.xml is changed. This can be configured in the **[CONFIGURATION > Phone Setting > SIP Options]** menu.

Name	Description
Use TLS Certificate Download	If used, TLS certification is used in signaling encryption for VoIP connections by SIP phones.
TLS Certificate Download Protocol	If used, the phone uses this protocol to get TLS certification from IPX-S300B server. (/tftpboot/sec_cert)
TLS Certificate Download Port	If used, the protocol uses port.
UDP Port Number	Port number to use SIP UDP signal.
TCP Port Number	Port number to use SIP TCP signal
TLS Port Number	Port number to use SIP TLS signal
Audio RTP Start Port Number	Audio RTP start port number
Audio RTP End Port Number	Audio RTP end port number
Video RTP Start Port Number	Video RTP start port number
Video RTP End Port Number	Video RTP end port number
Use TOS	If used, enable type of service.
TOS Type	select TOS Type
TOS Control value	Specify a number to use as TOS control value
TOS Media Value	Specify a number to use as TOS control value
Lowest Retransmission Timer	Specify a number to use lowest retransmission timer value
Highest Retransmission Timer	Specify a number to use highest retransmission timer value
T4 Timer	Specify a number to use T4 retransmission timer value

Name	Description
General request Timeout	Specify a number to use general request timer value
Register Expire Timer	Specify a number to use register expire timer value
Session Expire Time	Specify a number to use session expire timer value
User Expire Time	Specify a number to use subscriber expire timer value
TLS Certificate Format	Select Certificate Format PEM/DER
Use TLS Encrytion	Select whether to user the encrypted key Disable/Enable

4.1.3.8 Service Options

You can configure the phone options. When the settings are changed, sec_phone.xml is changed. This can be configured in the **[CONFIGURATION > Phone Setting > Service Options]** menu.

Name	Description
Use Premium CID Service	If used, the phone displays detail caller information.
Premium CID Service Address	Specify the CID server address to connect.
Use VCS Service	If used, the phone uses VCS service.
VCS Server Address	Specify the VCS server address to connect.
VCS Server Port Number	Specify the VCS server port value
Use Presence Service	If used, the phone uses presence service
Presence Server Address	Specify the presence server address to connect.
Use XML service	If used, the phone uses XML service
XML server URL	Specify the XML server address to connect.
Use LDAP Service	If used, the phone uses LDAP service
LDAP Server URL	Specify the LDAP server URL to connect.
LDAP Server Domain	Specify the LDAP server domain to connect.
LDAP Server ID	Specify the LDAP server ID to connect
LDAP Server Password	Specify the LDAP server password to connect
LDAP Server public URL	Specify the LDAP server public URL to connect.
Download Additional Tone Type	If used, the phone use dial tone file (download file from IPX-S300B system)
Special Tone File Folder	Specify the Dial tone file folder
Special Tone File 1-Name	Specify the Dial tone file name
Special Tone File 2-Name	Specify the Dial tone file name
Special Tone File 1-format	Specify the Dial tone file format (PCM, G729)
Special Tone File 2-format	Specify the Dial tone file format (PCM, G729

Name	Description
Ring Type Priority	Specify Ring Type Priority
Hotspot Channel (Country)	Specify Hotspot Channel for Wifi configuration phone use.
Phone Wallpaper	Specify the Phone's Wallpaper Setting - None: none - Download Only: phone will download the wallpaper from the SCM - Download and Apply: phone will download the wallpaper from the IPX-S300B and apply the wallpaper as phone's wallpaper. refer to 'File Upload' Menu for uploading wallpaper to IPX-S300B.
Phone Ring	Specify the Phone's Ring Setting - None: none - Download Only: phone will download the ring file from the SCM - Download and Apply: phone will download the ring file from the IPX-S300B and apply the ring file as phone's basic ring. refer to 'File Upload' Menu for uploading ring file to IPX-S300B.
Syslog Server URL	Specify the Syslog Server URL where phone send log to.
Syslog Server Public URL	Specify the Syslog Server Public URL where phone send log to.
Use Line Lock Out	Specify Line-Lock-Out (LLO) Function Use.
Line Lock Out ALarm	Specify playing of Line-Lock-Out tone when phone is in Line-Lock_Out State.
Hook Flash Detect Time (10 ms)	Specify Hook Flash Detect Time.
Hook On Detect Time (10 ms)	Specify Hook On Detect Time.
Hook Off Detect Time (10 ms)	Specify Hook Off Detect Time.

4.1.3.9 Tone Setting

You can configure the phone tone value. When the settings are changed, sec_phone.xml is changed. This can be configured in the **[CONFIGURATION > Phone Setting > Service Options]** menu.

Name	Description
Tone Type	Select a tone type for which the value will be configured
Tone Frequency 1	Specify the primary frequency.
Tone Frequency 2	Specify the second frequency
First On Time	Specify the first cadence on time.
First Off Time	Specify the first cadence off time.
Second On Time	Specify the second cadence on time.
Second Off Time	Specify the second cadence off time.

Name	Description
Third On Time	Specify the third cadence on time.
Third Off Time	Specify the third cadence off time.

4.1.3.10 Phone Configuration

You can set the additional configuration used by the phone
This can be configured in the **[CONFIGURATION > Phone Setting > Phone Configuration]** menu.

Name	Description
User Group	Select a user group
Call Log in Busy	Select to whether to logging a Call in Busy
Call Log for Group-Cll-Pickup	Select to whether to logging a Call for Group-Cll-Pickup
Call Log for Direct-Call-Pickup	Select to whether to logging a Call for Direct-Call-Pickup
Call Log for Parked-Call-Retrieve	Select to whether to logging a Call for Parked-Call-Retrieve
Call Log for Shared-Call-Retrieve	Select to whether to logging a Call for Shared-Call-Retrieve
Input Number Display	Select to whether to display number pressed by user during conversation
Call Log for Redial	Select to whether to logging a Call for Redial
Display for Profile Update	Select to whether to display the changing information on Phone

4.2 Configuring IPX-G500B Gateway

This section explains the configuration for linking the IPX-G500B (IPX-G520/G540) gateway to the IPX-S300B and also explains how to update and reset software.

4.2.1 Configuring IPX-G500B

You need to set up all of the followings: the IP address of the IPX-G5X0 series, the IP address of the IPX-S300B server to be linked, the registered gateway name, and the order for the IPX-G520S/540S.

4.2.1.1 Connecting to CLI

To connect to the CLI, execute the following steps:

- Connect a serial cable to the system's console port.
- Run the serial terminal program on your PC and set the baud rate (bps) at 38,400.
- When the system has completed booting, log in using the ID 'admin' and the password 'samsung'.
- Type 'cli' and press Enter.

```
Welcome to Samsung IPX-G500
IPT-NPU login: admin
Password:


#################################################################
#                                                               #
#                                                               #
#                  Samsung IPTGW-NPU                            #
#                                                               #
#          Enter 'cli' to set basic configuration               #
#                                                               #
#                                                               #
#################################################################
S A M S U N G  - I P T G W
-------------------------------------------------------
admin@IPT-NPU:~#
```

4.2.1.2 IPX-G500B (CMU) Basic Configuration

- Connect a serial cable to the IPX-G500B's serial port (baud rate: 38,400 bps).
- Log in to the system and enter 'bash shell'.
- Enter 'cli'.
- Select '1. Setup basic configuration'.
- Select the network mode, LAN or NAT.

- If you select NAT
  - Enter WAN IP address, gateway and subnet mask.
  - Enter the LAN IP address, gateway and subnet mask.
- If you select LAN
  - Enter the LAN IP address, gateway and subnet mask.
- Enter the IP address of the IPX-S300B following the ensuing instructions.
- Enter the name of the gateway you will use following the ensuing instructions.
- After entering all information, you are asked if you want to reboot the system to apply the changes. Select 'yes' if you want to reboot the system and select 'no' if you don't

[NAT]

```
######           Please enter network mode of this GW           ######
 1. Enter Networkmode [0:LAN / 1:NAT]    (Enter Key : NAT) :

######    Please enter [WAN IP/Gateway/Subnet Mask] of this GW    ######
 1. Enter WAN IP Address         (Enter Key : 10.251.0.10 ) :
 2. Enter Gateway Address        (Enter Key : 10.251.0.1   ) :
 3. Enter WAN IP Subnet Mask     (Enter Key : 255.255.255.0  ) :

######        Please enter [LAN IP/Subnet Mask] of this GW       ######
 1. Enter LAN IP Address         (Enter Key : 192.168.0.1    ) :
 2. Enter LAN IP Subnet Mask     (Enter Key : 255.255.255.0  ) :

######           Please enter the IP Address of SCME            ######
 1. Enter SCME IP Address        (Enter Key : 10.251.0.100 ) :

######       Please enter the name of this GW (Max 30char)       ######
 1. Enter the Gateway Name, as configured in SCM,
    to be used for this gateway  (Enter Key : G500_default   ) :


End of Configuration. To apply it, please restart the system.
Do you want to restart? (yes/no) :
```

[LAN]

```
######           Please enter network mode of this GW           ######
 1. Enter Networkmode [0:LAN / 1:NAT]    (Enter Key : NAT) : 0

######    Please enter [LAN IP/Gateway/Subnet Mask] of this GW    ######
 1. Enter LAN IP Address         (Enter Key : 192.168.0.10    ) :
 2. Enter Gateway Address        (Enter Key : 192.168.0.1   ) :
 3. Enter Subnet Mask            (Enter Key : 255.255.255.0  ) :

######            Please enter the IP Address of SCME           ######
 1. Enter SCME IP Address        (Enter Key : 192.168.0.100 ) :

######       Please enter the name of this GW (Max 30char)       ######
 1. Enter the Gateway Name, as configured in SCM,
```

```
   to be used for this gateway  (Enter Key : G500_default   ) :


End of Configuration. To apply it, please restart the system.
Do you want to restart? (yes/no) :
```

4.2.1.3 IPX-G520S/540S (ECU) Basic Information

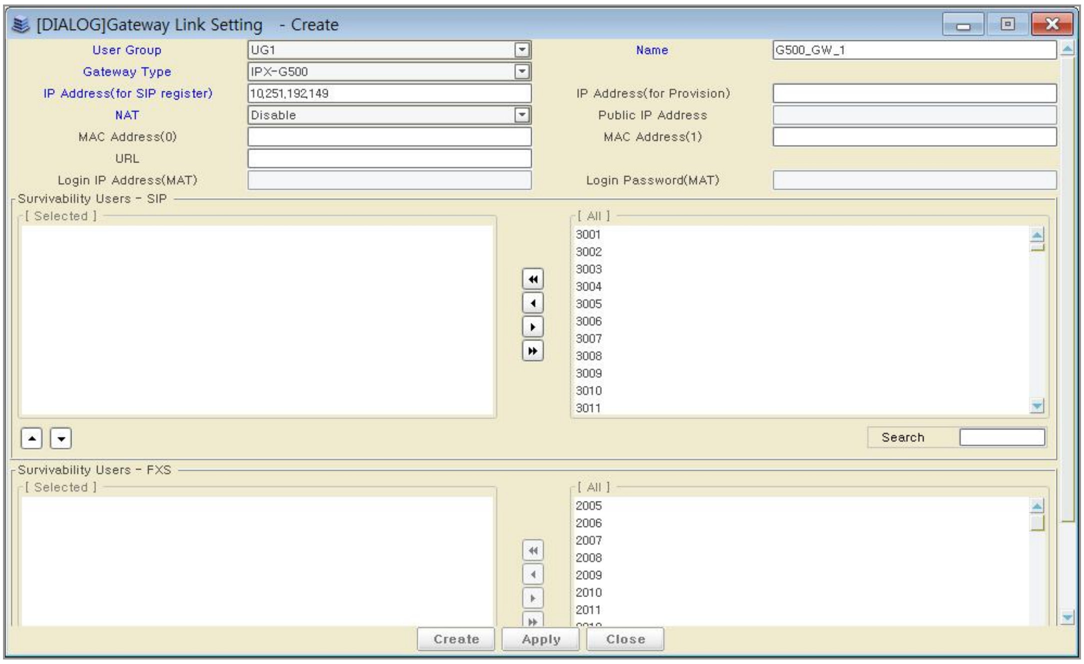
- Connect a serial cable to the IPX-G520S/540S's serial port (baud rate: 38,400 bps).
- Log in to the system and enter 'bash shell'.
- Enter 'cli'.
- Select '1. Setup basic configuration'.
- Enter the IP address, gateway, and subnet mask of the ECU following the instructions.
- Enter the IP address of the connected CMU following the ensuing instructions.
  If you want to enter 2nd CMU, enter yes and enter IP address of 2nd CMU.
- Following the ensuing instructions, enter one of the numbers 1-6 to specify what number ECU the ECU is.
- After entering all information, you are asked if you want to apply the changes.
  Select 'yes' if you want to apply the changes and select 'no' if you don't.

```
######        Please enter [IP/Gateway/Subnet Mask] of this ECU
######
 1. Enter ECU IP Address      (Enter Key : 10.251.191.177 ) :
 2. Enter Gateway Address     (Enter Key : 10.251.191.1   ) :
 3. Enter Subnet Mask         (Enter Key : 255.255.255.0  ) :

######        Please enter the IP Address of CMU              ######
 1. Enter CMU IP Address      (Enter Key : 10.251.192.162 ) :
 Do you want enter 2nd NPU IP  (yes/no)                    :

######        Please enter the ECU Number of this ECU         ######
 1. Enter this ECU number     (Enter Key : 1)              :



End of Configuration. Do you want to apply it? (yes/no) :
```

4.2.2 Configuring IPX-S300B

4.2.2.1 Gateway Link Setting

To add a gateway, create a new gateway link at **[Configuration > Gateway > Gateway Link Setting]**.



Item	Description
User Group	Specify a user group which will be assigned to Gateway
Name	Enter a name for Gateway. This name should be unique and it should be same with the name configured by gateway CLI.
Gateway Type	Model type for Gateway. (IPX-G500)
IP Address (for SIP register)	Enter the IP address of Gateway for SIP registration.
IP Address (for Provision)	Enter the IP address of Gateway for provision. Generally this should be same address for SIP register.
NAT	If the gateway is running under NAT, Enables this option.
Public IP Address	When the gateway is running under NAT, enter the public IP address of Gateway.
MAC Address (0)	Enter the gateway MAC address. If it is set, it only allows the system that has this MAC address.
MAC Address (1)	Do not use.
URL	Do not use. To connect OfficeServ or Ubigate web admin, enter the URL of gateway.
Gateway Reconnect	Select the option whether reconnect the link between gateway and

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
	SCM or not.
Upgrade	Upgrade the software of this gateway according to the upgrade settings. (Not saved)
Login IP Address (MAT)	Do not use
Login Password (MAT)	Do not use
Survivability Users-SIP	This is the list of SIP phones for survivability to allocate to the voice gateway. When SCM creates the profile for a SIP phone, it includes the IP address and the port of the interoperating gateway in survival mode. When the SIP phone is disconnected from IPX-S300B, it sends a register message to the gateway included in the profile.
Survivability Users-FXS	This is the list of FXS phones of the voice gateway. If the gateway type is IPX-G500, this item is not selective list.

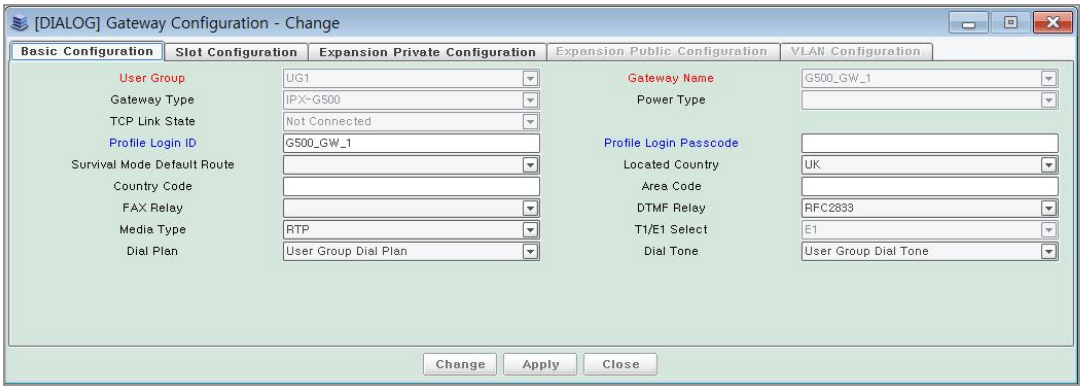
4.2.2.2 Gateway Configuration

You can configure the new gateway at **[Configuration > Gateway > IPX Setting > Gateway Configuration]**. It is able to set the followings.

- Profile Login ID/Password
- IP Settings
- Slot/Expansion Settings

4.2.2.2.1 Basic Configuration

In Basic Configuration, you enter basic configuration information about the gateway.



Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

<Read Only>

It shows basic informations assigned by Gateway Link Setting menu and you cannot change it.

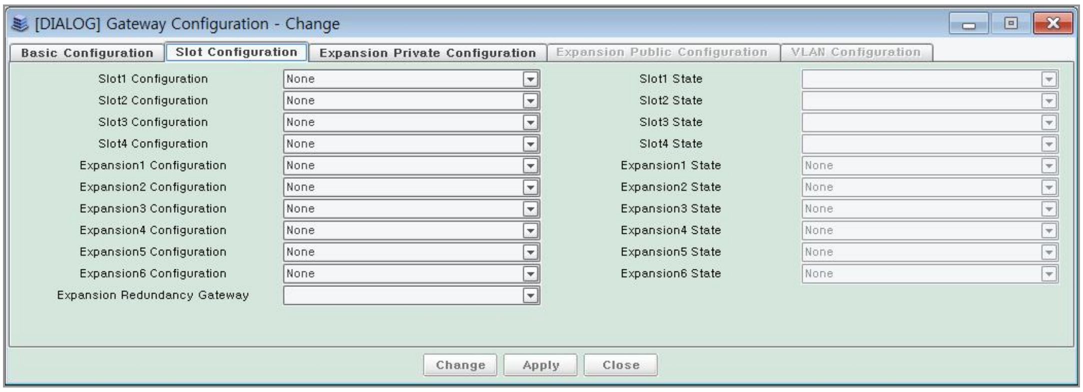
Name	Description
User Group	User Group
Gateway Name	Gateway name assigned by Gateway Link Setting menu.
Gateway Type	Gateway Type (IPX-G500)
Power Type	Gateway Power Type - IPX-G500B (Single Power) - IPX-G500BP (Dual Power)
TCP Link State	TCP Link State between Gateway and SCM
Dial Plan	Dial plan of selected gateway
Dial Tone	Dial tone of selected gateway

<other items>

Item	Description
Profile Login ID	Profile Login ID that is used when gateway requests provisioning to get its profiles from IPX-S300B.
Profile Login Passcode	Profile Login Passcode that is used when gateway requests provisioning to get its profiles from IPX-S300B.
Survival Mode Default Route	Default Route of Survival Mode. If user dials invalid access code in survival mode, gateway will make a call with default route.
Located Country	Country the gateway is located
Country Code	Country code the gateway is located.
Area Code	Area code the gateway is loated
FAX Relay	Fax Relay (T.38, Pass Through)
DTMF Relay	DTMF Relay (Outband, RFC2833, In Voice)
Media Type	Media Type RTP, SRTP (AES/ARIA128), SRTP (ARIA128/AES), SRTP (AES/ARIA192), SRTP (ARIA192/AES), SRTP (AES), SRTP (ARIA128), SRTP (ARIA192)
Dial Plan	Shows the dial plan used by the gateway. (Default: User Group Dial Plan.)
Dial Tone	Shows the dial tone used by the gateway. (Default: User Group Dial Tone.)

4.2.2.2.2 Slot Configuration

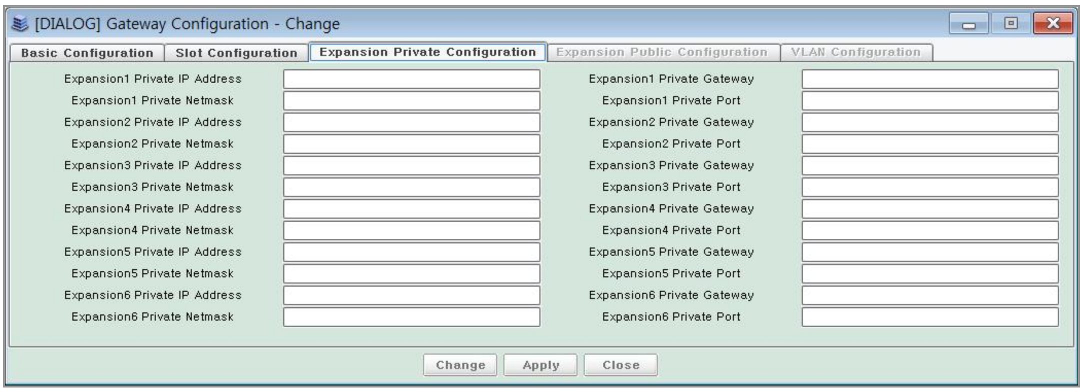
In Slot Configuration, enter slot information that matches the actual configuration of the gateway.



Item	Description
Slot1~Slot4 Configuration	Select GWU slot1~4 card type. (4FXO, 4FXS) Selected card should be same as actual installed card. But if card is not installed yet, you should select card which is scheduled to be installed on the designated slot.
Slot1~Slot4 State (read only)	Actually installed card type
Expansion1~6 Configuration	Select Expansion1~6 type. (IPX-G520S (20FXS), IPX-G540S (40FXS)) Selected Expansion should be same as actual installed Expansion. But if Expansion is not installed yet, you should select Expansion which is scheduled to be installed.
Expansion 1~6 State (read only)	Actually installed Expansion type

4.2.2.2.3 Expansion Private Configuration

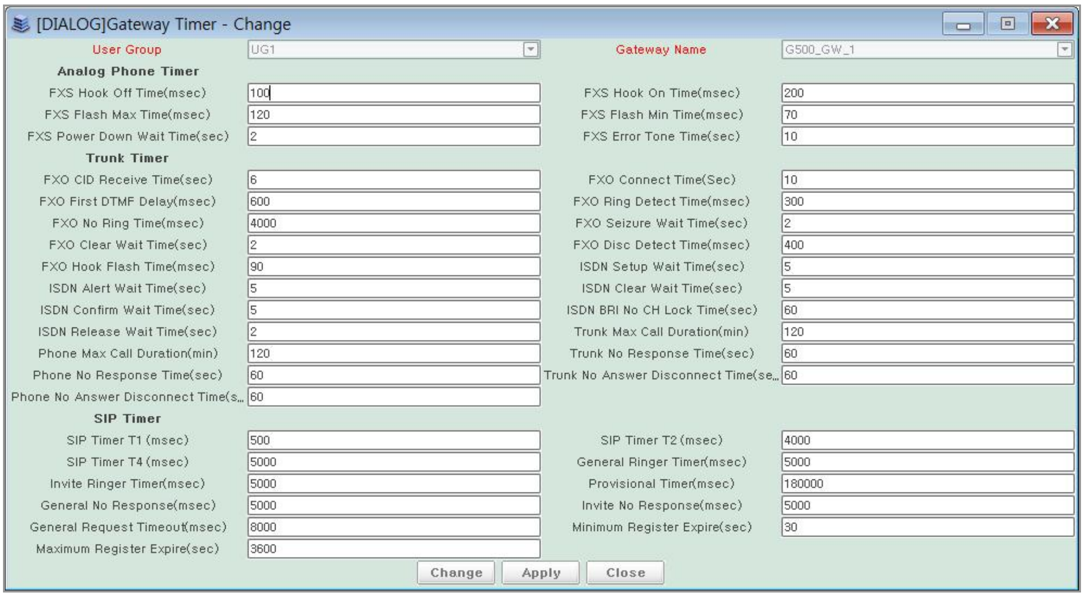
Enter private IP addresses of expansions in this menu.



Item	Description
Expansion1~6 Private IP Address	Expansion1~6 Private IP Address. This IP address should be same as ECU IP address which was set on ECU board by cli.
Expansion1~6 Private Gateway	Expansion1~6 Private Gateway Address.
Expansion1~6 Private Netmask	Expansion1~6 Private Subnet mask.
Expansion1~6 Private Port	Expansion1~6 Private Port number.

4.2.2.3 Gateway Timer

In Gateway Timer, you configure information about timers used for the gateway to provide analog phone, analog trunk, and SIP features.



4.2.2.3.1 Analog Phone Timer

Item	Description
FXS Hook Off Time (ms)	The time based on which whether the analog phone is in off-hook status is determined.
FXS Hook On Time (ms)	The time based on which whether the analog phone is in on-hook status is determined.
FXS Flash Max Time (ms)	The maximum time for determining whether the analog phone is in hook flash status.
FXS Flash Min Time (ms)	The minimum time for determining whether the analog phone is in hook flash status.
FXS Power Down Wait Time (sec)	The length of time for which Loop Open (Power Down) will be sustained when the FXS Loop Open is used.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
FXS Error Tone Time (sec)	The maximum length of time for which the error tone will play.

4.2.2.3.2 Trunk Timer

Item	Description
FXO CID Receive Time (sec)	The timeout time for CID receipt after a 1st ring signal is detected from the FXO port.
FXO Connect Time (sec)	When making a FXO trunk call without PRS, the point in time at which the trunk connect time starts being calculated for billing purposes.
FXO First DTMF Delay (ms)	The time delayed until the first DTMF is transmitted after dialing after finding an analog trunk.
FXO Ring Detect Time (ms)	The minimum length of time for which the ring signal that comes in the FXO port is sustained for determination of the ring signal's validity.
FXO No Ring Time (ms)	The point in time at which the ring signal that has been coming in the FXO port is judged not to be coming in any longer.
FXO Seizure Wait Time (sec)	When making a FXO trunk call, the wait time until the receipt of a response after a request for seizure is made to the port.
FXO Seizure Clear Wait Time (sec)	When terminating a FXO trunk call, the wait time that it takes to get back to idle status after a request for release is made to the port.
FXO Disc Detect Time (ms)	When the feeding voltage of the CO line connected to the FXO port has been cut off, the minimum length of time for deciding on Disconnect.
FXO Hook Flash Time (ms)	The time that it takes to transmit a hookflash signal to the analog trunk.
ISDN Setup Wait Time (sec)	The wait time after sending a Setup message.
ISDN Alert Wait Time (sec)	The wait time after sending an Alerting message.
ISDN Clear Wait Time (sec)	The wait time after sending a Disconnect message.
ISDN Confirm Wait Time (sec)	The wait time until receiving the next message after sending a Connect and Release message.
ISDN BRI No CH Lock Time (sec)	Not used.
ISDN Release Wait Time (sec)	The wait time until becoming available again after terminating a call.
Trunk Max Call Duration (min)	The length of time after which a trunk call is terminated.
Phone Max Call Duration (min)	The length of time after which an inter-extension call is terminated.
Trunk No Response Time (sec)	The time after which the call is terminated if a response to a request for making a trunk call has not been received from

Item	Description
	the SCM in Normal mode before the time has passed.
Phone No Response Time (sec)	The time after which the call is terminated if a response to a request for making an inter-extension call has not been received from the SCM in Normal mode before the time has passed.
Trunk No Answer Disconnect Time (sec)	The time after which the call is terminated if a response has not been made from the other end before the time has passed, even though the phone at the other end has been ringing after a trunk call was made.
Phone No Answer Disconnect Time (sec)	The time after which the call is terminated if a response has not been made from the other end before the time has passed, even though the phone at the other end has been ringing after an inter-extension call was made.

4.2.2.3.3 SIP Timer

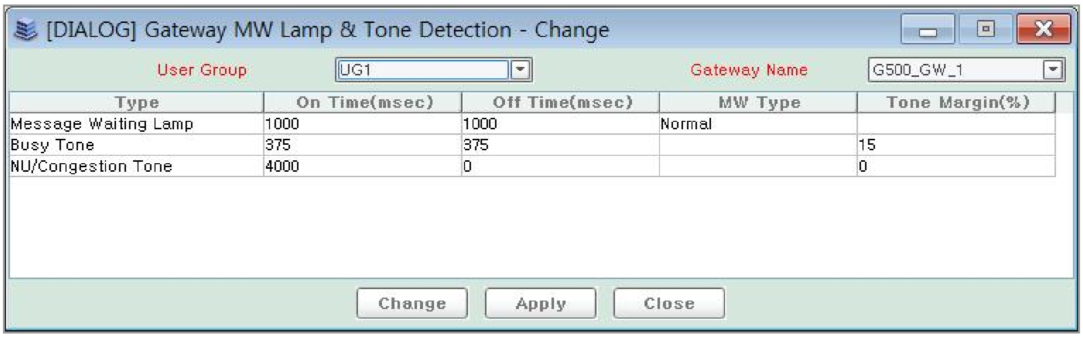
Item	Description
SIP Timer T1 (ms)	The length of time for which the SIP T1 (ms) timer lasts.
SIP Timer T2 (ms)	The length of time for which the SIP T2 (ms) timer lasts.
SIP Timer T4 (ms)	The length of time for which the SIP T4 (ms) timer lasts.
General Ringer Timer (ms)	The length of time for which the General Ringer timer lasts.
Invite Ringer Timer (ms)	The length of time for which the Invite Ringer timer lasts.
Provisional Timer (ms)	The length of time for which the Provisional timer lasts.
General No Response (ms)	The length of time for which General No Response lasts.
Invite No Response (ms)	The length of time for which Invite No Response lasts.
General Request Timeout (ms)	The length of time for which General Request Timeout lasts.
Minimum Register Expire (sec)	The minimum length of time for the expiration of the SIP phone’s registration.
Maximum Register Expire (sec)	The maximum length of time for the expiration of the SIP phone’s registration.

4.2.2.4 Gateway Ring Cadence

Ring Type	1st On (msec)	1st Off (msec)	2nd On (msec)	2nd Off (msec)	3rd On (msec)	3rd Off (msec)
Gateway1 Ring	400	200	400	3000	0	0
Gateway2 Ring	1000	3000	0	0	0	0
Gateway3 Ring	400	100	400	2000	0	0
Gateway4 Ring	200	200	200	2000	0	0
Gateway5 Ring	200	200	200	4000	0	0
Gateway6 Ring	400	300	400	1000	0	0

Item	Description
Ring Type	The list of rings used for the gateway
1st on (ms)	The length of time for which the first ringing continues.
1st off (ms)	The duration of silence after the first ringing stops.
2nd on (ms)	The length of time for which the second ringing continues.
2nd off (ms)	The duration of silence after the second ringing stops.
3rd on (ms)	The length of time for which the third ringing continues.
3rd off (ms)	The duration of silence after the third ringing stops.

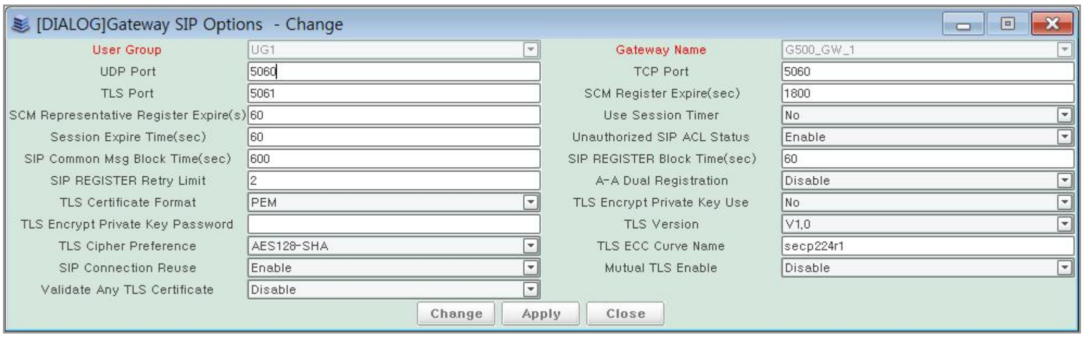
4.2.2.5 MW Lamp & Tone Detection



Type	On Time(msec)	Off Time(msec)	MW Type	Tone Margin(%)
Message Waiting Lamp	1000	1000	Normal	
Busy Tone	375	375		15
NU/Congestion Tone	4000	0		0

Item	Description
On Time (ms)	The length of time for which the Message Waiting Lamp remains on/the length of time for which the tone that will be detected remains on.
Off Time (ms)	The length of time for which the Message Waiting Lamp remains off/the length of time for which the tone that will be detected remains off.
Message waiting type	Normal (a common method that uses a ring signal)/PRS (a method that uses reverse polarity)
Tone Margin (%)	The additional margin value for the on and off times set up for Busy and NU/Congestion Tone Detection.

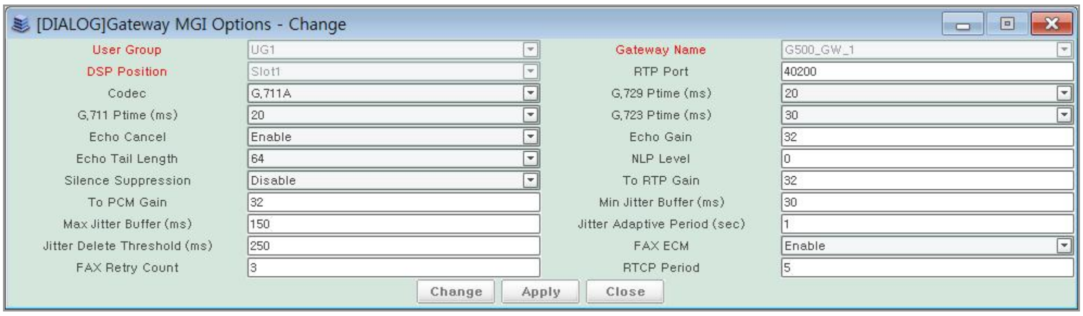
4.2.2.6 Gateway SIP Options



Item	Description
UDP Port	The port number that will be used for UDP signaling.
TCP Port	The port number that will be used for TCP signaling.
TLS Port	The port number that will be used for TLS signaling.
SCM Register Expire (sec)	The registration expiration time used for when registering the FXO and the FXS separately.
SCM Representative Register Expire(s)	The registration expiration time used for when registering a main number.
Use Session Timer	Whether to Use the Session timer.
Session Expire Time (sec)	The expiration time for when using the Session timer.
Unauthorized SIP ACL Status	Select whether to use ACL blocking for unauthorized SIP messages.
SIP Common Msg Block Time (sec)	The length of time for which ACL blocking by unauthorized SIP messages (excl. REGISTER messages) lasts.
SIP REGISTER Block Time (sec)	The length of time for which ACL blocking by REGISTER messages lasts.
SIP REGISTER Retry Limit	The maximum number of REGISTER messages allowed
TLS Certificate Format	Select between PEM and DER.
TLS Encrypt Private Key Use	Select whether to use authentication key encryption.
TLS Encrypt Private Key Password	The password for when using authentication key encryption.
TLS Version	Sets V1.0/V1.2.
TLS Cipher Preference	Select between AES128-SHA/DES-CBC3-SHA and ECDSA-AES128-SHA.
TLS ECC Curve Name	Specify the ECC curve name.
SIP Connection Reuse	Select whether to enable connection reuse.
Mutual TLS Enable	Specify whether to enable mutual authentication TLS.
Validate Any TLS Certificate	Specify whether to enable Validate Any TLS certificate.

4.2.2.7 Gateway MGI Options

The Media Gateway Interface (MGI) converts a PCM voice to an IP voice packet and an IP voice packet to a PCM voice. You can set up features related to it in MGI Options.
Note that IP voice packets use the Real-Time Transport Protocol (RTP).

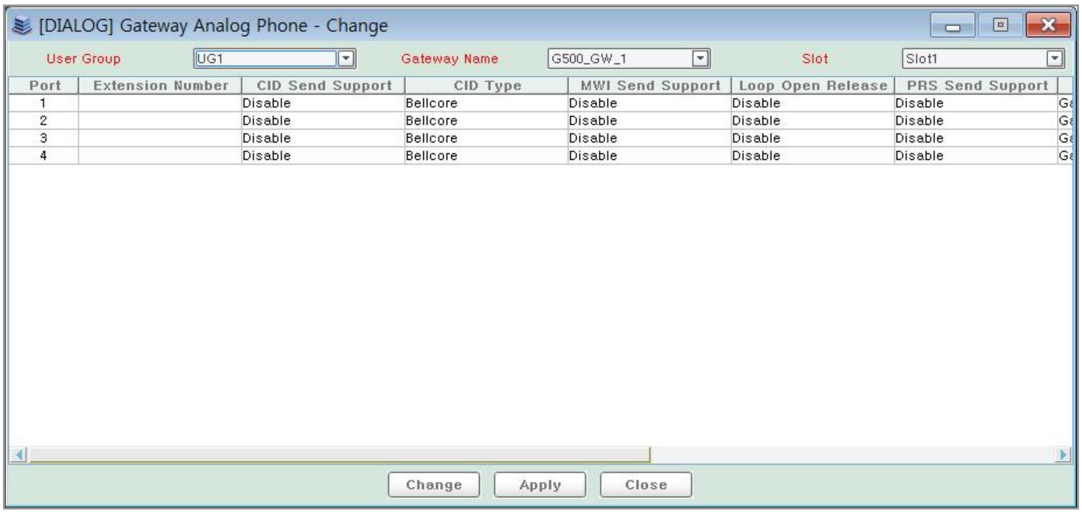


Item	Description
RTP Port	The start number of the RTP to be transmitted.
Codec	Voice codec information about the RTP to be transmitted. (G.729, G.729a, G.711a, G.711u, G.723)
G.729 P-time (ms)	The transmission interval for G.729/G.729a RTP packets.
G.711 P-time (ms)	The transmission interval for G.711a/G.711u RTP packets.
G.723 P-time (ms)	The transmission interval for G.723 RTP packets.
Echo Cancel	Cancels echoes generated during a call by packet delays or by sound reflections occurring on the PSTN.
Echo Gain	Set call sensitivity for Echo Cancel. (18-38 (-14-+6 dB))
Echo Tail Length	Set the tail length for Echo Cancel (8-128).
NLP Level	Specify the Non-Linear Processor (NLP) control value (0, 1, or 2). This feature is only available when Echo Cancel is enabled. 0. The normal (default) level of NLP engagement. 1. NLP Tune Option 2 (reduced level of NLP engagement) 2. NLP Tune Option 1 (increased level of NLP engagement) NLP Tune Option 2 (reduced level of NLP engagement): You can select the NLP Tune Option 2 value to reduce the NLP level when the voice heard on the phone sounds choppy in the beginning due to double-talk. NLP Tune Option 1 (Increased level of NLP engagement): You can select the NLP Tune Option 1 to increase the NLP level when a 2-to-4 wire hybrid/line condition occurs.
Silence Suppression	Select this feature to enable detecting silence during a call. If silence is detected, voice packets are not transmitted.
To RTP Gain	Specify the call sensitivity of the PCM voice inputted through DSP.

Item	Description
	(18-38 (-14-+6 dB))
To PCM Gain	Specify the call sensitivity for when turning a voice packet into a PCM voice. (18-38 (-14-+6 dB))
Min Jitter Buffer (ms)	Specify a minimum jitter delay value (0-150 ms).
Max Jitter Buffer (ms)	Specify a maximum jitter delay value (30-200 ms).
Jitter Adaptive Period (sec)	Specify the jitter adaptive period (1-10).
Jitter Delete Threshold (sec)	Specify the jitter delete threshold (150-500 ms).
Fax ECM	Enable this feature to further restore errors that occur when using the T.38 Fax.
Fax Retry Count	Specify the retry count for errors that occur when using the Fax (0-3).
RTCP Period	Specify the time interval for RTCP transmission (2-10).
QoS	Configure QoS features used with the ToS field of the IPV4 packet header. (ToS, IP Precedence, and DSCP.) You specify priority values for packets used at the routers, switches, etc. on an external network. If the network does not support this feature, you must use default values.
ToS	Specify a priority value using an 8-bit field called ToS included in the IPv4 packet header (binary number: 00000000-11111111).
IP Precedence	Specify a priority value using the first 3 bits of the 8-bit Type of Service (ToS) field. A higher value means a higher priority (0-7).
DSCP	Specify a priority value using the first 6 bits of the 8-bit Type of Service (ToS) field. A higher value means a higher priority (0-63).

4.2.2.8 Analog phone

Assign created phone number to the port of the gateway at **[Configuration > Gateway > IPX Setting > Analog Phone]**.



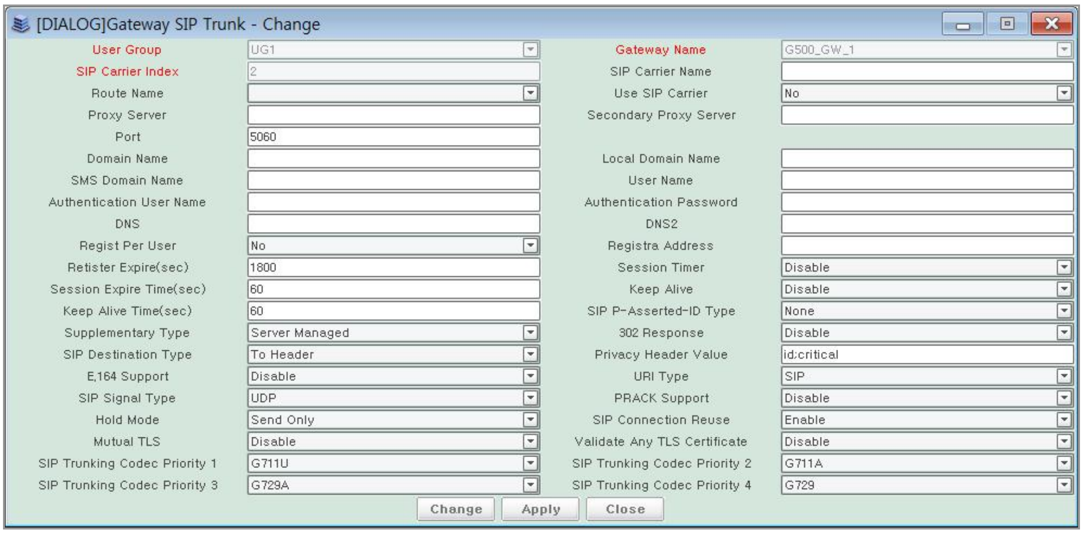
Item	Description
Extension Number	The extension number of the analog phone.
CID Send Support	CID Send Support (Enable/Disable)
CID Type	The type of FSK CID (ITU-T/Bellcore)
MWI Send Support	MWI Send Support (Enable/Disable)
Loop Open Release	Loop Open Release feature. (Enable/Disable) If this feature is enabled, FXS Loop will be disconnected (Power Down) for a while when the opponent hang up.
PRS Send Support	PRS Send Support feature. (Enable/Disable) If this feature is enabled, the PRS signal will be generated when the opponent answer.
Ring Type	Station Ring Type (Gateway1~6 Ring)
Tx Gain	The Tx gain of this extension (-4~+4 dB)
Rx Gain	The Rx gain of this extension (-4~+4 dB)
Feature Tone detecation	When Call Forward or DND or Extension Lock feature is enabled, special dial tone will be played instead of normal dial tone.

4.2.2.9 Analog Trunk

Assign a created FXO route to the FXO port of gateway.

Name	Description
Route Name	The trunk route name to which this trunk belong.
Link State	Current connection state of FXO.
CID Receive Support	CID Receive Support (Enable/Disable)
Ring Destination Number	The destination number to be rung when this trunk port detects ring signal.
PRS Receive Support	- Disable - Answer-Disconnect • 1st PRS Detect: Opponent answer • 2nd PRS Detect: Opponent hang up and the line will be released automatically. - Disconnect • 1st PRS Detect: Opponent hang up and the line will be released automatically.
Tone Detection	- Disable - Busy Tone - NU/Congestion Tone - Both (Busy Tone, NU/Congestion Tone) If selected tone is detected, the line will be released automatically.
Loop Type	- Short (~1 Km) - Mid (1~3 Km) - Long (3 Km~)
TX Gain	The Tx gain of this TRK (-4~+4 dB)
RX Gain	The Rx gain of this TRK (-4~+4 dB)

4.2.2.10 Gateway SIP Trunk

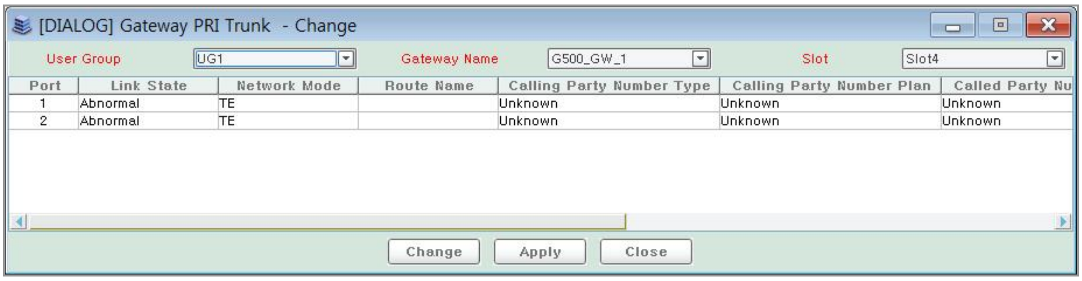


Item	Description
SIP Carrier Index	The number of the ISP. Up to 4 ISPs can be set for the gateway.
SIP Carrier Name	The name of the ISP.
Route Name	Specify the trunk route to which the SIP trunk will belong.
Use SIP Carrier	Specify whether to link the ISP to the SIP trunk.
Proxy Server	Specify the primary proxy server of the ISP that the gateway will link to the SIP trunk.
Secondary Proxy Server	Specify the proxy server that will be used instead of the primary proxy server when the primary proxy server is down.
Port	Specify the port number of the SIP server.
Domain name	The domain name used by the ISP's proxy server.
Local Domain name	Specify the local domain name.
SMS Domain Name	Specify the SMS domain name.
User Name	Specify a main number when linking with an external ISP that uses a main number.
Authentication User Name	Enter the ID that will be used for authentication upon the registration of a main number when linking with an external ISP that uses a main number.
Authentication Password	Enter the password that will be used when authenticating the main number.
DNS	Specify the IP address of the name server that tells the IP address of the SIP server when executing a DNS query using the domain name.
DNS2	The IP address of the secondary DNS server.
Register Per User	Enabling this feature performs registration for each user, and disabling this feature perform registration using the main number.
Register Address	Specify the address to which the gateway will send the REGISTER

Item	Description
	message when linking the ISP to the SIP trunk.
Register Expire (sec)	Specify the value that will be included in the Expire header of the REGISTER message which is transmitted by the gateway when linking the ISP to the SIP trunk.
Session Timer	Specify whether to use the Session timer for the gateway.
Session Expire Time (sec)	Specify the time that will be included in the Session timer header.
Whether to enable Keep Alive	Some external ISP servers need to receive the Options message periodically from the gateway for link tests. Specify whether or not to send the Options message.
Keep Alive Time (sec)	Specify the interval at which the gateway transmits the Options message.
SIP P-Asserted-ID Type	When linking to an external ISP that uses a main number, enabling this features puts the main number in the P-Asserted-ID header of the SIP message that the gateway transmits and puts the individual number in the FROM header.
Supplementary Type	In the SIP standard setting, the Re-Invite message is used to Hold or Resume during the SIP's supplementary service. However, in the PBX MANAGED 2 setting, Samsung's own method is used so that the Re-Invite message is not transmitted. In addition, in the PBX MANGED 2 setting, the Refer message commonly used for Transfer is made not to be transmitted out.
302 Response	If you have set up All Call Forward for a station terminal, when an INVITE message is received for the terminal, if the 302 RESP option is set to Enable, the gateway does not process Forward but transmits a 302 response to the server so that the server transmits an INVITE message again to the Forwarded terminal.
SIP Destination Type	Specify whether to recognize the callee in To-Header or in Request-URI when an INVITE message is received from the gateway.
Privacy Header Value	Specify a value for the Privacy header (e.g. id;critical).
E.164 Support	Specify whether to use the E.164 format.
URI Type	Specify the type of the SIP URI used for the SIP message. As per the SIP standard, the gateway supports TEL-URI, and SIPS-URI in addition to SIP-URI.
SIP Signal Type	Specify the type of the transport used for the SIP message. (UDP, TCP, or TLS.)
PRACK Support	Specify whether to support PRACK.
Hold Mode	Select the Hold mode. (SENDONLY, SENDRECV, or INACTIVE.)
SIP Connection Reuse	Specify whether to reuse SIP Trunk connection.
Mutual TLS	Specify whether to use TLS mutual authentication.
Validate Any TLS Certificate	Specify whether to enable Validate Any TLS certificate.

Item	Description
SIP Trunking Codec Priority 1-4	Codecs 1-4 that can be used for the SIP trunk.

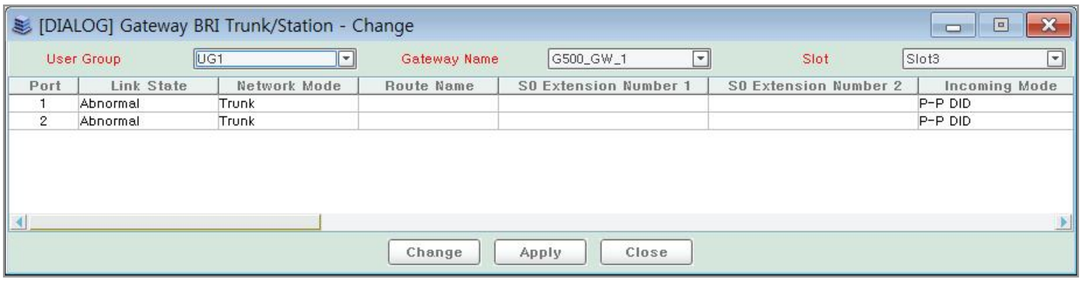
4.2.2.11 Gateway PRI Trunk



Item	Description
Link State	Shows current link state.
Network Mode	Specify the mode of the network to which the PRI port will connect. (TE: Terminal/NT: Network.)
Route Name	Specify the trunk route to which the PRI trunk will belong.
Incoming Mode	Specify the incoming mode. - Normal - DID
Calling Party Number Type	Specify the calling party number type. - Unknown - International - National - Network - Subscriber - Abbreviate - Extension
Calling Party Number Plan	Specify the calling party number plan. - Unknown - ISDN - Data - Telex - Nation - Private - Extension
Receiving Party Number Type	Specify the receiving party number type. - Unknown - International - National - Network - Subscriber

Item	Description
	- Abbreviate - Extension
Receiving Party Number Plan	Specify the receiving party number plan. - Unknown - ISDN - Nation
CRC	Select whether to create a Cyclic Redundancy Check (CRC) and whether to detect errors.
Caller ID Restriction	Select whether to show the originating number to the other party when making a call.

4.2.2.12 Gateway BRI Trunk/Station



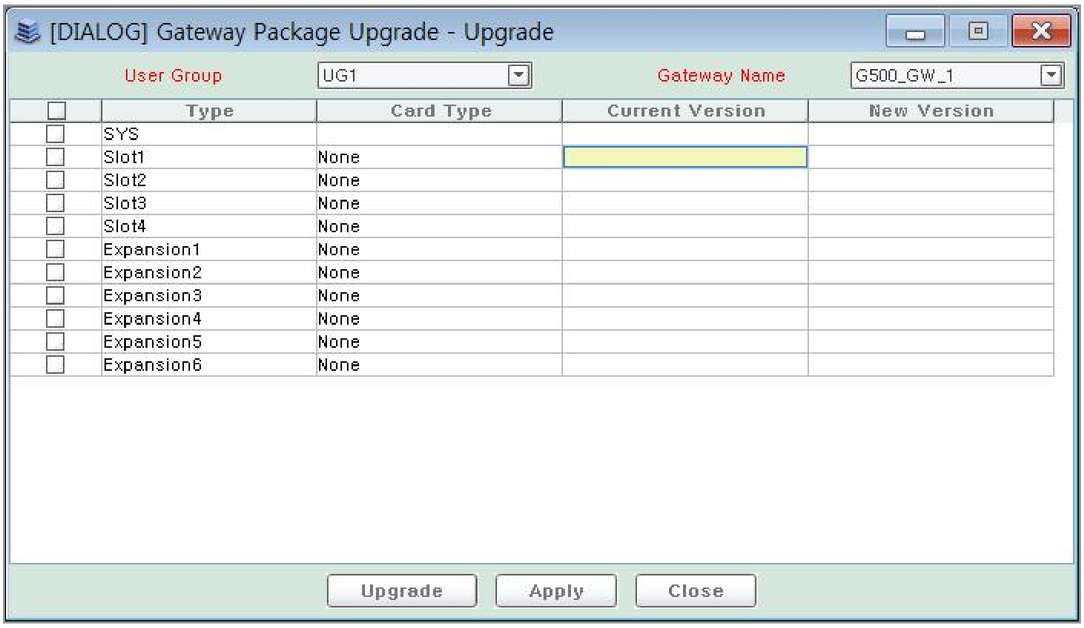
Item	Description
Link State	Shows current link state.
Network Mode	Specify the mode of the network to which the BRI port will connect. (Trunk/Station: Connects to the BRI Station.)
Route Name	Specify the trunk route to which the BRI trunk will belong.
The 1st S0 extension number.	The 1st S0 extension number.
The 2nd S0 extension number.	The 2nd S0 extension number.
Incoming Mode	Specify the incoming mode. - Point to Point Normal - Point to Point DID - Point to Multi Point
Normal Ring Calling Number	Specify the number where a ringing will occur when a call is received to the BRI trunk after the incoming mode has been set to Point to Point Normal.
Calling Party Number Type	Specify the calling party number type. - Unknown - International - National - Network

Item	Description
	- Subscriber - Abbreviate - Extension
Calling Party Number Plan	Specify the calling party number plan. - Unknown - ISDN - Data - Telex - Nation - Private - Extension
Receiving Party Number Type	Specify the receiving party number type. - Unknown - International - National - Network - Subscriber - Abbreviate - Extension
Receiving Party Number Plan	Specify the receiving party number plan. - Unknown - ISDN - Nation
Caller ID Restriction	Select whether to show the originating number to the other party when making a call.

4.2.2.13 Gateway Profile Sync.

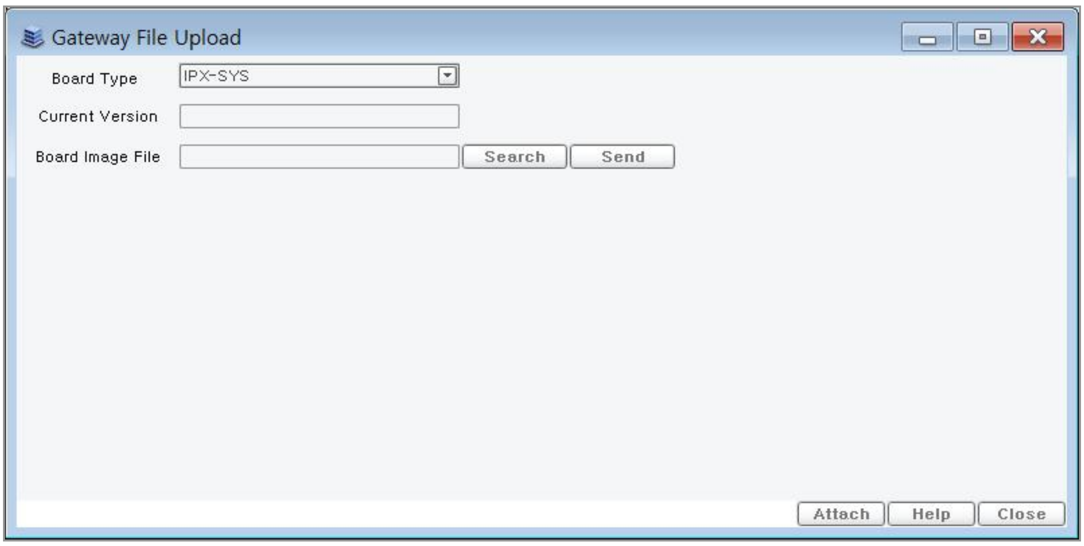
Item	Description
Profile Type	Specify the profile type that will be synced. Select from among All, Normal Mode Profile, and Survival Mode Profile.
Sync. Type	Specify the point in time where the sync will be performed. Select between Day and Now.
Sync. Time (HHMM)	Specify the sync time that will be used if the Sync. Type has been set to Day.
Notification Interval (ms)	The time interval at which multiple GWs are synced in sequence.

4.2.2.14 Package Upgrade



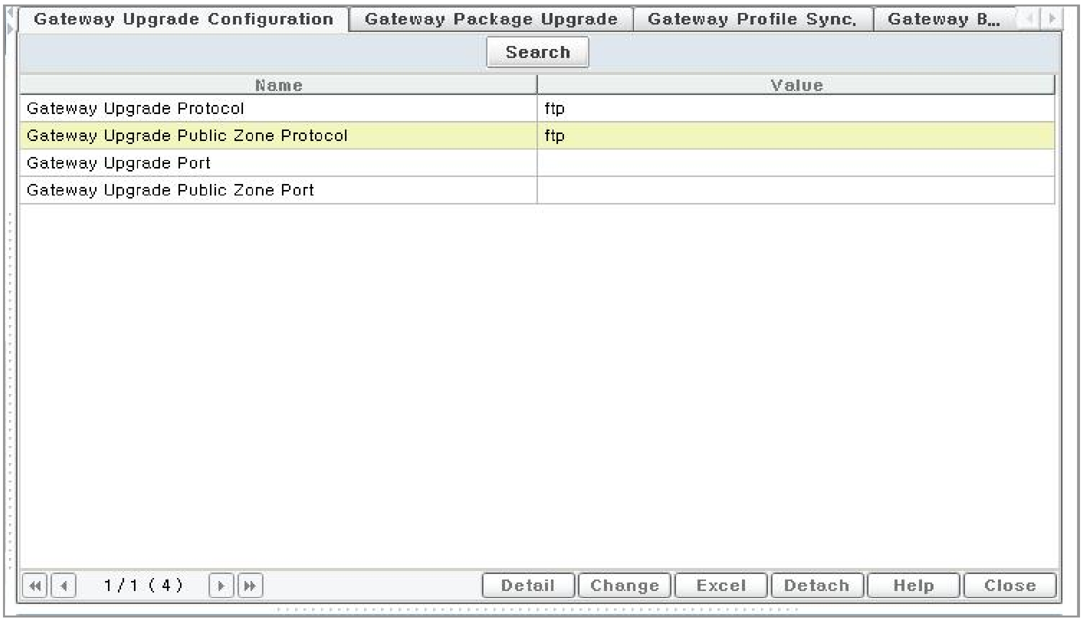
Item	Description
Type	The location of the board whose package will be upgraded.
Card Type	The type of the card whose package will be upgraded.
Current Version	The current version of the board.
New Version	The version of the image file stored in the system with relevance to the type of the board.
Upgrade	Select the location of the board you want to upgrade (check the check box '□') and click 'Upgrade,' and then the board is upgraded.

4.2.2.15 File Upload



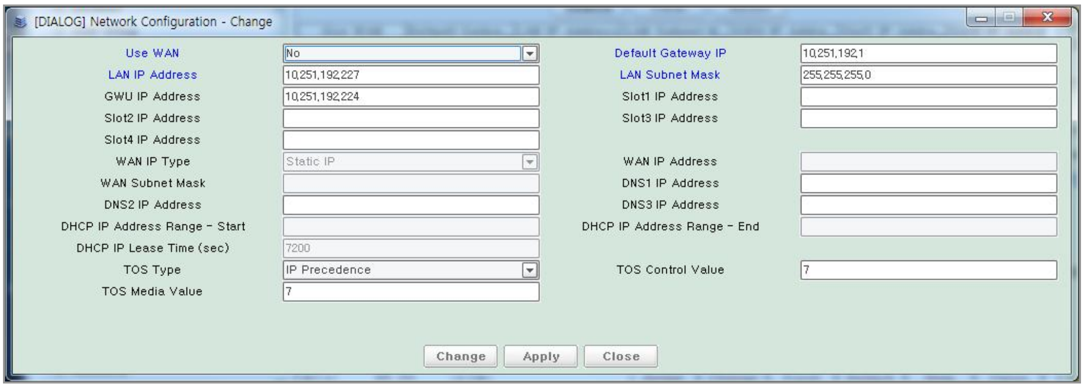
Item	Description
Board Type	Select the board type that matches the image file that want to upload. (IPX-SYS, IPX-ECU, or IPX-PRI.)
Current Version	Displays the version of the image file stored in the system with relevance to the type of the selected board.
Board Image File	Select the board image file stored on your PC by clicking 'Search' and click 'Send' and then the file is uploaded to the IPX-S300B.

4.2.2.16 Upgrade Option



Item	Description
Gateway Version (IPX-G500)	Enter the version of the board image file stored in the IPX-S300B; for example, V1.0.0.0.
Gateway Upgrade Protocol	Select the protocol that will be used for package updates (ftp or tftp).
Gateway Upgrade Public Zone Protocol	Select the Public Zone protocol that will be used for package updates (ftp or tftp).
Gateway Upgrade Port	Enter the port that will be used for package updates (FTP: 21/TFTP: 69; if the field is left blank, the default port is used.)
Gateway Upgrade Public Zone Port	Enter the Public Zone port that will be used for package updates. (FTP: 21/TFTP: 69; if the field is left blank, the default port is used.)
Gateway Upgrade FTP Login ID	Enter the ID that will be used for package updates via FTP. (If the field is left blank, the FTP ID for the IPX-S300B is used.)
Gateway Upgrade FTP Login Password	Enter the password that will be used for package updates via FTP. (If the field is left blank, the FTP ID for the IPX-S300B is used.)

4.2.2.17 Configuring Network



Item	Description
Use WAN	Select whether to use the NAT feature.
Default Gateway IP	The IP address of the default gateway.
LAN IP Address	The IP address of the LAN. (If Use WAN is set to No, the LAN IP address becomes the IP address with which the system connects to a network. (If Use Wan is set to Yes, the LAN IP address is the IP address of the network within the system.)
LAN Subnet Mask	The subnet mask of the LAN IP address.
GWU IP Address	The IP address to be assigned to the GWU board. This must be in the same band frequency as the LAN IP address.
Slot1 IP Address	The IP address of the board to be installed in Slot 1. This must be in the same band frequency as the LAN IP address.
Slot2 IP Address	The IP address of the board to be installed in Slot 2. This must be in the same band frequency as the LAN IP address.
Slot3 IP Address	The IP address of the board to be installed in Slot 3. This must be in the same band frequency as the LAN IP address.
Slot4 IP Address	The IP address of the board to be installed in Slot 4. This must be in the same band frequency as the LAN IP address.
WAN IP Address	The WAN IP address for when using the NAT feature.
WAN Subnet Mask	The subnet mask of the WAN IP address.
DNS IP Address (1-3)	The DNS IP address that will be transmitted by the DHCP client. The DHCP server is by default run by the system.
DHCP IP Address Range-Start	The start address of the IP Pool that will be used by the DHCP server. The DHCP server is by default run by the system.
DHCP IP Address Range-End	The end address of the IP Pool that will be used by the DHCP server. The DHCP server is by default run by the system.
DHCP IP Lease Time (sec)	The lease time for the IP address that is assigned via DHCP.

4.2.2.18 Configuring Port Forwarding

Item	Description
Rule Name	The name of the port forwarding rule that you will create newly or switch to.
Internal IP Address	The destination address of the internal network to which packets that meet the port forwarding rule will be transmitted.
Protocol	The protocol condition for packets to be port forwarded. (TCP or UDP.)
Internal Port	The port number of the destination address of the internal network to which packets that meet the port forwarding rule will be transmitted.
External Port	The destination port number condition for packets that will be port forwarded.

4.2.2.19 Configuring Packet Filtering

Item	Description
Rule Name	The name of the packet filtering rule that you want to create newly or switch to.
Protocol	The protocol condition for packets that will be packet filtered. (TCP or UDP.)
Source IP Address	The source IP address of packets that will be packet filtered.
Destination IP Address	The destination IP address of packets that will be packet filtered.
Destination Port-Start	The start value of the destination port of packets that will be packet filtered.
Destination Port-End	The end value of the destination port of packets that will be packet filtered.

4.2.3 Initializing

To initialize all configurations of gateway, do the followings.

- Connect the serial cable to the console port of system to initialize. (baud rate: 38,400 bps)
- Enter ID and Password to log on.
- Enter “cli”.
- Select ‘4. Factory Reset’.
- It will ask you it is sure. It will remove all configurations. Enter ‘yes’.

```
##################################
#                                #
#        GW Setup Manager        #
#                                #
##################################
 1. Setup basic configuration
 2. Show current configuration
 3. Show version
 4. Factory Reset
 5. Exit
Please select the number : 4

This will clear all configurations. Do you want to continue?
(yes/no) :
```

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

CHAPTER 5. Call Service

This chapter describes the call processing services provided by IPX-S300B and how to configure them.

IPX-S300B provides the following three types of call processing services.

- System features: Determine the overall operation of the system. You can configure a system service for the entire system or a user group.
- User features: Configured for each user.

5.1 System Features

System features are performed according to the configuration of the system, not individual user setting.

5.1.1 Anonymous Call Reject

The anonymous call reject service rejects anonymous incoming trunk calls without caller IDs.
An anonymous call's SIP message has anonymous@anonymous.invalid in the From header.
To use the anonymous call reject service, set 'Anonymous Call Reject' to Anonymous Reject in the **[CONFIGURATION > Trunk Routing > Route]** Trunk Service tap.
To use the no number call reject service, set 'Anonymous Call Reject' to No Number Reject. Also, to use both type rejections, set 'Anonymous Call Reject' to Both Reject.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.1.2 Call Admission Control (CAC)

Since system resources are limited, a service is required to set the maximum number of calls at any one time. The Call Admission Control (CAC) service provided by IPX-S300B includes CAC by call counts, CAC by location bandwidth, CAC by system resources, and CAC by trunk call counts.
All CAC is independently operated by each node.

CAC Override for Emergency Call

All CAC can override for emergency call.
When the emergency call occurs, IPX-S300B drop the other call.
For using CAC override, Trunk need to configured the maximum outbound call and maximum call.

CAC by Call Counts

CAC by call counts restricts calls when the maximum usage ratio set for the maximum number of calls supported by IPX-S300B is exceeded.
If the maximum usage ratio for CAC by call counts is 100, the maximum number of calls supported by IPX-S300B is allowed. The default maximum usage ratio for CAC by call counts is 100.
You can view the maximum number of calls simultaneously supported by IPX-S300B under 'Maximum Call' in the **[CONFIGURATION > Resource > System Capacity]** menu.
You can set the maximum usage ratio for CAC by call counts under 'Local CAC Threshold' in the **[MANAGEMENT > Call Admission Control > Local CAC]** menu.
The number of calls allowed simultaneously by CAC by call counts is calculated by the following formula: (Maximum Call) * (Local CAC Threshold)/100.

CAC by Location Bandwidth

The CAC by location bandwidth service restricts the calls made in excess of the bandwidth set for each location.
When calls are made between users or endpoints in different locations, the system calculates the bandwidth based on the codec used for the calls. Any calls exceeding the maximum bandwidth set for each location are restricted.
You can set the maximum bandwidth for locations under 'Bandwidth' in the **[CONFIGURATION > Location > Location]** menu.
You can view the bandwidth required for each codec under 'Codec Bandwidth' in the **[CONFIGURATION > Resource > System Capacity]** menu. This data cannot be changed.
You can set the maximum usage ratio for CAC by location bandwidth under 'Bandwidth' in the **[MANAGEMENT > Call Admission Control > Location Based CAC]** menu.

CAC by System Resources

The CAC by system resources service restricts calls made in excess of the maximum usage ratio set for the system CPU and memory.
You can restrict calls based on CPU and memory usage.
You can also check the CPU and memory usage level in the following three ways:
Use System Viewer in the lower-left corner of SCM Administrator.
View the real-time usage charts in the **[PERFORMANCE > Main Monitor]** menu.
Use the **[PERFORMANCE > Server Resources > System]** menu.
You can set the maximum usage ratio for CAC by system resources under 'CPU Threshold for Resource Based CAC' and 'Memory Threshold for Resource Based CAC' in the **[MANAGEMENT > Call Admission Control > Resource Based CAC]** menu.

CAC by Trunk Call Counts

The CAC by Trunk Call Counts service restricts calls made in excess of the maximum trunk call count for each route.
You can restrict calls based on inbound call, outbound call, and total trunk call.
You can also 'Maximum call', 'Maximum Inbound Call', and 'Maximum Outbound Call' in the **[CONFIGURATION > Trunk Routing > Route]** Trunk Service tap. If you don't set the maximum count, calls are not restricted.

DID Maximum Limitation

In the case of DID-specific incoming trunk calls, you can specify a maximum allowed number of calls for each ring plan and allow as many as the specified number of calls to be received. You can set this by using the **[CONFIGURATION > Trunk Routing > DID Routing]** menu's 'Maximum Call'.

5.1.3 Least Cost Route (LCR)

IPX-S300B performls the LCR service in various ways.

LCR by Location

The LCR by location feature allows you to assign one of the three LCR methods (priority-based LCR, time based LCR, and Load Balance LCR) for each location of the calling party. You can create route partitions in the **[CONFIGURATION > Trunk Routing > Location Based Routing]** menu. The following items are mandatory. This menu is used for entering the detailed routes in the route partition.

Mandatory Item	Description
User Group	Select a user group to which the route partition belongs.
Location Based Routing Name	Select a route partition in which to enter the data. If you want to create new partition, select <new> and enter the route partition name.
Location Select	- Specify whether to select location

Mandatory Item	Description
	- Disable: do not select location - Enable: specify a location mandatory.
Location	Select a location to set the routes.
Routing Type	Select a type of LCR for the location. - Time-Based-Route: Selects the time-based route select Routing. - Normal-Route: Select to use a route sequence. - Route-Set: Select to use a route set.

To use Location based routing, One of the following items is mandatory, depending on the type of the selected LCR type.

Mandatory Item	Description
Time Based Routing Name	If the LCR type is set to Time Based Routing, select a Time Based Routing to use.
Priority Routing Name	If the LCR type is set to Priority Routing, select a Priority Routing to use.
Load Balance Routing Name	If the LCR type is set to Load Balance Routing, select a Load Balance Routing to use.

Priority Routing

The Priority Routing feature allows automatic selection of alternative routes when the endpoint set as the default LCR is not available for call connections. Routes are assigned with priorities so that the route with the highest priority among those available is selected. You can create route sequences in the **[CONFIGURATION > Trunk Routing > Priority Routing]** menu. The following items are mandatory. This menu is used for entering the detailed routes in the route sequence.

Mandatory Item	Description
User Group	Select a user group to which the route sequence belongs.
Name	Select a route sequence where the data is entered. If you want to create new route sequence, select <new> and enter the route sequence name.
Route Priority	Assign priority to the route. - Direct Route: Specify the top priority route. - Alternative Routes 1 through 8: Select the routes according to their priority levels.
Route Name	Select a route for the route priority level.

The following menu is used to change calling number or called number when make an outgoing call through this route.

Item	Description
Outbound DOD Delete Length	Specifies the length of digits to delete from the first position of the called number for outbound call.
Outbound DOD Insert Digits	Specifies the digits to insert from the first position of the called number for outbound call.
Outbound CLI Delete Length	Specifies the length of digits to delete from the first position of the calling number for outbound call.
Outbound CLI Insert Digits	Specifies the digits to insert from the first position of the calling number for outbound call.
Outbound MCN	Specify MCN rules for outbound call.

Time Based Routing

The time based Routing feature allows each service group to use its own route sequence features based on its time and rate conditions.
You can create special route sequences in the **[CONFIGURATION > Trunk Routing > Time Based Routing]** menu. The following items are mandatory. This menu is used for entering the detailed route sequences into the special route sequence.

Mandatory Item	Description
User Group	Select a user group to which the time based routing belongs.
Time Based Routing Name	Select a time based routing where the data is entered. If you want to create new time-based routing, select <new> and enter the time based routing name.
Service Group	Select a service group to which the time based routing belongs.
Default Route Routing	Select a basic priority routing which will be used when a call of a service group, which is configured as an entry, does not satisfy the conditions of time or rate.. You must select one of the route sequences created in the **[CONFIGURATION > Routing > Priority Routing]** menu.

When entering the data into the time based routing, you can enter the time information optionally as needed. Without this information, the LCR would then behave the same as a route sequence, defeating the purpose.

Item	Description
Day Type 1~4	When LCRs are selected according to time, you can specify up to four types of days in which to reference the route sequences.
Start Time 1~4 End Time 1~4	When LCRs are selected according to time, you can specify the start time and the end time for up to four time periods in which to reference the route sequences.

Item	Description
Priority Routing 1~4	Specify up to four Priority Routing referenced by rate or date.

Load Balance Routing

The feature allows the use of the set routes in an equally distributed manner according to the set ratio. Calls are distributed only between the routes identified as available for calls, and therefore there is no need for configuring alternative routes, as in other LCR methods. You can create load balance routing in the **[CONFIGURATION > Trunk Routing > Load Balance Routing]** menu. The following items are mandatory. This menu is used for entering the detailed routes in the route set.

Mandatory Item	Description
User Group	Select a user group to which the route set belongs.
Load Balance Routing	Select a load balance routing where the data will be entered. If you want to create new load balance routing, select <new> and enter the load balance routing name.
Rate	Specify the default ratio in which to distribute calls between the endpoints configured in the route set. The ratio for each route can be changed in the Change menu.
Route	Select a route to include in the route set.

5.1.4 Call Restriction

IPX-S300B supports the following three types of call restriction policies.

Extension Lock

You can restrict outgoing calls or incoming calls for users. You can also restrict both outgoing and incoming calls. This setting is applied to all calls, whether internal or external. To restrict incoming calls or outgoing calls for users, set 'Extension Lock' in the **[CONFIGURATION > User > Single Phone User]** menu.

- NONE: No restriction is applied (all incoming/outgoing calls allowed).
- Answering-only: All incoming calls are restricted (no incoming calls).
- Dialing-only: All outgoing calls are restricted (no outgoing calls).
- Both: All incoming calls and outgoing calls are restricted (no incoming/outgoing calls).

Route Lock

You can specify whether to use a route for external calls coming through the endpoint connected to the route.
To restrict the use of a route, set 'Route Lock' in the **[CONFIGURATION > Trunk Routing > Route]** Trunk Service tap.

- NONE: All incoming and outgoing calls through the route are allowed.
- Outbound Locked: Outgoing calls are restricted and only incoming calls through the route are allowed.
- Inbound Locked: Incoming calls are restricted and only outgoing calls through the route are allowed.
- All Locked: All incoming and outgoing calls through the route are restricted.

Restriction Policy

Call restriction policies can be applied by analyzing the calling number or called number when external calls are made to the users or external calls are made by the users through the trunk.
The call restriction tables created and configured in the menu described below can be applied to specific users, service groups, or user groups for call restriction.
If multiple call restriction policies are applied to a user, the policies are applied in the priority of user, service group, and user group.
In Tandem call case, Trunk is restricted by incoming trunk's restriction policy.
You can create call restriction tables in the **[CONFIGURATION > Trunk Routing > Toll Restriction List]** menu.

Item	Description
User Group	Select a user group to which the call restriction table belongs.
Name	Select a call restriction table in which the data is entered. If you want to create new toll restriction list, select <new> and enter toll restriction list name.
Restriction Digit	Enter a prefix number to restrict calls. It supports longest prefix match.
Restriction Type	Specify the direction of calls to restrict. - Incoming: Incoming trunk calls are restricted. - Outgoing: Outgoing trunk calls are restricted. - Both: Both the outgoing and incoming trunk calls are restricted.
Restriction Enable	If you want to allow calls for number range with a prefix, set ENABLE.

You can configure time-based restriction policies in the **[CONFIGURATION > Toll Routing > Toll Restriction Policy]** menu.

Item	Description
User Group	Select a user group to which the time-based restriction policy table belongs.
Name	Enter a name for the time-based restriction policy table.
Default Restriction Name	Select a call restriction table to use outside the set time periods.
Ring Plan Schedule	Select a ring plan schedule for this toll restriction policy.
RP1~15 Toll Restriction	Specify the toll restriction list to act on this ring plan.

Call Forward Restriction Policy

When calls are forwarded to external trunk, call forward restriction policy can be applied by analyzing the called number.
The call restriction tables created and configured in the menu described below can be applied in the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu.
If multiple call restriction policies are applied to a user, the policies are applied in order by class of service for user, class of service for service group, default class of service.
You can create call restriction tables in the **[CONFIGURATION > Trunk Routing > Toll Restriction List]** menu.

Item	Description
User Group	Select a user group to which the call restriction table belongs.
Name	Select a call restriction table in which the data is entered. If you want to create new toll restriction list, select <new> and enter toll restriction list name.
Restriction Digit	Enter a prefix number to restrict calls. It supports longest prefix match.
Restriction Type	Specify the direction of calls to restrict. - Incoming: Incoming trunk calls are restricted. - Outgoing: Outgoing trunk calls are restricted. - Both: Both the outgoing and incoming trunk calls are restricted.
Restriction Enable	If you want to allow calls for number range with a prefix, set ENABLE.

You can configure time-based restriction policies in the **[CONFIGURATION > Toll Routing > Toll Restriction Policy]** menu.

Item	Description
User Group	Select a user group to which the time-based restriction policy table belongs.
Name	Enter a name for the time-based restriction policy table.
Default Restriction Name	Select a call restriction table to use outside the set time periods.
Ring Plan Schedule	Select a ring plan schedule for this toll restriction policy.
RP1~15 Toll Restriction	Specify the toll restriction list to act on this ring plan.

Call Transfer Restriction Policy for Outbound call

When calls are transferred to external trunk, call transfer restriction policy can be applied by analyzing the called number.
The call restriction tables created and configured in the menu described below can be applied in the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu.

If multiple call restriction policies are applied to a user, the policies are applied in order by class of service for user, class of service for service group, default class of service.
You can create call restriction tables in the **[CONFIGURATION > Trunk Routing > Toll Restriction List]** menu.

Item	Description
User Group	Select a user group to which the call restriction table belongs.
Name	Select a call restriction table in which the data is entered. If you want to create new toll restriction list, select <new> and enter toll restriction list name.
Restriction Digit	Enter a prefix number to restrict calls. It supports longest prefix match.
Restriction Type	Specify the direction of calls to restrict. - Incoming: Incoming trunk calls are restricted. - Outgoing: Outgoing trunk calls are restricted. - Both: Both the outgoing and incoming trunk calls are restricted.
Restriction Enable	If you want to allow calls for number range with a prefix, set ENABLE.

You can configure time-based restriction policies in the **[CONFIGURATION > Toll Routing > Toll Restriction Policy]** menu.

Item	Description
User Group	Select a user group to which the time-based restriction policy table belongs.
Name	Enter a name for the time-based restriction policy table.
Default Restriction Name	Select a call restriction table to use outside the set time periods.
Ring Plan Schedule	Select a ring plan schedule for this toll restriction policy.
RP1~15 Toll Restriction	Specify the toll restriction list to act on this ring plan.

Outside Work Restriction Policy

When user call to external number during outside work, Outside Restriction Policy can be applied by analyzing the called number.
The call restriction tables created and configured in the menu described below can be applied in the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu.
If multiple call restriction policies are applied to a user, the policies are applied in order by class of service for user, class of service for service group, default class of service.
You can create call restriction tables in the **[CONFIGURATION > Trunk Routing > Toll Restriction List]** menu.

Item	Description
User Group	Select a user group to which the call restriction table belongs.

Item	Description
Name	Select a call restriction table in which the data is entered. If you want to create new toll restriction list, select <new> and enter toll restriction list name.
Restriction Digit	Enter a prefix number to restrict calls. It supports longest prefix match.
Restriction Type	Specify the direction of calls to restrict. - Incoming: Incoming trunk calls are restricted. - Outgoing: Outgoing trunk calls are restricted. - Both: Both the outgoing and incoming trunk calls are restricted.
Restriction Enable	If you want to allow calls for number range with a prefix, set ENABLE.

You can configure time-based restriction policies in the **[CONFIGURATION > Toll Routing > Toll Restriction Policy]** menu.

Item	Description
User Group	Select a user group to which the time-based restriction policy table belongs.
Name	Enter a name for the time-based restriction policy table.
Default Restriction Name	Select a call restriction table to use outside the set time periods.
Ring Plan Schedule	Select a ring plan schedule for this toll restriction policy.
RP1~15 Toll Restriction	Specify the toll restriction list to act on this ring plan.

The Outside Work Restriction Policy is feature for WE VoIP only and it is used at next case.

- When user used WE VoIP is moved to external network from internal network, the Outside Work Restriction Policy can be applied during Outside Work Duration (hour) in **[CONFIGURATION > User > Single Phone User/Multi-Phone User]** menu.
- If above duration is not configured, the Outside Work Restriction Policy cannot be applied.
- The priority of Restriction Policy in Class of Service is like next.
- Call Forward Restriction Policy > Call Transfer Restriction Policy > Outside Work Restriction Policy > mVoIP Restriction Policy > IP Zone A Restriction Policy > IP Zone B Restriction Policy > Hot Spot Restriction Policy > Restriction Policy

Samsung APC is required to use this feature and next configuration is needed to connecting APC.

- Select a ACP to Check Mobile Access Network in **[CONFIGURATION > User Group > Change User Group > Options]** menu.
- Create a APC list in **[CONFIGURATION > Wireless Enterprise > APC List]** menu.

This feature is available, even if there are no Samsung APC. But there are limitations as next.

- If next setting is like No use or Only outgoing call use, this feature cannot used for Android WE VoIP.
  - Network Settings > Public wifi setting
  - Network Settings > mVoIP setting
- Next setting is required to use WE VoIP for iOS. But battery consumption will be faster.
  - Select a Register to Check Mobile Access Network in **[CONFIGURATION > User Group > Change User Group > Options]** menu.
  - Specify a duration for Registration Push on Push Reg-Request Interval (min) in **[CONFIGURATION > User Group > Change User Group > Timers]** menu.

5.1.5 Number Translation

IPX-S300B can translate numbers for incoming calls, outgoing calls and local calls

Calling/Called Number Translation for Inbound Call

IPX-S300B provides two kinds of number translation service for the calling number and the called number of the inbound call. The configuration is served at the **[CONFIGURATION > Trunk Routing > Route]** Number Translation tap.
First, you can make multiple rules to convert number to number in the **[CONFIGURATION > Trunk Routing > Inbound MCN]** menu. After that, select some rules for a trunk as you want to the 'Inbound MCN' in the **[CONFIGURATION > Trunk Routing > Route]** Number Translation tap. Only one matching rule can be applied call by call.

Item	Description
User Group	Select User Group which the number translation for inbound call is defined.
Name	Enter the name of Inbound MCN. When you make Inbound MCN list, you can see these Inbound MCN Names in the **[CONFIGURATION > Trunk Routing > Route]**. This name cannot be changed.
Number Type	Select a number type to convert. - Calling Number: Number Translation is applied to the calling number of the inbound call. - Called Number: Number Translation is applied to the called number of the inbound call.
Find Digits	Specifies the number replaced by another number from the first digit of the number of inbound call. 0-9, '*', '#', 'X', and '?' is possible for Find Digits. '#', '*' is treated the same as the number. 'X' means any number. If you want to use the Find Digits as a prefix not a full number, '?' is followed the desired pattern. - '1234': must be '1234'. - '1XX4': all four digits starting with 1 and ending with 4

Item	Description
	- '1234?': all numbers starting with '1234'. - '1XX4?': all numbers prefixed by four digits starting with 1 and ending with 4. Any digit between 'X's is not allowed but duplicated 'X' are supported. '?' is placed at the last digit of Find Digits. - '1X3X5': 3 is located between 'X' and 'X', so this pattern is not supported. Duplicated 'X' without any interrupt by other digits are possible as like 'XX345', '1XXX5', and '123XX' - '123XX???': Multiple '?' are not supported. If the compared target number matched to several Find Digits, the most exactly matched rule is selected with number pattern and the pattern length. - The compared number is '12345' and there are two Find Digits, '12345' and '123XX'. In this case, the Find Digits, '12345' is determined because it is matched with more digits. - The compared number is '12345' and there are two Find Digits, '123?' and '123XX'. In this case, the Find Digits, '123XX' is determined because it is matched with the length including pattern. - The compared number is '12345' and there are two Find Digits, '123?' and '12345?'. In this case, the Find Digits, '12345?' because its matched digit length is longer.
Modified Digits	Specifies the converted number by the Find Digits. 0-9, '*', '#', 'X', and '?' can be entered to the Modified Digits. The count of 'X' in the Find Digits and that of the Modified Digits must be same. If you use '?' in the Modified Digits, the digits corresponding to the '?' in the Find Digits are reused at the Modified Digits. - The compared number is '1234567'. The matched Find Digits is '123XXX7', and the Modified Digits is '00XXX111'. In this case, the number is converted to '00456111'. - The compared number is '1234567'. The matched Find Digits is '123?' and the Modified Digits is '00?'. In this case, the number is converted to '004567'. - The compared number is '1234567'. The matched Find Digits is '123?' and the Modified Digits is '00'. In this case, the number is converted to '00'

After the number translation by Inbound MCN, simple delete and insert to a number is supported trunk by trunk for inbound call. If delete digits and insert digits are configured both, the delete procedure is precedence over insert procedure.
The number translation mentioned above is set in the **[CONFIGURATION > Trunk Routing > Route]**.

Item	Description
Inbound DID Delete Length	Specifies the length of digits to delete from the first position of the called number for inbound call.
Inbound DID Insert Digits	Specifies the digits to insert from the first position of the called number for inbound call.

Item	Description
Inbound CLI Delete Length	Specifies the length of digits to delete from the first position of the calling number for inbound call.
Inbound CLI Insert Digits	Specifies the digits to insert from the first position of the calling number for inbound call.

Calling/Called Number Translation for Outbound Call

IPX-S300B provides two kinds of number translation service for the calling number and the called number of the outbound call. The configuration is served at the **[CONFIGURATION > Trunk Routing > Priority Routing]** menu **or** **[CONFIGURATION > Trunk Routing > Load Balance Routing]** menu. According to routing path, different number translation can be applied even if calls direct to a same trunk route.
First, you can make multiple rules to convert number to number for outbound call in the **[CONFIGURATION > Trunk Routing > Outbound MCN]** menu. After that, select some rules for a trunk as you want to the 'Outbound MCN' in the **[CONFIGURATION > Trunk Routing > Priority Routing]** menu or **[CONFIGURATION > Trunk Routing > Load Balance Routing]** menu. Only one matching rule can be applied call by call.

Item	Description
User Group	Select User Group which the number translation for outbound call is defined.
Name	Enter the name of Outbound MCN. When you make Outbound MCN list, you can see these Outbound MCN Names in the **[CONFIGURATION > Trunk Routing > Priority Routing & Load Balance Routing]** This name cannot be changed.
Number Type	Select a number type to convert. - Calling Number: Number Translation is applied to the calling number of the outbound call. - Called Number: Number Translation is applied to the called number of the outbound call.
Find Digits	Specifies the number replaced by another number from the first digit of the number of inbound call. 0-9, '*', '#', 'X' and '?' is possible for Find Digits. '#', '*' is treated the same as the number. 'X' means any number. If you want to use the Find Digits as a prefix not a full number, '?' is followed the desired pattern. - '1234': must be '1234'. - '1XX4': all four digits starting with 1 and ending with 4 - '1234?': all numbers starting with '1234'. - '1XX4?': all numbers prefixed by four digits starting with 1 and ending with 4. Any digit between 'X's is not allowed but duplicated 'X' are supported. '?' is placed at the last digit of Find Digits. - '1X3X5': 3 is located between 'X' and 'X', so this pattern is not supported.

Item	Description
	Duplicated 'X' without any interrupt by other digits are possible as like 'XX345', '1XXX5', and '123XX' - '123XX???': Multiple '?' are not supported. If the compared target number matched to several Find Digits, the most exactly matched rule is selected with number pattern and the pattern length. - The compared number is '12345' and there are two Find Digits, '12345' and '123XX'. In this case, the Find Digits, '12345' is determined because it is matched with more digits. - The compared number is '12345' and there are two Find Digits, '123?' and '123XX'. In this case, the Find Digits, '123**' is determined because it is matched with the length including pattern. - The compared number is '12345' and there are two Find Digits, '123?' and '12345?'. In this case, the Find Digits, '12345?' because its matched digit length is longer.
Modified Digits	Specifies the converted number by the Find Digits. 0-9, '*', '#', 'X' and '?' can be entered to the Modified Digits. The count of '*' in the Find Digits and that of the Modified Digits must be same. If you use '?' in the Modified Digits, the digits corresponding to the '?' in the Find Digits are reused at the Modified Digits. - The compared number is '1234567'. The matched Find Digits is '123XXX7', and the Modified Digits is '00XXX111'. In this case, the number is converted to '00456111'. - The compared number is '1234567'. The matched Find Digits is '123?' and the Modified Digits is '00?'. In this case, the number is converted to '004567'. - The compared number is '1234567'. The matched Find Digits is '123?' and the Modified Digits is '00'. In this case, the number is converted to '00'

After the number translation by Outbound MCN, simple delete and insert to a number is supported according to the routing path for outbound call. If delete digits and insert digits are configured both, the delete procedure is precedence over insert procedure.
The number translation mentioned above is set in the **[CONFIGURATION > Trunk Routing > Priority Routing]** or **[CONFIGURATION > Trunk Routing > Load Balance Routing]**.

Item	Description
Outbound DOD Delete Length	Specifies the length of digits to delete from the first position of the called number for outbound call.
Outbound DOD Insert Digits	Specifies the digits to insert from the first position of the called number for outbound call.
Outbound CLI Delete Length	Specifies the length of digits to delete from the first position of the calling number for outbound call.
Outbound CLI Insert Digits	Specifies the digits to insert from the first position of the calling number for outbound call.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Local Based Number Translation

IPX-S300B provides number translation service for the called number of the local call.
This feature is activated in same location and user group.
It can provide when using call-forward, blind-transfer and individual-speed-dial service.

First, you can make multiple rules to convert number to number for local call in the **[CONFIGURATION > Location > Local Based MCN]** menu. After that, select some rules for a location as you want to the 'Local Based MCN' in the **[CONFIGURATION > Location > Location]** menu. Only one matching rule can be applied call by call.

Item	Description
Name	Enter the name of Local Based MCN. When you make Local Based MCN list, you can see these Local Based MCN Names in the **[CONFIGURATION > Location > Local Based MCN]**. This name cannot be changed.
Number Type	Select a number type to convert. Only Called Number is allowed. - Called Number: Number Translation is applied to the called number of the local call.
Find Digits	Specifies the number replaced by another number from the first digit of the number of inbound call. 0-9, '*', '#', 'X' and '?' is possible for Find Digits. '#', '*' is treated the same as the number. 'X' means any number. If you want to use the Find Digits as a prefix not a full number, '?' is followed the desired pattern. - '1234': must be '1234'. - '1XX4': all four digits starting with 1 and ending with 4 - '1234?': all numbers starting with '1234'. - '1XX4?': all numbers prefixed by four digits starting with 1 and ending with 4. Any digit between 'X's is not allowed but duplicated 'X' are supported. '?' is placed at the last digit of Find Digits. - '1X3X5': 3 is located between 'X' and 'X', so this pattern is not supported. Duplicated 'X' without any interrupt by other digits are possible as like 'XX345', '1XXX5', and '123XX' - '123XX???': Multiple '?' are not supported. If the compared target number matched to several Find Digits, the most exactly matched rule is selected with number pattern and the pattern length. - The compared number is '12345' and there are two Find Digits, '12345' and '123XX'. In this case, the Find Digits, '12345' is determined because it is matched with more digits. - The compared number is '12345' and there are two Find Digits, '123?' and '123XX'. In this case, the Find Digits, '123XX' is determined because it is matched with the length including pattern. - The compared number is '12345' and there are two Find Digits, '123?' and '12345?'. In this case, the Find Digits, '12345?' because its matched digit

Item	Description
	length is longer.
Modified Digits	Specifies the converted number by the Find Digits. 0-9, '*', '#', 'X' and '?' can be entered to the Modified Digits. The count of 'X' in the Find Digits and that of the Modified Digits must be same. If you use '?' in the Modified Digits, the digits corresponding to the '?' in the Find Digits are reused at the Modified Digits. - The compared number is '1234567'. The matched Find Digits is '123XXX7', and the Modified Digits is '00XXX111'. In this case, the number is converted to '00456111'. - The compared number is '1234567'. The matched Find Digits is '123?' and the Modified Digits is '00?'. In this case, the number is converted to '004567'. - The compared number is '1234567'. The matched Find Digits is '123?' and the Modified Digits is '00'. In this case, the number is converted to '00'

Number Translation

Each group of users, number translation service, is the ability to convert to number there is a certain pattern number or international number (E.164).
Number translation rules can be created in the **[Configuration> User Group> Change User Group> Number Translation]**.
Operator can be the length of the number translation and adjust it through the **[Minimum Number Translation Length]** from **[Configuration> User Group> Change User Group> Options]**.

The IPX-S300B is allow one '+' character and allow '*' and '#'.
Other character is automatically deleted.
Delete Function is activated for all number.

This feature applies to the called number and calling number of all.
Following items are required for creating **[Number Translation]** Rule.

Item	Description
User Group	select the user group which operator wants to use a number conversion rules.
Check Digit	It is the digit to compare the incoming number.
Delete Length	The IPX-S300B remove digits from the number that matches the **[Check Digit]**. Maximum value allow 40.
Insert Digit	After Delete Length, Insert Digits activated. Maximum Insert Digit is allowed 20 digits.

5.1.6 Call Button

The call button feature allows directing multiple calls to the one phone number when the user's number is a single device and the call waiting service is in use.
If you are using a phone with programmable buttons, such as a Samsung phone, you can assign up to eight call buttons. If there are eight call buttons, the phone can control up to eight calls simultaneously. If there is no call button on the phone, it is treated as having two call buttons.
To create call buttons on the phone, set 'Key' to CALL in the **[CONFIGURATION > User > Phone Key Programming]** menu.

5.1.7 CLI Number for Internal Call

Basically, a user makes a call by dialing Extension number within same User Group. Extension Number should be assigned in unique within a User Group and it used as CLI when make a call and Destination number when finding a user.
But, exceptional cases like below, different number may be used instead of Extension Number.

- In case that a user makes a call to the users in a different User Group by dialing just Extension number, the call will be failed. Extension Number is a reachable number within same User Group.
  When a user makes a call to the users in a different User Group, the user should dial Extension Number followed by User Group Code. When a user makes a call among different User Groups, both Calling and Called Number will be displayed as 'User Group Code + Extension Number'.
- Within same user group, extension number is used normally, but if there is 'Service Group Code' in a User group, the users in same Service Group can make a call each other just using 'Service Group Local Number' that is called 'Station Number'.
  In other words, Extension Number includes 'Service Group Code' and 'Service Group Local Number'.
  To use 'Service Group Local Number', create a service group and specify 'Service Group Code' and then assign users to the service group. Only the Extension Numbers the prefix is starting with Service Group Code can be assigned to the service group. 'Service Group Local Number' field in each User menu displays the digits which exclude the Service Group Code from Extension Number automatically. If there is Service Group Code, this field displays nothing. The Users within same service group can make a call using Extension number or Service Group Local Number.
- To display CLI with not Extension Number but 'Service Group Local Number', operator should configure 'Service Group Local CLI Number' to 'Station Number' in **[CONFIGURATION > User > Single Phone User]** or **[CONFIGURATION > User > Multi-Phone User]**.
  If a user does not want to use Extension Number as CLI, use 'Send Extension Number' instead.

'User Group Code + Extension Number' and 'Service Group Local CLI Number' are reachable number but 'Send Extension Number' is not reachable and display only, so be careful to set this field.

5.1.8 CLI Number for Outbound Call

Several 'CLI Number' are served for the outbound call to a trunk as follows.

- 'CLI Number' is a virtual number; it can be used for CLI but receiving call with the CLI number as called number is not available. Each 'CLI Number' can be configured independently, but the priority is shown in the following description.
- The same CLI number can be used for multi-users assigned to a Multi-Extension Phone even if the users have different CLI numbers for each user. The CLI for the Multi-Extension Phone is set to 'Send CLI Number' in the **[CONFIGURATION> User> Multi-Extension Phone]** menu.
- The 'Send CLI Number' of Multi-Extension Phone is not set, each user can use their own 'Send CLI Number'. Single phone users use 'Send CLI Number' in the **[CONFIGURATION> User> Single Phone User]** menu. In the case of multi-line users, 'Send CLI Number' is configured in the **[CONFIGURATION> User> Multi-Phone User]** menu.
- If the 'Send CLI Number' of a user is not set, the 'CLI Number' of service group can be used. The 'CLI Number' is configured in the **[CONFIGURATION > User Group > Service Group]** menu.
- If the 'Send CLI Number' of a service group is not set, the 'CLI Number' of a user group can be used. The 'CLI Number' is configured in the **[CONFIGURATION > User Group > Service Group]** menu.
- If there is no configuration for CLI number, the extension number by prefixing the 'Outbound CLI Prefix' in the **[CONFIGURATION> Trunk Routing> Route]** Number Translation tap can be used.

The default CLI number is the extension number without any configuration as describes above.
The CLI number for outbound call, by default, is applied in the order mentioned above. Depending on your needs, you can designate the CLI number type for a specific trunk.
It is served by 'Forced Send CLI Number' in the **[CONFIGURATION > Trunk Routing > Route]** menu. If you set the 'Forced Send CLI Number' to none, outbound CLI number has priority according to the order mentioned above.

5.1.9 CLI Name for Outbound Call

It is a virtual calling name for outbound call at particular use. It is mainly used to insert a number to the 'Display Name' in the From header of the SIP message by the request of SIP ISP. There are two cases as described below.
Each user has 'Send CLI Name' in the **[CONFIGURATION > User > Single Phone User]** or **[CONFIGURATION > User > Multi-Phone User]** menu, which can be used for 'Display Name' for a specific trunk. If 'Send CLI Name for User' in the **[CONFIGURATION > Trunk Routing > Route]** Number Translation tap set to Send CLI Name for the trunk, the 'Send CLI Name' of each user is used for display name, or 'Extension Name' in the **[CONFIGURATION > User > Single Phone User]** and **[CONFIGURATION > User > Multi-Phone User]** menu is used by default.
Tandem Call means the outbound call which is originating from a trunk. To configure the 'Display Name' of the tandem call, the 'Send CLI Name for Inbound Cal' is provided.
If the 'Send CLI Name for Inbound Call' in the **[CONFIGURATION > Trunk Routing > Route]** Number Translation tap set to Receive CLI Number, the original received calling number is copied to the 'Display Name' regardless of the number translation.

5.1.10 Internal CLI Name

It is a virtual calling name for inbound call at particular use. It is mainly used to insert a name to verify which GW's call.
This name displays only in subscriber and service group. This name does not display in tandem call.
To use the Internal CLI Name, specify in the **[CONFIGURATION > Trunk Routing > Route]** Number Translation tap **[Send CLI Name for Internal Call]** menu.

5.1.11 Premium CID

The Premium CID service provides supplementary information about an opposite user. Photo is provided basically and other information can be configured with the priority.

To use Premium CID, License is necessary and each user has to set Premium CID to Enable.
You can check the Premium CID license in the **[CONFIGURATION > Miscellaneous > Activation Key]** menu, which is named Directory Service. Each user using Premium CID is configured Premium CID option to Enable which cannot exceed the license.

If the Premium CID is out of license, it is not working.

The information provided by Premium CID is as follows.

- Additional 6 Information on opposite user
- Location Service
- Photo

Additional 6 information is configured in the **[CONFIGURATION** > **User Group** > **Change User Group** > **Premium CID List]**. The information is displayed according to the priority of each data whenever the phone supports this feature.

Item	Description
User Group	Select a user group for which the Premium CID is provided.
Priority 1~6	Configure the supplementary information which can be up to 6. - Name: show a Name in the **[CONFIGURATION > User > User Profile]** menu. - Position: show a Position in the **[CONFIGURATION > User > User Profile]** menu. - Department: show a Department in the **[CONFIGURATION > User > User Profile**] menu. - Company: show a Company Name in the **[CONFIGURATION > User Group > Change User Group > Information]** menu. - Reserv1~Reserv4: is defined in the **[CONFIGURATION > User Group > User Profile Field Name]** and is configured in the **[CONFIGURATION > User > User Profile]** menu.

If Mobile Phone Location Service is set to Yes in the **[CONFIGURATION** > **User Group** > **Change User Group** > **Options],** IPX-S300B include location information to the Premium CID. Each phone can be display the location by the Update Location (Private/Wi-Fi Hotspot/mVoIP) option in the **[CONFIGURATION** > **Wireless Enterprise** > **Mobile Phone Profile]**, which is included in a profile.

Photo file for each user can be changed in the **[CONFIGURATION** > **User** > **User Profile]** menu. Single photo file can be updated with Administrator, Personal Web page and FMC Phone. In this case, the file name is changed to an application Id. You can change a file name only. If you want to change photo files concurrently, refer the Photo File Management chapter.

5.1.12 CLI Service

The Calling Line Identification (CLI) service notifies the user of the caller's phone number and name for incoming calls.

Calling Line Identification Presentation (CLIP)

The Calling Line Identification Presentation (CLIP) service displays the caller's phone number and name on the called user's phone for incoming calls.

Calling Line Identification Restriction (CLIR)

The Calling Line Identification Restriction (CLIR) service restricts display of the caller information for the calls made by the user.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

To use the CLIR feature, click **[Activate]** and enable 'Caller ID Block' in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu.
If a user set with CLIR calls another user set with CLIP, the calling user's information is restricted to the called user, as CLIR has precedence over CLIP.

CLI Routing

The CLI routing feature allows special processing of incoming trunk calls according to the caller number.
When entering a calling number, you can use wild cards (entered by X) to enter multiple numbers at a time.
Incoming trunk calls with caller numbers only and without caller names can be supported by caller name.
Incoming calls from specified callers can be rejected.
Incoming calls from specified callers can be assigned called numbers; regardless of the called DID numbers. Called numbers can also be assigned by time period. Time periods are defined by ring plans. For more information, see the section on 'Ring Plans'.
You can configure CLI routing using the **[CONFIGURATION > Trunk Routing > CLI Routing]** menu. The following items are specified.

Item	Description
User Group	Select a user group for which the CLI routing is defined.
CLI Number	Enter a caller number for incoming calls given special treatment. This is used as an identifier in the CLI routing table and cannot be changed. If necessary, you must delete it and enter it again.
CLI Name	You can enter the caller's name for the selected caller number and service the calls by the name.
Call Reject	Specify whether the incoming calls with the selected caller number is rejected.
ACD Queuing Level	Specify a level of priority for a call queues in ACD. You can select between Level 0 and Level 9. If higher the number, higher the priority and a waiting time to connect an agent is shorter.
Default Destination	You can specify a called number for incoming calls from the selected caller number, regardless of the DID number if the call is received outside the time periods defined by ring plans 1 through 15 or if the called number is not specified in the ring plan.
Ring Plan Schedule	Select a ring plan schedule for this toll restriction policy.
RP1~RP15 Destination	You can specify a called number for incoming calls from the selected caller number, regardless of the DID number if the call is received within the time periods defined by ring plans 1 through 15.
Logging	Specify whether to logging a SPAM Call. It logs with three cases like normal case, with announcement case, and without announcements case.

It is often necessary to register a lot of CLI routing information at once. Using SCM Administrator's batch feature, you can prepare the CLI routing information offline in an Excel spreadsheet and batch register the information with IPX-S300B.
You can batch register the CLI routing information prepared in an Excel spreadsheet using the **[Tool > Customer Data Import/Export > CLI Routing]** menu in the upper-left corner of SCM Administrator.
For more information on the Customer Data Import/Export, see the data import/export section of '2.3 Changing All Users Data'

5.1.13 Direct Inward Dialing (DID) Routing

The Direct Inward Dialing (DID) routing feature allows incoming trunk calls to be directed to different called numbers according to the DID number.
When entering a DID number, you can use wild cards (entered by X) to represents the multiple numbers at a time.
The system also performs translation of the DID number, the translated DID number can be used as the called number.
The called number according to the DID number can be assigned with the user number, the hunt group number, the ACD group number, and various feature codes, including the VMS access code and access code + external number. They can also be assigned by time period. Time periods are defined by ring plans. For more information, see the section on 'Ring Plans'.
You can configure DID routing using the **[CONFIGURATION > Routing > DID Routing]** menu. The following items are specified.

Item	Description
User Group	Select a user group to which the calls are directed.
DID Number	Enter a DID number for incoming trunk calls. This is used as an identifier in the DID routing table and cannot be changed. If necessary, you must delete it and enter it again.
DID Name	Specify a name for the DID number. The DID name makes it easy to identify the number dialed by the caller.
Default Destination	Specify a called number to which the incoming calls with the selected DID number is directed when calls are received outside the time periods defined in ring plans 1 through 15 or if the called number is not specified in the ring plan. If the called number is set to 'B', a translated DID number is used as the called number. In the case of 'E' for the called number, tandem call is not allowed. If the call is not termination at IPX-S300B, the call is rejected with invalid number response. However, the tandem calls by Call Forward or Transfer is not restricted.
Delete Length	DID number can be translated to be used as the called number. Here, you can specify the number of digits to delete from the first part of the DID

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
	number.
Insert Digit	DID number can be translated to be used as the called number. Here, you can specify the digits to insert into the first part of the DID number.
MOH ID	A specific on-hold tone can be played when incoming calls with the selected DID number are put on hold. Here, you can specify the ID of the MOH file to play for each DID number.
ACD Queuing Level	Specify a level of priority for a call queues in ACD. You can select between Level 0 and Level 9. If higher the number, higher the priority and a waiting time to connect an agent is shorter.
Ring Plan Schedule	Select a ring plan schedule for this toll restriction policy.
RP1~RP15 Maximum Call	You can specify a maximum incoming call limitation each Ring Plan. If Maximum Call is blank, it means no limitation.

It is often necessary to register a lot of DID routing information at once. Using SCM Administrator's batch feature, you can prepare the DID routing information offline in an Excel spreadsheet and batch register the information with IPX-S300B.
You can batch register DID routing information prepared in an Excel spreadsheet using the **[Tool > Customer Data Import/Export > CLI Routing]** menu in the upper-left corner of SCM Administrator.
For more information on the Customer Data Import/Export, see the data import/export section of '2.3 Changing All Users Data'

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.1.14 Directory Service

It provides the ability to dial by searching the number and name for the extension, hunt group and system speed dial at the phones.
You can search directory by name or number.

- When you try to search a user by number, User's extension number, Hunt Group number should be used.
- When you try to search a user by name, User name, Hunt Group name and System Speed dial name should be used.
- When you search by number, it required at least 3 digits, search by name, it requires at least two characters.

You should configure options for Directory Service using the **[CONFIGURATION > Phone Setting > Directory Service Control]** menu.

Item	Description
Use Directory Service	Select option for Using Directory Service In System. This option applies for all users.
Directory Service Protocol	Select interworking Protocol between IPX-S300B and phone. - HTTP: Use HTTP Protocol - HTTPS: Use HTTPS Protocol
Directory Service HTTP Port	Select interworking Port between IPX-S300B and phone. If you select HTTP Protocol, you must set the Port Number 80. Otherwise, Selecting HTTPS is required Port Number 443.

You can configure Directory Service Display Option in phone using the **[CONFIGURATION > Service > Directory Service Display]** menu. The following items are specified.

Item	Description
User Group	It displays the User Group of a user. It will be displayed by default.
Display Name	It displays the Extension name of a user. It will be displayed by default.
Display Number	It displays the Extension Number of a user. It will be displayed by default.
Display Position	Specify whether to display the position of a user.
Display Department	Specify whether to display the department of a user.
Display Number Type	Specify the type of a user. It will be displayed among User, Hunt Group, Speed Dial.
Display Mobile Number	Specify whether to display the Mobile Number of a user. This Option is provided for user only.

5.1.15 Direct Trunk Selection

IPX-S300B supports Direct Trunk Selection (DTS) to use a specific trunk, which provides services according to a status of the designated trunk.

To use DTS feature, the following configuration is required.

- Set a feature code for 'Direct Trunk Selection' in the **[CONFIGURATION > Service > Feature Service > Feature Code]** menu.
- Check the 'Class of Service' for a user which uses a DTS feature. The service permission of 'Direct Trunk Selection' has to be included.
- Set 'DTS Mode' to Enable and enter 'Access Number' mandatorily when you make a Route for DTS in the **[CONFIGURATION > Trunk Routing > Route]** interworking tap.
  A same number with an extension cannot be used for 'Access Number'.
- If you need, 'Toll Restriction Policy', 'Dial Tone' and 'Dial Plan' can be applied for DTS feature.

How to use the DTS feature is described below.

- A User dials to 'DTS feature code + Access Code to select trunk' and Enter a called party number by DTMF after Gateway is connected. One-touch dial key is also available if 'DTS feature code + Access Code to select trunk + a called party number' pre-sets to the one-touch dial key.
- To control for some user to use a DTS trunk call, apply a 'Toll Restriction Policy'. For more detail about 'Toll Restriction Policy', refer to Restriction Policy part of Call Restriction feature.
- Account for DTS can be adjusted by setting 'SMDR Timer'/'SMDR Time (sec)' in the **[CONFIGURATION > Miscellaneous > System Options]**. The connect start time is delayed 10 seconds if 'SMDR Timer' sets to Enable and 'SMDR Time (sec)' sets to 10 sec.

DTS Trunk Status	LED Display	Key Service	Description
Idle	Off	Outgoing to the trunk	Makes a call to the trunk.
Seize	On	Outgoing to the trunk	Ends a call after playing an announcement for Busy state.
Incoming	Blink rapidly	Pickup	Picks up the call
Outgoing	On	Outgoing to the trunk	Ends a call after playing an announcement for Busy state.
Hold	Blink slowly	Retrieve	Retrieves the hold call.
Conv	On	Barge-in with tone	Make a conference with the call if 'Barge-in with tone' is permitted for the user.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.1.16 Emergency Group

In case of emergency, if user makes a call with pre-configured emergency group number, system automatically make a call to the members with emergency type access code.
When a member of emergency group response the call. The system automatically calls to the manager of the emergency group. The manager can only hear.
To use Emergency Group feature, 'Emergency Type Access Code' is required in the **[CONFIGURATION > Trunk Routing > Access Code]** menu.
And the following Items are required in the **[CONFIGURATION > Service > Group Service > Emergency Group]** menu.

Item	Description
User Group	Select a user group for which the emergency group will be configured
Location	Select a location for which the emergency group will be configured
Group Number	Enter a number for the emergency group A user dials this number when emergency.
Group Name	Specify a name for the Emergency group. The emergency group name is useful for identifying the purpose of the emergency group.
Ring Type	Specify a ring type for manager's phone in emergency
Manager	Specify a manager for the Emergency group IPX-S300B automatically dial to managers in emergency. It can be assigned to maximum of 3 managers.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.1.17 History Log

IPX-S300B provides history logging capability for the events like a SPAM call, incoming call, paging on answer call, wakeup call, feature set, registration fail, dispatch call etc.
You can configure Logging Service, using the **[CONFIGURATION > User Group > Change User Group > Detailed Event Logging Option]** Menu.
Administrator can review logs at **[Performance > Detailed History]** menu.
Follows are detailed description of **[CONFIGURATION > User Group > Change User Group > Detailed Event Logging Option]** menus.
All options should be configured, and it was disabled by default.

Item	Description
User Group	It displays the user group.
SPAM Call Logging	Specify whether to logging the SPAM calls. It logs a call from external only.
Call Logging	Specify whether to logging the incoming/outgoing/all calls from extension or trunk.
Paging On Answer Call Logging	Specify whether to logging the Paging On Answer calls.
Wakeup Call Logging	Specify whether to logging the Wakeup calls.
Wakeup Set Logging	Specify whether to logging the Wakeup Setting calls.
Call Block Feature Logging	Specify whether to logging the calls blocked by Cal Block feature.
ACL Block Logging	Specify whether to logging the calls blocked by ACL feature. This option is not available.
Register Logging	Specify whether to logging for Register Failure. - Disable: It does not logging. - Phone Only: It logging the Register failure for Phone only. - All Devices: It logging the Register failure for all devices.
Notice Board Logging	The administrator can collect the notice board history by this option.

SPAM Call History

SPAM Call History feature provides a user to logging the call with specific SPAM number from external judged as SAPM call.
You should enable the ‘SPAM Call Logging’ option of **[CONFIGURATION > User Group > Change User Group > Detailed Event Logging Option] and [CONFIGURATION > Trunk Routing > CLI Routing]** menu.
You can review the history record of SPAM call **at [Performance > Detailed Event History > Incoming Call History]** menu.

Item	Description
Date	It displays the time when the SPAM Call event occurred.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Call Type	It displays the type of a call. Reject: It just reject the call. - Announcement: It plays an announcement for SPAM Call. - Routing: Normal Case, Not SPAM Call
User Group	It displays the User Group name of a user when the event occurred.
Extension	It displays the Extension number of a user when the event occurred.
CLI Number	It displays the CLI Number of the call from trunk when the event occurred.
DID Number	It displays the DID Number of the call from trunk when the event occurred.
Destination	Only if call type is routing case, Called Number is displayed.

Call History

Call History feature provides a user to logging the call.

You should enable the 'Call Logging' option of **[CONFIGURATION > User Group > Change User Group > Detailed Event Logging Option]** and **[CONFIGURATION > Service > Feature Service > Service Activation]** menu.

Item	Description
Call Logging	Specify whether to logging an incoming call. - No Answer Only (Incoming): it logging the incoming calls not answered only - All Call (Incoming): It logging the incoming calls. - No Answer Only (Outgoing): it logging the outgoing calls not answered only - All Call (Outgoing): It logging the outgoing calls. - No Answer Only (Both): it logging the calls not answered only - All Call (Both): It logging the whole calls.

Administrator can review the history record of Incoming Call at **[Performance > Detailed Event History > Call History]** menu.

Item	Description
Date	It displays the time when Incoming Call event occurred.
Call Type	It displays the type of a call. - Abandon: A caller cancelled the call. - Answer: the call was connected. - No Answer: The call was not answered by user. - Call Failure: The call was not complete by other reason.
Direction	It display the Call Attempt Direction. - Incoming: Extension receive a call

Item	Description
	- Outgoing: Extension makes a call.
User Group	It displays the User Group name of a user when the event occurred.
Extension	It displays the Extension number of a user when the event occurred.
CLI Number	It displays the CLI Number of the call from trunk when the event occurred. If direction is outgoing, It displays subscriber number.
DID Number	It displays the DID Number of the call from trunk when the event occurred. If direction is outgoing, It displays destination number.
Duration	It displays talk duration time. If Not Answer case, Ring time is displayed.

Paging On Answer Call History

Administrator can review the history record of Paging On Answer Call at **[PERFORMANCE > Detailed Event History > Paging On Answer History]** menu.

Item	Description
Date	It display the time when Paging On Answer Call event was occurred.
Call Type	It displays five call Types for Paging On Answer Call. - COMMAND: The caller of the Paging On Answer call. - NOANSWER: In case that member has no answered. - ANSWER: In case that member was answered. - BUSY: In case that member was busy. - UNREACHABLE: In case that member was unreachable.
User Group	It displays the User Group name of a user when the event occurred.
Extension	It displays the Extension number of a user when the event occurred.
Group Number	It displays the Group Number of the Paging On Answer Group.
Ring Number	It displays the members of the group who is in ringing state.

Wake-up Call History

Administrator can review the history record of wake-up set, reset and working history at **[PERFORMANCE > Detailed Event History > Wakeup Call History]** menu.

Item	Description
Date	It displays the time when the event occurred.
Call Type	It displays three call Types for Wake-up Call. - FAIL: In case of alarm call failure. - NOANSWER: In case that a user does not answered for an alarm call. - ANSWER: In case that user answered alarm call.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
User Group	It displays the User Group name of a user when the event occurred.
Extension	It displays the Extension number of a user when the event occurred.

Feature Set History

Administrator can review the history record of wake-up set, reset and working history at **[PERFORMANCE > Detailed Event History > Feature Set History]** menu.

Item	Description
Date	It displays the time when Feature Set event occurred.
Feature	It displays the Types of Features.
User Group	It displays the User Group name of a user when the event occurred.
Event	It displays the Types of events.
Method	It displays who generate this event.
Index	It displays the index of Wake-up call.
Destination	It displays the destination number for the services like Forward All, Follow Me, and Remote Office.
Login ID	It displays the Login ID of a user for this event.
Login IP	It displays the Login IP Address of a user for this event.

Register History

Administrator can review the history record of Incoming Call at **[PERFORMANCE > Detailed Event History > Register History]** menu.

Item	Description
Date	It displays the time when the event occurred.
Device Type	It displays the type of device when the event occurred.
User Group	It displays the User Group name of a user when the event occurred.
Device	It displays the Name of device when the event occurred.
Reason	It displays the reason when the event occurred.
IP	It displays the IP Address of device when the event occurred.
ID	It displays the Authentication ID of device when the event occurred.
MAC	It displays the MAC Address of device when the event occurred.

Notice Board History

Administrator can review the history record of Notice Board at **[PERFORMANCE > Detailed Event History > Notice Board History]** menu.

Item	Description
Author	This item represents the author, which is application user name who makes the posts.
Action	This item represents a user action to the posts. - Add - Delete - Modify - Expire
Type	This item represents the posts type. - Normal - Emergency
Title	This item represents the posts title.
Write Time	This item represents a time when the posts is made.
Start Time	This item represents the starting time when the posts is noticed.
End Time	This item represents the close time when the posts is noticed.

5.1.18 Home Worker Support

IPX-S300B provides the same user services to home workers. The services are provided whether the home worker’s phone is connected to the public IP network or to a private IP network within NAT (an IP router).
For a phone connected to a private IP network on NAT, the source port number used for transmitting SIP messages must be symmetric or can be set as symmetric.

When IPX-S300B Is on the Public IP Network

If IPX-S300B is connected to the public IP network, services can be provided to home workers with additional settings. In general, if both the phones on a call are connected to the public IP network, they exchange voice and video data (RTP/SRTP) directly. If either of the two or both are connected to private IP networks on NATs, they exchange voice and video data through IPX-S300B using SBC feature.

When IPX-S300B Is on a Private IP Network

If IPX-S300B is connected to a private IP network on NAT, a SBC system is required.
To use the SBC, set configuration in the **[CONFIGURATION > Network > SBC]** menu.
When a call is made between two home workers’ phones connected to the public IP network outside IPX-S300B’s NAT, they exchange the voice and video data directly.
But when a call is made between a phone connected to the private IP network inside IPX-S300B’s NAT and a home worker’s phone, the data is exchanged through the SBC.

5.1.19 Hotel Service

There are additional menus for hotel services.

> **NOTE**
>
> Refer to the 'SCM Hotel Service Guide' for detailed information.

5.1.20 PMS Interface

IPX-S300B should interworks with PMS for hotel services.

> **NOTE**
>
> Refer to the 'SCM Hotel Service Guide' for detailed information.

5.1.21 Hot Desking

The hot desking feature allows a user to log in from a phone shared by multiple users.
The user can use a phone in the logged out status to enter his/her ID and password to log in and use the phone as his/her own phone until logged out. The user can log out when using Samsung phones' menu. If the user leaves the phone without logging it out, it is automatically logged out after a set period of time, preventing unauthorized users from using the phone. The default login expiration time is 8 hours. If the user is already logged in through a phone but requests for login again through another phone using the same user ID, the new login request is processed by logging the previous phone out.

To use the hot desking feature, the following items must be configured.
In the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu, hot desking must be enabled in the 'Service Permission' section. In the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu, hot desking service must be enabled for the extension number used as a hot desking phone. Here, you can enter the login expiration time in 'Hot Desk Expire Time (hour)'.

5.1.22 Hot Line and Warm Line

The hot line feature allows automatic connection to a specified number when the handset of the selected phone is lifted. If the call is connected without delay when the handset is lifted, it is called a hot line. If the call is automatically connected when the handset is lifted but no number is dialed for a set period of time, it is called a warm line.
To use the hot line feature, the following items must be configured.
In the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu, hot line must be enabled in the 'Service Permission' section.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

In the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu, hot line service must be enabled for the extension number used as a hot line phone. Here, you can enter 'Hot Line Expire Time' to use the warm line service.

5.1.23 Hunt Group

The hunt group service directs calls received by the pilot number of a hunt group appropriately within the hunt group using various routing methods.
When calls are received for a hunt group, the available member list excludes members unable to receive calls because they are unavailable, are subject to incoming call restriction policies, have logged out of the system, have user information that is locked out, or do not have their phones connected.
Called parties for calls received for a hunt group are determined in the following four ways.

Sequential

The call is always directed to the first member in the hunt group. The call is directed to the next member only if the previous member is on the line or unavailable.

Circular

When a call is received for the hunt group, the call is directed to the person on the hunt group member list after the one who answered the previous call. If the member the call is directed is on the line or unavailable, the call is re-directed to the next member.

Parallel or Broadcast

The call is directed to all the members in the hunt group. When one of the members answers the call, the call is canceled for all other members.

Random

The call is randomly directed to one member in the hunt group at random.
When a call is received for the hunt group, the call is directed to a selected member. But if the call is not answered for a specified period of time, the call is canceled and directed to the next member randomly.

Equal

All extension numbers of members can be pilot number. And the call is distributed sequentially starting from pilot number if pilot number is member extension.

You can create hunt groups using the **[CONFIGURATION > Service > Group Service > Hunt Group]** menu. The following items are mandatory.

Item	Description
User Group	Select a user group for which the hunt group is defined.
Group Number	Enter a pilot number used for calling the hunt group.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Group Name	Specify a name for the hunt group. The hunt group name is useful for identifying the purpose of the hunt group.
Hunt Type	Specify the method of determining the member to whom the incoming hunt group call is directed. You can use one of the four methods described above.
Hunt Member	Select a user to add as a member of the hunt group.

When configuring a hunt group, you can enter the following data as necessary.

Item	Description
All Busy/Unavailable Destination	You can specify an alternative called number to use when the call cannot be directed to any member in the hunt group.
Member No Answer Time (sec)	Specify a period of time during which the call must be answered by current ringing member in the hunt group. If the call is not answered during this period, it is directed to the next member in the hunt group.
No Answer Destination	You can specify an alternative called number to use when the call is not answered by any member in the hunt group for a specified period of time. This works in the same way as no answer call forwarding.
No Answer Time (sec)	Specify a period of time during which the call must be answered by any member in the hunt group. If the call is not answered during this period, it is directed to the specified alternative called number.
External Ringback Tone Use	Specify whether to use external ringback tone. - None: Do not use Ringback tone - Internal: Use Ringback tone only for the originating call from user - External: Use Ringback tone only for the inbound call from the trunk. - Both: User Ringback tone for both, the originating call from user and the inbound call from the trunk
External Ringback Tone Server	Select Application Server for Ringback tone. To select Ringback tone server, make External Ringback tone Server at **[CONFIGURATION > Application > Other Application Server]** menu first.
Group Member Call Forward	If a member has been set the call forwarding, it is operated as follows. - Override: Member's call forwarding is ignored and member is received a call. - Next Member: It is directed to the next number in the hunt group.
Voice Mail Server	Select the application server to use when call is forwarded by no answer.
Auto Attendant Plan Schedule	Select a Ring Plan Schedule to use when call is transferred to Auto Attendant.
Maximum Waiting Call Count	If there are no idle member in hunt group, specify a max count to wait in hunt group. If there are no idle member, the wait call is waited in hunt group during

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
	No Answer Time. After No Answer Time, the wait call is transferred to All Busy/Unavailable Destination. If All Busy/Unavailable Destination is blank, the wait call is disconnected.
Second External Ringback Tone Server	Select Application Server for Ringback tone. When External Ringback Tone Server was unregisterd, it is serviced. To select Ringback tone server, make External Ringback tone Server at **[CONFIGURATION > Application > Other Application Server]** menu first.

'Maximum Waiting Call Count' is used to waited in hunt group during 'No Answer Time (sec)' when there are no idle member in hunt group. 'Maximum Waiting Call Count' is worked each IPX-S300B system based. If IPX-S300B is configured to Active-Active system, received call is waited in hunt group of each node. It is possible to waited incoming call to each extension group per node until 'Maximum Waiting Call Count'.

Hunt Group Login/Logout

You can temporarily prevent a hunt group member from receiving incoming calls for the hunt group. If a member logs out of his or her hunt group, the member is excluded from the available member list, and incoming hunt group calls are not directed to the member.
If the member logs in again, he or she can receive incoming hunt group calls normally.
To use the hunt group login/logout feature, you must set the 'Station Group-In/Station Group-Out' feature code in the **[CONFIGURATION > Service > Feature Service > Feature Code]** menu.
If the user dials feature code + hunt group number to set Station Group-In/Station Group-Out, the member is logged in to or out of the selected hunt group only. If the user only dials the feature code without a hunt group number to set Station Group-In/Station Group-Out, the member is logged in to or out of all his or her hunt groups.

Hunt Group Member Service

When incoming call is to the pilot number, without regard to whether the setting of the service-call forwarding, multi-ring, the group call forwarding-hunt group member causes the incoming call. If you change the Enable an item of 'Hunt Group Member Services' from the menu **[Configuration > User Group > Change User Group > Options]**, hunt group member who setting call forwarding, multi-ring, the group call forwarding, works as configured service for the user.
Services for these users if that will run only if the hunt group that you set 'sequential', 'distribution', 'random' except 'parallel'. If the first hunt group member does not respond for no answer time, the call is terminated without forwarding to next members. The hunt group members who set 'Absent', 'DND', 'lock', 'log out', are excluded from the available list of the hunt group regardless of the setting of 'Hunt Group Member Service'.

5.1.24 Location Codec Negotiation

Codec negotiation takes place between two Internet phones when a call is made between them using the SIP protocol. IPX-S300B can change codec priority by intervening in the codec negotiation process.
You can specify the default audio codec, the default video codec, and the announcement codec for each location in order to change the priority of the codec list between the calling phone and the called phone.
Since codec negotiation only takes place between Internet phones, codec negotiation priority settings cannot be forced on calling or called phones which do not support the selected codec.
You can set the preferred codec for locations in the **[CONFIGURATION > Location > Location]** menu.

Item	Description
Name	Specify a name to identify the location.
Bandwidth	Specify the maximum available bandwidth for the location.
Intra-Location Video Codec	Select a video codec preferred for the Intra-location.
Inter-Location Video Codec	Select a video codec preferred for the Inter-location.
Intra-Location Audio Codec	Select an audio codec preferred for the Intra-location.
Inter-Location Audio Codec	Select an audio codec preferred for the Inter-location.
Intra-Location Forced Codec	Select specific audio codec by Administrator for the Intra-location.
Inter-Location Forced Codec	Select specific audio codec by Administrator for the Inter-location.
Announcement Codec	Select an announcement codec preferred for the location.

Location Codec

You can designate Calling Location, Called Location, Video codec, Forced Audio Codec and Audio Codec at Location codec. You can specify the default audio codec, the default video codec, and the announcement codec for each location in order to change the priority of the codec list between the calling phone and the called phone. When you select forced Audio Codec, IPX-S300B sends only audio codec in Forced Audio Codec field to the destination.
Since codec negotiation only takes place between Internet phones, codec negotiation priority settings cannot be forced on calling or called phones which do not support the selected codec.
You can set the preferred codec for locations using the **[CONFIGURATION > Location > Location Codec]** menu.

Item	Description
Calling Location	Select calling location.
Called Location	Select called location.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Video Codec	Select a video codec preferred for the location.
Forced Audio Codec	Select specific audio codec by Administrator for the location.
Audio Codec	Select an audio codec preferred for the location.

Audio Codec

When IPX-S300B receives a sent message, IPX-S300B finds the default audio codec set for the calling phone's location in the audio codec list of the sent message and moves it to the top-priority position of the list before resending the message. Codec negotiation is performed as the specified audio codec is selected by the called phone if it can service the codec.
This process is skipped if the default audio codec set for the location is not found in the codec list of the sent message.

Video Codec

When IPX-S300B receives a sent message, IPX-S300B finds the default video codec set for the calling phone's location in the video codec list of the sent message and moves it to the top-priority position of the list before resending the message. Codec negotiation is performed as the specified video codec is selected by the called phone if it can service the codec.
This process is skipped if the default video codec set for the location is not found in the codec list of the sent message.

Announcement Codec

IPX-S300B can connect its sound source to the phone put on hold during a call and play an on-hold tone. It can also play an announcement for the phone of the calling party in case of call failure or any other errors.
When IPX-S300B sends a sent message for connecting the sound source to the phone for which an on-hold tone is played while the call is put on hold or an announcement is played for an error, IPX-S300B moves the announcement codec set for the phone's location to the top-priority position of the audio codec list in the sent message before resending the message.
Codec negotiation is performed as the specified audio codec is selected by the called phone if it can service the codec.

Default System Codec

Default System Codec menu is used to set the default value of the codec used in Location menu. If codec is set to System Codec in Location menu, it is mean to use the Default System Codec.

5.1.25 Multiple Appearance

There are the following two multiple appearance services: assigning one phone number to multiple phones, or assigning multiple phone numbers to one phone.
These two services can be set independently or collectively.
You can configure multi-device and multi-number using the **[CONFIGURATION > User > Multi-Extension Phone]** menu and the **[CONFIGURATION > User > Multi-Phone User]** menu.

Multi-Device

The multi-device service assigns one user (phone number) to multiple devices (phones). IPX-S300B performs the service regardless of the phone being used. One phone number can be assigned to maximum of 32 phones regardless of the phone type.

Multi-Number

The multi-number service assigns multiple users (phone numbers) to one device (phone). The service is performed collectively by IPX-S300B and by the phone. Since the phone must be able to differentiate the lines and select them, the maximum number of phone numbers allowed varies by the phone type. In case of SMT-i5243, the phone with the most service capacity, up to 8 phone numbers can be assigned per phone.

5.1.26 Music On Hold

When a call is put on hold, IPX-S300B can connect its built-in sound source and play a tone or music for the phone or the trunk.
To use music on hold (MOH), administrator should set ‘MOH Enable’ to ‘Enable’ and specify the ID of the sound source file in ‘MOH ID’ in the **[CONFIGURATION > User Group > Change User Group > Information]** menu.
It is necessary to enable or disable MOH for each user group because there is limited number of channels for IPX-S300B’s built-in sound source device. When there are too many calls put on hold, the MOH may not be played for some of the calls. In this case, it might be better not to play the MOH at all than to have the MOH played for some calls while the MOH is not played for other calls.

To support a specific music source per each subscribers, set ‘MOH Announcement ID’ in the **[User > Single Phone User]** or **[User > Multi-Phone User]** menu.
The MOH ID of subscriber has priority over that of user group.

5.1.27 Missed Call Display

Missed Call Display is a notice service to the phone to inform the call is answered by other user. IPX-S300B can activate or de-activate the function for the Multi-Device Calls, Hunt Group, Multi-ring calls and Pickup.

Missed Call Display by Multi-Device

Multi-Device means that several phones share single phone number. When a user makes a call to an extension of Multi-Device, multiple phones are ringing. Set **[Missed Call by Multi-Device]** in the **[CONFIGURATION > User Group > Change User Group > Options]** menu to 'Display Enable' if you want to leave a Missed Call Display for no-answer phones when one phone of them is answered. Default value for the 'Missed Call by Multi-Device' is 'Display Disable'.

Missed Call Display by Hunt Answer

IPX-S300B provides Missed Call Display feature for Hunt Service. How to display the missed call depends on the Hunt Type. If Hunt Type is 'Parallel', Missed Call Display is determined by Answer action according to the 'Missed Call by Hunt Answer' in the **[CONFIGURATION > User Group > Change User Group > Options]**. For other hunt type, for example 'Sequential', 'Circular' and 'Random', Missed Call Display is determined by the option regardless of the Answer action.

Missed Call Display by Multi-ring Answer

When a user which activates Multi-ring service and has Multi-ring members receives a call, the phones of the user and members are ringing. By the 'Missed Call by Multiring Answer' option in the **[CONFIGURATION > User Group > Change User Group > Options]**, you can determine the missed call display on the no-answer phones when one phone is answered. Default value is 'Display Disable'. If you want to leave the missed call play on the no-answer phones, change the option to 'Display Enable'.

Missed Call Display for Pickup

When a user picks up another user's incoming call IPX-S300B does not leave a missed call display on the original called party phone by default. If you want to display missed call to the original called party, change 'Missed Call for Pickup' to 'Display Enable' in the **[CONFIGURATION > User Group > Change User Group > Options]** menu.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.1.28 Operator Group

An operator group is a special hunt group made up of members who act as operators. Different hunt groups can be specified as operator groups by time periods. In general, a user selected as a member of an operator group uses the phone in parallel to a PC application or uses a PC-based soft phone.
You can configure operator groups using the **[CONFIGURATION > Service > Group Service > Operator Group]** menu. When specifying an operator group, you can select one of the hunt groups configured in the **[CONFIGURATION > Service > Group Service > Hunt Group]** menu.

Item	Description
User Group	Select a user group for which the operator group will be defined.
Access Number	Enter a pilot number used for calling the operator group.
Operator Name	Specify a name for the operator group. The operator group name is useful for identifying the purpose of the operator group.
Default Operator Group	Specify a hunt group to use as the operator group when ring plans 1 through 15 are not applied.
Ring Plan Schedule	Select a ring plan schedule for this toll restriction policy.
RP1~15 Operator Group	Specify a hunt group to use as the operator group when ring plans 1 through 15 are applied.

Operator Recall

When a call is transferred or parked and then directed back to the original called party but the connection was not established, the operator recall service directs the call back to the operator group.
Calls are redirected to the operator group in the following cases:
Reconnection failure after call transfer failure: When call transfer fails for an incoming call for an extension number, the call is redirected to the extension number. Here, if the call is not answered by the extension number, it is redirected to the operator group.
Reconnection failure after call park: When an incoming call for an extension number is put on hold (call park) and the call is not answered for a set period of time, the call is redirected to the extension number. Here, if the call is not answered by the extension number, it is redirected to the operator group.
To use operator recall, set 'Operator Recall' to Enable in the **[CONFIGURATION > User Group > Change User Group > Options]** menu.
To use operator recall, the following time settings are required in the **[CONFIGURATION > User Group > Change User Group > Timers]** menu.

Item	Description
Park Recall Time (sec)	When a trunk call is put on hold and if it is not answered for a set period of time, it will be redirected to the user who put it on hold.

Item	Description
	Specify this time period.
Transfer Recall Time (sec)	When a trunk call is forwarded and if it is not answered for a set period of time, it will be redirected to the user who forwarded it. Specify this time period.
Operator Recall Time (sec)	When a call is redirected to the user after it was forwarded or put on hold and if the call is not answered for a set period of time, the call is redirected to the operator group. Specify this time period.
Operator Recall Wait Time (sec)	When a call is redirected to the user after it was forwarded or put on hold and if the user is busy, the call is redirected to the operator group after waiting for a set period of time for the user to become available. Specify this time period.
Recall Disconnect Time (sec)	If a call redirected to the operator group is not answered by any member of the operator group for a set period of time, the call is terminated. Specify this time period.

Operator Transfer Recall

When call transfer by operator fails or the call is not answered, it is redirected to the operator.

Calls are redirected to the operator in the following cases:

- No answer by transferee: When an operator transfers a call to a user, the call is redirected to the operator if the call is not answered during 'Transfer Recall Time'.
- Call reject by transferee: When an operator transfers a call to a user, the call is redirected to the operator if the transferee activates DND service.

To work Operator Transfer Recall, 'Transfer Recall' has to set to 'Enable All' or 'Enable Operator Only' in the **[CONFIGURATION > User Group > Change User Group > Options]** menu.

5.1.29 Registration Status

You can view the current registration statuses of the phone, trunk route, and application server in the [Performance > Registration Status > Registration Status] menu. You can view user group information and user information. If you have not selected a specific user group, you are shown the registered users from all groups. You can view Register Type, Register State, and, if the user is unregistered, Reason for Unregister.

Item	Description
User Group	Shows the user, the trunk route, and the user group to which the application belongs.

Item	Description
User Info	User Information of SIP URI.
Register State	Shows the registration status of the phone/gateway. If registration is unnecessary, Not Required is displayed. If the phone/gateway is found to be already registered, Reg is displayed. Unreg means that the phone/gateway is unregistered, calls cannot be made to the phone/gateway.
IP Address	Shows the IP address displayed on the REGISTER message's Contact header.
Port	Shows the port number displayed on the REGISTER message's Contact header.
MAC address	Shows the MAC address of the phone. It shows only the MAC addresses of phones that include MAC information in the REGISTER message.
Register Type	Shows the user, the trunk route, and the registration type of the application.
Protocol	Shows the transmission protocol (TCP, UDP, or TLS).
Expires	Shows the time when the phone's registration expires.
Regi/Unreg Time	Shows the time when the phone was registered.
User Agent Info	Shows the REGISTER message's USER-AGENT.
Phone Type	Show the subscribers' phone type
Source IP	Shows the source IP address of the IP packet that transmits the REGISTER message.
Source Port	Shows the source port number of the IP packet that transmits the REGISTER message.
Fail Reason	Shows the reason for no registration if the phone is unregistered.

5.1.29.1 Registering a SIP Phone Using the SIP REGISTER Message

Registering a SIP Phone

The SIP phone periodically transmits the REGISTER message to the IPX-S300B, and the IPX-S300B authenticates it.

If authentication has succeeded, you can view the SIP phone in the **[Performance > Registration Status > Registration Status]** menu, and Register Type is displayed as Subscriber.

Deregistering Upon the SIP Phone's Request

The SIP phone transmits the IPX-S300B the REGISTER message with the expire time set as '0,' and the IPX-S300B deregisters the SIP phone.

Deregistration by Expire Time

The IPX-S300B periodically checks the SIP phone's expire time. If the SIP phone has not re-transmitted the REGISTER message within the expire time, the system waits for the time specified in the **[CONFIGURATION > User Group > Change User Group > Timer Option]** menu's 'Expire Time Deviation (sec)' and then deregisters the phone.

5.1.29.2 Registering a Gateway FXS Using the SIP REGISTER Message

Registering a Gateway FXS

The gateway FXS periodically transmits the REGISTER message to the IPX-S300B, and the IPX-S300B authenticates it. If authentication has succeeded, you can view the gateway FXS in the **[Performance > Registration Status > Registration Status]** the menu, and Register Type is displayed as Subscriber.

Deregistering Upon the Gateway FXS's Request

The gateway FXS transmits the IPX-S300B the REGISTER message with the expire time set as '0,' and the IPX-S300B deregisters the gateway FXS.

Deregistration by Expire Time

The IPX-S300B periodically checks whether the gateway FXS's registration has been renewed. If the gateway FXS has not re-transmitted the REGISTER message within the expire time, the system waits for the time specified in the **[CONFIGURATION > User Group > Change User Group > Timer Option]** menu's 'Expire Time Deviation (sec)' and then deregisters the gateway FXS.

5.1.29.3 Registering a SIP Gateway Using the SIP REGISTER Message

Registering a SIP Gateway

The SIP gateway periodically transmits the REGISTER message to the IPX-S300B, and the IPX-S300B authenticates it. If authentication has succeeded, you can view the SIP gateway in the **[Performance > Registration Status > Registration Status]** the menu, and Register Type is displayed as EndPoint RegReceive.

Deregistering Upon the SIP Gateway's Request

The SIP gateway transmits the IPX-S300B the REGISTER message with the expire time set as '0,' and the IPX-S300B deregisters the SIP gateway.

Deregistration by Expire Time

The IPX-S300B periodically checks the SIP gateway's expire time. If the gateway FXS has not re-transmitted the REGISTER message within the expire time, the system waits for the time specified in the **[CONFIGURATION > User Group > Change User Group > Timer Option]** menu's 'Expire Time Deviation (sec)' and then deregisters the SIP gateway.

5.1.29.4 Registering a Trunk Route Using the SIP REGISTER Message

Registering a Trunk Route

If 'Register Type' under the Basic Configuration tab in the [CONFIGURATION > Trunk Routing > Route] menu has been set to Send REGISTER, the IPX-S300B transmits the REGISTER message to the 'Proxy Server' address under the Basic Configuration tab in the **[CONFIGURATION > Trunk Routing > Route]** the menu. If necessary, the system registers the trunk route after receiving a 200 OK message through authentication.
The registered information can be viewed in the **[Performance > Registration Status > Registration Status]** menu, and Register Type is displayed as 'EndPoint RegSend'.

Deregistering a Trunk Route

The IPX-S300B periodically transmits the REGISTER message, and, if it receives an abnormal response to the message, attempts to re-transmit the message as many times as 'Register Retry Count' under the SIP Register Configuration in the **[CONFIGURATION > Trunk Routing > Route]** menu. If all retransmission attempts fail, the registration gets canceled, it attempts to make a request for registration again after waiting for the 'Register Retry Interval (sec)' under the SIP Register Configuration tab in the **[CONFIGURATION > Trunk Routing > Route]** menu.

If 'Keep Alive' under the SIP Register Configuration tab in the **[CONFIGURATION > Trunk Routing > Route]** menu is set to Enable with the trunk being already registered, the system periodically transmits the OPTIONS message. If there is no response to the OPTIONS message transmitted, the system attempts to re-transmit the message as many times as **'Keep Alive Retry Count' under the SIP Register Configuration tab in the [CONFIGURATION > Trunk Routing > Route]** menu. If all retransmission attempts fail, it transmits the REGISTER message to check registration state again. The response to the REGISTER message is processed in the same way as above.
If the license displayed in the **[CONFIGURATION > Resource > License]** has expired, the system cannot transmit the REGISTER message anymore, and the trunk route gets deregistered by the expire time.

5.1.29.5 Registering a Trunk Route Using the SIP OPTIONS Message

Registering a Trunk Route

If 'Register Type' under the Basic Configuration tab in the [CONFIGURATION > Trunk Routing > Route] menu is set to None, and **'Keep Alive' under the SIP Register Configuration tab in the [CONFIGURATION > Trunk Routing > Route]** is set to Enable, the system periodically transmits the OPTIONS message to the 'Proxy Server' address under the Basic Configuration tab in the **[CONFIGURATION > Trunk Routing > Route]** menu. If the system receives a 200 OK message in response to the OPTIONS message, it registers the trunk route. The registration information can be viewed in the **[Statistics/Performance > Phone/Gateway Register State]** menu, and Register Type is displayed as 'EndPoint Option'.

Trunk Route Deregistration

If there is no response or a failure message occurs to OPTIONS messages, retry as many times as ‘Keep Alive Retry Maximum Number’ specified in **[CONFIGURATION > Trunk Configuration > Trunk Route]** menu in SIP Registration Configuration tab, and deregister after all retries fail.
If deregistered, wait for as long as ‘Retry Wait Time (Sec)’ specified in **[CONFIGURATION > Trunk Configuration > Trunk Route]** menu in SIP Registration Configuration tab, and retry sending OPTIONS messages.

5.1.29.6 Application Registration using SIP OPTIONS

Application Server Registration

IPX-S300B periodically transmits OPTIONS messages to an application server if ‘Enable Keep Alive’ is enabled in the application server menu in **[CONFIGURATION > Application]** menu. If 200 OK message is received in response to OPTIONS messages, the application server will be registered.

Application Server Deregistration

If there is no response or a failure message occurs to OPTIONS messages, retry as many times as ‘Keep Alive Retry Maximum Number’ specified in the application server menu in **[CONFIGURATION > Application]** menu. Deregister after all retries fail.
If deregistered, wait for as long as ‘Retry Pause Time (Sec)’ specified in the application server menu in **[CONFIGURATION > Application]** menu, and retry sending OPTIONS messages.

5.1.30 Ring Plans

When processing calls on a PBX, it is often necessary to provide different services for different days of the week or time of the day. Different services also may be required for public holidays. To accommodate such needs, the services are configured for different days of the week, different time of the day, and different dates. But the problem is that the settings become too complicated.
IPX-S300B provides different services for different days of the week, different time of the day, and different dates by utilizing a feature known as ring plans. IPX-S300B can be created a ring plan schedule by maximum 300. IPX-S300B supports a total of 16 ring plans, including the 15 ring plans (ring plans 1 through 15) which can be assigned their own dates, days of the week and time of the day, and the default ring plan which is used when none of the former 15 ring plans is applied.
The maximum counts of ring plan schedule are as follows.

System Capacity	Count of Ring Plan Schedule
512 User System	Maximum 20

Calendar Exceptions

When User Group is created, calendars of twenty years are created by default.
Specify holidays for the year and special days for the site in the calendars.
You can make detailed calendar settings using the **[CONFIGURATION > Time Schedule > Calendar Exceptions]** menu.

Item	Description
User Group	Select a user group for which the calendar will be changed.
Year	Specify a year for which the calendar will be changed.
Day Type	Specify a type of the day to set. You can use one of the following types of days. - Holiday1: Set a type 1 holiday. - Holiday2: Set a type 2 holidays. - User1: Set a special day for type 1 site. - User2: Set a special day for type 2 sites.
Date	Specify a date of the day to set.

Ring Plan Schedule

A ring plan schedule is a table containing data which specifies ring plans by days of the week, dates, and time of the day. To configure ring plan, select ring plan which you want to set, then drag time table in the **[CONFIGURATION > Time Schedule > Ring Plan Schedule]**. The ring plan schedule is used for CLI Routing, DID Routing, Group Call Forward, and Operator Group. The default ring plan schedule is created automatically when user group is created.

Item	Description
User Group	Select a user group for which the ring plan schedule will be configured.
Name	Specify name of ring plan schedule.
ID	Specify ID of ring plan schedule. ID is used for the ring plan override function.
Current Ring Plan	Current ring plan is displayed.
Default Ring Plan	Display whether or not to default ring plan scheduling of user groups. If this is Yes, this is default ring plan schedule in user group.

Ring Plan Override

IPX-S300B provides a manual override service which allows temporary use of a particular ring plan regardless of the current time. When using ring plan override, you can use the override temporarily by specifying an expiration time or use it permanently by not specifying an expiration time.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Once a permanent ring plan override is set, you can delete the ring plan override list created in the **[CONFIGURATION > Time Schedule > Ring Plan Override]** menu or change 'Override Ring Plan' to 'None' to clear the ring plan override and use the ring plans again. You can set ring plan override using the **[CONFIGURATION > Time Schedule > Ring Plan Override]** menu.

Item	Description
User Group	Select a user group for which the ring plan override will be configured.
Ring Plan Schedule	Select a ring plan schedule for this toll restriction policy.
Override Ring Plan	Specify a ring plan to override. You can select None or one from Ring Plan 1 through Ring Plan 15. Selecting None clears the ring plan override.
Expire Time	Set the time when the ring plan override is cleared and the ring plans are put in effect again. Leaving this option empty allows the ring plan override to stay in effect permanently.

You can change the current ring plan by feature code manually. To use this function, Ring Plan Override-Set and Ring Plan Override-Cancel feature code must be set in **[CONFIGURATION > Service > Feature Service > Feature Code]** menu.
To use ring plan override, dial next number in phone.

- To set Ring Plan Override:
  Ring Plan Override-Set Feature Code + Ring Plan (00~15) + Ring Plan Schedule ID
- To cancel Ring Plan Override:
  Ring Plan Override-Cancel Feature Code + Ring Plan (00~15) + Ring Plan Schedule ID

If ring plan schedule ID is not dialed, set ring plan override for default ring plan schedule.

Service Schedule

A service schedule is a table containing data which specifies service plans by days of the week, dates, and time of the day. There is a service schedule of up to 16 per user group. To configure service plan, select service schedule which you want to set, then drag time table in the **[CONFIGURATION > Time Schedule > Service Schedule]**. The current status of service schedule can be confirmed in **[CONFIGURATION > Time Schedule > Service Schedule]** menu. The service schedule is used for system call forward.

Item	Description
User Group	Select a user group for which the ring plan schedule will be configured.
Service Schedule	Select a service schedule.
Service Schedule Status	Display current service schedule status.

5.1.31 Group Call Forward

This service is used for forwarding all incoming calls for the phone in a group to another number according to the ring plan. If Group Call Forward is activated by feature code or by 'Forced Forward Number' set in the **[CONFIGURATION > Service > Group Call Forward]** menu, it overrides the forward number from ring plan. Group Call Forward can follow the ring plan only when forced Group Call Forward is deactivated. You can check feature code for Group Call Froward in the **[CONFIGURATION > Service > Feature Service > Feature Code]** menu. Group Call Forward is configured in the **[CONFIGURATION > Service > Group Call Forward]** menu.

Item	Description
User Group	Select a user group for which the group call forward will be configured.
Name	Enter the name of group call forward.
Ring Plan Schedule	Select a ring plan schedule for this toll restriction policy.
RP1 Forward Number~RP15 Forward Number	Enter the forward Number by ring plan.
Forced Forward Number	Enter the forward Number if you want to override the Forward Number by ring plan.
User Member Toll Restriction	Set to 'Enable' if you want to apply Toll Restriction to the members of the Call Forward Group.
Assigned Group Number	Enter Call Forward Group ID to indicate the Call Forward Group. It is used to set Forced Forward Number of the Call Forward Group with Feature Code.
Group Member	Select the members for the call forward group.

5.1.32 Service Group Local Number

Normally, call setup is made with an extension number. Service Group Local Number Feature provides a shorter number to call within a service group. An extension number in a IPX-S300B consists of 2 parts: the service group code and the service group local number. Each service group local number in a service group is unique and can be used directly without the service group code. This service group local number can have meaning only when Service Group Code in the **[CONFIGURATION > User Group > Service Group]** menu is configured. IPX-S300B provides a phone display method for a service group local number. An extension number or a service local number can be used for phone display.
It is configured at the service group local CLI number in the **[CONFIGURATION > User > Single Phone User]** menu. This information is provisioned to the Samsung phone and used to determine phone number display.
The followings are limitations about a service group local number.
A service group code in the **[CONFIGURATION > User Group > Service Group]** menu can be changed only when no users are assigned to the service group.

To use a service group local number, a service group code for the user must be configured. You can check the service group local number in the **[CONFIGURATION > User > Single Phone User]** menu. It is displayed automatically.
When a user is assigned to a service group which has a service group code, the extension number of the user has to start with the service group code. An extension consists of the service group code and the service group local number.
A service group local number is available in the service group. When you setup a call to other service group, use an extension number for the user.

5.1.33 System Call Forward

This service performs call forwarding based on the system settings regardless of the call forwarding settings of users. The call forward feature can have limitation by Restricted Call Forward in the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu. If Restricted Call Forward is enabled, call cannot be forwarded to trunk.

Preset Call Forward All

This service performs preset call forward all based on the settings made by the system administrator even if preset call forward all is not set by users.
When administrator sets preset call forward all for users, this feature can be used to forward the call to the voice mail system for voice announcement, etc.
If the user has set call forwarding, the user settings have precedence over the system call forward settings. In other words, if the user has set call forward all, the system preset call forward preset all setting is ignored.
You can use the system preset call forward all feature by enabling ‘Preset Call Forward All’ for selected extension numbers in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu. If you select a service schedule when activate the service, call be forwarded by service schedule plan. Service schedule plan is got ‘Set’ status and ‘Unset’ status. System call forward is working only when service schedule plan’s status is ‘Set’.
Service schedule can be configured in **[CONFIGURATION > Time Schedule > Service Schedule]** menu.

Preset Call Forward Busy

This service performs call forward preset busy based on the settings made by the system administrator even if call forward preset busy is not set by users.
When the called user is busy, this feature can be used to forward the call to the voice mail system for voice announcement, etc.
If the user has set call forwarding, the user settings have precedence over the system call forward settings. In other words, if the user has set call forward all or call forward preset busy, the system call forward preset busy setting is ignored.
You can use the system call forward preset busy feature by enabling ‘Call Forward Preset Busy’ for selected extension numbers in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu.

If you select a service schedule when activate the service, call be forwarded by service schedule plan. Service schedule plan is got 'Set' status and 'Unset' status. System call forward is working only when service schedule plan's status is 'Set'. Service schedule can be configured in **[CONFIGURATION > Time Schedule > Service Schedule]** menu.

Preset Call Forward No Answer

This service performs call forward preset no answer based on the settings made by the system administrator even if call forward preset no answer is not set by users.
When the called user does not answer a call, this feature can be used to forward the call to the voice mail system for voice announcement, etc.
If the user has set call forwarding, the user settings have precedence over the system call forward settings. In other words, if the user has set call forward preset no answer, the system call forward preset no answer setting is ignored.
You can use the system call forward preset no answer feature by enabling 'Call Forward Preset No Answer' for selected extension numbers in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu. If you select a service schedule when activate the service, call be forwarded by service schedule plan. Service schedule plan is got 'Set' status and 'Unset' status. System call forward is working only when service schedule plan's status is 'Set'. Service schedule can be configured in **[CONFIGURATION > Time Schedule > Service Schedule]** menu.

Preset Call Forward Unreachable

This service performs preset call Forward Unreachable based on the settings made by the system administrator even if preset call Forward Unreachable is not set by users.
When administrator sets preset call Forward Unreachable for users, this feature can be used to forward the call to the voice mail system for voice announcement, etc.
If the user has set call forwarding, the user settings have precedence over the system call forward settings. In other words, if the user has set call Forward Unreachable, the system preset call forward preset unavailable setting is ignored.
You can use the system preset call Forward Unreachable feature by enabling 'Preset Call Forward Unreachable' for selected extension numbers in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu. If you select a service schedule when activate the service, call be forwarded by service schedule plan. Service schedule plan is got 'Set' status and 'Unset' status. System call forward is working only when service schedule plan's status is 'Set'. Service schedule can be configured in **[CONFIGURATION > Time Schedule > Service Schedule]** menu.

Preset Call Forward DND

This service performs preset call forward DND based on the settings made by the system administrator even if preset call forward DND is set by user.
When administrator sets preset call forward DND for users, this feature can be used to forward the call to the voice mail system for voice announcement, etc.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

If the user has set DND service and administrator has set preset call forward DND service, incoming call is forwarded to the destination by setting of administrator.
You can use the system preset call forward DND feature by enabling 'Preset Call Forward DND' for selected extension numbers in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu. If you select a service schedule when activate the service, call be forwarded by service schedule plan. Service schedule plan is got 'Set' status and 'Unset' status. System call forward is working only when service schedule plan's status is 'Set'. Service schedule can be configured in **[CONFIGURATION > Time Schedule > Service Schedule]** menu.

5.1.34 VoIP Security

Signaling Encryption

The signaling encryption feature encrypts signaling information required for calls such as the SIP protocol. TLS is used in signaling encryption for VoIP connections between IPX-S300B and SIP phones and between IPX-S300B and endpoints.
Specifications of TLS serviced by IPX-S300B are as follows:

- Uses OpenSSL library and supports TLS v1.0 and TLS v1.2.
- AES and ARIA are supported as media encryption algorithms.
- Key management method is RSA and ECC, and supports both 1024 and 2048 bits as key length.

To use TLS, it must be enabled for phones and endpoints in the following ways:

- You can enable TLS of a Single Phone User by setting Protocol to TLS in the **[CONFIGURATION > User > Single Phone User]** menu.
- You can enable TLS of a Multi-Extension Phone by setting Protocol to TLS in the **[CONFIGURATION > User > Multi-Extension Phone]** menu.
- You can enable TLS for endpoints by setting Protocol to TLS in the **[CONFIGURATION > Trunk Routing > Route]** Additional SIP tap.

Signaling encryption is not used for calls between phones or endpoints enabled with encryption and phones or endpoints not enabled with encryption.

Media Encryption

The media encryption feature provides encryption for the voice data exchanged between the caller and the called party for calls established with signaling encryption.
Media encryption can be enabled to calls between SIP phones or between a phone and an endpoint by applying secure RTP (sRTP), in which case, IPX-S300B performls signaling for sRTP.
IPX-S300B supports media encryption for calls with phones, IPX-S300B's built-in conference system, IPX-S300B's built-in voice mail system, endpoints, and IPX-S300B's built-in MOH system.

IPX-S300B supports AES and ARIA as media encryption algorithms.
ARIA is a block encryption algorithm developed in Korea in 2003 for protection of information for public administration services. This is used as the TLS and sRTP encryption algorithm.
You can enable media encryption for a Single Phone User by setting 'Media' in the **[CONFIGURATION > User > Single Phone User]** menu. You can enable media encryption for a Multi-Extension Phone by setting 'Media' in the **[CONFIGURATION > User > Multi-Extension Phone]** menu.

- RTP: No media encryption.
- sRTP (AES/ARIA128): Encrypts media into the ARIA128 or AES protocol, and uses AES first.
- sRTP (ARIA128/AES): Encrypts media into the ARIA128 or AES protocol, and uses ARIA128 first.
- sRTP (AES/ARIA192): Encrypts media into the ARIA192 or AES protocol, and uses AES first.
- sRTP (ARIA192/AES): Encrypts media into the ARIA192 or AES protocol, and uses ARIA192 first.
- sRTP (AES): Encrypts media into the AES protocol.
- sRTP (ARIA128): Encrypts media into the ARIA128 protocol.
- sRTP (ARIA192): Encrypts media into the ARIA192 protocol.

Media encryption is not used for calls between phones or endpoints enabled with encryption and phones or endpoints not enabled with encryption.

5.1.35 Feature Services

Class of Service

IPX-S300B allows the administrator to set privileges for each user. But setting privileges for all the users individually is very complicated, so, it allows creating Class of Services which includes a set of service privileges and users can be assigned to their appropriate Class of Service.
When a user group is created, a default service group is automatically created for the user group. If you wish to apply a different service class than the default service class, you can create a new service class and use it.
Newly created Class of Service can be applied to user groups, service groups and users. Service classes are applied in the priority order of users > service groups > user groups.
Each service class can be set with different service privileges. It also supports override levels and privacy levels, whereby a particular service is provided only if the overriding user's override level is higher than the privacy level of the user being serviced.
The services involving override and privacy levels include the DND override feature and the barge-in with/without tone feature.

You can set Class of Service in the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu.

Item	Description
User Group	Select a user group for which the service class will be configured.
Name	Specify a name for the service class. This data is used as an identifier when selecting the service class in other menus.
Override Level Privacy Level	You can enter a level used for the barge-in with/without tone feature, etc. The override level is applied to the user using the service and the privacy level is applied to the user provided with the service. Available in levels 0 through 5. Higher levels have precedence over lower levels. A service is allowed only when the override level is higher than the privacy level.
Call Limitation Level for Emergency	In an emergency situation, the call can be restricted by setting Call Limitation Level (Level 0~Level 5). The priority of Call Limitation Level are as follows: Level 5 > Level 4 > Level 3 > Level 2 > Level 1 > Level 0 If the call level is lower than Call Limitation Level, this call is limited. For example, if the value of Call Limitation Level is level 3, all call of level 2 or level 1 or level 2 will be rejected including incoming and outgoing. The changed policy is applied to new calls only.
Second Class of Service	Operator can configure the Second Class of Service for providing to user. It provides only Restriction Policy.
Leaving Office Class of Service	Operator can configure the Leaving Office Class of Service for providing to user. It requires Service Schedule configuration. Also requires to assigned the Service Schedule to Subscribers. It provides only Restriction Policy.
Restriction Policy	Specify a restriction policy to apply to the users belonging to class of service. A restriction policy only applies to trunk calls. You can select one of the existing restriction policies. You can create restriction policies in the **[CONFIGURATION > Routing > Restriction Policy List]** menu.
Call Forward Restriction Policy	Specify a call forward restriction policy to apply to the users belonging to class of service. A call forward restriction policy only applies to trunk forwarded calls. You can select one of the existing restriction policies. You can create restriction policies in the **[CONFIGURATION > Routing > Restriction Policy List]** menu.
Hot Spot Restriction Policy	Specify a Hot Spot restriction policy to apply to the users belonging to class of service. A Hot Spot restriction policy only applies to trunk calls when subscriber located in Hot Spot zone. You can select one of the existing restriction

Item	Description
	policies. You can create restriction policies in the **[CONFIGURATION > Routing > Restriction Policy List]** menu.
mVoIP Restriction Policy	Specify a mVoIP restriction policy to apply to the users belonging to class of service. A mVoIP restriction policy only applies to trunk calls when subscriber located in mVoIP zone. You can select one of the existing restriction policies. You can create restriction policies in the **[CONFIGURATION > Routing > Restriction Policy List]** menu.
IP Zone A Restriction Policy	Specify a IP Zone A restriction policy to apply to the users belonging to class of service. A IP Zone A restriction policy only applies to trunk calls when subscriber located in IP Zone A. You can select one of the existing restriction policies. You can create restriction policies in the **[CONFIGURATION > Routing > Restriction Policy List]** menu.
IP Zone B Restriction Policy	Specify a IP Zone B restriction policy to apply to the users belonging to class of service. A IP Zone B restriction policy only applies to trunk calls when subscriber located in IP Zone B zone. You can select one of the existing restriction policies. You can create restriction policies in the **[CONFIGURATION > Routing > Restriction Policy List]** menu.
Call Transfer Restriction Policy	Specify a call transfer restriction policy to apply to the users belonging to class of service. A call transfer restriction policy only applies to trunk transfer calls. You can select one of the existing restriction policies. You can create restriction policies in the **[CONFIGURATION > Routing > Restriction Policy List]** menu.
Smart Handover Type	Specify allowing type for Smart Handover function. - Blank: Smart Handover function is not working. - Handover Out: Smart Handover Out function is only working. - Handover Both: All Smart Handover function is only working.
Service Permission	Specify allowed/inhibited settings of individual services for the service class. To allow a service, select the checkbox of a corresponding service. Only the services allowed in the **[CONFIGURATION > User Group > User Group]** menu can be set for permission.
Restriction Class	Even if user group is same, the call between service groups can be restricted.

Feature Lists

The administrator can assign privileges for Class of Service or individual users for use of the services listed below.

Service	Description
Absence	If enabled, when there is an incoming call, the absent announcement is played for the caller and the call is terminated.
Add-On Conference	It is including 2 types of conferences First. Conference member is added one by one. There are Ad-hoc, Conference On Answer, Barge-In and Multi-Device Conference. Second, conference services features related with UMS. There is Call Recording/AME.
AME	If enabled, when there is an incoming call, the caller's voice is heard over the speaker as the caller leaves a voice message, just like an answering machine. You can also press the button to start the answering machine while ringing.
Auto Answer	If enabled, when there is an incoming call, the call is automatically answered in speaker mode.
Auto Record	If enabled, when a call is connected, voice mail system is automatically connected for recording the call.
Auto Retry	If used when the number you dialed is busy, the number is automatically redialed repeatedly.
Barge-In with Tone	If used when the other person is engaged in a call, a three-way call is established.
Barge-In without Tone	If used when the other person is engaged in a call, a three-way call is established but your voice is muted.
Callback	If used when the number you dialed is busy, your phone will ring when the other person finishes the previous call. Answering the phone automatically redials the number.
Call Forward All	If enabled, all incoming calls are forwarded to the number specified.
Call Forward Busy	If enabled, incoming calls when busy are forwarded to the number specified.
Call Forward No Answer	If enabled, unanswered incoming calls are forwarded to another number.
Call Forward Unreachable	If enabled, incoming calls are forwarded to another number if the phone being called is not registered or otherwise unavailable.
Call Intercept	Feature code is used for an user intercept a call after barged-in
Forced Call Release	Feature code is used for an user release the call by force after barged-in.
Call Park Extension	If used during a call, the call is parked for an extension number.
Call Park Orbit	If used during a call, the call is parked for an orbit park ID.
Call Recording	If used during a call, the call conversation is recorded in voice mail.
Outbound Call Lock	Feature code is used for which inhibits outgoing calls to the trunk. If this

Service	Description
	feature code is set, IPX-G300B does not let user make a outgoing call.
Call Transfer	This feature allows you to transfer a call.
Call Waiting	If used when the number you dialed is busy, this feature allows you to wait for the called party. If there is a call waiting, the phone will alert the user through the LED, LCD or tone.
Caller ID Block	If enabled, when there is an incoming call, the caller's number is not displayed.
Caller ID Display	If enabled, when there is an incoming call, the caller's number is displayed.
DND	If enabled, when there is an incoming call, the DND announcement is played for the caller and the call is terminated.
DND Override	If used when the number you dialed is in DND mode, the DND setting is ignored and the call is made.
Follow Me	This feature allows you to use another phone to answer incoming calls for your phone.
Group Call Forward	This service is used for forwarding all incoming calls for all the phones in one hunt group to another number.
Hot Desking	This feature allows you to use any phone in any location by logging in with your user ID.
Hot Line	If enabled, when you lift your handset, the phone automatically dials a specified number.
Hotel Inter-Room Call Lock	If enabled, it blocks room to room calls in Hotel environments.
Individual Speed Dial Insert/Delete	A user can register pairs of 3-digit Individual Speed Dial ID and the destination number. By dialing the IDs, the user can make a phone call to the destination.
Malicious Call Trace	If used when there is a malicious call or an otherwise unwanted call, signaling will remain connected even if the caller has hung up the phone, so that the caller can be traced.
Music on Hold	This service plays the MOH when the call is put on hold.
Multi-ring	If enabled, when there is an incoming call, multiple phones ring so that one of the phones can be used to answer the call.
Multi-Device Conference	Even if one multi-device already joined a conference, the other multi-device using same number can also join the same conference.
No Ring	If enabled, when there is an incoming call, the phone does not ring but the call can be answered.
No Ring Override	If enabled, incoming call is allowed, although no ring service is activated.
One-Step Conference	Conference master calls Multiple members at once. It is including Predefined Conference, Progressive Conference, Intercom Conference, Dispatch Conference, Click 2 Conference and Emergency Conference.
Preset Call Forward	This service performs preset call forward all based on the settings made

Service	Description
All	by the system administrator even if preset call forward all is not set by users.
Preset Call Forward Busy	This feature allows the administrator to forward incoming calls for a user without the call forward busy setting to another number when the user is busy.
Preset Call Forward No Answer	This feature allows the administrator to forward incoming calls for a user without the call forward no answer setting to another number when the user does not answer the phone.
Preset Call Forward Unreachable	This service performs preset call Forward Unreachable based on the settings made by the system administrator even if preset call Forward Unreachable is not set by users.
Preset Call Forward DND	This service performs preset call forward DND based on the settings made by the system administrator even if preset call forward DND is set by user.
Remote Office	This feature allows you to answer incoming calls for your phone from another location.
Restricted Call Forward	This feature allows you to exclude particular numbers from call forwarding.
Station Paging	This feature allows paging extension numbers.
Temporary CID Restriction	If used when making a phone call, your number is not displayed in the called party's phone unless recovered.
Call Transfer	This feature allows you to transfer a call.
Wake-Up Call	If set with a time, an alarm will ring at the set time.
Mobile Extension (MOBEX)	Feature allows incoming calls to be directed not only to the landlines and mobiles phones registered with IPX-S300B but also to external phone numbers
Paging On Answer	Feature code is used for make a paging on answer call, The member of paging on answer group should answer to listen paging announcement.
Ring Plan Override	This feature allows you to set ring plan manually.
Remote Extension Set	This feature allows you to set the DB of remote extension.
Premium CID Service	If enabled, the detailed Information of caller will be shown on the callee display.
Operator Call	If enabled, operator call is allowed
BLF Key Create	If enabled, BLF key can be created.
No Use Transferee Restriction Policy	If enabled, Call transfer to external number is allowed by restriction policy of held party phone.
Call Bridge	A gateway FXS user to join the conversion of bridged user by hook-off.
Move To Mobile	If enabled, the call can be moved to mobile. The call conversation is continued.
Mobile Auto Answer	If enabled, the mobile can answer automatically in case of move to mobile

Service	Description
	call.
Smart Routing Activate	Enable or disable Smart Routing service. It will be applied only to WE VoIP Subscribers.
Smart Routing Including Access Code	When using Smarting Routing service, specifies whether to provide the service for numbers including Access Code It will be applied only to WE VoIP Subscribers.
Smart Routing Call Reject	It is the ability to choose to call directly to the mobile phone through the Trunk. Subscribers that use this item will not be able to receive the smart routing service. It will be applied only to WE VoIP Subscribers.
Service After Smart Routing	When converted into an extension through Smart Routing service, specifies whether to provide the service set for the extension. It will be applied only to WE VoIP Subscribers.
Receving Call in Hot Spot Zone	Specifies whether to allow incoming calls in a hot spot region. It will be applied only to WE VoIP Subscribers.
Receving Call in mVoIP Zone	Specifies whether to allow incoming calls in a mVoIP region. It will be applied only to WE VoIP Subscribers.
Un-registed BLF On	Specifies whether to light up in order to notice the un-registration when unregistered subscriber is specified by the BLF key.
Deny International Call Forward/Transfer	Specify the International call Forward and Transfer block.
Conference Member Eject	Conference owner or a user which setup the conference can delete another user which is joined the conference.

Feature Codes

A user can use features in one of the following four ways:

- The administrator enables a feature for the individual user.
- The user enables a feature on SCM Personal Assistant.
- The user presses a feature button on the phone.
- The user dials a feature code from the phone.

If the user wishes to use a feature temporarily without having it configured in advance, the user must use a feature button or a feature code.
Since IPX-S300B uses a standard protocol between phones and IPX-S300B, arbitrary feature buttons cannot be created. Therefore IPX-S300B uses feature codes when configuring features or when using features temporarily. If you press a feature button on a Samsung phone, IPX-S300B is also designed to process the feature code assigned to the feature button.
You can configure feature codes using the **[CONFIGURATION > Service > Feature Service > Feature Code]** menu.

You must create new feature codes after the initial installation.

Item	Description
User Group	Select a user group for which the feature code will be configured.
Feature Code Digit	Specify a number to use as the feature code. No duplicates are allowed. Specify up to 8 digits. # is allowed only for the first part of the code. (Can be used consecutively in the first part. Examples: #, #1, #12, ##, ##2, ###34, etc)
Service Type	Select a service to which the feature code will be assigned.
Minimum Digit Length Maximum Digit Length	Specify the minimum digit length and the maximum digit length for the feature code to be executed. An error will be generated if the range is exceeded.

Feature codes can be configured for the following service types. When used in pairs-such as for enable and disable, request and cancel, and login and logout-two feature codes are registered.

- Feature Code + 1: Feature codes for enable, request, login, etc.
- Feature Code + 0: Feature codes for disable, cancel, logout, etc.

Service	On/Off		Description
Absence	0	Cancel	This feature code requests for registration or cancelation of the absence service. If the service is enabled, when there is an incoming call, the absent announcement is played for the caller and the call is terminated.
	1	Set	
Account Code Voluntary	-	-	This feature code is used for entering an account code during a call.
ACD Agent Break	0	Cancel	This feature code requests for setting or unsetting of an ACD agent's break status.
	1	Set	
ACD Agent Login	0	Logout	This feature code requests for an ACD agent's login or logout of an ACD group.
	1	Login	
ACD Agent Wrap-up	0	Cancel	This feature code requests for setting or unsetting of an ACD agent's wrap-up status after an agent call.
	1	Set	
All Feature Clear	-	-	This feature code requests resetting of all features assigned to your number. Call Forward, DND, Absence, Extension Lock, etc. will be cleared.
AME Enable	0	Cancel	This feature code enables or disables answering machine emulation. If the service is enabled, when there is an incoming call, the caller's voice is heard over the speaker as the caller leaves a voice message.
	1	Set	
AME Mode	0	Stop	This feature code, when there is an incoming call, directs the call to answering machine while ringing, or stops answering machine recording of a call and directs the
	1	Start	

Service	On/Off		Description
			call to the user.
Attendant Continuous Call	-	-	This feature code is used for requesting a service which allows a call transferred by an IP attendant to be redirected to the IP attendant after called party hangs up the phone.
Auto Answer Mode	0	Cancel	This feature code requests for registration or cancelation of a service which automatically connects incoming calls in speaker mode.
	1	Set	
Auto Retry	0	Cancel	This feature code is used for requesting or canceling auto redials. If the service is used when the number you dialed is busy, the number is automatically redialed repeatedly.
	1	Set	
Barge-In with Tone	-	-	This feature code is used for requesting the barge-in service. If the service is used when the other person is engaged in a call, a three-way call is established.
Barge-In without Tone	-	-	This feature code is used for requesting the barge-in without tone service. If the service is used when the other person is engaged in a call, a three-way call is established but your voice is muted.
Callback	0	Cancel	This feature code is used for requesting registration or cancelation of the callback service. If the service is used when the number you dialed is busy, your phone will ring when the other person finishes the previous call. Answering the phone automatically redials the number.
	1	Set	
Call Forward All	0	Cancel	This feature code is used for requesting registration or cancelation of a service which forwards all incoming calls to another number.
	1	Set	
Call Forward Busy	0	Cancel	This feature code is used for requesting registration or cancelation of a service which forwards incoming calls to another number when busy.
	1	Set	
Call Forward No Answer	0	Cancel	This feature code is used for requesting registration or cancelation of a service which forwards incoming calls to another number when not answered.
	1	Set	
Call Forward Busy/ NoAnswer	0	Cancel	This feature code is used for requesting registration or cancelation of Call Forward Busy and Call Forward No Answer simultaneously.
	1	Set	
Call Forward Unreachable	0	Cancel	This feature code is used for requesting registration or cancelation of a service which forwards incoming calls to another number if the phone being called is not registered or otherwise unavailable.
	1	Set	
Call Forward Clear All	-	-	This feature code is used for requesting cancelation of all call forward services configured for your number.

Service	On/Off		Description
Call Intercept	-	-	This feature code is used for an user intercept a call after barged-in.
Call Park Extension	-	-	This feature code is used for requesting parking of the current call for an extension number. If you dial the feature code without an extension number, the call will be parked for your number.
Call Park Orbit	-	-	This feature code is used for requesting parking of the current call for an orbit park ID.
Call Recording	-	-	This feature code is used for requesting recording of the current call conversation in voice mail.
Cancel Move	-	-	This feature code is used for cancel the ringing call for ‘Move to Mobile’ or ‘Move to Number’ service.
Outbound Call Lock	0	Cancel	This feature code is used for requesting registration or cancelation of a service which inhibits outgoing trunk calls.
	1	Set	
Call Waiting	0	Cancel	This feature code is used for requesting registration or cancelation of a service which allows you to wait for the called party when the number you dialed is busy.
	1	Set	
Change Language	-	-	This feature code is used for requesting change of the language used for your number.
Change Password	-	-	This feature code is used for requesting change of the password used for your number.
Conference	-	-	This feature code is used for requesting a conference call.
Conference On Answer	-	-	This feature code is used for requesting a service which immediately starts a three-way conference call when answered by the called party during a call.
Direct Call Pickup	-	-	This feature code is used for requesting a service which allows you to dial into another number and answer its incoming call when the phone is ringing.
DND	0	Cancel	This feature code is used for requesting for registration or cancelation of the DND service. If the service is enabled, when there is an incoming call, the DND announcement is played for the caller and the call is terminated.
	1	Set	
DND Override	-	-	This feature code is used for requesting the DND override service. If the service used when the number you dialed is in DND mode, the DND setting is ignored and the call is made.
Dispatch Conference	-	-	This feature code is used for making conference by using CSTA application. It calls multiple members at once.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Service	On/Off		Description
Extended Alarm Reminder	0	Cancel	One subscriber can have multiple wakeup calls by using this feature code. This feature is aimed to support hotel services.
	1	Set	
Follow Me	0	Cancel	This feature code is used for requesting registration or cancelation of a service which allows you to use another phone to answer incoming calls for your phone.
	1	Set	
Forced Call Release	-	-	This feature code is used for an user release the call by force after barged-in.
FXS Make Call with CW Disable	-	-	This feature code is used for disabling call waiting during a call which was dialed from a gateway FXS phone.
FXS Call Waiting Disable	-	-	This feature code is used for disabling call waiting during a call on a gateway FXS phone.
FXS make Conference Call	-	-	This feature code is used for requesting a conference call during a call on a gateway FXS phone.
FXS Last Member Drop	-	-	This feature code is used for requesting deletion of the last joined attendee of a conference call on a gateway FXS phone.
Group Call Forward	0	Cancel	This feature code is used to activate or deactivate the forwarding feature of a Call Forward Group.
	1	Set	
Group Call Pickup	-	-	This feature code is used for requesting a service which allows you to dial into the Pickup group number of another number and answer its incoming call when the phone is ringing. If you dial the feature code without a Pickup group number, the call currently ringing for the Pickup group will be answered.
Hotel COS Change	-	-	This feature code is used for changing COS of a room phone. It is aimed to support hotel services.
Hotel Service	-	-	This feature code is used for changing status of room. It is aimed to support hotel services.
Hotel Staff Locate	-	-	This feature code is used for informing the location of staff. It is aimed to support hotel services.
Hunt Group Login	0	Out	This feature code requests for login or logout for a hunt group. When there is an incoming call for the hunt group, the logged out members are excluded when determining the called party.
	1	In	
Intercom	-	-	This feature code requests for the intercom service which allows one-touch dialing and automatic answering of calls between the numbers registered with the intercom feature, such as executives and secretaries.
Intercom Conference	-	-	If user dial this feature code and hunt group number, caller and hunt group members can join conference.

Service	On/Off		Description
			And the hunt group members answer automatically.
Last Incoming Redial	-	-	This feature code requests for redialing of the last incoming call's CLI number.
Last Outgoing Redial	-	-	This feature code requests for redialing of the last dialed number.
Individual Speed Dial	0	Insert	This feature code is used to add an Individual Speed Dial number.
	1	Delete	This feature code is used to delete an Individual Speed Dial number.
Individual Speed Dial-Call	-	-	This feature code is used for support speed dial per subscribers.
Malicious Call Trace	-	-	This feature code is used for requesting the malicious call trace service. If the service is used when there is a malicious call or an otherwise unwanted call, signaling will remain connected even if the caller has hung up the phone, so that the caller can be traced.
Meet Me Conference Join	-	-	This feature code is used for participating in a meet me conference, which is set up by reserving the time and channel of the conference and joined by calling into the conference room at the conference time.
MOBEX Call Pickup	-	-	This feature code is used for requesting the MOBEX on desk pick up service. If the service is enabled, when the office phone and the mobile phone ring simultaneously by the multi-ring feature, you can answer the call on the mobile phone and then continue the call on the office phone.
Move to Mobile	-	-	This feature code is used for moving the call on the desk phone to mobile phone.
Moved Call Pickup			After moving the conversation to a mobile, the desk phone can pick up the call through 'Moved Call Pickup' soft menu. This feature code is used for supporting it.
Multi-Device Conference	-	-	Even if one multi-device already joined a conference, the other multi-device using same number can also join the same conference. This feature code is used for supporting it.
Multi-ring Enable	0	Cancel	This feature code registers or cancels the multi-ring service. If the service is enabled, when there is an incoming call, multiple phones ring so that one of the phones can be used to answer the call.
	1	Set	
Multi-ring Member	0	Delete	This feature code requests adding/removing of members to/from the multi-ring service. If the service is enabled, when there is an incoming call, multiple phones ring so
	1	Insert	

Service	On/Off		Description
			that one of the phones can be used to answer the call.
Move Key – To Mobile	-	-	If the desk phone has move button, the call can be moved to mobile through move button. This feature code is used for supporting it.
Move Key – To Desk	-	-	If the mobile installed Samsung NFC application (SDM) is on conversation, the call can be moved to the desk phone through soft menu. This feature code is used for supporting it.
Move Key – Pick Up	-	-	If the desk phone has move button, when call is ringing, the mobile installed application (SDM) can answer the call instead of the ringing desk phone. This feature code is used for supporting it.
No Ring	0	Cancel	This feature code requests for registration or cancelation of the no ring service. If the service is enabled, when there is an incoming call, the phone does not ring but the call can be answered.
	1	Set	
Outbound Call Lock	0	Cancel	If this feature code is set, system does not let user make a outgoing call.
	1	Set	
Parked Call Retrieve	-	-	This feature code is used for picking up a call parked for the extension number or the orbit park ID.
Predefined Conference	-	-	This feature code is used for paging all the members registered in the conference system simultaneously for a conference.
Predefined Text Message	-	-	If users want to send text messages set by the administrator, user should dial this feature code with destination number.
Progressive Conference	-	-	This feature code is used for entering numbers of multiple members in the conference system and then paging them simultaneously for a conference.
Remote Extension Lock	-	-	This feature code is used for an IP attendant to set incoming/outgoing call lock for another extension number.
Remote Extension Set	-	-	This feature code is used for an IP attendant to set features for another extension number.
Remote Office	0	Cancel	This feature code is used to activate or deactivate the Remote Office feature.
	1	Set	
Ring Plan Override	0	Cancel	This feature code is used to override time-based auto ring plan manually.
	1	Set	
Shared Call Retrieve	-	-	This feature code is used for picking up a call parked for another phone when using multi-device.
Station Paging	-	-	This feature code is used for requesting an extension

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Service	On/Off		Description
			announcement service.
Temporary CID Restriction	-	-	If used when making a phone call, your number is not displayed in the called party's phone.
VM Access	-	-	This feature code is used for dialing to access the voice mail system.
VM Administration	-	-	This feature code is used for accessing the voice mail system and changing it settings.
VM Memo	-	-	This feature code is used for accessing the voice mail system and leaving a message for another number or in your own mailbox.
VM Message	-	-	This feature code is used for accessing the voice mail system and listening to messages in your mailbox.
VM Transfer	-	-	This feature code is used for directing the current call to the voice mail system and connecting it the mailbox for another number.
Trunk Redial			This feature code is used for making a new call adding prefix 0 after cancel an original call when subscriber attempts a call.
Wake-Up Call	0	Cancel	This feature code requests for registration or cancelation of the wake-up call service, which, if set with a time, an alarm will ring at the set time.
	1	Set	
Conference Member Eject	-	-	This feature code requests to delete another user which is joined the conference. Dial with 'Conference Member Eject Feature Code + a user number to delete'.
PRI MCID	-	-	The user who receives a call with a malicious intent.can report the call through using PRI connections to the PSTN. While the user is connected to the call, the user can invoke the MCID feature by using this feature code.

Feature Activation

Those services not performed temporarily by user actions but configured in the database can be configured by the administrator.
The administrator can change service settings for individual users using the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu.
Enter a user group and an extension number and press the **[Search]** button to change the current settings for the user.
Select an inactive service and press the **[Activate]** button to activate the service.
Select an active service and press the **[Deactivate]** button to deactivate the service.
When activating a service, you may be required to enter additional information depending on the service type.

Item	Description
User Group	Select a user group.
Extension Number	Specify a user number for which the service will be activated.
Service Type	Specify a service to activate.
Destination	Specify a destination number if registering call forwarding, etc.
Use Notification	Specify whether the phone needs to be notified of the status if registering call forwarding, etc.
Start Time	Time based call forward all function’s start time.
End Time	Time based call forward all function’s end time.
Service Date	Specify a service date if registering wake-up call, etc.
Allow Other Ring	Specify whether incoming calls when busy will be forwarded to the multi-ring members if registering multi-ring.
Preset Call Forward Type	Select a type of preset call forward type. - Internal: only incoming call from internal is served. - External: only incoming call from external is served. - Both: all incoming call from internal and external is served.
Auto Record Mailbox	Specify a mailbox where the recorded files will be stored if registering auto record.
Auto Record Call Type	Specify a type of calls to auto record if registering auto record. - Internal: Incoming calls from extension numbers are automatically recorded. - External: Incoming calls from trunks are automatically recorded. - Both: Incoming calls from both extension numbers and trunks are automatically recorded.
Call Logging	Select a type of call logging.
Service Schedule	Select a service schedule to use preset call forward.

User Service Timers

User Service Timer can be changed in **[CONFIGURATION > User > User Service Timers]** menu.
Types of service timer are as follows.

Item	Description
User Group	Select a user group.
Extension Number	Specify a user number for which the service will be activated.
No Answer Forward Time (sec)	Specify the time period used for determining no answer when registering no answer call forward.
No Answer Preset Forward Time (sec)	Specify the time period used for determining no answer when registering no answer preset call forward.
Hot Line Delay (sec)	Specify a time period for which to wait before the preset number is automatically dialed after the handset is lifted if registering hot line.

Item	Description
Hot Desk Expire Time (hour)	Specify a time period for which to wait before a logged in user is automatically logged out if registering hot desk.
DeskPhone Simultaneous RingDelay(s)	Specify a time period for desk phone's ring delay when desk phone is ringing with mobile phone or WE VoIP at the same time.
Speaker Disconnect Tone Time (sec)	Specify a time period for disconnect tone of speaker after forced auto answer service like Intercom, paging or paging on answer.
Handset Disconnect Tone Time (sec)	Specify a time period for disconnect tone of handset after forced auto answer service like Intercom, paging or paging on answer.

5.1.36 User Authentication

IPX-S300B performls digest authentication. It authenticates a SIP phone in different ways when the SIP phone sends a REGISTER message. For more information on digest authentication, see the SIP standard documents RFC3261 and REC2617.

Local Authentication

IPX-S300B performls internal authentication in the following order:

1) The SIP phone transmits REGISTER without authentication header to IPX-S300B.
2) IPX-S300B transmits 401 Unauthorized with challenge information to the SIP phone.
3) SIP phone transmits REGISTER without authentication header to IPX-S300B.
4) After IPX-S300B executes Digest Authentication, it transmits 200 OK to the SIP phone.

RADIUS Authentication

IPX-S300B supports RADIUS digest authentication and acts as a RADIUS client for remote RADIUS authentication of users' phones. RADIUS digest authentication is performed in Scenario 1 and Scenario 2. Both are supported by IPX-S300B.
IPX-S300B acts as a relay between the user phone and the external RADIUS server. Authentication is performed in the following order:

1) When IPX-S300B receives a REGISTER message from the user phone, it sends Access-Request to the RADIUS server.
2) When IPX-S300B receives Access-Accept or Access-Reject from the RADIUS server, it sends the authentication result to the user phone and finishes the authentication procedure.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

LDAP Authentication

IPX-S300B acts as an LDAP client for remote LDAP authentication of user' s phones.
It provides LDAP and LDAPS (LDAP over SSL) for this task.
IPX-S300B interoperates with the external LDAP server and fetches the password from the user phone by using LDAP protocol. Authentication is performed in the following order:

1) When IPX-S300B receives a REGISTER message including a password from the user, it sends a Search-Request message to the LDAP server.
2) The user's password stored in the LDAP server is received through a Search-Result message.
3) The user phone's password received from the LDAP server is compared with the password received with the REGISTER message from the user phone.
   IPX-S300B sends the authentication result to the user phone and finishes the authentication procedure.

5.1.37 Boss/Secretary

The boss/secretary feature allows a boss and a secretary to share one user number while using their own individual numbers and the intercom feature.
Bosses and secretaries can be connected 1:1 or M:N.

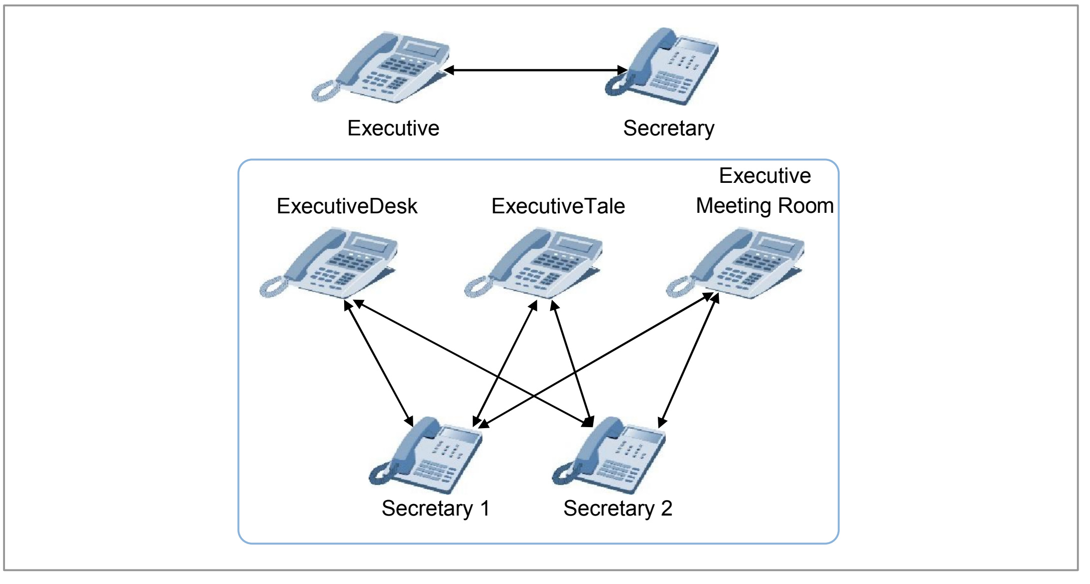


Figure 15. Boss/Secretary

While configuration may vary by administrators, we recommend that you configure the feature in the following order for maximum efficiency.

1) Create a device in the **[CONFIGURATION > User > Multi-Extension Phone]** menu.
   - Phone Verification: Select IP Address or MAC Address for authentication method.
   - User Type: Select Manager or Secretary for device.

2) Create a user password for the manager/secretary in the **[CONFIGURATION > User > Multi-Phone User]** menu.
   - Call Appearance: Select SCA (SCA: Shared Call Appearance, MCA: Multiple Call Appearance).
3) Configure intercom between the individual numbers of the manager and the secretary in the **[CONFIGURATION > Service > User Service > Intercom]** menu.

The boss/secretary feature allows the phones to share the status of the shared line so that the users are informed of the current status of the shared line.

Example) Redirect an incoming call to the Boss

- Boss and Secretary have shared the same SCA number and each has its own private number also.
- A call is incoming to the SCA number, after the Secretary answered the call, Secretary using its private number make an intercom call to the private number of Boss to explain who is calling to the Boss.
  - If the Boss wants to talk, secretary make the Boss pickup the held SCA call by pressing SCA line button of the Boss's Phone.
  - Secretary redirect the SCA call by just pressing the BLF button pre-configured with the private number of the Boss.

5.1.38 Busy Lamp Field (BLF)

IPX-S300B provides the Busy Lamp Field (BLF) service which indicates the status of a particular service or the status of the user number using the LED on the buttons of the phone. To use the BLF service, you must first configure the feature to the buttons of the phone. Buttons can be configured in the following ways:
Use the **[CONFIGURATION > User > Phone Key Programming]** menu to configure the built-in buttons on the phone.
Use the **[CONFIGURATION > User > AOM Key Programming]** menu to configure the buttons on the button extension box.

Fields	Description
User Group	Select a User Group
Phone Name	Select a phone number or Mac Address of AOM
#	Displays number and order of the buttons.
Display Name	Specify the Name of the button. It may be displayed in BLF button depends on Phone Model.
Key	Select the BLF type.
Value	Specify the Extension number want to monitor.
Extension Number	Multi-phone user case, you can specify one of the extension numbers which extension number wants to monitor.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.1.39 DTMF Detection Service

During a call between SIP phones on an IP PBX, all data except SIP signaling for call connection is exchanged by the phones. Therefore, the numbers dialed for services-except the phone number included in the INVITE message for call connection-cannot be sent to the system using the standard protocol.
In order to receive the numbers dialed on the phone-except the INVITE message-IPX-S300B connects the call to its built-in voice announcement system and collects the numbers dialed on the phone according to the voice announcement.
User interaction services provided in this way include account code, call authentication code, and DISA user authentication.

5.1.39.1 Account Code

This feature allows the user to enter his/her account code in the account information when making an external call through the trunk. Account codes can be entered in the following two ways.

Forced Account Code

When a trunk call is made from a phone set with forced account code input, a registered account code must be entered. The account code entered will be saved in the charging data record (CDR), which can be used for calculating call charges for the user.
※ Entering account code is displayed with *.

Voluntary Account Code

When a trunk call is made from a phone set with voluntary account code input, you can press the account code button and enter an account code when outbound call connected. The account code entered will be saved in the charging data record (CDR), which can be used for calculating call charges for the user.
You can set the account code input method of a Single Phone User under 'Account Code Use' in the **[CONFIGURATION > User > Single Phone User]** menu.
You can set the account code input method of a Multi-Phone User under 'Account Code Use' in the **[CONFIGURATION > User > Single Phone User]** menu.

- None: No account code needs to be entered when making a trunk call.
- Force: An account code registered with IPX-S300B must be entered when making a trunk call.
- Voluntary: An account code can be entered by pressing the account code button when making a trunk call.

To use the voice announcement system for entering account codes, you should enable the item whose service type is 'ACC' in the **[CONFIGURATION > Service > DTMF Detection Service > Activation]** menu.

The account codes entered for forced account code input can be registered in the **[CONFIGURATION > Service > DTMF Detection Service > Account Code]** menu.
When a trunk call is made on a phone set with forced account code input, actions are performed in the following order:

1) The user dials an access code and an external number.
2) If the user's phone is set with forced account code input, IPX-S300B connected the call to its built-in voice announcement system.
3) The voice announcement system plays an announcement for the user to enter a registered account code.
4) The user enters an account code as instructed. The voice announcement system verifies that a valid account code has been entered.
5) If the account code entered is valid, IPX-S300B uses the access code received in step (1) to select a route and makes a call for the external number specified.
6) If the account code is invalid, IPX-S300B plays an error announcement and terminates the call.

5.1.39.2 Authorization Code

Those users restricted from making external calls can make external calls by dialing the number for the built-in voice announcement system which authenticates external calls.
To use the voice announcement system for authenticating call, you should enable the item whose service type is 'AUTH' in the **[CONFIGURATION > Service > DTMF Detection Service > Activation]** menu.
The number for the voice announcement system which authenticates call authentication codes can be registered in the **[CONFIGURATION > Service > DTMF Detection Service > Service Number]** menu.
The call authentication codes used for call authentication can be registered in the **[CONFIGURATION > Service > DTMF Detection Service > Authorize Code]** menu.
When a user restricted from making external calls attempts to make a trunk call by using a call authentication code, actions are performed in the following order:
Entering account code is displayed with *.

1) The user dials the number for the call authentication system.
2) IPX-S300B connects the call to the call authentication system.
3) The voice announcement system plays an announcement for the user to enter a registered call authentication code.
4) The user enters a call authentication code as instructed. The voice announcement system verifies that a valid call authentication code has been entered.
5) If the call authentication code entered is valid, IPX-S300B temporarily suspends the external call restriction set for the user.
6) The user can now dial an access code and an external number and the trunk call will be made.

The authorize code entered will be saved in the charging data record (CDR), which can be used for calculating call charges for the user.

5.1.39.3 DISA User Authentication

When using the Direct Inward System Access (DISA) feature, the user can call IPX-S300B from outside to get authenticated as instructed by the voice announcement so that he/she can make a trunk call through the system.
When there is an incoming DISA call, IPX-S300B connects the call to its built-in DISA user authentication announcement system and plays a voice announcement for the external caller.
To use the voice announcement system for authenticating DISA users, you should enable the item whose service type is 'DISA' in the **[CONFIGURATION > Service > DTMF Detection Service > Activation]** menu.

The number for the voice announcement system which authenticates DISA users can be registered in the **[CONFIGURATION > Service > DTMF Detection Service > Service Number]** menu. To connect incoming DISA calls from trunks to the voice announcement system for authenticating DISA users, set the called number to the number of the DISA user authentication announcement system in the **[CONFIGURATION > Trunk Routing > DID Routing]** menu.
IPX-S300B can allow incoming DISA calls from registered callers to use DISA calls without user authentication. If you register caller numbers in the **[CONFIGURATION > Service > DTMF Detection Service > DISA Approved CLI Number]** menu in advance, the incoming DISA calls from the registered caller numbers are provided with the DISA service without having to enter user numbers or passwords.

5.1.40 System Speed Dial

This feature allows you to assign maximum 16-digit shortcut number to a phone number frequently dialed not by individual users but by all users of the system or to a lengthy phone number so that the number can be dialed just by using the shortcut number when necessary.
You can register system speed dial numbers using the **[CONFIGURATION > Service > Speed Dial > System Speed Dial]** menu.

Item	Description
User Group	Select a user group for which the system speed dial will be registered.
Speed ID	Specify an ID for the system speed dial. The ID must be a number of two or longer digits. The ID must not be overlapped with other extension numbers and conference channel numbers.
Speed Name	Specify a name for the system speed dial.
Speed Number	Specify an actual phone number to be dialed by the system speed dial.

You can specify , using system speed dials, the display the number or name to calling subscribers using ‘System Speed Dial Display ‘ in **[CONFIGURATION > User Group > Change User Group > Option]** menu.

5.1.41 RTP Call Restriction

This function limits calls that are not encrypted. This can be set by the user group.
To use this function, you should set the item whose ‘Media Option’ is ‘SRTP Only’ in the **[CONFIGURATION > User Group > Change User Group > Information]** menu.
Afer this option is changed, you should create ‘phone profile’ in the **[CONFIGURATION > Phone Setting > Create Phone Profile]** menu and ‘User Profile User group’ in the **[CONFIGURATION > Phone Setting > Upgrade Software]** menu

5.1.42 Mline service

This is the function that contains the 2 mline (AVP/SAVP) in the INVTIE message of sRTP Calls. To use this function, you should set the item whose ‘SRTP Mline offer option’ is ‘RTP SRTP BOTH’ in the **[CONFIGURATION > User Group > Change User Group > Information]** menu.
After this option is changed, you should create ‘phone profile’ in the **[CONFIGURATION > Phone Setting > Create Phone Profile]** menu and ‘User Profile User group’ in the **[CONFIGURATION > Phone Setting > Upgrade Software]** menu.

5.1.43 Caller Ring Type

One of the phone’s ringtones be used as Caller distinct Ring
Caller Ring Type can be changed via the Admin **[CONFIGURATION > User > Single Phone User/Multi-Phone User > Caller Ring Type]** menu. If the value is greater than that supported by the phone Caller Ring Type operates in the default Ring

5.1.44 System SPAM Call Block Service

IPX-S300B provides the ability to shut off the incoming Trunk Call to the same calling number at a specific time.

The operator must be able to use the **[SPAM Call Block Service]**, a set of the following items from the **[CONFIGURATION > Miscellaneous > System Options]** menu.

Item	Description
Maximum System SPAM Call Block Activate	Select whether to use the **[SPAM Call Block Service]**. Default Option value is disable.
Maximum System SPAM Call Count	It is the option to set the number of incoming Call to the same calling number to whether to allow. count range is 3 to100.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
System SPAM Call Expire Timer (Second)	It's time to keep the Block to the number that is blocked. Timer range is 10 to 600 second.

5.1.45 Gateway Channel Display

GW channel display function is a function that will display information to the user of the Channel of GW that you are using.

To use the GW channel display function, set the items in the **[CONFIGURATION > Trunk Routing > Route > Interworking]**.

Item	Description
Gateway Channel Display	Operator will be able to choose whether to use the GW channel display.

When user uses the GW channel display function, displays the name of the Trunk and the channel information of PRI Trunk currently using.

5.1.46 Default Access Code Use List

'Default Access Code Use List' is the service to use a Default Access Code for a specific incoming number. If you change the **[Use Default Access Code Use List]** to **[Yes]**, Routing Process is performed as follows.
This option menu is located the **[CONFIGURATION > User Group > Change User Group > Options]**.

1) First, Find the Default Access Code Use List using Called Number.
2) If called number exist in the Default Access Code Use List, Attach the Default Access Code.
3) If not, using called number only.
4) Original Routing Process.

If you change the **[Use Default Access Code Use List]** to **[No]**, Routing Process is performed as follows.

1) First, Check the Called Number including Access Code.
2) If called number includes the Access Code, skip No.3 Process.
3) If called number not include the Access Code, attach the Default Access Code.
4) Original Routing Process.

For use List in the Default Access Code Use List, follows adding process is required.

1) Adding Called Number to **[CONFIGURAITON > Trunk Routing > Default Access Code Use List]**.
2) Default Access Code is configured in **[CONFIGURATION > User Group > Change User Group > Options]**.

Mandatory Item	Description
User Group	Select the User Group for use a List.
Match Number	IPX-S300B is compared the Called Number and the Match Number. The Number has rules below. - 0311233000 ・Exact Match: Check Length and all digits. - 031123xxxx, 031123XXXX ・Prefix Match: Check Length and prefix. - 031123? ・Longest Match: Check Prefix Only.

5.1.47 All Hot Desking Logout

IPX-S300B provides forced logout service to All Hot Desking user in User Group.
To use this service, operator should configured following items in **[CONFIGURATION > User Group > Change User Group > Options]** menu.

Item	Description
All Hot Desking Logout	Option to use “All Hot Desking Logout”.
All Hot Desking Logout Time	Specify a activation time for All Hot Desking Logout. IPX-S300B provides “All Hot Desking Logout” service by using a configured time each day.

All Hot Desking Logout service don’t provide to specific user, who was set a value 0 “Hot Desk Expire Time (hour)” in **[CONFIGURATION > User > User Service Timers]** menu.

5.1.48 Leaving Office Class of Service

IPX-S300B provides a leaving office class of service when users leave an office.
This class of service provides restriction policy only.
To use Leaving office class of service, operator should configure following steps.

1) Create a service schedule (**users work time pattern**) in **[CONFIGURATION > Time Schedule > Service Schedule]** menu. Operator can create a pattern of working hours per week.
2) Assign a service schedule to user in **[CONFIGURATION > User > Single Phone User/Multi-Phone User]** menu.

Item	Description
Service Schedule	A service schedule is a table containing data which specifies service plans by days of the week, dates, and time of the day. There is a service schedule of up to 32 per user group. The service schedule is used for system call forward and Leaving Office Class of Service. Leaving Office Class of Service is activated when Service Schedule status unset.

1) Create Toll restriction list in **[CONFIGURATION > Trunk Routing > Toll Restriction List]** menu.
2) Create Toll Restriction Policy in **[CONFIGURATION > Trunk Routing > Toll Restriction Policy]** using no.3.
3) Create class of service in **[CONFIGURATION > Service > Feature Service > Class of Service]** menu. Then, assign restriction policy above.
4) Assign a leaving office class of service for user's class of service.

Item	Description
Leaving Office Class of Service	Operator can configure the Leaving Office Class of Service for providing to user. It requires Service Schedule configuration. Also requires to assigned the Service Schedule to Subscribers. It provides only Restriction Policy.

5.1.49 Trunk Redial

IPX-S300B provides redial feature when trunk call attempts.

IPX-S300B makes a new call adding prefix 0 after cancel an original call, when subscriber uses Trunk Redial.
To use this service, operator need to make a feature code in **[CONFIGURATION > Service > Feature Service > Feature code]** menu.

This service provides only IP-Desk Phone and communicator.

5.1.50 International Call Service

IPX-S300B provides International Call Service.

To use whole International Call Service, "International call prefix" configuration is necessary.

- Menu Location**: [CONFIGURATION > User Group > Change User Group > International Call Prefix]**.

Name	Description
User Group	It displays the user group.
Prefix	Specifies the prefix to verify international call.
Prefix Minimum Length	Specifies the minimum length to use prefix.

International call prefix needs to match an outbound call prefix without trunk number translation.

5.1.50.1 International Call Inform Service

IPX-S300B sends notification to operator when International Call Attempt count reaches configured counts.

International Call Inform count and duration can configure following menu.

- Menu Location: **[CONFIGURAITON > Miscellaneous > System Options]**

Item	Description
International Call Check Count	Specify the attempt call counts, which is used for international call check. (default: 1) The input range can be from 0 to 100.
International Call Check Duration (minute)	Specify a call logging time to check international call count. (default: 1) The input range can be from 1 to 60.

5.1.50.2 Deny International Call Forward/Transfer

IPX-S300B provides deny service when user make an international call forward or a call Transfer.
This service needs to configure “class of service” configuration.

- Menu Location: **[CONFIGURATION > SERVICE > FEATURE SERVICE > CLASS OF SERVICE]**

Item	Description
Deny International Call Forward/Transfer	Specify the International call Forward and Transfer block.

5.1.51 Display DTMF Detection Code

IPX-S300B provides a display option for code in CDR when subscriber makes a outbound call using an account code or an authorize-code.

To use this option, following menu configuration is required.

- Menu Location: **[MANAGEMENT > CDR Storage Options > User Group CDR Option]**

ITEM	Description
Display DTMF Detection Code	Specify the option to display code in **[CONFIGURATION > Service > DTMF Detection]** menu. - Not Display: IPX-S300B hides all DTMF Detection codes. - Account Code: IPX-S300B displays an account code. - Auth Code: IPX-S300B displays an authorize code. - Both: IPX-S300B displays an account code and an authorize code.

5.1.52 Transfer CID for VM/AA

IPX-S300B provides Semi-blind Transfer CID options to VM/AA.
Operator can specify the Transfer CID Number to VM/AA using following menu.

- Menu Location: **[CONFIGURATION > User Group > Change User Group > Options]**

Item	Description
Transfer CID for VM/AA	Select the CID Number which is provided to VM/AA when VM/AA receives the Semi-Blind Transfer Call. - Transferer: Use Transferer CID for VM/AA. - Transferee: Use Transferee CID for VM/AA

5.1.53 Multiple SIP Account

Multiple SIP Account to one ISP is supported. To use this feature, called number range has to be assigned to each route. Under multiple SIP account, receiving route is determined by 'User Number Range'. If a call with unassigned called number is received through a specific route, the call is rejected even if the call is authenticated.

'Multiple Registration' and 'User Number Range' have to be configured in the **[CONFIGURATION > Trunk Routing > Route]** menu.

Item	Description
Multiple Registration	If multiple registration to one ISP has to be handled, configure 'Enable'. - Multiple Registration cannot mixed with Normal Registration which is

Item	Description
	configured Multiple Registration to Disable for one ISP. - Configuration is possible only when Register Type is 'Send REGISTER'.
User Number Range	Enter the called number range which is used to find receiving Rout. - 'Number', '**[-]**' and 'X' can be entered for 'User Number Range'. - 'User Number Range' has to be started with number. Range can be expressed with '**[-]**'. 'X' means any number. - User Number Range is supported 3 digits and the range is possible within 100. ex) Example · 0312791000: In case a called number is same to 0312791000 · 0312791XXX: In case a called number is starting with 0312791 and its length is 7. · 031279[1-4]XXX: In case a called number is starting with 0312791, 0312792, 0312793 and 0312794 and its length is 7.

5.1.54 Common Route

Common Route is used to share a route between several user groups. Every user group has a permission to use the common route whereas other route can be accessed only by the user group the route belongs to. A call from the common route can be delivered to any user group and all user group can send a call to the route.

Following configuration is needed to use the Common Route.

- 'Route Type' and 'User Group' has to be set to COMMON in the **[CONFIGURATION > Trunk Routing > Route]** menu.
- **[CONFIGURATION > Trunk Routing > Common Route Prefix]** is configured to select a user group which a call is processed based on.
- When the route is used as a gateway, 'User Group' had to be set to COMMON in the **[CONFIGURATION > Gateway > Gateway Link Setting]**.

There are limitations for the common route for a gateway (REGISTER Type: Receive Register).

- 'Domain Name' in the route configuration cannot duplicate with 'Host' of any user group.
- The common gateway does not support FXS users and Survival mode. It's a exclusive trunk gateway.

5.1.55 Noticeboard Service

IPX-S300B provides NoticeBoard Service to WE VoIP terminal.
WE VoIP terminal can add, modify, delete and search a notice.
SCM Administrator can manage notices totally.

To use Noticeboard service, configure following steps.

- Configure 'Directory Service Activation' of WE VoIP terminal settings.
- Configure 'Noticeboard Service Activation' of WE VoIP terminal settings.
- Configure 'Noticeboard Service Authority' of WE VoIP terminal settings.
- Management of Noticeboard Service by SCM Administrator.

Activation of Directory Service

Set following items in the **[CONFIGURATION > Wireless Enterprise > Mobile Phone Profile]** menu.

Item	Description
Directory Service-Private (Office)	Select option for Using Directory Service when Private WiFi Network Zone register (Default: Disable)
Directory Service-Wi-Fi Hotspot	Select option for Using Directory Service when public WiFi Hotspot Network Zone register (Default: Disable)
Directory Service-mVoIP	Select option for Using Directory Service when LTE Network Zone register (Default: Disable)

Activation of Noticeboard Service

set following items in the **[CONFIGURATION > Wireless Enterprise > Mobile Phone Profile]** menu.

Item	Description
NoticeBoard Service-Private (Office)	Select option for Using NoticeBoard Service when Private WiFi Network Zone register (Default: Disable)
NoticeBoard Service-Wi-Fi Hotspot	Select option for Using NoticeBoard Service when public WiFi Hotspot Network Zone register (Default: Disable)
NoticeBoard Service-mVoIP	Select option for Using NoticeBoard Service when LTE Network Zone register (Default: Disable)

Authority of Noticeboard Service

set following items in the **[CONFIGURATION > User > User Profile]** menu.

Item	Description
NoticeBoard Service Authority	Select a authority on NoticeBoard Service.

Management of Noticeboard Service by SCM Administrator

SCM Administrator can add/modify/delete and read a notice in **[CONFIGURATION > Service > Notice Board]** Menu.
Operator can use Noticeboard Service according to Account Class.

- Engineer Class: Create/Search/Modify/Delete
- Technician Class: Create/Search

File Attachment Function

When creating notice, user can attach one image file.
Size limit is 1 MB (1 Mbyte).

User Group Category for NoticeBoard.

SCM Administrator can configure the category of user group in **[CONFIGURATION > User Group > NoticeBoard Category]** Menu.

Item	Description
User Group	User Group
Category Name	Noticeboard Category Name
Category Priority	Input a priority level for order of Category in Mobile Phone.

Max Notice Count.

SCM Administrator can configure the max count of notice in **[CONFIGURATION > User Group > Change User Group > Options]** Menu.

Item	Description
Max Count of Notice	Select the Max Count of Notice per User Group - range: 0~1000 * Max count of System: 10,000

5.1.56 Service Limitation

The service limit is the control function about the resource which the User Group's use commonly.

A internal voice mail/auto attendant service server and a internal conference server are generated by user group, and can be limited through the number of maximum calls in **[CONFIGURAITON > Application > VM/AA Server]** menu and **[CONFIGURATION> Application > Conference Server]** menu.

5.1.57 Minimum Call Limit

IPX-S300B provides the way to limit a call connection by call limitation level. 'Call Limitation Level' is assigned to a call by CoS. If the call limit level is lower than that of IPX-S300B, the call is denied. The level ranges from 0 to 5 and higher number means higher level.
Each call has a call limit level by user's CoS. Under urgent situation, administrator can set Minimum Call Limit to control the number of calls. For example, with the 'LEVEL 2 ANSWER ONLY', users with Level 0 ~ Level 1 cannot make and receive a call and ones with Level 2 cannot make but receive a call. Only the user with Level 3~Level 5 is available both. Minimum Call Limit policy is applied to new calls not connected call. IPX-S300B does not cut a call.

To use this feature following configuration is needed.

- The call limit level is decided by the 'Call Limitation Level' in the **[CONFIGURATION > Service > Class of Server]** menu.
- To control the number of calls by level, 'Minimum Call Limit' is selected in the **[CONFIGURATION > Miscellaneous > System Options]** menu.

5.1.58 Multicast Paging

IPX-S300B has a paging feature using multicast. In the case of internal paging, a maximum of 128 people per group is supported. Now, it supports a maximum of 512 people.

Telephones and versions, which support multicast, are as follows.

- SMT-i5230/SMT-i5210S/SMT-i5220/SMT-i2205: V2.84 or above
- SMT-i601x/2x: V1.30 or above

Multicast requires the following setup.

- [CONFIGURATION > Service > Feature Service > Class of Service > Multicast Paging] must be checked.

- [CONFIGURATION > Service > Feature Service > Feature Code > Multicast Paging] must be specified.
- [CONFIGURATION > Service > Group Service > Multicast PagingGroup]must be created, and the following data must be set up.

Item	Description
User Group	Select the user group to generate the multicast group.
Group Name	Input the multicast group name.
Group Number	Input the multicast group number.
Multicast Address	Input the multicast address. An IP address only in Class D (224.0.0.0-239.255.255.255) can be input. However, an address from 224.0.0.0 to 224.0.0.255, 228.228.228.229, and 228.228.228.229 cannot be input as they are reserved for other use. Therefore, 239.x.x is recommended for a multicast address.
Multicast Port	Input the audio port for multicast. It is recommended to use the default value, 35998.
Multicast Codec	Select the codec for multicast paging. PCMU, PCMA, and G.729 can be selected. The default value is PCMU.
Multicast Group Member	Select the multicast group member among users. Only Samsung desktop phones can be selected.

The multicast paging host can be broadcasted by pressing the multicast code and the group number. The service by pressing only the group number is also supported.
In IPX-S300B, if a member is taking a call and could not receive the multicast paging request, the member can receive the multicast paging after the call is ended. Also, when more than one multicast paging is on hold, the multicast paging requested first will be received. The telephone version must be checked for this.

Multicast has limitations such as below.

- IPX-S300B does not guarantee the reception to all multicast members. If a member has no IGMP protocol support or is located in another subnet, the multicast reception may fail. When multicasting to a member in another subnet, a special network device is required.
- The device processes a multicast reception with the lowest priority. In other words, if a member is taking a call or doing other operations, the member will ignore the multicast paging request.
- If a member receives a multicast paging request while already receiving the multicast paging, the member ignores the second request.
- If a member receives the same multicast request while receiving the multicast paging, SCM declines the second request.

- Multicast paging does not support SRTP. When SRTP is required, the internal paging service using conference server is used.
- When the call monitoring service is set on to the multicast host, the call monitoring service is not activated during the multicast paging.
- When multiple devices are the multicast host and selected as multicast members, the multicast paging is not received.
- If the multicast host is on WAN when IPX-S300B is on WAN, multicast members on LAN cannot receive the multicast paging. On the other hand, if the multicast host is on LAN, multicast members on WAN cannot receive the multicast paging.

5.1.59 System Information Display

IPX-S300B offers a feature to check the system information on an IP phone.
The corresponding information is updated every minute.

- Supporting device model: SMT-i6000 Series

The system Information that can be checked on an IP phone is as follows.

Item	Description
Version Info.	The version of the system by each module is displayed.
Network Info.	The network setting information of the system is displayed.
Slot Info.	The setting type and the card status of the system by each slot are displayed. Abbreviation in menu - C: Card setting information - S: Card installation status
Registration Info.	The registration status of telephones and gateways, which are interworking with the system, are displayed. Abbreviation in menu - O: The number of items registered. - X: The number of items not registered.
Etc.	Additional information of the system is displayed. - The number of current alarms is displayed.

5.1.60 Nursing Home Equipment Interworking

IPX-S300B offers a nursing home equipment interworking service through the SCM OpenTSP Driver that is separate API/SDK. Emergency messages from a nursing home device can be transmitted to a designated device using API provided in the SCM OpenTSP Driver. The detailed information on API for emergency messages is explained in the SCM OpenTSP Driver user manual.

5.1.61 Emergency Call Service (E911)

IPX-S300B provides the PSAP (Public Safety Answering Point) with the caller ID in an emergency call so that they can identify the location of the emergency caller.

The emergency zone members in proximate location, and the group members share one emergency call information.
The emergency zone can be created in the [Configuration > Service > Emergency Call Service > Emergency Zone] menu, and the detailed items are as follows.

Name	Description
User Group	Select the user group using the emergency call service.
Emergency Zone Name	Select the emergency call area name.
Emergency Zone Description	Input the information to identify the emergency call area.
Emergency CLI Number	When a member in the emergency zone makes a call, the corresponding number is provided as the caller number.
Emergency CLI Name	When a member in the emergency zone makes a call, the corresponding name is provided as the caller name.

The menu, which the emergency zone created is to be assigned, and the priority ranking are as follows.

1) User: [CONFIGURAITON > User > Single Phone User/Multi-Extension Phone]
2) IP MAP: [CONFIGURAITON > Service > Emergency Call Service >Emergency Zone IP MAP]
3) User group: [CONFIGURAITON > User Group > Change User Group > Information > Default Emergency Zone]

In the case of #2 IP MAP, the emergency zone is determined by the registered IP address of the telephone. Therefore, if the registered IP address is changed when the location of the telephone is changed, the emergency caller number may be changed. The detailed information for the IP MAP setup is as below.

Name	Description
User Group	Select the user group using the emergency call service.
Priority	The IP MAP table order, which attempts IP mapping.
Source IP Address	The caller’s registered IP and the IP address for mapping.
Subnet Mask Length	The caller’s registered IP and the subnet mask length for mapping.
Emergency Zone	Assign the value created in the [Configuration > Service > Emergency Call Service > Emergency Zone] menu.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

If the location is not identifiable with the registered IP address, it must be assigned to the #1 User menu.
If there is no emergency zone applicable to the #1 User and #2 IP MAP menu, the emergency caller number will be determined by the emergency zone assigned to #3 User Group.

The emergency call service provided by IPX-S300B has the following limitations.

1) When a real-time mobile device makes a emergency call, IPX-300B does not guarantee the exact emergency caller ID.
2) A callback call from the PSAP does not guarantee the callback call from the emergency caller. Receiving calls from an attendant console or another member is possible with the DID setup.

The above limitations can be resolved by a 3rd party emergency service provider.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.2 User Features

User features are an executed features by users.
A user features are an available service to the users with privilege for the service which is setting by the administrator. Since setting service privileges for all the users individually could be very complicated, Class of Service are be created with their own set of service privileges and users are then assigned to their appropriate Class of Service.
CoS can be applied to each user groups, service groups and users. CoS are applied in the priority order of users > service groups > user groups.
You can set Class of Service in the **[CONFIGURATION > Service > Feature Service** > **Class of Service]** menu.
To use a user features defined by the user in advance, you can use the feature on SCM Personal Assistant. The user can also use a feature-including those services used temporarily-by pressing the feature button on the phone or dialing the feature code.
To use a feature by pressing the feature button or dialing the feature code, the feature code must be defined in advance.
You can define feature codes using the **[CONFIGURATION > Service > Feature Service > Feature Code]** menu.
For more information on feature services, see the ‘4.1.36 Feature Service’.

5.2.1 Absence

The absence feature is used for notifying that the user is absent. If the absence feature is enabled, when there is an incoming call, an announcement is played to notify the caller of the absence status and the call is terminated.

To use the absence feature, the following items must be configured.

- The ‘Absence’ service must be enabled in Class of Service.
- The ‘Absence-Set’ and ‘Absence-Cancel’ feature codes must be defined.

The user can register or cancel the absence feature in the following ways:

- The user can dial the ‘Absence-Set’ feature code on the phone to register the absence status.
- The user can dial the ‘Absence-Cancel’ feature code on the phone to cancel the absence status.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the absence feature. You can also specify the time for enabling the absence feature.

5.2.2 Auto Answer

The auto answer feature is used when the user wishes to have his/her incoming calls answered automatically. If the auto answer feature is enabled, when there is an incoming call, the speaker will be turned on and the call will be answered automatically.
To use the auto answer feature, the following items must be configured.

- The 'Auto Answer' service must be enabled in Class of Service.
- The 'Auto Answer Mode-Set' and 'Auto Answer Mode-Clear' feature codes must be defined.

The user can register or cancel the auto answer feature in the following ways:

- The user can dial the 'Auto Answer Mode-Set' feature code on the phone to enable auto answer.
- The user can dial the 'Auto Answer Mode-Clear' feature code on the phone to disable auto answer.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the auto answer feature.

You can use an attention tone which is used for indicating that an incoming call has been answered automatically. The attention tone setting is applied to all users of the selected user group.
Auto answer attention tone can be configured in the **[CONFIGURATION > User Group > Change User Group > Options]** menu.

- Auto Answer Attention Tone: Specify whether to play the attention tone when a call is answered automatically.
- Auto Answer Attention Tone Count: Specify how many times the 100 ms tone should be repeated when the auto answer attention tone is played.

5.2.3 Automatic Retry

When the user makes an outgoing trunk call but the called party is busy or does not answer the call, the automatic retry service can be used to automatically redial the number after a set period of time. If the automatic retry is enabled, the phone's speaker is automatically turned on at a set interval and the last dialed number is dialed again.
To use the automatic redial feature, the following items must be configured.

- The 'Auto Retry' service must be enabled in **[CONFIGURATION > Service > Class of Service]**.
- The 'Auto Retry-Set' and 'Auto Retry-Cancel' feature codes must be defined in the **[CONFIGURATION > Service > Feature Service > Feature Code]**.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

The user can register or cancel the automatic retry feature in the following ways:

- When the user makes an outgoing trunk call but the called party is busy or does not answer the call, the user can press the automatic redial button to use the automatic redial feature. When the automatic redial button is pressed on the phone, the phone dials the 'Auto Retry-Set' feature code to IPX-S300B.
- When the phone dials the 'Auto Retry-Cancel' feature code, the automatic redial is canceled.

When the automatic redial feature is serviced, the following three timers are activated. The timers required for the automatic redial feature can be configured using the **[CONFIGURATION > User Group > Change User Group> Timers]** menu.

- Auto Retry No Answer Time (sec): When a trunk number is redialed by automatic retry and the called party does not answer the call, the phone waits for this period of time before it terminates the call as an unanswered call.
- Auto Retry Interval (sec): When a number is automatically redialed but is still busy or does not answer, the phone waits for this period of time before it redials the number.
- Auto Retry Service Duration (min): When automatic retry continues to fail, the phone tries for this period of time before it terminates the service.

5.2.4 Barge-In

This feature allows you to intrude into a user's current call for a three-way conference call. The call intrusion feature is also known as Barge In or Call Override.
The call intrusion feature is providing with warning or without warning depending on whether the user is notified that the call has been intruded into. IPX-S300B services both types of the feature.
IPX-S300B uses its VPU option card for three way conference calls. IPX-S300B does not support the barge in feature though its built-in conference system. Therefore, the basic settings for using the VPU conference must be configured.
The administrator needs to set 'Application Type' to 'VPU Conference' in the **[CONFIGURATION > Application > Conference Server]** menu to create a connection to the built-in conference system. This conference server should be included 'Add-On Conference' in service list.
To use the call intrusion feature, both 'Override Level' and 'Privacy Level' must be defined in Class of Service. Call intrusion is allowed only when the override level is higher than the privacy level. The override level is applied to the user intruding and the privacy level is applied to the user being intruded into.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Barge-In with Tone

When you intrude into a call and establish a three-way conference, this service periodically plays a tone to notify the user being intruded into that the call has been intruded into.
To use the call intrusion feature, the following three items must be configured.

- The 'Barge-In with Tone' service must be enabled in Class of Service.
- The 'Barge-In with Tone' feature code must be defined in Class of Service.

The user can use the call intrusion feature in the following ways:

- The user calls an extension number and if the called party is busy, the user can press the call intrusion button to intrude into the call.
- The user can dial the 'Barge-In with Tone' feature code + number of the user currently in a call to intrude into the call.

Barge-In without Tone

When a three-way conference call is established by intruding into a call, the user being intruded into is not given any notification and the intruding user's voice is muted so that the call can be monitored in secret.
To use the call intrusion without tone feature, the following three items must be configured.

- The 'Barge-In without Tone' service must be enabled in Class of Service.
- The 'Barge-In without Tone' feature code must be defined in Class of Service.

The user can use the call intrusion without tone feature in the following ways:

- The user calls an extension number and if the called party is busy, the user can press the call intrusion without tone button to intrude into the call without tone.
- The user can dial the 'Barge-In without Tone' feature code + number of the user currently in a call to intrude into the call without tone.

5.2.5 Change Password

This service provides a user can change the PIN Number which is used for services by himself.
The Change Password Feature code must be configured.
User can dial the Change Password feature Code with previous Password once and news password twice like below:

Example) * 33 + (old Password) + (New Password) + (New Password)

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.2.6 Callback

When a user calls another user but if the called party is busy or does not answer, the caller can enable the callback feature so that when the called party becomes available, the caller's phone will ring, and if the caller answers the phone, the called party's number is redialed. To use the callback feature, the following items must be configured.

- The 'Callback' service must be enabled in Class of Service.
- The 'Callback-Set' and 'Callback-Cancel' feature codes must be defined.

The user can register or cancel the callback feature in the following ways:
When the user calls another user but the called party is busy or does not answer the call, the user can press the callback button to use the callback feature. When the callback button is pressed on the phone, the phone dials the 'Callback-Set' feature code to IPX-S300B.

- The user can dial the 'Callback-Cancel' feature code on the phone to cancel the callback feature.

When the callback feature is serviced, the following two timers are activated. The timers required for the callback feature can be configured using the **[CONFIGURATION > User Group > Change User Group > Timers] menu**.

- Callback Ring No Answer Time (sec): When the caller is called back by the callback feature but the caller does not answer, this call which is made to notify the caller that the called party has now become available will be processed as a failed call after ringing for this period of time. If the callback notification call fails, the system waits until the called party uses the phone and become available again.
- Callback Service Duration (min): When callback is enabled, if the callback service is not executed successfully during this period of time, the callback service is automatically terminated.

In case a user who was set the callback feature makes or receives a call, and then becomes an idle states, the system determines that the user becomes available for the callback call. If the **[CONFIGURATION > User Group > Change User Group > options > Line Seize Option]** is set to **[Send Message]**, the system only needs the hook-off/on actions through a handset of the user's phone.

5.2.7 Call Forward

When there is an incoming call, this feature is used for forwarding the call to another number specified by the user. The call forward feature can have limitation by Restricted Call Forward in the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu. If Restricted Call Forward is enabled, call cannot be forwarded to trunk.

5.2.7.1 Call Forward All

If the call forward all feature is enabled for a user, all incoming calls for the user are automatically forwarded to a specified number.
Even if the user has not enabled call forward all, the administrator can configure all incoming calls for the user in specific time periods to be forwarded to another number.
For more information, see the ‘4.1.32 Group Call Forward’.
To use the call forward all feature, the following items must be configured.

- The ‘Call Forward All’ service must be enabled in Class of Service.
- The ‘Call Forward All-Set’ and ‘Call Forward All-Cancel’ feature codes must be defined.

The user can register or cancel the call forward all feature in the following ways:

- The user can dial the ‘Call Forward All-Set’ feature code + destination phone number on the phone to enable call forward all.
- The user can dial the ‘Call Forward All-Cancel’ feature code on the phone to cancel call forward all.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the call forward all feature. If call forward is enabled on SCM Personal Assistant, the user can select two options, time and notification. Without time configuration, call forwarding is enabled permanently until the user disables the feature. However, if the user configures the time, call forward is worked during the time period. The user can know how many calls are forwarded with the notification option.

5.2.7.2 Call Forward Busy

If the call forward busy feature is enabled for a user, incoming calls for the user while the user is busy are automatically forwarded to a specified number.
Even if the user has not enabled call forward busy, the administrator can configure the incoming calls for the user while the user is busy to be forwarded to another number.
For more information, see the ‘Preset Call Forward Busy’ section of ‘4.1.34 System Call Forward’.
To use the call forward busy feature, the following items must be configured.

- The ‘Call Forward Busy’ service must be enabled in Class of Service.
- The ‘Call Forward Busy-Set’ and ‘Call Forward Busy-Cancel’ feature codes must be defined.

The user can register or cancel the call forward busy feature in the following ways:

- The user can dial the 'Call Forward Busy-Set' feature code + destination phone number on the phone to enable call forward busy.
- The user can dial the 'Call Forward Busy-Cancel' feature code on the phone to cancel call forward busy.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the call forward busy feature.

5.2.7.3 Call Forward No Answer

If the call forward no answer feature is enabled for a user, the incoming calls for the user which are not answered for a specified period of time are automatically forwarded to a specified number.
Even if the user has not enabled call forward no answer, the administrator can configure the incoming calls not answered by the user to be forwarded to another number. For more information, see the 'Preset Call Forward No Answer' section of '4.1.34 System Call Forward'.
You can use SCM Personal Assistant to set the time period for each user which is used by the call forward no answer service to determine the user's an incoming call as an unanswered call and forward it to a specified number. If call forward no answer is enabled by pressing the feature code on the phone, the default time of 15 seconds is used.
To use the call forward no answer feature, the following items must be configured.

- The 'Call Forward No Answer' service must be enabled in Class of Service.
- The 'Call Forward No Answer-Set' and 'Call Forward No Answer-Cancel' feature codes must be defined.

The user can register or cancel the call forward no answer feature in the following ways:

- The user can dial the 'Call Forward No Answer-Set' feature code + destination phone number on the phone to enable call forward no answer.
- The user can dial the 'Call Forward No Answer-Cancel' feature code on the phone to cancel call forward no answer.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the call forward no answer feature. Here, you can also specify the time period used for determining the no answer status.

5.2.7.4 Call Forward Unavailable

If the call Forward Unreachable service enabled for a user, all incoming calls for the user are automatically forwarded to a specified number when the user's phone is not registered, does not respond to signaling, or otherwise unavailable.
To use the call Forward Unreachable feature, the following items must be configured.

- The 'Call Forward Unavailable' service must be enabled in Class of Service.
- The 'Call Forward Unreachable-Set' and 'Call Forward Unreachable-Cancel' feature codes must be defined.

The user can register or cancel the call Forward Unreachable feature in the following ways:

- The user can dial the 'Call Forward Unreachable-Set' feature code + destination phone number on the phone to enable call Forward Unreachable.
- The user can dial the 'Call Forward Unreachable-Cancel' feature code on the phone to cancel call Forward Unreachable.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the call Forward Unreachable feature.

If you select a 'CFUR service schedule' in the **[CONFIGURATION > User group > Service group]** menu, call be forwarded by service schedule plan. Service schedule can be selected for each service group. Service schedule plan is got 'Set' status and 'Unset' status. Call forward is working only when service schedule plan's status is 'Set'. Service schedule can be configured in **[CONFIGURATION > Time Schedule > Service Schedule]** menu.

5.2.7.5 Selective Call Forward

This service only forwards a user's incoming calls from specified numbers.
Selective call forward can be serviced in the following two ways. If two features are enabled both, Call Forward Restriction has precedence over Call Forward Allowance.

Call Forward Allowance

If this feature is enabled, only the calls from specified numbers are forwarded and calls from unspecified numbers are not forwarded.

- The user can use the 'Selective Call Forward Allowance' menu on SCM Personal Assistant to register a list of caller numbers to allow selective call forward.
- The administrator can use the **[CONFIGURATION > Service > Selective Service > Call Forward Allowance]** menu to register a list of caller numbers to allow selective call forward for each user.

Forward Restriction

If this feature is enabled, the calls from specified numbers are not forwarded and calls from unspecified numbers are forwarded.

- The user can use the **[Call Forward Restriction]** menu on SCM Personal Assistant to register a list of caller numbers to restrict selective call forward.
- The administrator can use the **[CONFIGURATION > Service > Selective Service > Call Forward Restriction]** menu to register a list of caller numbers to restrict selective call forward for each user.

5.2.7.6 Call Forward No Response

No response from a User

It is one of cases of Call Forward Unavailable. It is different with Call Forward No Answer which is working after ring back tone.
If there is no response from a User within the specific time right after sending call setup message (initial INVITE), Call Forward Unavailable service is functioning.
If Call Forwarding Unavailable service was not set, the call will be dropped with announcements.
Administrator can configure the specific time for User at **[CONFIGURATION > User Group > Change User Group > Timers > Internal Call No Response Time (sec)]** and it is 3 seconds by default.

No Response from a Trunk

If there is no response from a Route within the specific time right after sending call setup message (initial INVITE), Call Forward Unavailable service is functioning.
If there is no Alternative Route for the Route, the call will be dropped with announcements.
Also, administrator can configure the specific time for Trunk at **[CONFIGURATION > User Group > Change User Group > Timers > External Call No Response Time (sec)]** and it is 10 seconds by default.

5.2.8 Call Hold

The call hold feature allows the user to park the current call and make another call for transfer or conference and then retrieve the hold call.
If the other person on the line is handling multiple calls (including call park, transfer, and conference), the call cannot put on hold.

5.2.9 Call Park

The call park feature allows the user to park the current call so that it can be retrieved on another phone by pressing the button or the feature code.
A park ID must be entered when parking a call, so that the call can be identified when picked up. There are the following two types of call park service depending on the park ID input method.

Call Park Extension

The call park service can be used by using an extension number as the park ID. Since an extension number is used as the park ID, only one call can be parked per extension number.
To use the call park feature, the following items must be configured.
The ‘Call Park Extension’ service must be enabled in **[CONFIGURATION > Service > Class of Service]** menu.
The ‘Call Park Extension’ feature code must be defined in **[CONFIGURATION > Service > Feature Code]** menu.
While on a call, the user can press the park button and then dial the call park feature code + an extension number to park the call for the extension number entered. Or, the user can just dial the call park feature code without an extension number to park the call for his/her own number.

Call Park Orbit

The call park orbit service can be used by using an independent orbit park number as the park ID. Since independent numbers are used as the park ID, multiple calls can be parked for each extension number. For Call Par Orbit, the park ID has a range from 01~99.
Only one call can be parked with one park ID. If you want to use empty orbit park ID, use press 00 instead of the park ID. In that case, IPX-S300B finds an empty park ID and the call is parked with the park ID. You can check the parked ID on the display of the phone. Therefore, this feature is useful for users who need to park many calls, such as operators.
To use the orbit park feature, the following items must be configured.
The ‘Call Park Orbit’ service must be enabled in **[CONFIGURATION > Service > Class of Service]** menu.
The ‘Call Park Orbit’ feature code must be defined in **[CONFIGURATION > Service > Feature Code]** menu.
While on a call, the user can press the park button and then dial the orbit park feature code + an orbit park number to park the call for the orbit park number entered.

Parked Call Retrieve

This feature allows a parked call to be reconnected on the phone for it had been parked or on another phone.
To use the Parked Call Retrieve feature, the ‘Parked Call Retrieve’ feature code must be defined in the **[CONFIGURATION > Service > Feature Code]** menu.
The user can dial the parked call retrieve feature code + park ID to pick up a parked call.

Park Recall

This feature allows a parked call to be redirected to the user who parked the call if the call is not picked up after a specified period of time.
The administrator can use 'Park Recall Time (sec)' in the **[CONFIGURATION > User Group > Change User Group > Timers] menu** to specify the time for which parked calls will remained parked before being redirected.
If the redirected call is not answered, the call is redirected to the operator. For more information, see the 'Operator Recall' section of '5.1.28 Operator Group'.

5.2.10 Call Pickup

This feature allows the user to answer another user's incoming call. Call Pickup can be serviced in the following two ways.

Direct Call Pickup

This feature allows you to pick up another user's incoming call by specifying the user's number.
To use the direct call Pickup feature, the following items must be configured.
The 'Direct Call Pickup' feature code must be defined in the **[CONFIGURATION > Service > Feature Code]**.
The user can press the direct call Pickup feature code + the number of the user whose phone is ringing to pick up the other user's incoming call which is currently ringing.

Group Call Pickup

If this feature is enabled, you can specify a call Pickup group number (instead of a user number) for which an incoming call is ringing to pick up the current incoming call for the selected group. You can also Pickup an incoming call ringing for your own call Pickup group.
To use the group call Pickup feature, the following three items must be configured.
The 'Group Call Pickup' feature code must be defined in the **[CONFIGURATION > Service > Feature Code]**.
Call Pickup groups must be configured in the **[CONFIGURATION > Service > Group Service > Pickup Group]** menu. The following data must be configured.

Item	Description
User Group	Select a user group for which the call Pickup group will be created.
Group Number	Enter a number for the call Pickup group.
Group Name	Enter a name for the call Pickup group.
Ping Ring	Select a notice about no-answer for an incoming call during some period. - Disable: There is no notice about no-answer call. - Enable: There is a notice about no-answer call.
Ping Ring Time (sec)	Enter time to wait to make a notice to the members about no-answer call.

Item	Description
Group Member	Select members for the call Pickup group. A user can belong to one call Pickup group only.

The user can press the group call Pickup feature code + the number of the call Pickup group whose phone is ringing to pick up the group's incoming call which is currently ringing. Or, the user can dial just the group call Pickup feature code without a call Pickup group number to pick up the current incoming call for his/her own call Pickup group.

Ping Ring

IPX-S300B provides Ping Ring service for Group Call Pickup. When nobody picked up an incoming call ringing for own call Pickup Group during Ping Ring Time, members of the Pickup Group receive a notify to let know there is a incoming call to pickup. The Ping Ring Service set 'Enable' in the **[CONFIGURATION > Service > Group Service > Pickup Group]** menu.
Ping Ring Time can be configured in the 'Pickup Group' as described below. If there is no configuration, IPX-S300B uses Ping Ring Time in the **[CONFIGURATION > User Group > Change User Group > Timer]**.
Each member of Pickup Group can select a notify type for Ping Ring. It can be configured in the 'Ping Ring Type' of **[CONFIGURATION > User > Single Phone User]**.

5.2.11 Outbound Call Lock

The Outbound Call Lock feature allows a user to request for restriction of outgoing trunk calls from his/her own number.
To use Outbound Call Lock feature, the following items must be configured.

- The 'Outbound Call Lock' service must be enabled in **[CONFIGURATION > Service > Feature Service > Class of Service]**.
- The 'Outbound Call Lock-Set' and 'Outbound Call Lock-Cancel' feature codes must be defined in the **[CONFIGURATION > Service > Feature Service > Feature Code]**.

The user can register or cancel the Outbound Call Lock feature in the following ways:

- The user can dial the 'Outbound Call Lock-Set' feature code + password to enable call restriction.
- The user can dial the 'Outbound Call Lock-Cancel' feature code + password to cancel call restriction.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the call restrict by user feature.

The user can use the **[User Information]** menu on SCM Personal Assistant to change the password entered on the phone when enabling or canceling call restrict by user.

5.2.12 Call Transfer

The call transfer feature allows the user to put on hold the current call and transfer it to another number. If call transfer fails, the call is reconnected to the user who transferred the call.
To use the call transfer feature, the 'Call Transfer' service must be enabled in **[CONFIGURATION > Service > Feature Service > Class of Service]**.

5.2.12.1 Call Transfer Methods

The user can transfer calls in the following three ways.

Blind Transfer

This feature allows the user to transfer the call directly to another number without hold it. Although IPX-S300B and Samsung SIP phones support blind transfer, this has the same effect as semi-blind transfer from the user's point of view. Therefore no separate feature code is defined.

Semi-Blind Transfer

This feature allows the user to put on hold the current call by pressing the transfer button, call another number, and then transfer the call by pressing the transfer button again while the phone is ringing.

Consultative Transfer

This feature allows the user to put on hold the current call by pressing the transfer button, call another number, and then transfer the call by pressing the transfer button again after the call is established.

5.2.12.2 Transfer Recall

This feature allows the transferred call to be redirected to the user who transferred the call when call transfer fails or when the transferred call is not answered.
The administrator can use 'Transfer Recall Time (sec)' in the **[CONFIGURATION > User Group > Change User Group > Timers] menu** to specify the time after which the transferred call is redirected. If the user to whom the call is transferred does not answer the call during this period of time, the call is redirected to the user who transferred the call.
If the redirected call is not answered, the call is redirected to the operator. For more information, see the 'Operator Recall' section of '5.1.28 Operator Group'.

5.2.12.3 Transfer Target Display for Transfer Recall

The transfer target information can be displayed on the phone that is receiving the recalling call.
This administrator can enable this option at 'Transfer Target Display for Recall' in the **[CONFIGURATION > User Group > Change User Group > Options]** menu. According to the options, the transfer target information can be displayed to all transferors, only to the operator, or only the normal user transferor.

5.2.12.4 Transfer Ring Back Tone

IPX-S300B can play MOH itself or transparently deliver the media played from remote side to held party when blind or semi-blind transfer services.
The administrator needs to set 'Transfer Ringback Tone' in the **[CONFIGURATION** > **User Group > Change User Group > Options]** menu.

Item	Description			
	When transfer target is subscriber		When transfer target is trunk	
Options	Use external ringback tone server	Not using external ringback tone server	Trunk side provides ringback tone	Trunk side doesn't provides ringback tone
Internal Ringback Tone	External server plays	IPX-S300B plays ringback tone	IPX-S300B plays ringback tone	IPX-S300B plays ringback tone
MOH	Music On Hold	Music On Hold	Music On Hold	Music On Hold
External Ringback tone	External server plays	IPX-S300B plays ringback tone	Trunk plays ringback tone	IPX-S300B plays ringback tone

5.2.12.5 Call Log For Transfer Target

The user can transfer calls in the three ways (Blind Transfer/Semi-Blind Transfer/Consultation Transfer). In that case, the transfer-party or held party information remains in the call log of the transfer target. It does not depend on the ways of transfer if the **[Call Log for Transferred Party]** option in the **[Configuration > User Group > Change User Group > Options]** is set to 'Held Party' or 'Transfer Party'.

5.2.13 Call Waiting

If the call waiting feature is enabled, when there is an incoming call while the user is already engaged, the call is not terminated as a call when busy, but instead the user is notified that a call is waiting so that the user can park or end the previous call and Pickup the new call.
If there is an incoming call while the user is already engaged, a brief call waiting tone will be played for the user. If the user presses the call button to answer the new call, the previous call is automatically parked.
If the call waiting feature is enabled for a phone, the phone can receive all the calls it can accommodate. But if the call waiting feature is not enabled, all incoming calls while the phone is engaged are terminated as calls when busy.
If the call waiting feature is enabled for a phone, the phone can accommodate as many calls as the call buttons configured. If no call button is configured, all incoming calls while the phone is engaged are treated as calls when busy.
To use the call waiting feature, the following items must be configured.

- The ‘Call Waiting’ service must be enabled in Class of Service.
- The ‘Call Waiting-Set’ and ‘Call Waiting-Cancel’ feature codes must be defined.

The user can register or cancel the call waiting feature in the following ways:

- The user can dial the ‘Call Waiting-Set’ feature code on the phone to register the call waiting feature.
- The user can dial the ‘Call Waiting-Cancel’ feature code on the phone to cancel the call waiting feature.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the call waiting feature.

To use a ‘Call Waiting’ feature when Wireless Enterprise Services phone’s option is ‘Ring Only’ or ‘Both’, it is needed a next setting.

- ‘Multi-ring’ feature must be activated in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu.
- ‘Allow Other Ring’ must be enabled when ‘Multi-ring’ feature is activated.

5.2.14 Call Intercept

Call Intercept can be provided under the 3-way conference by the Barge-In service.
A User can make 2-way call with the party which is barged in by pressing Call Intercept soft key.
The menu ‘Call Intercept’ and ‘Forced Call Release’ will be displayed at the user who barged-in. When the user presses ‘Intercept’ button, the opponent of barged-in user will be dropped and the barged-in target and the user who barged-in is directly connected.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.2.15 Forced Call Release

Forced Call Release can be provided under the 3-way conference by the Barge-In service. By pressing Force Call Release soft key, a user can make 2-way call by releasing the party barged in.
The menu ‘Call Intercept’ and ‘Forced Call Release’ will be displayed at the user who barged-in. When the user presses ‘Forced Call Release’ button, the barged-in target will be dropped and the opponent of barged-in target and the user who barged-in is directly connected.

5.2.16 CLI Control

Temporary CID Restriction

The temporary CID restriction feature allows the user to request that his/her number is not shown to the called party for a particular call.
To use the temporary CID restriction feature, the following items must be configured.

- The ‘Temporary CID Restriction’ service must be enabled in Class of Service.
- The ‘Temporary CID Restriction’ feature code must be defined.

When making a call, the user can dial the temporary CID restriction feature code + called party’s phone number to request temporary CID restriction.

Distinctive Ring by CLI

The distinctive ring by CLI feature allows incoming calls to be distinguished by ringing different rings depending on the caller numbers.
The user can use the **[Distinctive Ring]** menu on SCM Personal Assistant to register a list of caller numbers for which distinctive ring by CLI will be used.
The administrator can use the **[CONFIGURATION > Service > User Service > Distinctive Ring]** menu to register a list of caller numbers for which distinctive rings will be serviced.

5.2.17 Do Not Disturb (DND)

When the Do Not Disturb (DND) feature is enabled for a user, IPX-S300B rejects all incoming calls for the user. When there is an incoming call for a user with DND, an announcement is played to notify the caller of the DND status and the call is terminated.
To use the DND feature, the following items must be configured.

- The ‘DND (Do Not Disturb)’ service must be enabled in Class of Service.
- The ‘DND (Do Not Disturb)-Set’ and ‘DND (Do Not Disturb)-Cancel’ feature codes must be defined.

The user can register or cancel the DND feature in the following ways:

- The user can dial the ‘DND (Do Not Disturb)-Set’ feature code on the phone to register the DND feature.
- The user can dial the ‘DND (Do Not Disturb)-Cancel’ feature code on the phone to cancel the DND feature.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the DND feature.

5.2.17.1 DND White List

When there is an incoming call for a user with DND, this service prevents the call from getting rejected if the call originates from one of the caller numbers specified in advance.
The user can use the **[DND White List]** menu on SCM Personal Assistant to register a list of caller numbers to exclude from the DND service.
The administrator can use the **[CONFIGURATION > Service > User Service > DND White List]** menu to register a list of caller numbers which will be excluded from the DND service.

Item	Description
User Group	Select a user group for which the DND white list will be created.
Extension Number	Select an extension number for which the DND white list will be created.
White List	Enter the caller numbers for the DND white list.

5.2.17.2 DND Override

When there is an incoming call for a user with DND, this service allows the caller, while listening to the DND announcement, to ignore the DND status and have his/her call connected.
To use the DND override feature, both ‘Override Level’ and ‘Privacy Level’ must be defined in Class of Service. DND override is allowed only when the override level is higher than the privacy level. The override level is applied to the user overriding DND and the privacy level is applied to the user with DND.
To use the DND override feature, the following items must be configured.

- The ‘DND Override’ service must be enabled in Class of Service.
- The ‘DND Override’ feature code must be defined.

When there is an incoming call for a user with DND, the caller, while listening to the DND announcement, can press the DND override button to override DND and the called party’s phone will ring. Even when the DND override feature is used, the call may not be connected if the called party is busy or otherwise unavailable.

5.2.18 Follow Me

When the caller has temporarily moved to another location, this service allows the caller to answer all incoming calls to his number by using another phone.
To use the follow me feature, the following items must be configured.

- The ‘Follow Me’ service must be enabled in Class of Service.
- The ‘Follow Me-Set’ and ‘Follow Me-Cancel’ feature codes must be defined.

The user can register or cancel the follow me feature in the following ways:

- The user can dial the ‘Follow Me-Set’ feature code + his/her password + his/her phone number on another phone to enable follow me.
- The user can dial the ‘Follow Me-Cancel’ feature code + his/her password + his/her phone number on another phone to cancel follow me.
- If follow me is enabled, it will be shown as enabled in the **[Supplementary Service]** menu on SCaM Personal Assistant, where the feature can be canceled.

The user can use the **[User Information]** menu on SCM Personal Assistant to change the password entered when enabling or canceling follow me.

5.2.19 Individual Speed Dial

This feature allows you to assign a shortcut number to a phone number frequently dialed.
The user should dial ‘Individual Speed Dial’ feature code + ‘Speed Dial ID’.
A speed dial ID is just a one digit. IPX-S300B supports 10 individual speed dial for each users.
To use this feature, it needs feature code configurations as follow:

- ‘Individual Speed Dial-Call’ feature code
- ‘Individual Speed Dial Number-Insert’ feature code
- ‘Individual Speed Dial-Delete’ feature code

To set individual speed dial using phone, user should dial ‘Individual Speed Dial Number-Insert’ feature code and speed dial ID (0~9) and dialing number on the phone.
To delete individual speed dial set, user should dial ‘Individual Speed Dial Number-Delete’ feature code and speed dial ID (0~9) on the phone.

5.2.20 Intercom

When a call is made between the users for whom intercom is enabled, the call is automatically answered through the speaker. When using the manager/secretary feature, the intercom feature is used together.
To use the intercom feature, the following items must be configured:

- An intercom number must be specified for each user using the **[CONFIGURATION > Service > User Service > Intercom]** menu.
- The 'Intercom' feature code must be defined.

A user with an intercom number can dial the intercom feature code + user number to page the selected user for a call.
The following items must be configured in the **[CONFIGURATION > Service > User Service > Intercom]** menu.

Item	Description
User Group	Select a user group for which the intercom will be created.
Extension Number	Select an extension number for which the intercom will be created.
Name	Specify a name for the intercom.
Intercom	Select an extension number which will be connected with the intercom.

5.2.21 Language Selection

This service allows the user to change the language displayed on their phone.
The user can use the **[User Information]** menu on SCM Personal Assistant to change his/her language.
The administrator can change the language of a Single Phone User by changing 'Language' in the **[CONFIGURATION > User > Single Phone User]** menu.
The administrator can change the language of a Multi-Extension Phone by changing 'Language' in the **[CONFIGURATION > User > Multi-Phone User]** menu.

5.2.22 Last Number Redial

The last number redial feature allows the user to redial the caller or the called party number of the most recent call.
For Call Forward All and Multi-Ring services, the number user dialed initially will be used as called number not the number finally reached.
The last number redial service allows redialing the called number of the last outgoing call or the caller number of the last incoming call. The following must be configured.

- The 'Last Outgoing Redial' feature code must be defined in order to be able to redial the last dialed number.
- The 'Last Incoming Redial' feature code must be defined in order to be able to redial the caller number of the last incoming call.

The user can use the last call redial feature in the following ways:

- The user can dial the last outgoing feature code to redial the called number of the last outgoing call.
- The user can dial the last incoming redial feature code to redial the caller number of the last incoming call.
- Last Call Redial is independently operated by each node.

5.2.23 No Ring

The no ring feature prevents the phone from ringing when there is an incoming call for the user. This service is useful to prevent some phones from ringing when multiple phones are configured to ring at the same time by features such as multi-ring and multi-device.
To use the no ring feature, the following items must be configured.

- The ‘No Ring’ service must be enabled in Class of Service.
- The ‘No Ring-Set’ and ‘No Ring-Cancel’ feature codes must be defined.

The user can register or cancel the no ring feature in the following ways:

- The user can dial the ‘No Ring-Set’ feature code on the phone to register the no ring feature.
- The user can dial the ‘No Ring-Cancel’ feature code on the phone to cancel the no ring feature.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the no ring.

The Multi-extension phone user can register or cancel the no ring feature in each multi-line:

- The ‘No Ring for Multi-Device’ service must be enabled in **[Configuration > User Group > Change UserGroup > Options]**.
- select a multi-extension phone that you try to set a ring/no-ring feature in **[Configuration > User> Multi-Extension Phone Setting]** menu and press the ‘Change’ button, it can be changed by selecting No Ring

5.2.24 Multi-Ring

If the multi-ring feature is enabled, when there is an incoming call for the user, the call is directed to multiple phones at the same time, and when the call is answered by one of the phones, the call is connected to the phone and the ring on other phones are canceled.
This service is useful for incoming calls to ring the landline and the mobile phone to ring at the same time.
To use the multi-ring feature, the following items must be configured:

- The ‘Multi-Ring’ service must be enabled in Class of Service.
- The ‘Multi-Ring Enable-Set’ and ‘Multi-Ring Enable-Cancel’ feature codes must be defined.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Even if a user is set as a member on a multi-ring list, the multi-ring feature must be enabled for the user in order to use the multi-ring feature. The user can register or cancel the multi-ring feature in the following ways:

- The user can dial the 'Multi-Ring Enable-Set' feature code on the phone to enable the multi-ring feature.
- The user can dial the 'Multi-Ring Enable-Cancel' feature code on the phone to disable the multi-ring feature.

The administrator can add members to multi-ring lists by using the **[CONFIGURATION > Service > User Service > Multi-ring List]** menu.
The user can use the feature code to add members to multi-ring lists. For the user to add multi-ring members, the 'Multi-Ring Member-Insert' and 'Multi-Ring Member-Delete' feature codes must be defined.

The user can add or remove multi-ring members by using feature codes in the following ways:

- The user can dial the 'Multi-Ring Member-Insert' feature code + phone number to add a multi-ring member.
- The user can dial the 'Multi-Ring Member-Delete' feature code + phone number to remove a multi-ring member.

When there is an incoming call, services enabled for the master user who enabled multi-ring will be provided, but the services enabled for the multi-ring members will not be provided. Note that the no ring service is provided to all users.

For example, when the user number 2000 is set as a multi-ring member for the user number 1000, if there is an incoming call for the user number 1000:

- The call will be forwarded if call forwarding is enabled for the user number 1000, but the call will not be forwarded if call forwarding is enabled for the user number 2000.
- The call will be rejected if DND is enabled for the user number 1000, but the call will not be rejected if DND is enabled for the user number 2000.
- Only the user number 1000 will not ring if no ring is enabled for the user number 1000, and only the user number 2000 will not ring if no ring is enabled for the user number 2000.

When the master user who enabled multi-ring is busy, the incoming call is serviced according to the 'Allow Other Ring' setting which is accessible through the pop-up window when the Enable button is selected after selecting the **[CONFIGURATION > Service > Feature Service > Service Activation > Multi-ring]** menu.

- Master Busy Not Allow: If the master user who enabled multi-ring is busy, the incoming call is treated as a call when busy and is not directed to the multi-ring members.

- Always Alloow: If the master user who enabled multi-ring is busy, the incoming call is not treated as a call when busy and is directed to the multi-ring members. When there is no multi-ring member to ring, the call is treated as a call when busy.
- Any Member Busy Not Allow: If the multi-ring master or a member is busy, the incoming call is treated as a call when busy and is not directed to the multi-ring members or multi-ring master.

5.2.25 Mobile Extension (MOBEX)

The mobile extension (MOBEX) feature allows incoming calls to be directed not only to the landlines and mobiles phones registered with IPX-S300B but also to external phone numbers. This is one example of the multi-ring service.
The service also allows the user to answer the call with his/her mobile phone and then when the user returns to the office, the call can be transferred to the landline in the office and be picked up for continued conversation.

MOBEX Call Pickup

This service allows the call answered with an external mobile phone by the multi-ring feature to be transferred to the landline in the office and picked up for continued conversation.
To use the MOBEC Call Pickup feature, the 'Moved Call Pickup' feature code must be defined.
The user can dial the call Pickup on desk phone feature code on the master phone enabled with multi-ring to pick up the call from the mobile phone.
After answering an incoming call with a mobile phone enabled with multi-ring, the user can press the 'Moved Call Pickup' feature code on his/her master phone during the call to transfer the call to the master phone.

Transfer to Mobile Phone

This service allows the user to transfer a call to an external mobile phone specified as a multi-ring member without parking the call. It works in the same way as blind transfer.
The user can dial press the transfer button on the master phone enabled with multi-ring during a call to transfer the current call to the mobile phone.
To transfer a call, press the transfer button and a mobile phone number on the master phone during the call and end the call.

5.2.26 Remote Office

The remote office feature allows automatic forwarding of all incoming calls for a user to an internal number or an external number specified.
The remote office feature works in the same way as blind transfer but it is defined for remote use. It is also similar to the follow me to destination feature but it is different in that the calls can be forwarded to phone numbers outside the system.

To use the remote office feature, the following items must be configured.

- The 'Remote Office' service must be enabled in Class of Service.
- The 'Remote Office-Set' and 'Remote Office-Cancel' feature codes must be defined.

The user can register or cancel the remote office feature in the following ways:

- The user can dial the 'Remote Office-Set' feature code + the destination phone number on the phone to register the remote office feature.
- The user can dial the 'Remote Office-Cancel' feature code on the phone to cancel the remote office feature.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the remote office feature.

5.2.27 Wake-Up Call

The wake-up call feature allows the user's phone to ring at a wake-up time specified by the user. If the user answers the call, an announcement is played to notify that it is the wake-up time.
To use the wake-up call feature, the following items must be configured:

- The 'Wake-Up Call' service must be enabled in Class of Service.
- The 'Wake-Up Call-Set' and 'Wake-Up Call-Cancel' feature codes must be defined.

The user can register or cancel the wake-up call feature in the following ways:

- The user can dial the 'Wake-Up Call-Set' feature code + the wake-up retry type (1: Once/2: Repeat) + the wake-up time on the phone to enable the wake-up call feature.
- The user can dial the 'Wake-Up Call-Cancel' feature code + the wake-up time (HHMM) on the phone to cancel the wake-up call feature.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the wake-up call feature. When enabling wake-up call, you can also set the wake-up time and the number of times the phone will ring.

The following items should be configured for wake-up service by operator in the **[CONFIGURATION > Service > User Service > Wake-Up Call Lists]** menu.

Item	Description
User Group	Select a user group for which the wake-up call will be created.
Extension Number	Select an extension number for which the wake-up call will be created.
Wake-Up Index	If subscriber needs more than 2 wakeup services, 'Extended Alarm Reminder' feature can be used. This feature is aimed to support hotel service. It supports 4 wakeup services per subscriber. So there is an index (1-4) to separate which one is set or cancel.

Item	Description
	'1' is a default index for wakeup feature.
Wake-Up Retry Type	Select a service count for Once or Repeat
Wake-Up Time	Input wake-up ringing time. If service count is Once, should be input MMDDHHMM (Month, Date, Hour, Minute) or HHMM (Hour, Minute) type

5.2.28 Voice Mail Integration

IPX-S300B's built-in voice mail system is utilized for providing the basic voice mail services including answering machine emulation, call recording, deflection to voice mail, and transfer to voice mail.
The administrator needs to set Application Type to 'Internal UMS' in the **[CONFIGURATION > Application > VM/AA Server]** menu to create a connection to the built-in voice mail system.
There are three services for VM/AA server. The administrator can make several VM/AA servers. But only one service can only appoint one service.

- Voice Mail: All services except for Call Recording and Auto Attendant
- Call Recording
- Auto Attendant

Answering machine emulation and call recording services are provided as three-way conference calls. Since IPX-S300B utilizes its built-in conference system for establishing three-way conference calls, the basic settings for using the conference system must be configured.
The administrator needs to set Application Type to 'Internal Conference' in the **[CONFIGURATION > Application > Conference Server]** menu to create a connection to the built-in conference system. This conference server should be included 'Add-On Conference' in service list.

5.2.28.1 Answering Machine Emulation (AME)

If the AME feature is enabled, when there is an incoming call, the call is automatically answered by the voice mail system and the caller's message is recorded in the mailbox. The voice mail system announcement and the caller's voice message are heard over the phone's speaker.

AME Auto Start

This method allows the incoming calls to be connected to AME by configuring the AME feature in advance.
To use the AME auto start feature, the following items must be configured:

- The 'AME' service must be enabled in Class of Service.
- The 'AME Enable-Set' and 'AME Enable-Cancel' feature codes must be defined.

The user can register or cancel the AME auto start feature in the following ways:

- The user can dial the ‘AME Enable-Set’ feature code on the phone to enable the AME auto start feature.
- The user can dial the ‘AME Enable-Cancel’ feature code on the phone to cancel the AME auto start feature.
- The user can use the **[Supplementary Service]** menu on SCM Personal Assistant to register (enable) or cancel the AME auto start feature.

If the AME auto start feature is enabled, when there is an incoming call and the call is not answered, the call forward no answer feature is used for forwarding the call to the voice mail system to automatically start the AME.

AME Manual Start

This method allows the incoming calls to be connected to AME without configuring the AME feature in advance.
To use the AME manual start feature, the following items must be configured:

- The ‘AME’ service must be enabled in Class of Service.
- The ‘AME Mode-Start’ and ‘AME Mode-Stop’ feature codes must be defined.

When the user’s phone rings, the user can press the ‘AME Mode-Start’ button to process the call with call forward no answer and connect the call to the voice mail system.
The user can press the ‘AME Mode-Stop’ button on the phone while AME is in action, the caller will be connected to the user and AME will stop.

The Answering Machine Emulation (AME) feature allows a user to listen to a caller leaving a message in his/her voice mail box. It operates like a home answering machine.

- AME must be allowed in Class of Service
- To use AME feature, Call Forward No-Answer to Voice Mail must be set.
- Feature is enabled by a programmable ‘AME Enable’ button or feature access code.
  - Toggle button ON/OFF > Red LED when ON
  - Dial the assigned feature access code ‘xxx + 0/1’ 0 = Off and 1 = On
- The optional ‘AME Mode’ button can be assigned to any programmable button.
  - When AME is enabled and a call is ringing your phone, press the ‘AME Mode’ button to skip the call forward no-answer timer and immediately activate the AME feature.
  - To stop hearing the audio of the caller leaving a message in your mailbox:
    - Press the Speaker button or the ‘End’ button in the soft key menu. The caller is still connected to your mailbox.
    - To speak with the caller, lift the handset or press the ‘Answer’ button in the soft key menu. The mailbox is disconnected.

5.2.28.2 Call Recording

This feature allows the call conversation to be recorded during a call.
When call recording begins, the 'Recording' message will be shown on the phone display, and the CANCEL, PAUSE, and STOP soft menus will be displayed for use.

Auto Call Record

If call recording feature is enabled, this service automatically records calls whenever they are started.
To use the auto call record feature, the following items must be configured:

- The 'Call Recording' and 'Auto Record' services must be enabled in Class of Service.
- Auto Record should be enabled in [CONFIGURATION > Service > Feature Service > Service Activation] menu.

When a user for whom the auto call record feature is enabled is on a call, a three-way conference call will automatically be connected to the voice mail system and the call will be recorded.
When enabling the auto call record feature, you can specify a type of calls to record selectively.
The call type can be changed by 'Auto Record Type' in the **[CONFIGURATION > Service > Feature Service > Service Activation > Auto Record]** menu.

- Incoming: Incoming calls are recorded.
- Outgoing: Outgoing calls are recorded.
- Both: Both the incoming and outgoing calls are recorded.

Manual Call Record

This feature allows the call conversation to be recorded during a call by pressing the call record button.
To use the manual call record feature, the following items must be configured:

- The 'Call Record' service must be enabled in Class of Service.
- The 'Call Record' feature code must be defined.

If the user presses the 'Call Record' button + the mailbox number during a call, a three-way conference call will be established with the voice mail system and the call will be recorded in the selected mailbox. If a mailbox number is not entered, the call will be recorded in the user's mailbox.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.2.28.3 Deflect to Voicemail

This service forwards allows the currently ringing call to be forwarded to the voice mail system by using the call forward no answer feature.
The voice mail system answers the call immediately and plays the no answer announcement so that the caller can leave a voice mail.
To use the deflect to voice mail feature, the 'VM Transfer' feature code must be defined. If the user presses the deflect to voice mail button on the phone which is ringing, the call will be processed for call forward no answer and be connected to the voice mail system.

NOTE

Deflect to VM key can be generated in the program key item.
It can be generated in **[CONFIGURATION > User > Phone Key Programming]** menu.

5.2.28.4 Transfer to Voicemail

This feature allows the current call to be connected to a specified mailbox in the voice mail system so that the caller can leave a message.
If the current call is transferred to the voice mail system by a normal method, the voice mail system asks for the service code, mailbox number, password, etc. But if the transfer to voice mail feature is used for transferring the call, this step is skipped so that the caller can leave a voice message without entering anything.
To use the transfer to voice mail feature, the 'VM Transfer' feature code must be defined. When the user dials the transfer to voice mail feature code + a mailbox number during a call and ends the call, the call will be transferred to the voice mail system and the caller will be allowed to leave a voice mail in the selected mailbox.

NOTE

Transfer to VM key can be generated in the program key item.
It can be generated in **[CONFIGURATION > User > Phone Key Programming]** menu.

5.2.28.5 Other User MWI

This feature allows to connect to the other user's voice mail Box and verify the other user's MWI.
To use this feature, Other User MWI button must be created in phone. Other User MWI button can be created in **[CONFIGURATION > Users > Phone Key Programming]** menu. When created this button, input the other user's phone number in value field.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.2.29 Personal SPAM Number

This feature allow user to configure SPAM numbers. When a call comes in from internal or external to the user, if the number is matched with pre-configured SPAM list then reject the call.

To use Personal SPAM Number feature, following items are required in the **[CONFIGURATION > Service > User Service > Personal SPAM Number]** menu.

Item	Description
User Group	Select a user group for which the Personal SPAM number will be configured
Extension Number	Specify a user number for which the service will be activated.
SPAM Number	Enter the number specified as SPAM number.
Activation	Select 'Enable' or 'Disable' to use the Number as SPAM.

5.2.30 Pause Digit

In some cases, specific digits should be entered after making a call, It can be used for authentication. These digits can be assigned in speed dial menu after pause digits. IPX-S300B makes a call and send digits after pause delay time. Pause delay time depends on the number of pause digits ('p' or 'P')

This feature is served with the following services.

- Hot Line
- Speed Dial
- Call Forward
- Multi-Ring
- Paging on Answer
- Predefined Conference

In the **[CONFIGURATION > User Group > Timers > Pause Delay Time]**, the delay time between pause digits can be changed. And the delay time between normal digits after pause digit can be changed in the **[CONFIGURATION > User Group > Timers > DTMF Duration Time]**.

Item	Description
PAUSE Delay Time	Specify the delay time between pause digits.
DTMF Duration Time	Specify the delay time between normal digits after pause digit.

The destination number for services should include pause digits. For the detailed configuration for services, refer to the menus of each service.

5.2.31 Call Bridge

This feature allows the gateway FXS user to join the conversion of bridged user by hook-off. After that, if bridged user is hook-off, FXS user has a continued conversation.
If bridged user is not busy status, the gateway FXS user listens dial-tone and makes a call.

In the **[CONFIGURATION > Service > Feature Service > Service Activation > Hot Line],** the bridge feature code and the extension number of bridged user should be entered for the gateway FXS user. And the Hot Line Delay Time should be greater than 0.

5.2.32 Move to Mobile

The conversation can be continued through other phone. But there is no need to hold the call. It is the difference between Call Transfer and Call Move.

5.2.32.1 Call Move using Soft Menu

The conversation can be continued through other phone by using soft menus displayed on the phone.

Move to Mobile (Move to Number)

The ‘Move to Mobile’ soft menu will be displayed on the phone which is on conversation. If a user selects the soft menu, the conversation will be continued through the mobile phone which was set before.
In case a user selects the ‘Move to Number’ soft menu, the destination number for Call Move should be entered. And the call will be moved to the designated destination.
The ‘Move to Mobile’ feature code should be defined in the **[CONFIGURATION > Service > Feature Service > Feature Code]** menu.
The ‘Move to Mobile’ Service should be permitted in the **[CONFIGUARTION > Service > Class of Service]** menu.

Move to Multi-Device

If a multi-device is on conversation, the ‘Move to Multi-Device’ soft menu will be displayed on the phone. And the call can be moved to the other multi-device member, if a user selects the menu.
The ‘Move to Multi-Device’ feature code should be defined in the **[CONFIGARITON > Service > Feature Service > Feature Code]** menu.
The “Move to Multi-Device” feature provides to IP-Phone.

5.2.32.2 Call Move using Programmable Button

The conversation can be continued through other phone by pressing 'Move to Mobile' programmable button.
If a user press this button on conversation, the conversation will be continued through the mobile phone which was set before.
The 'Move to Mobile' feature code should be defined in the **[CONFIGURATION > Service > Feature Service > Feature Code]** menu.
The 'Move to Mobile' Service should be permitted in the **[CONFIGUARTION > Service > Class of Service]** menu.
The 'Move to Mobile' button should be created in the **[CONFIGURATION > User > Phone Key Programming]** menu.

5.2.32.3 Call Move using Move Key

The desk phone like SMT-i5343 supports 'Move' key. The conversation through the desk phone can be moved to the other phone by pushing Move Key.
First of all, the connection between a desk phone and a mobile should be configured for Call Move service. If the mobile name is assigned in the **[CONFIGURATION > User>Single Phone User (or Multi-Extension Phone)>NFC Mobile Phone Name],** the conversation of the desk phone can always be moved to the mobile.
Second, the 'NFC Service' should be permitted in the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu.
Third, the 'NFC Move to Desk', 'NFC Move to Mobile' and 'NFC Pickup' codes should be set in the **[CONFIGURATION > Service > Feature Service > Feature Code]**.

Move to Mobile

If a desk phone is on conversation, the call can be moved to the mobile phone by using Move key.

Move to Desk

If a mobile is on conversation or is ringing, the call can be moved to the desk phone by using Move key.

Mobile Pickup Desk Call

If a desk phone is ringing, the mobile phone can answer the call by using Move key.

5.2.32.4 Moved Call Pickup

After moving the conversation to a mobile, the desk phone can pick up the call through 'Moved Call Pickup' soft menu. Although the moved call is 3G call, it is possible. It is the difference between 'Moved Call Pickup' and 'Move to Desk' features.
The 'Moved Call Pickup' codes should be set in the **[CONFIGURATION > Service > Feature Service > Feature Code]**.

5.2.33 CSTA Line Seize

Line seize status is considered to the CSTA event. Phone sends NORIFY message when a status of receiver is changed and it is used to issue CSTA Event. Service Initiated Event is delivered when the receiver is hooked off and Connection Cleared Event is delivered when it hooked on. When a call tries in the state of hook off, Service Initiated Event from hook off is cleared by Connection Cleared Event and new Service Initiated Event by a call is created and other CSTA events are followed by the call state. Temporary CSTA Call ID is used for Service Initiated Event by hook off and Service Initiated Event by a call has a meaningful CSTA Call ID.

To CSTA Line Seize service, configure following steps. Configurations below is used for both CSTA Line Seize service and BLF service.

- SIP Phone Configuration
  'Line Seize Option' has to be set 'Send Message' in the **[CONFIGURATION > User Group > Change User Group]** menu.
- FXS Phone Configuration
  'FXS Line Seize Option' has to be set 'Enable' in the **[CONFIGURATION > User > Single Phone User: Service]** menu.

FXS Line Seize service has an effect on the performance of gateway. Refer the Performance Guide below before user the feature.

* Gateway Performance by FXS Line Seize Usage (OX7500/1000 Line)

FXS Line Seize Option Count	Notify Traffic	Call Traffic	Supported CPS	Option Usage.
0	0	16	8	0 %
1000	8	8	4	100 %
500	4	12	6	50 %
250	2	14	7	25 %

5.2.34 Universal Answer

Service which responds to a call incoming to other user instead of him, and functions the same as Pick-Up Response. Yet, it can include members redundant with the group of Pick-Up Response, and only one group can be generated per one user group.
Universal Answer feature is serving the same service with Call Pickup. It can include members of the Pickup Group as its own members, and only one group can be generated per one user group.

In order to use Universal Answer feature, the following two items should be set.

- 'Universal Answer' feature code should be defined.
- Universal Answer Group and the following data should be set in **[CONFIGURATION > Service > Group Service > Universal Answer Group]** menu.

Item	Description
User Group	Select the user group to generate the Universal Answer group.
Group Name	Enter the name of the Universal Answer group.
Group Member	Select the member of Universal Answer group, among users. One user can belong only to one Universal Answer group.

In order to pick up the calls to the members of the Universal Answer group, Universal Answer Feature Code should be dialed.

5.2.35 Ring Deflect

The Ring Deflect feature is to transfer receiving calls to another number.
Although the call forward always operates automatically under predetermined conditions, the Ring Deflect is useful when the user forwards calls selectively.

The Ring Deflect feature can be used as follows.

- Select 'Deflect' in the 'Function' menu of the telephone while receiving calls.
- In the **[Configuration > User > Phone Key Programming/AOM Key Programming]** menu, create the 'Ring Deflect (input)' or the 'Ring Deflect (Designation)' button, and input the button while receiving calls.

NOTE

Note: The 'Deflect' menu displayed in the 'Function' menu of the telephone operates the same as the 'Ring Deflect (input)' button.

5.2.36 Ring Preference

The Ring Preference feature is a setting that can choose to answer a call or dial a number when the receiver is picked up or the speaker key is pressed while receiving calls.
The default is 'ENABLE' that answers a call.

The Ring Preference feature is useful when the user wants to make a call in a condition where receiving calls constantly occur in a call center.

In the [Configuration > User > Single Phone User/Multi-Extension Phone] menu, you can change to 'Ring Preference'

5.3 Wireless Enterprise Service

Wireless Enterprise Service is the one of FMC (Fixed Mobile Convergence) services, which offers VoIP service to the smart phone users.
This chapter describes how to configure Mobile Services Options, Mobile Phone Profile, etc for Wireless Enterprise Services.

5.3.1 Basic Configuration

License Key Registration

To use Wireless Enterprise Service, a wireless enterprise solution license is required.
The license key can be registered in the **[CONFIGURATION > Miscellaneous > License]** menu or **[CONFIGURATION > Miscellaneous > Activation Key]** menu

Creating the user for Wireless Enterprise Service

Creating the user for Wireless Enterprise Service is similar to how to create single phone user.
The user configuration for Wireless Enterprise Service can be configured as making single phone user in the **[CONFIGURATION > User > Single Phone User]**.

Normal Configuration is same as creating single phone user. Phone Type should be set to Samsung-Mobile-Phone and Mobile Phone Number should be set.
In addition, Select the ‘Use Mobile Phone Number’ option.

The following describe ‘Use Mobile Phone Number’ option.

Use Mobile Phone Number Option	Description
None	Do not use the mobile phone number. Only ring to the extension.
Ring Only	Both extension and mobile phone are ringing simultaneously. Multi ring service with extension and mobile phone.
Dial Only	Allow a Remote Dial Service.
Both	Provides a Remote Dial Service and Multi-Ring Service

Mobile Service Option

Wireless AP (Access Point) SSID (Service Set Identifier) can be configured in the **[CONFIGURATION > Wireless Enterprise > Mobile Service Options]** menu.
This SSID configured must be same the SSID of WiFi Configuration in the mobile phone.
When FMC client receives a call, ‘Wait Call/Later Call’ menu is displayed on the FMC client UI in case that the FMC client user cannot receives that call. If the FMC client user selects ‘Wait Call or Later Call’ menu on ringing, caller can listen to the announcement and then the call is held in case of ‘Wait Call’ and is disconnected in case of ‘Later Call’.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Wait Call/Later Call can be configured in the **[CONFIGURATION > Wireless Enterprise > Mobile Service Options > Wait Call, Later Call]** menu.
In addition, a description of the settings for the Mobile service options that are listed in the following table.
Administrators can set an item of **[Configuration> Wireless Enterprise> Mobile Service Options]**.

Item	Description
Remote Dial Public IP Address	In the General Preferences in the Public IP address to be used at Remote Dial, this item could be the IP of the SBC. (Default: None)
Remote Dial Public Port	The port of Public address used at Remote Dial, Administrator needs to Port Forwarding to 9011 port of SCME. (Default: None)
Mobile DISA Number	This item is a DISA dial Phone number is calling out in the case of Remote Dial Service failure. (Default: None)
Mobile DISA Code	This item is Feature CODE to use in the DISA Phone is sending to Remote Dial Service failure. (Default: None)
Mobile VMS DISA Number	This item is a VMS DISA Dial Number to connect to when you use the Remote DISA Service from the external network. (Default: None)
WE Work Server IP Address	This field, enter the Address of WE Work Server if you want to work with WE Work Server. (Default: None)
WE Work Server Port	This field, enter the port of WE Work Server if you want to work with WE Work Server. (Default: 80)
WE Work Server Public IP Address	This field, enter the Public Address of WE Work Server if you want to work with WE Work Server. (Default: None)
WE Work Server Public Port	This field, enter the Public Port of WE Work Server if you want to work with WE Work Server. (Default: 80)
WE VoIP CID Server IP Address	Administrator puts the CID Server IP Address in conjunction with WE VoIP. (Default: None)
WE VoIP CID Server Port	Administrator puts the CID Server Port in conjunction with WE VoIP. (Default: 80)
WE VoIP CID Server Public IP	This field, enter the Public Address of CID Server. (Default: None)
WE VoIP CID Server Public Port	This field, enter the Public Port of CID Server (Default: 80)
WE Work Server Protocol	This field, select the Protocol of WE Work Server if you want to work with WE Work Server. (Default: HTTP)
WE VoIP CID Server Protocol	This field, select the Protocol of WE VoIP CID Server if you want to work with CID Server (Default: HTTPS)
WE VoIP CID Server Public Protocol	This field, select the Protocol of WE VoIP CID Server if you want to work with CID Server in Public Network. (Default: HTTP)
Wait Call, Later Call	Administrator decide whether user want to use to use the phone when receiving WE VoIP “Wait Call, the Later Call] option.

Item	Description
	(True/False, Default: False)
Wifi Band	set the Band of the AP user want to connect to. (Default: Auto) Administrator should be careful when you can choose to 2G/5G/Auto, WE VoIP to register, AP of the frequency as it scans.
Auto Answer CLI Number	Specify the calling number that is displayed when user use the Call Move Service and Remote Dial Service.
Auto Answer Profile Number	When use the Call Move Service and Remote Dial Service, Administrator specify automatic response received calling number. If you specify a value that is similar to the Auto Answer CLI Number, WE VoIP will automatically respond to the Call Move Service and Remote Dial Service is used.
Use 3G Call Only	Select this option to the calling party via a mobile phone unconditionally WE VoIP even if they are registered. (Default: No)
3G Call Prefix	If you begin with a number of specific, rather than WE VoIP, is an option that allows you to send to the mobile phone. (Default: None)
Private (Office) SSID List	When Private AP's SSID List value is not 'None', This value is used to check for the Private WiFi SSID List. (Default: None)
WiFi Hotspot SSID List	When WiFi Hotspot AP's SSID List value is not 'None', This value is used to check for the public WiFi Hotspot SSID List. (Default: None)
2.4G Channel List 5G Channel List	I will select the Channel you want to scan to WE VoIP use. Since the scan only to Channel WE VoIP use, Administrator need to be careful.

Mobile Phone Profile

The phones, which phone type is Samsung-Mobile-Phone are displayed in the **[CONFIGURATION > Wireless Enterprise > Mobile Phone Profile]** menu.
The parameters related to FMC is configured in the **[CONFIGURATION > Wireless Enterprise > Mobile Phone Profile]** menu.
The following items by setting default values are used except for the special site.

Item	Description
Roaming Trigger	Roaming Trigger Parameter for busy FMC client (Default: -70) Comparing with the RSSI, this value is used to scanning WiFi
Roaming Delta	Roaming Delta Parameter for busy FMC client (Default: 10) When this value is more than the minimum roaming delta, FMC clients scan WiFi and Roam
Roaming Scan Period	Roaming Scan Period Parameter for busy FMC client (Default: 3)
Noise Suppression RX	Select whether to use FMC Client Noise Suppression RX (Default: Disable)
Noise Suppression TX	Select whether to use FMC Client Noise Suppression TX (Default: Disable)
AECM	Select whether to use FMC Client AECM (Default: Speaker Phone)

Item	Description
Echo Suppression	Echo Suppression (Default: Enable)
Enable Swing Free RX	Select whether to use Swing Free RX (Default: Enable)
Enable Swing Free TX	Select whether to use Swing Free TX (Default: Enable)
Enable CNG	Comfort Noise Generation (Default: Enable)
Media Start Port	FMC Client Media Start Port
Media End Port	FMC Client Media End Port
Multiframe Enable	Select whether to use Voice Engine Multiframe with Samsung AP (Default: Disable)
Multicast Enable	Select whether to use Voice Engine Multicast with Samsung AP (Default: Disable)
TOS Media Value (DSCP)	You can set TOS (DSCP) value of RTP packets for IP header. The value controls priority of RTP packets in samsung AP/APC.
TOS Control Value (DSCP)	You can set TOS (DSCP) value of SIP packets for IP header. The value controls priority of SIP packets in samsung AP/APC.
JBC Threshold	Jitter Buffer Size of Voice Engine (Default: 4)
SIP Rx Port	FMC Client’s Rx Port of SIP Message (Default: 5080)
SIP Rx Public Port	FMC Client’s Rx Public Port of SIP Message (Default: 5080)
Protocol-public network	SIP Message Protocol for public network zone (Default: TLS)
TLS Connection-public network	SIP TLS Connection Mode for Public Network Zone (Default: Reuse)
Expire Time (sec)-private network	Registration Expire Time for Private WiFi Network Zone (Default:3600)
Expire Time (sec)-public network	Registration Expire Time for Public Network Zone (Default: 3600)
SIP Registration-WiFi Hotspot	Select whether to service for Public WiFi Hotspot Network Zone (Default: Enable)
SIP Registration-mVoIP	Select whether to service for LTE Network Zone (Default: Disable)
Directory Service-Private (Office)	Select option for Using Directory Service when Private WiFi Network Zone register (Default: Disable)
Directory Service-WiFi Hotspot	Select option for Using Directory Service when public WiFi Hotspot Network Zone register (Default: Disable)
Directory Service-mVoIP	Select option for Using Directory Service when LTE Network Zone register (Default: Disable)
Presence Service-Private (Office)	Select option for Using Presence Service when Private WiFi Network Zone register (Default: Disable)
Presence Service-WiFi Hotspot	Select option for Using Presence Service when public WiFi Hotspot Network Zone register (Default: Disable)
Presence Service-mVoIP	Select option for Using Presence Service when LTE Network Zone register (Default: Disable)
Location Service-Private	Select option for Using Location Service when Private WiFi Network

Item	Description
(Office)	Zone register (Default: Disable)
Location Service-WiFi Hotspot	Select option for Using Location Service when public WiFi Hotspot Network Zone register (Default: Disable)
Location Service-mVoIP	Select option for Using Location Service when LTE Network Zone register (Default: Disable)
Update Location-Min Distance (m)	Enter Minimum Distance Meters for Location Service (default: 500)
Update Location-Min Period (min)	Enter Minimum Period Minutes for Location Service (default: 10)
Update Location-Forced Period (hour)	Enter a Upload Period Hour for Location Information (default: 3)
Update Location-Start Time	Enter a Service's Start Time for Location Service (default: 08:00)
Update Location-End Time	Enter a Service's End Time for Location Service (default: 08:00)
Voice Quality Report-Private Network	Select whether to report FMC Client Voice Quality (Private Network)
Voice Quality Report-Wi-Fi Hotspot	Select whether to report FMC Client Voice Quality (Wi-Fi Hotspot)
Voice Quality Report-mVoIP	Select whether to report FMC Client Voice Quality (mVoiP)
Voice Quality Report-MOS	MOS value-when FMC Client' MOS value is under this MOS, FMC Client will report
NoticeBoard Service-Private (Office)	Select option for Using NoticeBoard Service when Private WiFi Network Zone register (Default: Disable)
NoticeBoard Service-WiFi Hotspot	Select option for Using NoticeBoard Service when public WiFi Hotspot Network Zone register (Default: Disable)
NoticeBoard Service-mVoIP	Select option for Using NoticeBoard Service when LTE Network Zone register (Default: Disable)

Mobile Service Server

For additional Mobile phone service, server configuration for that service can be configured in the **[CONFIGURATION > Wireless Enterprise > Mobile Service Server]** menu.

- Log File Upload Server
  The parameters related to log file which Mobile Phone writes.
  Configured in the [CONFIGURATION > Wireless Enterprise > Mobile Service Server > Log File Upload Server] menu.

Item	Description
User Group	User group to which the Log File Server belongs

Item	Description
Name	Server name
Service Type	- Internal: IPX-S300B is operated as Log server. IP addr/Port are inputted automatically. - External: External Log server.
IP Address	IP Address of Log Server
Port	Port of Log Server
Protocol	Protocol of Log Server
Public IP Address	Public IP Address of Log Server
Public Port	Public Port of Log Server
Public Protocol	Public Protocol of Log Server
File Path	Log file path (directory)
Server ID	Server ftp ID. The ID should be made in IPX-S300B OS.
Server Password	Server password

- Logo File Upload Server
  The parameters related to logo file which Mobile Phone use.
  Configured in the [CONFIGURATION > Wireless Enterprise > Mobile Service Server > Logo File Upload Server] menu.

Item	Description
User Group	User group to which the Logo File Server belongs
Name	Server name
Service Type	- Internal: IPX-S300B is operated as Logo server. IP Addr and Port are inputted automatically. - External: External Log server.
IP Address	IP Address of Logo Server
Port	Port of Logo Server
Protocol	Protocol of Logo Server
Public IP Address	Public IP Address of Logo Server
Public Port	Public Port of Logo Server
Public Protocol	Public Protocol of Logo Server
Logo File Name	Logo File Name
Logo File (Splash) Name	Logo File Name (Splash)
Logo File (Background) Name	Logo File Name (Background)

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

- Photo File Upload Server
  The parameters related to photo file which Mobile Phone download or upload for directory service.
  Configured in the [CONFIGURATION > Wireless Enterprise > Mobile Service Server > Photo File Upload Server] menu.

Photo file is available in the **[CONFIGURATION > User > User Profile]** menu.

Item	Description
User Group	User group to which the Server belongs
Name	Server name
Service Type	- Internal: IPX-S300B is operated as the photo server. IP Addr and Port are inputted automatically. - External: External photo server.
IP Address	IP Address of the Server
Port	Port of the Server
Protocol	Protocol of the Server
Public IP Address	Public IP Address of the Server
Public Port	Public Port of the Server
Public Protocol	Public Protocol of the Server

- Directory Service Server
  The parameters related to directory service which Mobile Phone use.
  Configured in the [CONFIGURATION > Wireless Enterprise > Mobile Service Server > Directory Service Server] menu.

In addition, configure the related option in the **[CONFIGURATION > Wireless Enterprise > Mobile Phone Profile]** menu for directory service activation on Mobile phone.

Item	Description
User Group	User group to which the Server belongs
Name	Server name
Service Type	- Internal: IPX-S300B is operated as the directory server. IP Addr and Port are inputted automatically. - External: External directory server.
IP Address	IP Address of the Server
Port	Port of the Server
Protocol	Protocol of the Server
Public IP Address	Public IP Address of the Server
Public Port	Public Port of the Server

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Public Protocol	Public Protocol of the Server

- Presence Service Server
  The parameters related to presence service which Mobile Phone use.
  Configured in the [CONFIGURATION > Wireless Enterprise > Mobile Service Server > Presence Service Server] menu.

In addition, configure the related option in the **[CONFIGURATION > Wireless Enterprise > Mobile Phone Profile]** menu for presence service activation on Mobile phone.

Check configurable presence type in the **[CONFIGURATION > User Group > Presence Field Name]** menu

Item	Description
User Group	User group to which the Server belongs
Name	Server name
Service Type	- Internal: IPX-S300B is operated as the presence server. IP Addr and Port are inputted automatically. - External: External presence server.
IP Address	IP Address of the Server
Port	Port of the Server
Protocol	Protocol of the Server
Public IP Address	Public IP Address of the Server
Public Port	Public Port of the Server
Public Protocol	Public Protocol of the Server

- Location Service Server
  The parameters related to location service which Mobile Phone use.
  Configured in the [CONFIGURATION > Wireless Enterprise > Mobile Service Server > Location Service Server] menu.

In addition, configure the related option in the **[CONFIGURATION > Wireless Enterprise > Mobile Phone Profile]** menu for location service activation on Mobile phone.

Check Mobile Phone Location Service option in the **[CONFIGURATION > User Group > Change User Group > Options]** menu for service activation.

Item	Description
User Group	User group to which the Server belongs
Name	Server name

Item	Description
Service Type	- Internal: IPX-S300B is operated as the location server. IP Addr and Port are inputted automatically. - External: External location server.
IP Address	IP Address of the Server
Port	Port of the Server
Protocol	Protocol of the Server
Public IP Address	Public IP Address of the Server
Public Port	Public Port of the Server
Public Protocol	Public Protocol of the Server

Mobile Configuration

Roaming parameter of each phone model can be configured in the **[CONFIGURATION > Wireless Enterprise > Mobile Configuration > Roaming Parameter]** menu. The priority of these setting values is higher than the setting values of each extension in Mobile Phone Profile menu.

Item	Description
Roaming Trigger	Roaming Trigger Parameter for busy FMC client Comparing with the RSSI, this value is used to scanning WiFi
Roaming Delta	Roaming Delta Parameter for busy FMC client When this value is more than the minimum roaming delta, FMC clients scan WiFi and Roam
Roaming Scan Period	Roaming Scan Period Parameter for busy FMC client

Codec Priority and Payload Type for FMC Client can be configured in the **[CONFIGURATION > Wireless Enterprise > Mobile Configuration > Codec Priority]** menu. Each codec does not have same codec priority.

When the FMC client supporting SILK or AMR-WB negotiates a codec with phone (example: Desk Phone) that does not support SILK or AMR-WB, the codec negotiated is PCMA or PCMU.
Parameters for SILK or ARM-WB are configured in the **[CONFIGURATION > Wireless Enterprise > Mobile Configuration > Codec Config]** menu. The following values by setting the default values is used except for special testing purposes.

Item	Description
SILK: Sampling Freq.	SILK Sampling Frequency (Default: 24000)
SILK: Max Packet Time	SILK Maximum Packet Time (Default: 100)
SILK: DTX Use	Select whether to use SILK DTX (Default: Off)
SILK: FEC Use	Select whether to use SILK FEC (Default: On)

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
AMR-WB: Bit Rate	Select whether to use AMR-WB Bit Rate (Default: 23850)
AMR-WB: DTX Use	Select whether to use AMR-WB DTX (Default: Off)
RTP Packet Interval (ms)-Private	Enter a Media Packet Interval for Private WiFi Network Zone (Default: 20)
RTP Packet Interval (ms)-WiFi Hotspot	Enter a Media Packet Interval for Public WiFi Hotspot Network Zone (Default: 20)
RTP Packet Interval (ms)-mVoIP	Enter a Media Packet Interval for Public LTE Network Zone (Default: 40)

For additional configuration of mobile, using the following menu.
[CONFIGURATION > Wireless Enterprise > Mobile Configuration >Miscellaneous Mobile Config]

Item	Description
Caller Screen Phrase-Left (Korean)	Set the Phrase on the caller screen. (left-korean)
Caller Screen Phrase-Left (English)	Set the Phrase on the caller screen. (right-english)
Caller Screen Phrase-Right (Korean)	Set the Phrase on the caller screen. (left-korean)
Caller Screen Phrase-Right (English)	Set the Phrase on the caller screen. (right-english)

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.3.2 Mobile Remote Dial

A WE VoIP client can make a call through IPX-S300B, even though the client is out of WIFI network. In this case, a WE VoIP client can request the ‘Mobile Remote Dial’ service through a data channel. If IPX-S300B receives the request, IPX-S300B makes a 3G call to the WE VoIP first. And if the WE VoIP answers the call, IPX-S300B makes a call to the destination which was requested before by the WE VoIP. After the destination answers the call, IPX-S300B lets them be connected.

The WE VoIP client can answer the 3G call automatically, if the **[CONFIGURATION > Wireless Enterprise > Mobile Service Options > Auto Answer CLI Number/Auto Answer Profile Number]** is configured.

The statistics information for the Mobile Remote Dial service is displayed in the **[PERFORMANCE > Statistics > System Service > Mobile Remote Dial]** menu.

5.3.3 Mobile DISA

If the WE VoIP client who requested the Mobile Remote Dial service receives ‘fail’ response, the client can make a Mobile DISA call through 3G network automatically.
After IPX-S300B let the DISA call be disconnected, IPX-S300B makes the 3G call for the WE VoIP.
If the WE VoIP answers the call, IPX-S300B requests the client enter the destination number after listening announcement. And IPX-S300B makes a call to the destination which was entered.
After the destination answer the call, IPX-S300B lets them be connected each other.

The WE VoIP client can answer the 3G call automatically, if the **[CONFIGURATION > Wireless Enterprise > Mobile Service Options > Auto Answer CLI Number/Auto Answer Profile Number]** is configured.

In the **[CONFIGURATION > Wireless Enterprise > Mobile Service Options > Mobile DISA Number/Mobile DISA Code]** menu, the ‘Mobile DISA Number’ and the ‘Mobile DISA Code’ should be set.

5.3.4 DID Number Service for FMC

IPX-S300B provides DID number of peer to FMC user during an internal call and FMC clients store this number in the call log. Therefore, FMC client can dial with this number within 3G network.

Select ‘DID Number Field’ in the **[CONFIGURATION > User Group > Change User Group > Options]**. Number types in the **[CONFIGURATION > User Group > User Profile Field Name]** is listed up for ‘DID Number Field’. If ‘DID Number Field’ is not selected, an extension number is stored in the call log in case of a internal call.

5.3.5 Handover

Handover is a function required to provide the continuity of calls when a subscriber of WE VoIP goes out of Wi-Fi network during a call.

5.3.5.1 Manual Handover

If the We VoIP subscriber has moved out of the WIFI network during WIFI call, WIFI call can be switched to 3G call by manually.
When you press a 'Switch to Mobile Phone' button on screen during a call in WE VoIP terminal, service will be working.
If IPX-S300B receives the request, IPX-S300B call to We VoIP subscriber through the 3G network.
If We VoIP subscriber receives the incoming 3G call, We VoIP subscriber can be connected to previous call by 3G.

5.3.5.2 Smart Handover

When the We VoIP subscriber has moved out of the WIFI network during WIFI call, this function is used to provide continuity of currency.

Smart handover is divided into the 'Handover-Out' and 'Handover-In'.
The Smart Handover-Out is that when WE VoIP subscribers move out of WIFI network during a call, WIFI call is switched to 3G by automatically before it goes out from the WIFI network. The Smart Handover-In is that when WE VoIP subscribers move in of WIFI network during on Smart Handover-Out, 3G call is switched to WIFI by automatically.

The statistics of success/failure for Smart Handover services is provided through the SCM Administrator. It is found in the **[PERFORMANCE > Statistics > System Service > Smart Handover]** menu.

The Smart Handover function is only available in WE VoIP device. So WE VoIP configurations must be required basically. Additional setting required are as follows.

Interworking APC

To use a Smart Handover function, TLS link should be established between IPX-S300B and APC.
Maximum 24 APC can be configured and can be confirmed the current connection status of APC in the **[CONFIGURATION > Wireless Enterprise > APC List]** menu.

Item	Description
User Group	Select a user group for which APC will be configured
Name	Specify the APC name.
IP Address	Specify the APC IP Address.

Item	Description
Status	Confirm the current connection status of APC.

Class of Service Setting

To use a Smart Handover function, creating of Class of Service is needed.
Class of Service can be set in the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu. When create a Class of Service, Smart Handover Type should be set. Each type as follows.

Item	Description
Blank	Handover services are not allowed.
Handover Out	Handover Out service is only allowed.
Handover Both	Handover Out/In service is allowed.

Class of Service created is allocated per user in the **[CONFIGURATION > User > Single Phone User/Multi-Phone User]** menu.

Other Settings

WE VoIP subscriber's available mobile number must be configured to each user. Mobile number can be set in the **[CONFIGURATION > User > Single Phone User/Multi-Phone User]** menu.

When WE VoIP is moved in the WIFI network, Smart Handover-In is available function after being WE VoIP registered with the IPX-S300B. There are needed a Service Guard Time because registering with the IPX-S300B are needed a few seconds.
So, Smart Handover-In is working after Service Guard Time.
Default Service Guard Time is 0 seconds and Service Guard Time can be changed in the **[CONFIGURATION > Miscellaneous > System Options > Smart Handover-In Waiting Time (second)]** menu.

To use Smart Handover-Out, it is needed the access code for Smart Handover-Out.
The access code can be configured in the **[CONFIGURATION > User Group > Change User Group > Information > Service Call Access Code]** menu.

When VoIP is moved out the WIFI network during external call, Smart Handover-Out is working. In this case, route type must be set to TIE for Smart Handover-Out operation.
That configuration can be set in the **[CONFIGURATION > Trunk Routing > Route > TIE Trunk]** menu.

Services Limitation

Smart Handover is only served basic internal/external call case. Hold call is not served the Smart Handover.

In Active-Active system case, WE VoIP must be enabled a dual registration option for Smart Handover operation. That configuration can be set in the **[CONFIGURATION > User > Single Phone User/Multi-Phone User]** menu.

5.3.6 Smart Routing

Extension subscribers when dialing a mobile phone number of WeVoIP subscriber, smart routing service is a service that analyzes the number and to call the extension number of WeVoIP subscriber without sending it to the outgoing trunk.

To use the smart routing service, please set the following items in **[Configuration > User Group > Change User Group > Information]** menu.

Item	Description
Smart Routing Activate	Operator will be able to choose whether or not to use smart routing service.
Smart Routing including Access Code	Operator will be able to choose whether to provide smart routing services to the number that contains the Access code.
Smart Routing Call Reject	It is the ability to choose to call directly to the mobile phone through the Trunk. Subscribers that use this item will not be able to receive the smart routing service.
Service After Smart Routing	If there is a service the subscriber set, you will be able to choose whether to apply the smart routing. The service that Operator set, Call Forward, Do Not Disturb, Absense, Follow Me is belong.

Operator can set **[Smart Routing Class of Service]** for the subscriber or Service Groups or User Groups.
Subscriber, Service Group, the User Groups is the order of priority of Class of Service.

If the WeVoIP subscriber has no response when receiving a call to an extension number of WeVoIP subscriber through a smart routing service, the system can offers a redial service to mobile phone number of the subscriber.

In order to use Redial service ‘Service Call Access Code’ should be set in **[CONFIGURATION > User Group > Change User Group > Information]** menu.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

5.3.7 Receiving Call in Hot Spot Zone

[Receiving Call in Hot Spot Zone] allows that a call from/to Hot Spot Zone treats as a internal call from subscriber.
[Hot Spot Zone] means that an external Wifi network.

To use the **[Receiving Call in Hot Spot Zone]**, need to set the following item in the **[Configuration > Service > Feature Service > Class of Service]** menu.

Item	Description
Receiving Call in Hot Spot Zone	Operator will be able to choose whether or not to use **[Receiving Call in Hot Spot Zone]** Service.

Class of Service for 'Receiving Call in Hot Spot Zone' can be set respectively for user group, service group and subscriber. The order of the priority is subscriber, service group and user group.
When a subscriber using WE VoIP does not respond to the call which tries to connect with extension number, redial will be performed to the mobile number of the User if the dialed number of the first outgoing call is a mobile phone number. In order to use Redial service, 'Service Call Access Code' should be set in **[CONFIGURATION > User Group > Change User Group > Information]**.

5.3.8 Receiving Call in mVoIP Zone

[Receiving Call in mVoIP Zone] allows that a call from/to mVoIP Zone treats as a internal call from subscriber.
[mVoIP Zone] is meaning that an external mobile data network.

To use the **[Receiving Call in mVoIP Zone]**, need to set the following item in the **[Configuration > Service > Feature Service > Class of Service]** menu.

Item	Description
Receiving Call in mVoIP Zone	Operator will be able to choose whether or not to use **[Receiving Call in mVoIP Zone]** Service.

'Receiving Call in mVoIP Zone' can be set respectively for user group, service group and subscriber. The order of the priority is subscriber, service group and user group.

5.3.9 IP Domain Based Call Restriction

IPX-S300B provides call restriction service by IP address interworking with iBG SBC.
(To configure IP address, refer to iBG SBC manual.)
IPX-S300B can configure restriction policy for 2 zone (ZONE A and ZONE B) in the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu

5.3.10 Desk phone Simultaneous Ring Delay

The WE VoIP, 3G Mobile and desk phones can ring simultaneously through multi-ring or multi-device service. In this case, the desk phone rings first. It could let the other phone pick up the call before WE VoIP or 3G mobile rings.
So the 'Ring Start Time' of the desk phone can be delayed. It operates only when WE VoIP or 3G mobile members exist for simultaneous ring.

The **[CONFIGURATION > User > Single Phone User (or Multi-Phone User) > Desk phone Simultaneous Ring Delay]** should be enabled.

The delay time can be changed in the **[CONFIGURATION > User > User Service Timers]** menu. The default value is 3 seconds.

5.3.11 Smart WLAN Link Configuration

This chapter describes the Smart WLAN Link Profile configuration for using the Smart WLAN Link Application on Mobile Phone.

License Key Registration

To use Smart WLAN Link Application, a Smart WLAN Link license is required.
The license key can be registered in the **[CONFIGURATION > Miscellaneous > Activation Key]** menu
In addition, configure the 'Use Smart WLAN Link' option in the **[CONFIGURATION >User >Single Phone User]** menu

Create Profile

To change the Smart WLAN Link User Profile, create Smart WLAN Link Site Profile, Smart WLAN Link Server Profile, Smart WLAN Link Network Profile and Smart WLAN Link Management Service Profile first.
After then assign the profile to each Smart WLAN Link user.

To change a number of Smart WLAN Link user profile, use the function **[Tool > Customer Data Import/Export]**.
Refer to '2.3 Changing All Users Data' in this Manual.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Smart WLAN Link Site Profile

The parameter of Server, Wi-Fi and Login Configuration for setup connection of Smart WLAN Link can be configured in the **[CONFIGURATION > Wireless Enterprise > Smart WLAN Link Configuration > Smart WLAN Link Site Profile]** menu.

30 Server profile can be configured.

Item	Description
Site Profile Info - Name	Name of Site Profile
Login Server Information - Server Address	Server Address of Login Server used to login
Login Server Information - Server Port	Server Port of Login Server used to login
Login Server Information - Public Server Address	Server Address of Login Server used to login after Primary Server Failure
Login Server Information - Public Server Port	Server Port of Login Server used to login after Primary Server Failure
Provision Server Information - Server Type	Server Type of Provision Server used to download provision file
Provision Server Information - Server Address	Server Address of Provision Server used to download provision file
Provision Server Information - Server Port	Server Port of Provision Server used to download provision file
Provision Server Information - Server Path	Server Path of Provision Server used to download provision file
Provision Server Information - Public Server Type	Server Type of Provision Server used to download provision file after Primary Server Failure
Provision Server Information - Public Server Address	Server Address of Provision Server used to download provision file after Primary Server Failure
Provision Server Information - Public Server Port	Server Port of Provision Server used to download provision file after Primary Server Failure
Provision Server Information - Public Server Path	Server Path of Provision Server used to download provision file after Primary Server Failure
Wi-Fi Information - SSID	SSID for connecting to Wi-Fi network
Wi-Fi Information - Security Mode	Security Mode for connecting to Wi-Fi network
Wi-Fi Information - Password	Password for connecting to Wi-Fi network
Login Information - Login Server	Configure Use of Login Server for User

Item	Description
Login Information - Common Information	Configure Use of Login Common Information for User
Login Information - Common Info ID	Common Information ID for Login
Login Information - Common Info Password	Common Information Password for Login
Login Information - Device Information	Configure Use of Device Information for User
Login Information - Device Info Password	Device Information Password for Login

Smart WLAN Link Server Profile

The parameter of Server Configuration for Smart WLAN Link Service can be configured in the **[CONFIGURATION > Wireless Enterprise > Smart WLAN Link Configuration > Wifi Server Profile]** menu.

30 Server profile can be configured.

Item	Description
Server Profile Information - Name	Name of Server Profile
Server Security Information - ID	ID used in Server Security Mode
Server Security Information - Password	Password used in Server Security Mode
Login Server Information - Server Address	Server Address of Login Server used to login
Login Server Information - Server Port	Server Port of Login Server used to login
Login Server Information - Public Server Address	Server Address of Login Server used to login after Primary Server Failure
Login Server Information - Public Server Port	Server Port of Login Server used to login after Primary Server Failure
Provision Server Information - Security Mode	Security Mode of Provision Server
Provision Server Information - Server Type	Server Type of Provision Server used to download provision file
Provision Server Information - Server Address	Server Address of Provision Server used to download provision file
Provision Server Information - Server Port	Server Port of Provision Server used to download provision file

Item	Description
Provision Server Information - Server Path	Server Path of Provision Server used to download provision file
Provision Server Information - Public Type	Server Type of Provision Server used to download provision file after Primary Server Failure
Provision Server Information - Public Address	Server Address of Provision Server to download provision file after Primary Server Failure
Provision Server Information - Public Port	Server Port of Provision Server to download provision file after Primary Server Failure
Provision Server Information - Public Path	Server Path of Provision Server to download provision file after Primary Server Failure
Upgrade Server Information - Security Mode	Security Mode of Upgrade Server
Upgrade Server Information - Server Type	Server Type of Upgrade Server to download upgrade files
Upgrade Server Information - Server Address	Server Address of Upgrade Server to download upgrade files
Upgrade Server Information - Server Port	Server Port of Upgrade Server to download upgrade files
Upgrade Server Information - Server Path	Server Path of Upgrade Server to download upgrade files
Upgrade Server Information - Public Type	Server Type of Upgrade Server to download upgrade files after Primary Server Failure
Upgrade Server Information - Public Address	Server Address of Upgrade Server to download upgrade files after Primary Server Failure
Upgrade Server Information - Public Port	Server Port of Upgrade Server to download upgrade files after Primary Server Failure
Upgrade Server Information - Public Path	Server Path of Upgrade Server to download upgrade files after Primary Server Failure
Log Server Information - Security Mode	Security Mode of Log Server
Log Server Information - Server Type	Server Type of Log Server to upload log files
Log Server Information - Server Address	Server Address of Log Server to upload log files
Log Server Information - Server Port	Server Port of Log Server to upload log files
Log Server Information - Server Path	Server Path of Log Server to upload log files
Log Server Information - Public Type	Server Type of Log Server to upload log files after Primary Server Failure
Log Server Information	Server Address of Log Server to upload log files after Primary

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
- Public Address	Server Failure
Log Server Information - Public Port	Server Port of Log Server to upload log files after Primary Server Failure
Log Server Information - Public Path	Server Path of Log Server to upload log files after Primary Server Failure

Smart WLAN Link Network Profile

The parameter of Network Configuration for Smart WLAN Link Service can be configured in the **[CONFIGURATION > Wireless Enterprise > Smart WLAN Link Configuration > Wifi Network Profile]** menu.

30 Server profile can be configured.

Item	Description
Network Profile Info - Name	Name of Network Profile
Wi-Fi Common - SSID	SSID for connecting to Service Wi-Fi
Wi-Fi Security - Security Mode	Security Mode for connecting to Wi-Fi Network
Wi-Fi Security - EAP Method	EAP Method for connecting to Wi-Fi Network
Wi-Fi Security - Provisioning	Provisioning Mode for connecting to Wi-Fi Network
Wi-Fi Security - Phase 2 Authentication	Phase 2 Authentication for connecting to Wi-Fi Network
Wi-Fi Security - CA Certificate	CA Certificate Path for connecting to Wi-Fi Network
Wi-Fi Security - User Certificate	User Certificate Path for connecting to Wi-Fi Network
Service Information - CM Service Activation	Configure Connection Manager Service Activation
Service Information - Roaming Service Activation	Configure Roaming Service Activation
Service Information - Roaming Service Method	Configure Roaming Service Method
Service Information - Roaming Service Mode	Configure Roaming Service Mode
Wi-Fi Parameter Information - Frequency Band	Frequency Band configured when Wi-Fi is connected

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Wi-Fi Parameter Information - Country Rev	Country Rev configured when Wi-Fi is connected
Config WES Wi-Fi Parameter - Activation	Configure WES Wi-Fi Parameter Activation (apply when Config 2G/5G WES Wi-Fi Parameter is not activated)
Config WES Wi-Fi Parameter - Roaming Trigger (dB)	Roaming Trigger configured when Wi-Fi is connected (A RSSI value to start channel scanning)
Config WES Wi-Fi Parameter - Roaming Delta (dB)	Roaming Delta configured when Wi-Fi is connected (RSSI Difference between Serving and Candidate AP that recognized as Target AP)
Config WES Wi-Fi Parameter - Roaming Scan Period (ms)	Roaming Scan Period configured when Wi-Fi is connected (Idle period after the each channel scanning iteration)
Config WES Wi-Fi Parameter - Roaming Scan Channels	Roaming Scan Channels configured when Wi-Fi is connected (A set of channel numbers for scanning)
Config WES Wi-Fi Parameter - Scan Channel Time (ms)	Scan Channel Time configured when Wi-Fi is connected (1 channel scanning time)
Config WES Wi-Fi Parameter - Scan Home Time (ms)	Scan Home Time configured when Wi-Fi is connected (The data Tx and Rx time with serving AP during channel scanning)
Config WES Wi-Fi Parameter - Scan Home Away Time (ms)	Scan Home Away Time configured when Wi-Fi is connected (The period of consecutive channel scanning)
Config WES Wi-Fi Parameter - Scan NProbes	Scan NProbes configured when Wi-Fi is connected (The number of probe requests for a channel scanning)
Config 2G WES Wi-Fi Parameter - Activation	Configure 2.4 GHz WES Wi-Fi Parameter Activation
Config 2G WES Wi-Fi Parameter - Roaming Trigger (dB)	Roaming Trigger configured when 2.4 GHz Wi-Fi is connected
Config 2G WES Wi-Fi Parameter - Roaming Delta (dB)	Roaming Delta configured when 2.4 GHz Wi-Fi is connected
Config 2G WES Wi-Fi Parameter - Roaming Scan Period (ms)	Roaming Scan Period configured when 2.4 GHz Wi-Fi is connected
Config 2G WES Wi-Fi Parameter - Roaming Scan Channels	Roaming Scan Channels configured when 2.4 GHz Wi-Fi is connected
Config 2G WES Wi-Fi Parameter - Scan Channel Time (ms)	Scan Channel Time configured when 2.4 GHz Wi-Fi is connected
Config 2G WES Wi-Fi	Scan Home Time configured when 2.4 GHz Wi-Fi is connected

Item	Description
Parameter - Scan Home Time (ms)	
Config 2G WES Wi-Fi Parameter - Scan Home Away Time (ms)	Scan Home Away Time configured when 2.4 GHz Wi-Fi is connected
Config 2G WES Wi-Fi Parameter - Scan NProbes	Scan NProbes configured when 2.4 GHz Wi-Fi is connected
Config 5G WES Wi-Fi Parameter - Activation	Configure 5 GHz WES Wi-Fi Parameter Activation
Config 5G WES Wi-Fi Parameter - Roaming Trigger (dB)	Roaming Trigger configured when 5 GHz Wi-Fi is connected
Config 5G WES Wi-Fi Parameter - Roaming Delta (dB)	Roaming Delta configured when 5 GHz Wi-Fi is connected
Config 5G WES Wi-Fi Parameter - Roaming Scan Period (ms)	Roaming Scan Period configured when 5 GHz Wi-Fi is connected
Config 5G WES Wi-Fi Parameter - Roaming Scan Channels	Roaming Scan Channels configured when 5 GHz Wi-Fi is connected
Config 5G WES Wi-Fi Parameter - Scan Channel Time (ms)	Scan Channel Time configured when 5 GHz Wi-Fi is connected
Config 5G WES Wi-Fi Parameter - Scan Home Time (ms)	Scan Home Time configured when 5 GHz Wi-Fi is connected
Config 5G WES Wi-Fi Parameter - Scan Home Away Time (ms)	Scan Home Away Time configured when 5 GHz Wi-Fi is connected
Config 5G WES Wi-Fi Parameter - Scan NProbes	Scan NProbes configured when 5 GHz Wi-Fi is connected

Smart WLAN Link Management Service Profile

The parameter of Management Service Configuration for Smart WLAN Link Service can be configured in the **[CONFIGURATION > Wireless Enterprise > Smart WLAN Link Configuration > Wifi Management Service Profile]** menu.

30 Server profile can be configured.

Item	Description
Name	Name of Management Service Profile.
User Permission	Configure User Permission. ADMIN user can configure more service (Network Diagnostics/Keep Trying to Connect/Keep Wi-Fi Enabled/Auto Reset).
Lock Screen	Configure Lock Screen Activation. When user executes WifiAgent, Lock Screen Popup is displayed.
Lock Screen Password	Configure Password for Lock Screen
Network Diagnostic	Configure Network Diagnostic Activation. Check Network Status and notify the result to user.
Network Diagnostic Type	Configure Network Diagnostic Type
Network Diagnostic Target URL	Configure Network Diagnostic Target URL
Keep trying to connect	Configure Keep trying to connect Activation. When Provisioned Wi-Fi Network is disconnected, try to connect periodically until Wi-Fi is connected.
Keep Wi-Fi Enabled	Configure Keep Wi-Fi Enabled Activation. When Wi-Fi is disabled, Enable Wi-Fi automatically.
Auto Reset	Configure Auto Reset Activation. If provisioned Wi-Fi network connection is not available for some period, forced initial login.
Auto Reset Check Count	Auto Reset Check Count
Forbid Other Connection	Configure Forbid Other Connection Activation. Forbid Wi-Fi Connection except provisioned Wi-Fi network.
Wi-Fi Frequency Band	Configure Frequency Band used when Wi-Fi is enabled

Smart WLAN Link User Profile

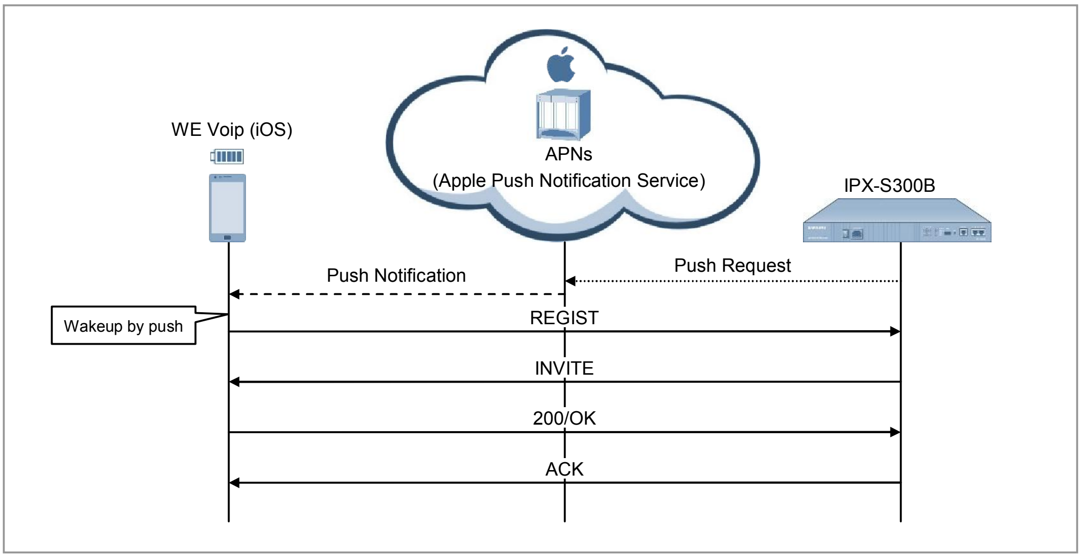
The parameter of User Configuration for Smart WLAN Link Service can be configured in the **[CONFIGURATION > Wireless Enterprise > Smart WLAN Link Configuration > Smart WLAN Link User]** menu.

Subscriber with 'Samsung-Mobile-Phone' as phone type in the **[CONFIGURATION > User > Single Phone User]** menu is displayed automatically in this menu.

Item	Description
User Group	User Group
Extension Number	Extension Number
Mobile Phone Number	Mobile Phone Number
Model Information	Model Information of Mobile Phone
Version Information	Version Information of Mobile Phone
Previous Login Time	Previous Login Time
Previous Logout Time	Previous Logout Time
Management Service Profile - Name	Name of Management Service Profile
Server Profile - Name	Name of Server Profile for user
Network Profile 0 - Name	Name of Network Profile 0 for user
Network Profile 0 - Wi-Fi ID	ID for connection to Wi-Fi Network of Network Profile
Network Profile 0 - Wi-Fi Anonymous ID	Anonymous ID for connection to Wi-Fi Network of Network Profile
Network Profile 0 - Wi-Fi Password	Password for connection to Wi-Fi Network of Network Profile
Network Profile 1 - Name	Name of Network Profile 1 for user
Network Profile 1 - Wi-Fi ID	ID for connection to Wi-Fi Network of Network Profile
Network Profile 1 - Wi-Fi Anonymous ID	Anonymous ID for connection to Wi-Fi Network of Network Profile
Network Profile 1 - Wi-Fi Password	Password for connection to Wi-Fi Network of Network Profile
Network Profile 2 - Name	Name of Network Profile 2 for user
Network Profile 2 - Wi-Fi ID	ID for connection to Wi-Fi Network of Network Profile
Network Profile 2 - Wi-Fi Anonymous ID	Anonymous ID for connection to Wi-Fi Network of Network Profile
Network Profile 2 - Wi-Fi Password	Password for connection to Wi-Fi Network of Network Profile

5.3.12 Mobile Push Service

Mobile Push Service must be configured when to use the WE VoIP (iOS).
To receive a incoming call, WE VoIP must be registered to IPX-S300B but WE VoIP (iOS) can send a register message to IPX-S300B when WE VoIP is wake up.
Push Service is used to request a registration to WE VoIP (iOS).



The Mobile Push Service can be used when using WE VoIP (Android) but it is not necessarily.

DNS Server and Firewall Configuration

To use a Mobile Push Service, DNS Server configuration must be needed in the **[CONFIGURATION > Miscellaneous > DNS Server]** menu.
And TLS link must be connected between APNs Server of Apple and IPX-S300B by using TCP 2195 port. Therefore TCP 2195 port must be opened for IPX-S300B in the firewall.
And HTTPs link must be connected between GCM Server of Google and IPX-S300B by using TCP 443 port. Therefore TCP 443 port must be opened for IPX-S300B in the firewall.

Mobile Push Service Activation

Mobile Push Service can be activated in the **[CONFIGURATION > Wireless Enterprise > Mobile Push Service]** menu.
If status is changed to CONNECTED after service activate, Mobile Push Service can be used. If status is displayed to DISCONNECTED, please check a firewall and DNS server.

Push Service for User

If Mobile Push Service is activated for System, specify push notification per device.
Specify the Push Notification about WE VoIP in the **[CONFIGURATION > User > Single Phone User/Multi-Extension Phone]** menu

CHAPTER 6. Application Features

This part describes the applications provided by IPX-S300B and how to configure them. IPX-S300B includes a basic conference system and a basic ACD server. IPX-S300B also includes an advanced conference system and a voice mail system.

6.1 Application Server Service Group

If IPX-S300B version is 3.2.4 or higher, an 'Application Server Service Group' should be configured to integrate with application server.

6.1.1 Configuring Application Server Service Group

First of all, operates should make an Application Server Service Group using **[SCM Administrator/CONFIGURATION/Application/Application Server Group]** menu.

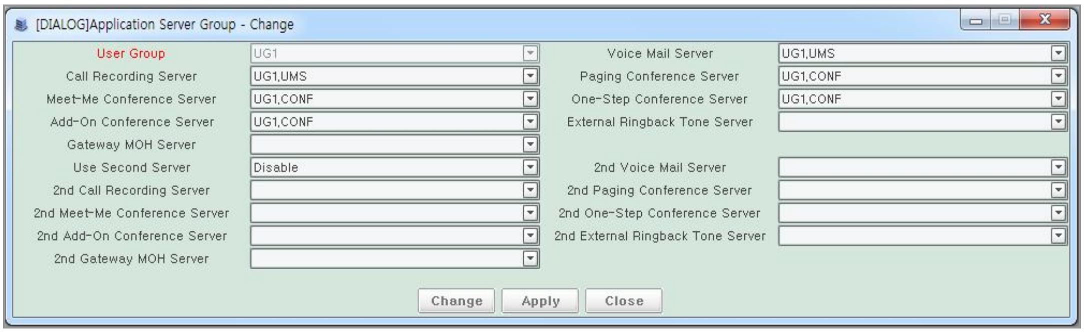


Figure 16. Application Server Service Group

6.1.2 Assigning Application Server Service Group

In order to provide a specific service using an application server, operators can assign an Application Server Service Group in user or service group or user group menu.
The application server specified in user menu has a higher priority than the application server of service group or user group. And the application server of service group has a higher priority than that of user group.
For example, if a user should be connected to application server for a specific service, IPX-S300B first tries to fine an application server in user menu. If there is no available application server, IPX-S300B checks an Application Server Service Group of service group. After that, if IPX-S300B fail to find an application server in service group, the Application Server Service Group of user group is checked. If it is also failed, the call will fail.

In User Menu

In **[SCM Administrator/CONFIGURATION/User/Single User]** menu, operator can assign Application Server Service Group.

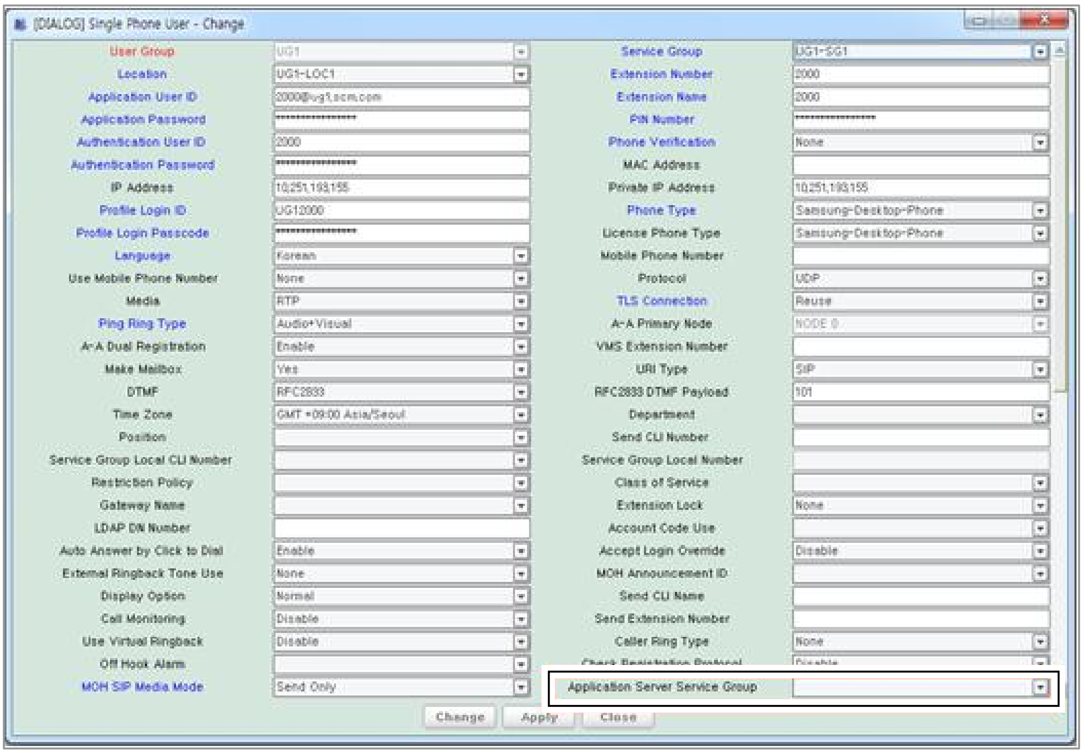


Figure 17. Assigning Application Server Service Group-User

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

In Service Group Menu

In **[SCM Administrator/CONFIGURATION/User Group/Service Group]** menu, operator can assign Application Server Service Group.

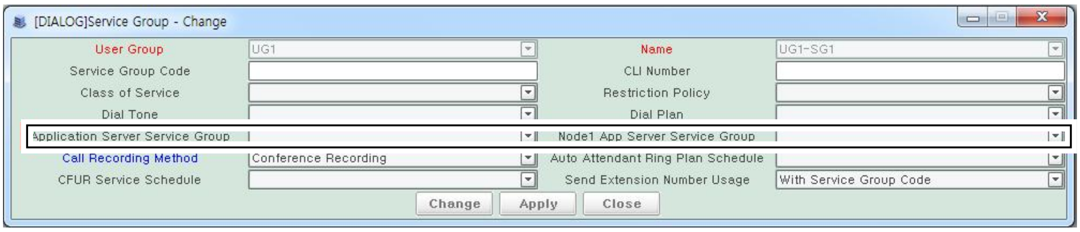


Figure 18. Assigning Application Server Service Group-Service Group

In User Group Menu

In **[SCM Administrator/CONFIGURATION/User Group/Change User Group/Options]** menu, operator can assign Application Server Service Group.

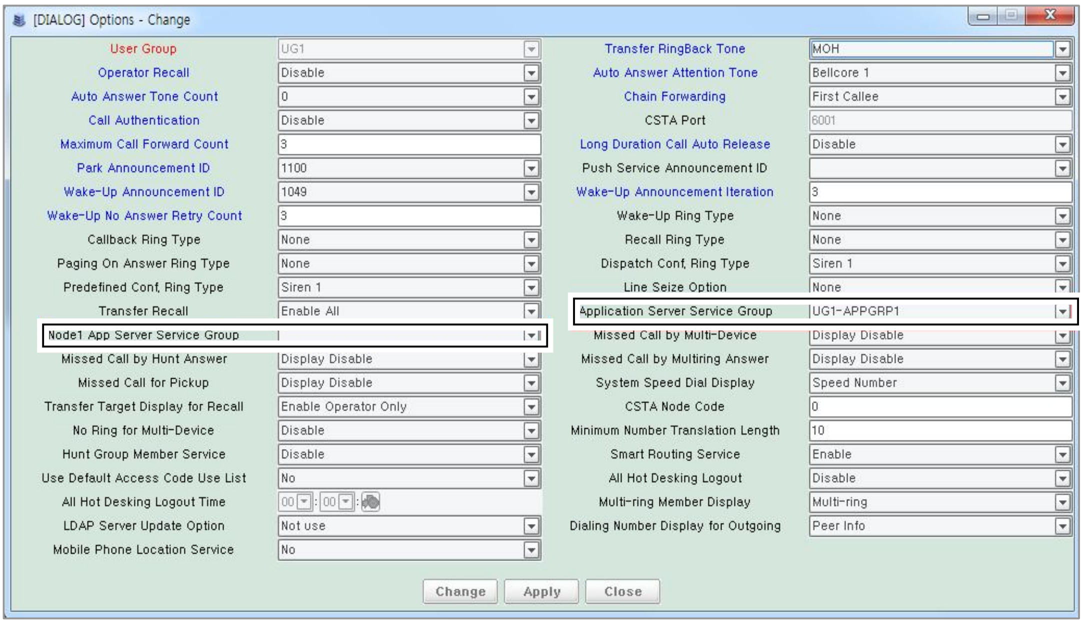


Figure 19. Application Server Service Group-User Group

6.2 Automatic Call Distribution (ACD)

The Automatic Call Distribution (ACD) service is useful when there are more incoming calls than the people available to answer them. If the ACD feature is enabled, callers do not need to hear the busy tone for a long time or get delayed in getting their calls answered. When a call is connected while the ACD group is busy, the call is put in waiting status until an agent becomes available, and a waiting announcement is played for the caller so that the caller can wait until an agent answers the call.

To use the ACD feature, three sets of information must be configured. Only when these items are configured, ACD calls can be supported with waiting when busy, waiting announcement, agent allocation, agent distribution, etc.

6.2.1 Creating ACD Agents

Agent IDs are specified for each user group regardless of the system extension number. An agent can use any phone to login with his/her agent ID and password to have the ACD group's incoming calls distributed to the phone used for login.

ACD agents can be created in the **[CONFIGURATION > Application > ACD > ACD Agent]** menu.

Item	Description
User Group	Select a user group for which the ACD agent will be created.
Agent ID	Enter the user ID used by the ACD agent to log into the ACD group. This is the key data identifying each agent and must not be entered in duplicate.
Password	Enter the password used by the ACD agent to log into the ACD group.
Agent Name	You can enter the name of the ACD agent. This is used for easy identification of the agent and can be entered in duplicate.

6.2.2 Configuring ACD Group

The way of distributing ACD calls and the configuration for ACD call processing are described as follow.

ACD groups can be configured in the **[CONFIGURATION > Application > ACD > ACD Group]** menu.
The following sets of information can be configured for ACD groups.

ACD Group Basic Information

The basic ACD group settings such as the list of agents logging into the ACD group and the method for distributing incoming calls to agents must be set.

Item	Description
User Group	Select a user group for which the ACD group will be created.
Name	Enter a name for the ACD group. This is used for easy identification of the ACD group and can be entered in duplicate.
Group Number	Enter a number to call the ACD group. This is the key data identifying each ACD group and must not be entered in duplicate with any extension number or extension group number.
Agent Select Method	When there are two or more available ACD agents, select the method for selecting an agent for call distribution. - Longest Idle Agent: When distributing an ACD group call to an agent, the call is distributed to the agent with the longest idle time. - Least Occupied Agent: When distributing an ACD group call to an agent, the call is distributed to the agent with the fewest calls since login. - Sequence Mode: Calls are distributed to the ACD group's agents in a sequence.
Group Members	Register the IDs of the agents allowed to log into the ACD group. You can select from the agent IDs created in the **[CONFIGURATION > Application > ACD > ACD Agent]** menu.

ACD Overflow

When there is an incoming ACD group call, the call will be connected to an agent if an agent is available. But if no agent is available, the call will standby for the ACD group and the caller will continue to hear the greeting and the MOH one after another. If no agent becomes available after a long time of waiting, the call can be forwarded to another ACD group, etc.
You can set the following information for forwarding calls to another number, etc.
When the calls waiting for the ACD group cannot be connected because no agent is available for a long time.

Item	Description
Overflow Time (sec)	When there is an incoming call for the ACD group, the settings must be made to forward the call to another ACD group, etc. if no agent is available to answer the call or an agent connected does not answer for a long time.
Overflow Destination	You can specify the phone number to which incoming calls for the ACD group will be forwarded if not answered by any agent during 'Next Destination Overflow Time (sec)'. When 'Next Destination Overflow Time (sec)' has exceeded and the call has to be forwarded to another number, but if the number set for this item is unavailable for receiving calls, the call is terminated. If no phone number is set for this item, the call is forwarded to Operator Group.
All Busy Destination	When there is an incoming ACD group call and all agents are busy, the call can be forwarded to another ACD group, etc. without waiting. If a phone number is set for this item, the call will be forwarded immediately. Take note that if the phone number is unavailable for receiving calls, the call will be terminated.
All Logout Destination	When there is an incoming ACD group call, if no agent is logged in, the call can be forwarded to another ACD group, etc. If a phone number is set for this item, the call will be forwarded immediately. If the phone number set is unavailable for receiving calls, the call will be terminated.
Agent No Answer Time (sec)	If a call connected to an agent is not answered by the agent for a specified period of time, the call can be connected the next agent. If the agent does not answer after 'Agent No Answer Time (sec)', the connection is canceled and the call is connected to the next agent. This setting is necessary as situations arise whereby agents are unable to answer calls. If no other agent is available after this period of time, the phone will keep ringing.

ACD Miscellaneous Information

The following additional information can be configured for processing ACD group calls.

Item	Description
Maximum Queuing Count	Greetings and MOH are played for callers waiting for ACD groups. Since IPX-S300B's built-in announcement system has limited number of channels, it is necessary to limit the number of calls waiting for ACD. You must use this item to specify the maximum number of calls allowed to wait for ACD.
Maximum Overflow Count	When forwarding an incoming ACD group call to another phone number, if the phone number is an ACD group number, you must specify the maximum number of calls allowed to be forwarded.
Queuing Wait Time (sec)	When there is an incoming ACD group call, the caller may find it strange if the greeting is played immediately. Therefore, you must specify a time

Item	Description
	period for which the caller will hear the ring-back tone before listening to the first greeting.
Minimum Greet Listen Time (sec)	When an agent becomes available while the greeting is being played, the caller may find it strange if the greeting suddenly stops and the call is connected to an agent. Therefore, you must specify a minimum time period for which the greeting will be played before connecting the call to an agent.
No Answer Wrap-Up Time (sec)	When an incoming ACD group call is connected to an agent but the agent did not answer the call and the call is connected to the next agent, the agent who became idle because he/she failed to answer the call is likely to be unable to answer the next connected call as well. Therefore, you must specify a time period during which the incoming calls for the ACD group will not be connected to the agent. Incoming ACD group calls will not be distributed to the agents who became idle because they did not answer ACD calls during this period of time.
Normal Wrap-Up Time (sec)	When an agent answers an ACD group call and ends the call, the agent needs some time to wrap up. You must specify the time period during which ACD group calls will not be connected to the agent after an ACD call.
Supervisor Number	You can specify the phone number of an ACD group supervisor, whom the agents can consult for urgent matters. This information can be downloaded for the agent program so that the supervisor can be connected with a single click.
Queuing Level-up Interval (sec)	'ACD Queuing Level' is incremented by one per **[Queuing Level-up Interval (sec). ACD Queuing Level 0-9 is available. The higher the 'ACD Queuing Level]**'is, the shorter the queue waiting time is. At the time of the first incoming, 'ACD Queuing Level' can be set in the **[CONFIGURATION > Trunk Routing > CLI Routing]** and **[CONFIGURATION > Trunk Routing > DID Routing]**.
Call Distribution Mode	Select a call distribution mode. - Auto: Determined the call distribution mode by CSTA application's condition. - SCM: Call is distribution by IPX-S300B. - CSTA: Call is distribution by CSTA.
Auto Attendant Plan Schedule	Select a Ring Plan Schedule to use when call is transferred to AA.

6.2.3 Configuring ACD Announcement

The way of connecting Greeting message for incoming ACD call can be configured.
It sets in the **[CONFIGURATION > Application > ACD > ACD Greeting Message]**.

ACD Announcement

When there is an incoming call for the ACD group, the caller is allowed to listen to the ring-back tone for a certain period of time and then the call is connected to IPX-S300B's announcement system for an announcement. If an agent is available at this point, the call is connected to the agent. If no agent is available, the call will continue to standby for the ACD group as the announcement and the MOH are played one after another. If an agent becomes available during the announcement or the MOH, the announcement or the MOH will stop and the call will be connected to the available agent. The caller will hear the ring-back tone or the MOH until the call is answered by the agent.
The following information can be configured for the announcement played for callers waiting for the ACD group.

Item	Description
First Greet Message (Available)	You can specify the ID of a first greeting to play when there is an incoming call for the ACD group and an agent is available to answer the call. The call will be connected to an agent when the first greeting finishes playing. If no first greeting is set, the call will be connected to an agent without any announcement.
First Greet Message (All Busy)	You can specify the ID of a first greeting to play when there is an incoming call for the ACD group and no agent is available to answer the call. When the first greeting finishes playing, the MOH and the greeting will be played one after another until an agent becomes available.
First MOH ID	You can specify the ID of the MOH to play after the first greeting and before the second greeting.
First Greeting Repeat	You can specify the number times to repeat the first greeting.
First MOH Duration (sec)	You can specify the period of time to play the first MOH.
Second Greeting Message 1~7	After the first greeting and the MOH are played, the second greeting and the second MOH will be played after one another until an agent become available to answer the call. You can specify the ID of a second greeting here. Second MOH is played in order (1~7).
Second MOH ID 1~7	After the first greeting and the MOH are played, the second greeting and the second MOH will be played after one another until an agent become available to answer the call. You can specify the ID of a second MOH here. Second MOH is played in order (1~7).
Second Greeting Repeat	You can specify the number times to repeat each the second

Item	Description
1~7	greeting.
Second MOH Duration 1~7 (sec)	You can specify the period of time to play each the second MOH.
Last Repeated Greeting Message	After all second greeting message (1-7) and all second moh (1-7) finish, if caller is still waiting for agent, IPX-S300B supports Last Repeated Greeting Message. You can specify the ID of a Last Repeated Greeting Message.
Last Repeated MOH ID	After all second greeting messages (1-7) and all second MOH (1-7) finish, if caller is still waiting for agent, IPX-S300B supports Last Repeated MOH. You can specify the ID of a Last Repeated MOH.
Last Repeated Greeting Repeat	You can specify the number times to repeat the last repeated greeting.
Last Repeated MOH Duration (sec)	You can specify the period of time to play the last repeated MOH.
Release Message	If a call is disconnected without any connection to agent, IPX-S300B support a specific Release Message. You can specify the ID of a Release Message.
Release Message Repeat	You can specify the number times to repeat the release message.

6.2.4 ACD Agent Status

This section describes the agent status, which is used for distributing incoming calls for ACD groups.
An agent's status can be in one of the following three:

- Logged In: The agent is available to take calls as a member of an ACD group.
- Wrap-Up: The agent is wrapping up after ending an ACD group call.
- Break: The agent is taking a break.

An agent can log in/out or register or cancel the wrap-up or break status by dialing the feature codes.
The feature codes for registering and canceling the agent status can be created in the **[CONFIGURATION > Service > Feature Service > Feature Code]** menu.

- Logged In: The 'ACD Agent Login-Login' and 'ACD Agent Login-Logout' feature codes must be configured.
- Wrap-Up: The 'ACD Agent Wrap-Up-Set' and 'ACD Agent Wrap-Up-Cancel' feature codes must be configured.
- Break: The 'ACD Agent Break-Set' and CD Agent Break-Cancel' feature codes must be configured.

Agents' login status can be viewed for each ACD group in the **[CONFIGURATION > Application > ACD > ACD Group Status]** menu.
Agents' wrap-up and break status can be viewed for each ACD agent in the **[CONFIGURATION > Application > ACD > ACD Agent]** menu.

6.2.4.1 Log In/Out

For an agent to be able to answer ACD group calls, the agent must log into an ACD group.

- The agent can log in by dialing '**[ACD Agent Login-Login]** feature code + agent ID + agent password + ACD group number' on the phone.
- If an ACD group number is entered when logging in, the agent will be logged into the selected group only. If no ACD group number is entered, the agent will be logged into all the groups the agent belongs to.

For an agent to be unavailable to answer ACD group calls, the agent must log out of an ACD group.

- The agent can log out by dialing '**[ACD Agent Login-Logout]** feature code + agent password + ACD group number' on the phone.
- If an ACD group number is entered when logging out, the agent will be logged out of the selected group only. If no ACD group number is entered, the agent will be logged out of all the groups the agent belongs to.

6.2.4.2 Wrap-up

When an agent does not answer an ACD group call and the call ends, or when the agent answers a call and the call ends, the agent automatically enters the wrap-up status. Although ACD group calls are not connected when an agent is in the wrap-up status, the status indicates that the agent is busy working.
If an agent wishes to extend the time for the wrap-up status, the agent can register the wrap-up status by dialing the 'ACD Agent Wrap-Up-Set' feature code.
If an agent wishes to end the wrap-up status and become available to answer ACD group calls, the agent can cancel the wrap-up statuses by dialing the 'ACD Agent Wrap-Up-Cancel' feature code.

6.2.4.3 Break

An agent can register the break status instead of logging out so that no ACD group calls are connected to the agent. The break status indicates that the agent is not busy working.
If an agent wishes to take a break, the agent can register the break status by dialing the 'ACD Agent Break-Set' feature code.
If an agent wishes to end the break status and become available to answer ACD group calls, the agent can cancel the break status by dialing the 'ACD Agent Break-Reset' feature code.

6.2.5 ACD Statistics

IPX-S300B provides statistics on the incoming calls processed by ACD groups and ACD calls processed by agents.
For more information on ACD statistics, see the '6.6 Statistics Reports' section.

6.2.6 ACD Agent Program

While it is possible for ACD agents to handle ACD calls just using phones, they can handle ACD calls more efficiently by using a dedicated agent program.
A separate license is required for using the dedicated ACD agent program. The program supports features such as real-time monitoring of calls waiting for the ACD group.
For more information on the ACD agent program, see the '5.4 External Applications' section.

6.2.7 ACD Wallboard ID

This menu provides ACD Wallboard ID configuration. Select CONFIGURATION > Application > ACD > ACD Wallboard ID

Item	Description
User Group	Specify a user group which the ACD Wallboard ID will be created.
ID	Enter the User ID used by the ACD Wallboard ID to log in
Password	Enter the password used by the ACD Wallboard to log in

6.3 Conference

A conference system built-up in IPX-S300B services six default voice conference channels, and additional 32 channels will be available with the installation of a VPU card.

6.3.1 Conference Features

IPX-S300B provides the types of conferences as follows. The functions other than Add-On conference can be provided only by a VPU card.

6.3.1.1 Basic Conference

Add-On Conference

During a call (including a conference call), the call can be put on hold and a new call is made to another attendee. If the new attendee answers the call, the conference button can be pressed to include the new attendee in the conference.

Conference On Answer (COA)

Similar to the Add-On method, a call is made to an attendee and when the called party answers the call, the called party is automatically included in the conference.

6.3.1.2 Meet-Me Conference

Meet-Me Conference

A conference room is reserved, and then the conference attendees call the conference system to enter the conference room.
When the conference moderator uses the Conference Reservation menu on PWP to set the conference room number, password, etc. and register the conference attendees, IPX-S300B sends an invitation mail to the conference attendees. The conference attendees can call the conference ID at the conference time as instructed by the invitation mail to enter the conference room.

6.3.1.3 One-Step Conference

Predefined Conference

A list of conference attendees are registered in advance and the attendees are paged using the conference group number. Those attendees answering the call are automatically included in the conference.

Intercom Conference

Initiate a conference by dialing feature code + station group number. The phones registered in the station group answer automatically to join the conference.

Dispatch Conference

You can specify the master and members during IDLE/call/conference and request a dispatch conference through CSTA (Computer Supported Telephony Application).
When the chairman is connected, the server automatically make calls to the members.
When the members respond, the conference will be configured.
The master can transfer the conference call.

Phonebook Conference

Similar to the dispatch conference, a user can initiate a conference using his/her own conference group from a phone. The dispatch conference should be specified prior to perform a phonebook conference, because it is handled as a kind of dispatch conference internally.

6.3.1.4 Paging Features

Station Paging

When extension numbers are registered to a paging group number in advance, the entire paging group can be paged. The call is automatically answered by the phones paged so that the subscribers can listen to the moderator’s announcement.

Paging On Answer

When the telephone numbers are registered to a paging on answer group number in advance, the entire group can be paged. When the called party answers, he can listen to the moderator’s announcement.

6.3.1.5 Conference Member Eject

Conference Member Eject

A member joined in a conference can be deleted by another member. The feature permission is allowed to conference owner or a user which setup the conference.

6.3.2 Call Processing System Configuration

IPX-S300B's built-in conference system cannot run independently of call processing. Therefore, to use the conference system, the necessary settings must be made for the call processing system.

6.3.2.1 License Key Registration

A separate embedded application license key is required to use the meet me conference feature, which allows attendees to join a conference by dialing a conference room number reserved in advance.
The license key can be registered in the **[CONFIGURATION > Resource > Activation Key]** menu.
After entering the activation key, please check the Meet-Me Channels value in the Misc Settings tab of **[CONFERECE > System Configuration]** menu. Out of the total 32 voice conference channels, this number of channels are available for meet me conferences.

6.3.2.2 Conference Server Registration

The connection information must be configured for the call processing system to connect to the conference system.
Conference system connection information can be configured in the **[CONFIGURATION > Application > Conference Server]** menu. To use IPX-S300B's built-in conference system, 'Application Type' must be set to Internal Conference.

Item	Description
User Group	Select a user group which will use the conference system.
Application Type	Specify a conference system type. - Internal Conference: IPX-S300B's built-in conference system. - External Conference: An independent conference system in an external server. - 3rd party Conference: A third-party conference system in an external server. Service provided may vary by conference systems.
Name	Specify a conference system name.
Access Number	Enter a phone number used for calling the conference system.
Location	Specify a location where the conference system is used.
Start Channel Tel Number Last Channel Tel Number	Extension numbers are required to identify one conference from another. For this, a continuous range of extension numbers is applied by the start channel and the end channel.
Keep Alive Retry Interval (sec)	By default, IPX-S300B sends the OPTIONS message to the application sever every 30 seconds to check the registration status. If there is no response to the OPTIONS message, IPX-S300B retries for the keep alive retry maximum time at the keep alive retry interval.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
	If there is still no response, the application server registration is canceled.
Retry Pause Time (sec)	If the conference system registration is canceled, IPX-S300B waits for the retry pause time before it retries sending the OPTIONS message.
Service List	Specify the type of conference services on this conference system. - Paging: Station Paging, Paging On Answer - Meet-Me Conference: Meet-Me Conference - One-Step Conference: Predefined, Progressive, Intercom, Dispatch Conferences - Add-On Conference: Add-On, COA Conferences

6.3.2.3 Service Class Settings

To use the conference features, the conference-related items must be enabled in the service class. For more information on service classes, see the '4.1.36 Feature Service'.
Service classes can be configured in the **[CONFIGURATION > Service > Feature Service > Class of Service]** menu. The following conference-related items must be enabled.

- Add-On Conference: If the service permission is set, the Add-On conference and COA, features are available.
- One-Step Conference: If the service permission is set, Predefined conference, Intercom conference, and Dispatch conference features are available.
- Station Paging: If the service permission is set, the station paging feature is available.
- Paging On Answer: If the service permission is set, the paging on answer feature is available.

6.3.2.4 Feature Code Settings

When a user dials a conference feature code, IPX-S300B connects the call to the conference system. The conference system uses the feature code to determine which type of conference should be serviced.
The feature codes can be configured in the **[CONFIGURATION > Service > Feature Code]** menu. The following conference-related feature codes must be configured.

- Conference: This feature code is used for starting an Add-On conference.
- Conference On Answer: This feature code is used for adding a conference participants by COA.
- Predefined Conference: This feature code is used for starting a predefined conference.
- Intercom Conference: This feature code is used for starting an intercom conference
- Dispatch Conference: This feature code is used for starting a dispatch conference from equipments through CTSA interface.

- Meet Me Conference Join: This feature code is used for joining a meet-me conference.
- Station Paging: This feature code is used for starting a station paging.
- Paging On Answer: This feature code is used for starting a paging on answer.

6.3.2.5 Extension Paging On Answer Configuration

When two IPX-S300Bs are connected through SIP trunk, subscribers to other IPX-S300Bs can be configured as paging on answer members. The following service will be applied equally both to the members directly connected to the IPX-S300B applied with the paging on answer service, and to the members to other IPX-S300Bs connected through SIP trunk.

- Automatic response when receiving a page on service call.
- Distinguishing ring type when receiving a page on service call.

Specify page on service members in **[CONFIGURATION > Service > Group Service > Page On Service Group]** menu. 'Automatic Response' among the menu is set to Enable, IPX-S300B trunk members as well as extension members will be configured with Automatic Response function. Also, IPX-S300B trunk members as well as extension members will be configured with the distinguishing ring type in **[CONFIGURATION > User Group > User Group Change > Detailed Option]** menu.
However, No Ring Ignore function will always be ignored irrespective of the settings of IPX-S300Bs trying the Page On Answer function in **[CONFIGURATION > Service > Group Service > Page On Service Group]** menu. To this end, 'Trunk Page On Answer Information Relaying' for IPX-S300B trunk should be enabled in **[CONFIGURAITON > Trunk Configuration > Trunk Route]** menu.

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6.3.3 Conference System Configuration

The conference system information can be configured by clicking the **[CONFERENCE]** icon in SCM Administrator.

6.3.3.1 System Configuration

Using system configuration, you can configure the essential settings for running the conference system.

Mixer Setting Tab

You can set the audio codec information used by the conference system.

Item	Description
Preferred Audio Codec	Specify priority for audio codec negotiation.
Outbound Media Options	Determine to provide dual media-both RTP and SRTP-when offering SRTP

Prompt/DTMF Setting Tab

You can set the information related to the prompt used by the conference system.

Item	Description
TIMEOUT	You can specify the digit input waiting time when the user is instructed to enter a digit during an interactive voice announcement such as for a progressive conference.
Error Prompt Repeat Count	You can specify the retry count for digit input error when the user is instructed to enter a digit during an interactive voice announcement such as for a progressive conference.
Max DTMF Input Length	You can specify the maximum digit length when the user is instructed to enter a digit during an interactive voice announcement such as for a progressive conference.
Default Language	You can view the language used for voice announcement. The language used by the conference system for voice announcement follows the language configured for announcement by the IPX-S300B.

Meet-Me Dial Number

You can save the number which will be set as telephone number to join the meet-me conference to insert into the invitation e-mail.
This number should be exist in the DID table to be translated into the Meet-Me Join feature code.

Misc Setting Tab

You can set the information by the conference system when reserving conferences, etc.

Item	Description
Meet-Me Channels	You can view the maximum number of channels allocated for meet me conferences. This is determined by the license key entered.
Alert attendee's in/out with sound	You can specify whether an alarm will be heard when joining or exiting conferences.
Overbooking Rate	You can specify whether overbooking will be allowed when reserving conferences. Overbooking allows reserving a conference for more attendees than the number of available channels, considering those attendees who may not be able to join the conference.
Allow Early Entrance	You can specify whether to allow attendees to enter the conference room even before the conference start time for a reserved conference. If you allow early entrance, you can also specify how many minutes before the start time the attendees will be allowed in.
Sole Participant Audio Type	You can specify the type of music played when an attendee is left alone in the conference room for a reserved conference.
Gain Controller Threshold (%)	You can specify the control range of audio decibel for conferences.
Jitter Buffer (ms)	You can specify the size of jitter buffer of the internal conference mixer.
Paging Setup Time (sec)	You can specify the waiting time for the callee's answer. If all the extensions answer prior to this timeout, the paging starts immediately.

6.3.4 Conference Management

You can configure the settings related to meet-me conferences and predefined conferences. You can also monitor the currently running conferences.

Meet Me Reservation

You can view reservation status of meet me conference by hours. Hours with reserved conference(s) are highlighted in different colors. Depending on the ratio of channels reserved, if less than 50 % are reserved, it is highlighted in green > yellow and if 50 % or more are reserved, it is highlighted in orange > red.

If you have the IPX-S300B in master-slave configuration, choose the node from the node combo box before you proceed.
When you over the mouse over the timetable, the number of remaining channels is displayed
You can reserve a conference and send the invitation mails by dragging a time period with available channels and clicking the **[Create]** button.

Item	Description
Date	This is the date for which the conference will be reserved.
Title	Enter a title for the conference. This is shown when querying reservation information and is saved in the conference log.
Subject	Enter a subject for the conference. This is notified to the attendees when the invitation mail is sent.
Conference ID	This is the conference ID used for joining the conference. Use a three-digit number from 100 to 999. Press the **[Check]** button to check whether the number is available.
Duration	This is the conference time.
Number of Attendees	Specify the number of attendees for the conference. You must enter within the range of the maximum number of invites shown on the right. This number determines how many channels will be reserved, and there may be no more channels left for others to use. Therefore, only enter the value you will actually use.
Owner	When the conference is running, only the conference owner can view the conference status on PWP.
Attendee List	Select conference attendees by clicking the **[Select]** button. If selecting an extension subscriber, select the attendee from the list. If selecting an external attendee, enter the name and the email address. The email address is used for sending the invitation mail.
Send Invitation letters	Check this if you wish to send the invitation mail to the email addresses of the attendees. The email field entered when reserving meet me conferences is used for sending the invitation mail. If you are not sending the invitation

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Item	Description
	mail, you do not have to enter the email addresses. If you are not sending the invitation mail, the conference owner must give separate instruction to the attendees on how to join the conference.
Password	This is the password used for joining the conference.
Recurrence	You can specify whether to repeat the conference. When changing the recurrence option, you must specify the recurrence period in the date item again.
Early Entrance	Early entrance before the conference starts time.
Stay Locked	Enable lock so that people cannot join the conference. This is useful when the conference owner does not want other attendees to join the conference. For example, if the conference owner wants to be the first person in the conference, the owner can unlock and start the conference when he/she is available to join the conference.

After entering all the information, click the **[Create]** button to create a meet me conference.
An email account is required for sending invitation mails. If a user ID and a password is set for auth login in the **[PERFORMANCE > Fault > E-mail Notification Setup]** menu in SCM Administrator, this account will be used. If the setting is not found or invalid, a window is displayed for entering the email information.
To allow early entrance and conference channel management, a meet me conference must be reserved at least an hour prior to the conference start time.
To view detailed reservation information, select an hour period for which the conference is reserved and click the **[Details]** button. You can change or cancel the reservation on the Details screen.

6.3.4.1 Meet-Me Status

You can view, edit or cancel the list of currently reserved meet-me conferences in a table.

6.3.4.2 Pre-defined

This menu will be displayed only when the Predefined Conference feature code has been created.
You can view, create, delete, or edit conference groups for predefined conferences.
If you select a conference group ID from the list on the left, a list of the conference owner and attendees is shown in the window on the right.
To create a new predefined conference group, click the **[Create]** button to be allocated with a conference group ID and register the phone numbers of the attendees to call for conference.

Item	Description
Type	Pre-defined
Group ID	This is the ID for the newly created conference group. A four-digit number from 1000 to 9999 can be used. Click the **[Check]** button to check if the ID is available.
Owner	This is the owner of the conference group.
Name	Enter a name for the conference group. This is used for identifying the conference group when viewing the information.
Select attendees from subscribers	If selecting extension subscribers, you can search for them by their phone numbers or names and enter them as attendees. Select attendees from the search results and click the **[Add]** button to add them to the attendees list.
Participant List	You can add attendees by searching for attendees or manually entering their names and phone numbers and then clicking the **[Add]** button. To remove some of the attendees, select them on the list and click the **[Remove]** selected button. To remove all attendees, click the **[Remove All]** button.

Select the conference group ID from the list on the left and click the **[Edit]** or **[Delete]** button to edit or delete the conference group.

6.3.4.3 Current Conference Status

The administrator can monitor the status of currently running conferences real-time. Here you can end a conference also.

6.3.5 Using Conference Features

The user can use the conference features in the following ways.

Add-On Conference

You can put the current call on hold and dial 'Add-On conference feature code + phone number' on your phone to call an attendee. When the attendee answers the call, you can dial the conference feature code to start an conference.

COA

You can put the current call on hold and dial 'conference on answer feature code + phone number' on your phone to call an attendee. When the attendee answers the call, a conference will start automatically.

Predefined Conference

You can dial 'predefined conference feature code + conference group ID' on your phone to connect to the conference system. The conference system will then call all the members registered in the conference group and the members answering the call are included in the conference. Members must be defined for the conference group.

Intercom Conference

You can dial 'Intercom conference feature code + hunt group telephone number' on your phone to connect to the conference system. The conference system will then call all the members registered in the hunt group and the call will be automatically answered by the members for conference. Members must be defined for the hunt group

Dispatch Conference

If you specify the master and members and request a dispatch conference through CSTA, the master is connected to the server, which automatically make calls to the members, and when they respond to the calls, the conference will be configured. The master can be connected during IDLE.

Station Paging

You can dial 'paging feature code + paging group telephone number' on your phone to connect to the conference system. The conference system will then call all the members registered in the paging group and the call will be automatically answered by the members for station paging. Members must be defined for the paging group.

Paging On Answer

You can dial 'paging on answer feature code + paging on answer group telephone number' on your phone to connect to the conference system. The conference system will then call all the members registered in the paging on answer group and the call will be automatically answered by the members for paging on answer. Members must be defined for the paging on answer group.

Conference Member Eject

A member joined in a conference can be deleted by another member. The feature permission is allowed to conference owner or a user which setup the conference. To delete a user of the conference, hold the conference call and dial with 'Conference Member Eject Feature Code + User Number'. The user which requests this service can know whether the service is failed or not with announcement.

6.3.6 Conference History Management

History Management

The administrator can use the **[CONFERENCE > History Management]** menu to view the conference system history.

- Select conference history to view the list of conferences processed by the conference system by dates, etc.
- Select system history to view history related to conference system operation and management including conference system start and stop. You can delete unnecessary history entries.

6.4 Voice Mail

The voice mail system is an application which interoperates with the call processing system to play voice announcements and perform various services such as forwarding calls to subscriber numbers (auto attendant) and allowing the caller to leave a voice mail when the subscriber is absent. Voice Mail administration GUI has Basic/Advanced mode.
Basic mode provides VM/AA feature partially. It does not provide Access Manager, System Report feature.

6.4.1 User Service Code Table

Code 1	Code 2
[11] Group New Messages [33] Group Saved Messages	[1] Group Urgent Messages
	[2] Group Callback Requests
	[3] Group Reminders
	[4] Group Private Messages
	[6] Group Voice Only Messages
	[8] Pause, Resume Menu Prompting
	[9] Group a Specific Sender
	[#] Play Message Inventory
	[0] Play Menu Options
	[*] Cancel, Return to Subscriber Menu
[1] Listen to New Messages [3] Review Saved Messages	[1] Play Message
	[11] Play Previous Message
	[2] Save Message
	[3] Discard Message
	[4] Reply to Sender
	[5] Place Call to Sender
	[6] Forward a Copy of Message
	[7] Rewind 5 Seconds
	[8] Pause, Resume Message Playback
	[9] Fast Forward 5 Seconds
	[#] Skip to Next Message
	[##] Scan Messages
	[0] Play Menu Options
	[00] Play Message Information
	[*] Cancel, Return to Subscriber Menu
[2] Record & Send Messages	[#] For a Directory of Subscribers
	[##] To Create a Reminder

Code 1	Code 2
	Enter the Recipient's Number
	[1] Review Recording
	[2] Stop, Append to Recording
	[3] Discard Recording and Re-record
	[4] Set Delivery Options
	[5] Specify Future Delivery
	[6] Send Message, Then Copy
	[7] Rewind 5 Seconds
	[8] Pause, Resume Record/Playback
	[9] Fast Forward 5 Seconds
	[#] Send Message, Then Exit Record
	[0] Play Menu Options
	[*] Cancel, Return to Subscriber Menu
[4] Access Manager	[1] Follow Me
	[3] Block All Calls
	[4] Call Forwarding
	[5] Call Screening
	[6] Find Me
	[7] Auto Set Night Intercept
	[8] Pause, Resume Menu Prompting
	[#] Play Access Coverage
	[0] Play Menu Options
	[*] Cancel, Return to Subscriber Menu
[5] Personal Greetings	[1] Assign/Edit Primary Greeting
	[2] Assign/Edit Busy Greeting
	[3] Assign/Edit Blocking Greeting
	[4] Assign/Edit Night Greeting
	[5] Assign/Edit Call Screen Greeting
	[6] Edit Only Personal Greetings
	[7] Edit Only Mailbox Greetings
	[8] Pause, Resume Menu Prompting
	[0] Play Menu Options
	[*] Cancel, Return to Subscriber Menu
[6] Mailbox Administration	[1] Change Message Alert
	[2] Change Pager Notification
	[3] Review Deleted Messages
	[4] Review Undelivered Messages

Code 1	Code 2
	[5] Auto Play New Messages
	[6] Auto Play Message Information
	[8] Pause, Resume Menu Prompting
	[9] Record & Send Broadcast Message
	[0] Play Menu Options
	[*] Cancel, Return to Subscriber Menu

Code 1	Code 2	Code 3
[8] Pause, Resume Subscriber Menu	-	-
[0] Play Menu Options	-	-
[#] Personal Services	[1] Review Workload	-
	[2] Edit Stored Telephone Numbers	-
	[3] Change Weekly Schedule	-
	[5] Place a Direct Call	-
	[7] Personal Administration	[1] Change Password
		[2] Record Name
		[3] Enter Directory Name
		[4] Extended Prompting
		[8] Pause, Resume Menu Prompting
		[0] Play Menu Options
		[*] Cancel, Return to Subscriber Menu
	[8] Pause, Resume Menu Prompting	-
	[#] Record a Reminder	-
	[0] Play Menu Options	-
	[*] Cancel, Return to Subscriber Menu	-
[*] Exit Subscriber Menu	-	-

Administrator Mode

Code 1	Code 2	Code 3
[1] To Edit Prompt	[2] To Record	[1] To Review
		[2] To Continue & Record
		[3] To Discard & Re-record
		[#] To Save
	[#] To Satisfy	-
	[*] To Cancel	-

6.4.2 Descriptions for User Service Codes

[11] Group New Messages

This menu is used for new messages. After logging into the mailbox, press [11] to select the Group New Messages menu.

```
[1] Group Urgent Messages: Listen to urgent messages.
[2] Group Callback Messages: Listen to messages with callback
requests.
[3] Group Reminders: Listen to reminders.
[4] Group Private Messages: Listen to private messages.
[6] Group Voice Only Messages: Listen to all voice messages.
[8] Pause, Resume Menu Prompting: Pause the current listening or
operation.  The default pause time is 60 seconds.
[9] Group a Specific Sender: Listen to messages from a specified
caller.
[#] Play Message Inventory: Check the list of all messages in the
mailbox.
[0] Play Menu Options: Check the list of all available menu items.
[*] Cancel, Return to Subscriber Menu: Cancel or go back to the parent
menu.
```

[33] Group Saved Messages

This menu is used for saved group messages. After logging into the mailbox, press [33] to select the Group Saved Messages menu.

```
[1] Group Urgent Messages: Listen to urgent messages.
[2] Group Callback Messages: Listen to messages with callback
requests.
[3] Group Reminders: Listen to reminders.
[4] Group Private Messages: Listen to private messages.
[6] Group Voice Only Messages: Listen to all voice messages.
[8] Pause, Resume Menu Prompting: Pause the current listening or
operation.  The default pause time is 60 seconds.
```

```
[9] Group a Specific Sender: Listen to messages from a specified
caller.
[#] Play Message Inventory: Check the list of all messages in the
mailbox.
[0] Play Menu Options: Check the list of all available menu items.
[*] Cancel, Return to Subscriber Menu: Cancel or go back to the parent
menu.
```

[1] Listen to Messages

You can listen to, save, delete or forward received messages.
After logging into the mailbox, press [1] to select the Listen to Messages menu.

```
[1] Play Message: Listen to a new message or listen to the current
message again.
[11] Play Previous Message: While listening to new messages, listen to
the previous message. If there is no previous message, the feature for
[11] is unavailable.
[2] Save Message: Save the current message.
[3] Discard Message: Delete the current message.
[4] Reply to Sender: After listening to a new message, reply the
caller.
[5] Place Call to Sender: Dial the callback number.
[6] Forward a Copy of Message: Forward a new message to another
location.
[7] Rewind 5 Seconds: While listening to a message, rewind it five
seconds.
[8] Pause, Resume Message Playback: Pause the current listening or
operation. The default pause time is 60 seconds.
[9] Fast Forward 5 Seconds: While listening to a message, fast forward
it five seconds.
[#] Skip to Next Message: Skip to the next message.
[##] Scan Messages: Listen to the first 10 seconds of all new
messages.
[0] Play Menu Options: Check the list of all available menu items.
[00] Play Message Information: Check the information for the current
message.
[*] Cancel, Return to Subscriber Menu: Cancel or go back to the parent
menu.
```

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

[3] Review Saved Messages

This menu is used for managing saved messages. After logging into the mailbox, press [3] to select the Review Saved Messages menu.

```
[1] Play Message: Listen to a new message or listen to the current
message again.
[11] Play Previous Message: While listening to new messages, listen to
the previous message. If there is no previous message, the feature for
[11] is unavailable.
[2] Save Message: Save the current message.
[3] Discard Message: Delete the current message.
[4] Reply to Sender: After listening to a new message, reply the
caller.
[5] Place Call to Sender: Dial the callback number.
[6] Forward a Copy of Message: Forward a new message to another
location.
[7] Rewind 5 Seconds: While listening to a message, rewind it five
seconds.
[8] Pause, Resume Message Playback: Pause the current listening or
operation. The default pause time is 60 seconds.
[9] Fast Forward 5 Seconds: While listening to a message, fast forward
it five seconds.
[#] Skip to Next Message: Skip to the next message.
[##] Scan Messages: Listen to the first 10 seconds of all new
messages.
[0] Play Menu Options: Check the list of all available menu items.
[00] Play Message Information: Check the information for the current
message.
[*] Cancel, Return to Subscriber Menu: Cancel or go back to the parent
menu.
```

[2] Record & Send Messages

This menu is used for recording messages and sending recorded messages. After logging into the mailbox, press [2] to select the Record & Send Messages menu.

```
[#] For a Directory of Subscribers: Find a recipient by directory
name.
[##] To Create a Reminder: Record a reminder.

Enter the Recipient’s Number
[1] Review Recording: Listen to the recorded message.
[2] Stop, Append to Recording: Stop recording, or continue recording
to append to current recording.
[3] Discard Recording and Re-record: Discard the recorded message and
record it again.
[4] Set Delivery Options: Specify a message delivery option.
    [1]: Urgent
    [2]: Return receipt required
    [3]: Call back requested
```


Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

```
   [4]: Private message
   [5]: Reply required
[5] Specify Future Delivery: Message is sent at an appointed time.
   [#]: Immediate delivery
   [1]: Some hours later (1 to 9 hours)
   [2]: At the end of the current work day
   [3]: At the beginning of the next work day
   [4]: At a specified time on a specified day of the week
   [5]: At a specified time on a specified date
[6] Send Message, Then Copy: Send the message and then copy it to
another mailbox.
[7] Rewind 5 Seconds: While listening to a message, rewind it five
seconds.
[8] Pause, Resume Message Playback: Pause the current listening or
operation. The default pause time is 60 seconds.
[9] Fast Forward 5 Seconds: While listening to a message, fast forward
it five seconds.
[#] Send Message, Then Exit Record: Send the message and then exit the
record menu.
[0] Play Menu Options: Check the list of all available menu items.
[*] Cancel, Return to Subscriber Menu: Cancel or go back to the parent
menu.
```

[4] Access Manager

Log into the mailbox and press [4] to select the Access Manager menu.

```
[1] Follow Me: Incoming calls are forwarded to another number.
[3] Block All Calls: Incoming calls are not connected but a recorded
announcement is played instead.
[4] Call Forwarding: Incoming calls are forwarded to another extension
number for a specified period of time.
[5] Call Screening: When there is an incoming call, the system
notifies who is the caller.
[6] Find Me: When the called party is absent, incoming calls are
connected to the multiple phone numbers specified in advance by the
called party. The specified phone numbers are dialed in the order they
were entered. The called party can decide whether to answer the call,
forward it to another number, or to reject the call.
[7] Auto Set Night Intercept: Incoming calls outside the work hours
are not connected but the night greeting is played instead.
[8] Pause, Resume Message Playback: Pause the current listening or
operation. The default pause time is 60 seconds.
[#] Send Message, Then Exit Record: Send the message and then exit the
record menu.
[0] Play Menu Options: Check the list of all available menu items.
[*] Cancel, Return to Subscriber Menu: Cancel or go back to the parent
menu.
```

[5] Personal Greetings

This menu is used for managing personal greetings.
After logging into the mailbox, press [5] to select the Personal Greetings menu.

```
[1] Assign/Edit Primary Greeting: Edit the primary greeting.
[2] Assign/Edit Busy Greeting: Edit the busy greeting.
[3] Assign/Edit Blocking Greeting: Edit the greeting when all incoming
calls are blocked.
[4] Assign/Edit Night Greeting: Edit the night greeting.
[5] Assign/Edit Call Screen Greeting: Set caller information to be
provided.
[6] Edit Only Personal Greeting: Record personal greetings.
[7] Edit Only Mailbox Greeting: Record mailbox greetings.
[8] Pause, Resume Message Playback: Pause the current listening or
operation. The default pause time is 60 seconds.
[0] Play Menu Options: Check the list of all available menu items.
[*] Cancel, Return to Subscriber Menu: Cancel or go back to the parent
menu.
```

[6] Mailbox Administration

Log into the mailbox and press [6] to select the Mailbox Administration menu.

```
[1] Change Message Alert: Set the system to alert a specified phone
number when there is a new message. Select the following sub-level
codes.
   [1]: Enable/disable the message alert feature.
   [2]: Set schedule alert.
   [3]: Set urgent message alert.
   [4]: Change alert number.
[2] Change Pager Notification: Set the system to alert a pager when
there is a new message. Select the following sub-level codes.
   [1]: Enable/disable the alert feature.
   [2]: Set schedule alert.
   [3]: Set urgent message alert.
   [4]: Change alert number.
[3] Review Deleted Messages: Review and/or restore deleted messages.
[4] Review Undelivered Messages: Review sent messages which are not
yet checked by the recipient. After reviewing the messages, you can
cancel their delivery.
[5] Auto Play New Messages: You can set the system to play new
messages each time you log into the mailbox.
[6] Auto Play Message Information: You can set the system to play the
new message information each time you log into the mailbox.
[8] Pause, Resume Message Playback: Pause the current listening or
operation. The default pause time is 60 seconds.
[9] Record & Send Broadcast Message: Record a message and broadcast it
to all mailboxes.
[0] Play Menu Options: Check the list of all available menu items.
[*] Cancel, Return to Subscriber Menu: Cancel or go back to the parent
menu.
```

[#] Personal Services

After logging into the mailbox, press '#' to select the Personal Services menu.

```
[1] Review Workload: Check reminders.
[2] Edit Stored Telephone Numbers: Save phone numbers or edit saved
phone numbers.
[3] Change Weekly Schedule: Set weekly schedule.
[5] Place a Direct Call: Make a call.
[7] Personal Administration: Change general settings for the mailbox.
Change password, name, etc. Select the following sub-level codes.
   [1] Change Password: Change the mailbox password.
   [2] Record Name: Change the record message name.
   [3] Enter Directory Name: Enter the directory name.
   [4] Extended Prompting: Specify the level for playing the mailbox
menu information.
[8] Pause, Resume Menu Prompting: Pause the current listening or
operation.  The default pause time is 60 seconds.
   [0] Play Menu Options : Check the current menu location.
   [*] Cancel, Return to Services Menu : Deselect or go back to the
parent menu.
[#] Record Reminder: Record a reminder.
   [0] Play Menu Options: Check the list of all available menu items.
   [*] Cancel, Return to Subscriber Menu: Cancel or go back to the
parent menu.
```

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

6.4.3 Call Processing System Configuration

IPX-S300B's built-in voice mail system cannot run independently of call processing. Therefore, to use the voice mail system, the necessary settings must be made for the call processing system.

6.4.3.1 License Key Registration

IPX-S300B includes a built-in voice mail system, which requires a license key for use. The license key can be registered or viewed in the **[CONFIGURATION > Resouce>Activation Key]** menu.
To use the voice mail system, a separate embedded application license is required. The license key can be registered in the **[CONFIGURATION > Resouce>Activation Key]** menu.

6.4.3.2 Voice Mail Server Registration

The connection information must be configured for connecting to the voice mail system. Voice mail system connection information can be configured in the **[CONFIGURATION > Application > VM/AA Server]** menu. To use IPX-S300B's built-in voice mail system, 'Application Type' must be set to Internal UMS. If you use VM/AA of VPU, you shoud set to VPU UMS.

In order to use a voice mail system built-up in IPX-S300B, you should set internal UMS to Enable in 'APP type' item. In order to use a VPU card as a voice mail system, set VPU UMS to Enable.

Specifies a voice mail server to use in **[CONFIGURATION > Application > Application Server Service Group]** menu.

Item	Description
User Group	Select a user group which will use the voice mail system. If you set User Group to COMMON, you can config application server as common type.
Name	Specify a voice mail system name.
Application Type	Specify a voice mail system type. - Internal UMS: IPX-S300B's built-in voice mail system. - VPU UMS: Voice mail system using a VPU card.
Use	Set "Enable" or "Disable"
Location	Specify a location where the voice mail system is used.
Call Number	Specifies the phone number to use to call the voice mail system.
Which To Use	Sets which to use among UMS or VPU UMS.
Keep Alive	Checks whether the normal response (200 OK) comes from the application server by sending OPTIONS messages.

Item	Description
Keep Alive Interval (sec)	Sets the interval at which to send OPTIONS messages as follows when receiving the normal response (200 OK) from the application server for OPTIONS messages.
Maximum Keep Alive Retry	Sets the maximum number of retires when receiving no response or error messages from OPTIONS messages.
Register Retry Interval (sec)	Sets the interval between retires when receiving no response or error messages from OPTIONS messages.
Retry wait time	Sets the retry wait time to wait when the normal response (200 OK) fails to arrive after retrying as many times as KeepAlive Retry Maximum number.

6.4.3.3 Service Class Settings

To use the voice mail features, the voice mail-related items must be enabled in the service class. For more information on service classes, see the ‘4.1.36 Feature Service’.
Service classes can be configured in the **[CONFIGURATION > Service > Feature Service > Service Class]** menu. The following voice mail-related items must be enabled.

- AME: If the service permission is set, the AME feature is available. If the AME feature is enabled, when there is an incoming call, the caller’s voice is heard over the speaker as the caller leaves a voice message, just like an answering machine.
- Auto Record: If the service permission is set, the auto record feature is available. If the auto record feature is enabled, when a call is connected, voice mail is automatically connected for recording the call.
- Call Recording: If the service permission is set, the call recording feature is available. The call recording feature records a call conversation in voice mail during the call.

6.4.3.4 Feature Code Settings

When using features related to the voice mail system or when accessing the voice mail system, the user can dial the feature code to have the call connected to the voice mail system. The voice mail system uses the feature code to determine the service type to be serviced.
The feature codes can be configured in the **[CONFIGURATION > Service > Feature Service > Feature Code]** menu. The following conference-related feature codes must be configured.

- AME-Enable, AME-Disable: This feature code enables or disables answering machine emulation. If the service is enabled, when there is an incoming call, the caller’s voice is heard over the speaker as the caller leaves a voice message.
- AME-Manual Start, AME-Manual Stop: This feature code, when there is an incoming call, directs the call to answering machine while ringing, or stops answering machine recording of a call and directs the call to the user.

- Call Record: This feature code is used for requesting recording of the current call conversation in voice mail.
- UMS Access: This feature code is used for dialing to access the voice mail system.
- VM Administration: This feature code is used for accessing the voice mail system and changing it settings.
- VM Memo: This feature code is used for accessing the voice mail system and leaving a message for another number or in your own mailbox.
- VM Message Listen: This feature code is used for accessing the voice mail system and listening to messages in your mailbox.
- VM Transfer: This feature code is used for directing the current call to the voice mail system and connecting it the mailbox for another number.

The administrator can set a particular subscriber's call to be recorded automatically.
To enable the auto record feature, enable 'Auto Record' for 'Service Type' in the **[CONFIGURATION > Service > Feature Service > Service Activation]** menu.
To enable the 'Auto Record' service, the following items must be configured as well.

Item	Description
Auto Record Mailbox	Specify a mailbox where the recorded files will be stored if registering auto record.
Auto Record Call Type	Specify a type of calls to auto record if registering auto record. - Internal: Incoming calls from extension numbers are automatically recorded. - External: Incoming calls from trunks are automatically recorded. - Both: Incoming calls from both extension numbers and trunks are automatically recorded.

6.4.4 VM/AA Basic Mode

VM/AA provides Basic/Advanced mode in the Administration GUI. Basic mode hides the part of Open block Table menu in order to GUI menu makes simple. So the part of feature can't be set on the GUI. Below list is Basic mode feature list. Refer the description of details on the next pages.

- Voice Mail
  - Answering Machine Emulation (AME): Supported by VPU only
  - Auto Call Record: Supported by VPU only
  - Call Back
  - Call forward to Voicemail
  - Call Back Requested
  - Future Delivery
  - Group And Sort Message Prior to Play
  - Individual Mailbox Greeting
  - Individual Mailbox Name
  - Message Delivery Options
  - Message Listen Options
  - Message Forward with Append
  - Message Retrieve
  - Message Reply
  - Message Scan
  - Message Skip
  - Message Undelete
  - One Touch Access
  - Reminder
- Auto Attendant (AA)
  - Auto Attendant Route
  - Interruptible Voice Prompts
  - Multi language Support
  - Operator Access
  - Single Digit Routing
- Voice Studio-Prompt
- Port Activity
- Status Screen

6.4.5 Auto Attendant

When the auto attendant feature is enabled, the voice mail system automatically answers the incoming call and one or more announcements are played for the caller depending on the digit dialed by the caller before connecting the call. The caller can dial a number to specify a particular person or a group.
The auto attendant answers the incoming call with a pre-recorded announcement and allows the caller to select one of the several available features (including selecting a phone number to connect, connecting to the voice mailbox, connecting to the attendant, and accessing pre-recorded information).

6.4.5.1 Multiple Alphabetic Directory

When making a call, if the caller does not know the extension number of the person he/she wants to call, using this feature, the caller can just enter the first few characters of the called party and the system will find the phone number and connect the call to the called party's phone or voice mailbox.

- To register a directory name, log into your mailbox and press '#' to enter the Personal Services menu.
- Press '7' to enter the Personal Services menu.
- Press '3' and use the keypad on the phone to enter your last name and first name as your directory name. After entering the name, press '#' to confirm it.
- Press '2' in the Personal Services menu to record your name.

When there is an incoming call, the Auto Attendant menu will play an announcement telling the caller to enter an extension number to connect or if the caller does not know the number, he/she can press 9 to use the directory feature. Press '9' to enter the Directory menu.
Enter the first few characters of the directory name of the extension number you wish to connect.
If only one extension number matches the condition, the call will be connected immediately. If there are multiple numbers, their names are played for the caller so that the appropriate called party can be selected for connection.

6.4.5.2 Auto Attendant Route

The auto attendant route feature allows calls to be transferred according to the number dialed by the caller. This feature is useful when the voice mail system transfers a call to a specific subscriber or when a call is connected to the voice mailbox.

- Use the Auto Attendant menu to press an extension number to connect.
- If the extension number exists, a voice announcement is played to verify the number and then the call is transferred.

6.4.5.3 Holiday and Special Events

This feature plays appropriate announcements to the callers on public holidays and the company's special days.
The administrator can use the **[VOICE MAIL > Schedule Table]** menu to specify holidays or days with special events based on schedules.

6.4.5.4 Interruptible Voice Prompts

The interruptible voice prompts feature allows the caller to select a service without having to wait for the current voice announcement or greeting to finish.
You can access the next menu by pressing a digit during an announcement or a greeting.

6.4.5.5 Multi language Support

Supports multiple languages.
You can change the service language by changing the language code in the **[VOICE MAIL > System Parameters]** menu.

6.4.5.6 Operator Access

The caller can press a specific digit any time to connect to an available operator.
You can configure operator groups using the **[CONFIGURATION > Service > Group Service > Operator Group]** menu.
In SCM Administrator, use the **[VOICE MAIL > Open Block Table > Menu]** menu to set operator access for a specific digit. Default digit is '0'.

6.4.5.7 Single Digit Routing

Menu Input Processor can be configured to recognize a number for routing to a specific routine. (For instance, it can be configured to connect the call to the extension number 2001 for the sales department when '1' is pressed.)
SCM Administrator can use the **[VOICE MAIL > Open Block Table > Menu]** menu to select a menu and configure in Menu Input Processor.

6.4.6 Voice Mail

The voice mail feature is mainly used for allowing the caller to leave a voice mail for the called subscriber when the subscriber is unable to answer the call. When the call is connected to the voice mail system, the caller will be connected to the voice mailbox either after hearing the ringback tone for some time or without having to hear the ringback tone.
The voice mail system can play pre-recorded announcements depending on the call status such as busy, no answer or DND and connect the call to the voice mailbox for the caller to leave a message.
The subscriber can access his/her own voice mailbox from any location to listen to the messages. Various options are provided for processing the messages.

6.4.6.1 Answering Machine Emulation (AME)

This feature allows the subscriber to use his/her phone's speaker to monitor the call being recorded in the voice mailbox. This is similar to an answering machine.

When AME is running, the caller can leave a message in the called party's mailbox and the called party can listen to the call being recorded. While recording, the called party can press the 'Stop AME' soft button on the phone to end recording and talk with the caller.

For more information on the AME feature, see the 'Voice Mail Interoperation' section in '5.2. User Features'.

> NOTE
>
> **VPU Features**
>
> The Answering Machine Emulation (AME) feature is provided by a VPU card.

6.4.6.2 Auto Record

This feature allows recording of the conversation between the caller and the called party in the called party's voice mailbox. Recording automatically starts when the call starts and recording ends when the call ends. The recording can be checked in the voice mailbox.

When the auto record feature is enabled, you can specify a mailbox in advance.

Depending on the auto record call types, you can record incoming calls only, outgoing calls only, or both the incoming and outgoing calls.

When a call is being recorded, the 'Recording' message will be shown on the screen and the subscriber can use the 'CANCEL', 'PAUSE', and 'STOP' soft menus.

For more information on the AME feature, see the 'Voice Mail Interoperation' section in '5.2. User Features'.

> NOTE
>
> **VPU Features**
>
> The Auto Record feature is provided by a VPU card.

6.4.6.3 Auto Forward

If this feature is enabled, any voice messages in the voice mailbox which are not checked after a specified period of time are automatically forwarded to the mailbox of another phone. The user can specify the delay time and the user can also specify whether the forwarded messages will be deleted from the mailbox or be left undeleted.
To use the auto forward feature, the following items must be configured in the **[VOICE MAIL > Open Block Table > MailBox]** menu.

> NOTE
>
> **VPU Features**
>
> The Auto Forward feature is provided by a VPU card.

Item	Description
Enable auto forward	Select 'Yes' to use the auto forward feature.
Auto forward delay (HH:MM)	Enter the delay time (in minutes) after which the messages are forwarded.
Auto-FWD in Call Director Tab	Select a mailbox to which the messages are forwarded.

> NOTE
>
> If there fails to appear **[Voice Mail/Automatic attendant service > Operation Configuration > Mailbox]** menu, check whether the mode tab is Advanced mode in **[Voice Mail/Automatic attendant service > System Variable]** menu.

6.4.6.4 Auto Login

This feature allows you to automatically log in to a voice mail box without entering the password and going through the authentication process.
To use the auto login feature, the following must be configured.
Administrator can set Auto Login to 'Yes' in General tab under the **[VOICE MAIL > Open Block Table > Extension]** menu.

> NOTE
>
> If there fails to appear **[Voice Mail/Automatic attendant service > Operation Configuration > Mailbox]** menu, check whether the mode tab is Advanced mode in **[Voice Mail/Automatic attendant service > System Variable]** menu.

6.4.6.5 Auto Message Play

This feature automatically plays any new messages that have arrived in a voice mailbox, when you log into the system. This helps minimize the operation of selecting and playing messages.
To use the auto message play feature, the following must be configured.
Administrator can set Auto Play New Messages to 'Yes' in the Authentication tab under the **[VOICE MAIL > Open Block Table > MailBox]** menu.

NOTE

If there fails to appear **[Voice Mail/Automatic attendant service > Operation Configuration > Mailbox]** menu, check whether the mode tab is Advanced mode in **[Voice Mail/Automatic attendant service > System Variable]** menu.

6.4.6.6 Broadcast

This feature allows a subscriber with administrator privileges to broadcast a voice message to all subscribers of the system.
In order to use the message sending function for all users for the mailbox, sets as follows.

- Administrator can set Broadcast Messages to 'Yes' in the Control tab under the **[VOICE MAIL > Open Block Table > MailBox]** menu.
- Administrator can Log in using the phone, press '6' to enter the Mailbox Management menu, press '9' to select the Broadcast Messages menu, record a message and broadcast it.

NOTE

VPU Features

The Broadcast feature is provided by a VPU card.

NOTE

If there fails to appear **[Voice Mail/Automatic attendant service > Operation Configuration > Mailbox]** menu, check whether the mode tab is Advanced mode in **[Voice Mail/Automatic attendant service > System Variable]** menu.

6.4.6.7 Call Back

This feature allows the subscriber, while listening to a message in the voice mailbox, to be connected automatically to the caller of the voice message. This works for both the extension numbers and trunks. For trunks, the phone must be able to recognize call IDs.
To use the call back feature, log into the mailbox, listen to a message and press '5' while listening to call the caller.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Call Forward to Voicemail

This feature allows incoming calls to be forwarded to the voice mailbox when the subscriber is busy or does not answer, or allows all incoming calls to be forwarded to the voice mailbox.
To use the call forward to voicemail feature, the following must be configured:

- To enable call forward busy, in the **[CONFIGURATION > Service > Service Activation]** menu, select a user group and an extension number, click the Search button, select call forward busy and then click the Enable button. Specify the voice mail system feature code for the phone number and click the Enable button.
- To enable call forward no answer, in the **[CONFIGURATION > Service > Service Activation]** menu, select a user group and an extension number, click the Search button, select call forward no answer and then click the Enable button. Specify the voice mail system feature code for the phone number, specify the time (seconds) to wait before forwarding the call as an unanswered call, and click the Enable button.

Call Back Requested

When the caller leaves a message for the called party, the caller can select the call back request option. When leaving a message, the caller can enter the phone number for the called party to call back. When the subscriber listens to the message, the subscriber is notified that call back has been requested. The subscriber only needs to press a number to call the person who left the message.
To leave a call back request message:

- Dial the voice mail system feature code to log into your voice mailbox.
- Press ‘2’, enter a mailbox where you wish to leave a voice mail, and record your message.
- When recording finishes, press ‘4’ and select call back requested for the delivery option.

Distribution List

This feature allows a subscriber to leave a voice message for multiple subscribers at once. A list can not only include subscribers’ phone numbers but also other lists which contain subscribers’ phone numbers. When you leave a message for a list, the same message is left for all the subscribers who belong to the list.
To create a list:

- Create a list using the **[VOICE MAIL > Open Block Table > List]** menu.
- Assign members to the list.
- If you send a voice mail to the list, the voice mail will be sent to the mailboxes of all the members on the list.

External Number Notification

When there is a new voice mail in the subscriber’s mailbox, the notification is sent to a home phone, a mobile phone or another phone registered in advance.
Set the message notification feature to ‘Yes’ in Alerts under the **[VOICE MAIL > Open Block Table > MailBox]** menu.
Specify an alert phone number.
When there is a new voice mail in the mailbox, a call will be made to the specified number.

> NOTE
>
> **VPU Features**
>
> The External Number Notification feature is provided by a VPU card.

Future Delivery

When a subscriber leaves a voice mail for another subscriber, this feature allows the message to be sent at an appointed time in the future.
To use the future delivery feature:

- Dial the voice mail system feature code to log into the voice mailbox.
- Press ‘2’, enter a mailbox number for which to leave a voice mail, and record your message.
- When recording finishes, press ‘5’ to schedule the delivery time.

```
[#]: Immediate delivery
[1]: Some hours later (1 to 9 hours)
[2]: At the end of the current work day
[3]: At the beginning of the next work day
[4]: At a specified time on a specified day of the week
[5]: At a specified time on a specified date
```

Group and Sort Message Prior to Play

This feature allows the subscriber to listen to the voice messages in his/her mailbox by grouping them into different types (urgent, call back, reply requested, alarm message, etc.).
To use the group and sort message prior to play:

- Log into your voice mailbox.
- Press ‘11’ to listen to the messages as grouped by types.

Individual Mailbox Greeting

This feature allows the subscriber to record a greeting for his/her own mailbox.
When a caller is connected to the subscriber’s mailbox for leaving a message, the recorded mailbox greeting will be played.
To record an individual mailbox greeting:

- Log into your voice mailbox.
- Press ‘5’ to record a greeting in the Individual Greeting menu.

Individual Mailbox Name

This feature allows the subscriber to record his/her name with his/her own voice and link it to the subscriber's personal mailbox.

- Log into your voice mailbox.
- Press '#' to enter personal services.
- Press '7' to enter personal management.
- Press '2' to record your name.

Message Delivery Options

This feature allows you to set different options when sending a voice message. Available options include urgent message, call back request, reply required, confidential message, and return receipt.

- Log into your voice mailbox.
- Press '2', enter a mailbox number for which to leave a voice mail, and record your message.
- When recording finishes, press '4' to specify a message delivery option.

```
[1]: Urgent
[2]: Return receipt required
[3]: Call back requested
[4]: Private message
[5]: Reply required
```

- Press '#' to send the message.

Message Listen Options

These are the options used for listening to voice messages in the voice mailbox. Available options include replay, save, delete, rewind, fast forward, and pause.

- Log into your voice mailbox.
- Listen to a voice message.
  - Press '1' to play the message from the beginning again.
  - Press '2' to save the message.
  - Press '3' to delete the message.
  - Press '7' to rewind the message by 5 seconds.
  - Press '8' to pause the message and press '8' again to resume playing.
  - Press '9' to fast forward the message by 5 seconds.

Message Forward With Append

When the subscriber forwards a voice message in his/her mailbox, this feature allows the subscriber to record additional information about the voice message which will be forwarded with the voice message.
To forward a voice message with append:

- While listening to a voice message on the phone, press ‘6’.
- Enter the number of a mailbox to which the message will be forwarded.
- To record an introduction, press ‘2’ and record it. After recording finishes, press ‘#’ to send the message.

Message Retrieve

After the subscriber has left a message in another subscriber’s voice mailbox, this feature allows the caller to cancel the voice message if the called party has not yet listened to the message.
To cancel a voice message delivery:

- Log into your voice mailbox.
- Press ‘6’ to enter the Mailbox Management menu.
- Press ‘4’ to enter the Review Undelivered Message menu. If asked to enter a mailbox number, enter the called party’s mailbox number.
- Listen to your message and press ‘2’ to have the message deleted from the called party’s mailbox and have the message sent back to your mailbox.

Message Reply

This feature allows the subscriber to press a button while listening to a voice message in the subscriber’s voice mailbox to leave a voice message for the caller.
To reply a voice message:

- Log into your voice mailbox.
- While listening to a voice message, press ‘4’ to leave a message in the caller’s mailbox.

Message Scan

This feature allows the subscriber to scan all the messages in the subscriber’s voice mailbox by listening to the beginning part (10 seconds) of each message.
To scan messages:

- Log into your voice mailbox.
- While listening to a voice message, press ‘##’ to listen to the beginning part only and skip to the next message.

Message Skip

When listening to a message in the voice mailbox, this feature allows the subscriber to listen to the next message without waiting for the current message to finish.

- Log into your voice mailbox.
- While listening to a voice message, press '#' to listen to the next message.

Message Undelete

This feature allows the subscriber to search for the messages which were previously deleted in the voice mailbox and listen to them or save them.

- Log into your voice mailbox.
- Press '6' to enter the Mailbox Management menu.
- Press '3' to select the Check Deleted Messages menu.
- You can listen to the deleted messages. You can also save, copy or forward them as you would with normal messages.

One Touch Access

This feature allows you to log into your mailbox or log in with administrator privileges with a single button.
To enable the one touch access feature and log in with administrator privileges:

- Register the 'UMS Administration' feature code in the **[CONFIGURATION** > **Subscriber > Device Key Programming]** menu.
- Press the key on the phone to log in with administrator privileges.

Reminder

The reminder feature allows the subscriber to leave a message for himself/herself. This is useful for recording important events or information.

- Register the 'UMS Memo' feature code + your mailbox number in the **[CONFIGURATION > Subscriber > Device Key Programming]** menu.
- Press the registered key and leave a message.

6.4.7 Access Manager

Access Manager controls how the callers are connected to individual subscribers.
A mailbox owner can specify settings to disable ringing for his/her extension number, forward incoming calls to another extension number, or scan calls before answering them. All these settings are valid until the time specified. You can also enable the find me feature which allows you to call stored phone numbers to connect to subscribers in multiple locations.

6.4.7.1 Call Blocking

While call blocking is enabled by the subscriber, the voicemail system does not connect callers to the subscriber's extension. Instead, the call blocking greeting prompt is played immediately for the caller. If the call blocking greeting has not been recorded, the voicemail system plays the subscriber's no answer greeting. If the no answer greeting has not been recorded either, the voicemail system plays an announcement informing the caller that the number dialed is currently unavailable and other options are given to the caller.
The subscriber can enable call blocking using Access Manager. After enabling call blocking, the subscriber can set the time period for call blocking.
This feature is enabled for the following conditions:

1) For the next 1 to 9 hours as specified
2) Until the end of the current work day
3) Until the beginning of the next work day
4) Until the next specified day of the week
5) Until the specified date

Call blocking can be enabled in the following way:

- Set Allow Blocking to 'Yes' in the Authentication tab under the **[VOICE MAIL > Open Block Table > Extension]** menu to give the subscriber the privilege to configure call blocking.
  - To record a greeting, set Greeting to Basic in the Caller Option Processor tab under the **[VOICE MAIL > Open Block Table > Extension]** menu.
  - Log into the mailbox and press '4' to select the Access Manager menu.
  - Press '3' to enable call blocking.
  - While logged into the mailbox, press '5' to enter the Personal Greeting menu.
  - Press '3' to record a blocking greeting.

NOTE

VPU Features

The Call Blocking feature is provided by a VPU card.

6.4.7.2 Call Forwarding

This feature directs callers to another extension number. (Directing to trunks is not possible.) When a call is connected to an extension, the caller will hear the prompt 'You are attempting to connect to person A in department B. This call has been forwarded to person C'. When the called party answers the call, the called party will hear an announcement explaining where the call is being forwarded from.
This feature is enabled for the following conditions:

1) For the next 1 to 9 hours as specified
2) Until the end of the current work day
3) Until the beginning of the next work day
4) Until the next specified day of the week
5) Until the specified date

Call forwarding can be enabled in the following way:

- Set Call Forwarding to 'Yes' in the Authentication tab under the **[VOICE MAIL > Open Block Table > Extension]** menu to give the subscriber the privilege to configure call forwarding.
- Use your phone to log into the mailbox. Press '4' to select Access Manager and press '4' again to select Call Forwarding, and then specify an extension number for forwarding and set a duration.

When there is an incoming trunk call, the caller will be informed that the call is being forwarded to another number and the call will be forwarded to the specified number.

> NOTE
>
> **VPU Features**
>
> The Call Forwarding feature is provided by a VPU card.

6.4.7.3 Find Me

If this feature is enabled, the voice mail system attempts to forward incoming calls to a location specified by the subscriber. The voice mail system firstly looks for the subscriber in the location specified by the subscriber, and then, if necessary, calls all of the numbers specified by the subscriber until the call is answered.
This feature is enabled for the following conditions:

1) For the next 1 to 9 hours as specified
2) Until the end of the current work day
3) Until the beginning of the next work day
4) Until the next specified day of the week
5) Until the specified date

Find me can be enabled in the following way:

- Set Allow Find Me and Phone Number Saving to 'Yes' in the Authentication tab under the **[VOICE MAIL > Open Block Table > Extension]** menu.
- After logging into the mailbox, press '#' to enter the Personal Services menu.
- Press '2' to enter phone numbers to use for the find me feature. You can enter up to 9 numbers.
- After logging into the mailbox, press '4' to enter Access Manager.
- Press '6' to select the Find Me menu and specify the duration for which the feature will be enabled.

NOTE

VPU Features

The Find Me feature is provided by a VPU card.

6.4.7.4 Follow Me

Any subscriber can pick up a call that is automatically forwarded to a designated location. This is called subscriber location designation. Location designation is possible for both extension numbers and external phone numbers.
This feature is enabled for the following conditions:

1) For the next 1 to 9 hours as specified
2) Until the end of the current work day
3) Until the beginning of the next work day
4) Until the next specified day of the week
5) Until the specified date

Follow me can be enabled in the following way:

- Set Allow Alternative Numbers to 'Yes' in the Authentication tab under the **[VOICE MAIL > Open Block Table > Extension]** menu.
- After logging into the mailbox, press '4' to enter Access Manager.
- Press '1' to specify phone numbers to connect and specify the duration for which the feature will be enabled. Specify both extension numbers and trunk numbers and test them repeatedly.

NOTE

VPU Features

The Follow Me feature is provided by a VPU card.

6.4.7.5 Multiple Personal Greetings

When the subscriber is unable to answer calls, the voice mail system answers them.
The voice mail system uses five different reasons for the subscriber's inability to answer calls. Different greetings can be used for different reasons. The voice mail system allows the subscriber to record greetings 1 to 9. The subscriber can link any of these greetings to appropriate reasons/conditions.
To record multiple personal greetings, the following must be configured:

- In the Authentication tab under the **[VOICE MAIL > Open Block Table > Extension]** menu, set Allow Blocking to 'Yes' to be able to record block greetings, set Allow Busy Greeting to 'Yes' to be able to record busy greetings, and set Call Screening to 'Yes' to be able to record call screening greetings.
- Log into the voice mailbox and press '5' to enter the Personal Greetings menu. Record the greetings.

6.4.7.6 Park and Paging

The voice mail system provides the park and paging feature to those users who frequently leave their desks. When the subscriber does not answer the call, the caller can select 'Park and Paging'. Then, the voice mail system turns on the speakers on all the phones in the same station paging group as the subscriber and makes an announcement that there is an incoming call for the subscriber.

Park and paging can be enabled in the following way:

- Create a conference group in the **[CONFERENCE > Conference management > Paging]** menu.
- Enter the number of the conference group created in the page zone in the Overhead Page tab under the **[VOICE MAIL > Open Block Table > EClass]** menu.
- Set Overhead Page When Unanswered to Y in System Caller Options in General under the **[VOICE MAIL > Open Block Table > EClass]** menu.

NOTE

VPU Features

The Park and Paging feature is provided by a VPU card.

6.4.8 Administration

The administration feature allows administration of the system using the essential operations data (including extension numbers, mailbox numbers, and various messages) as well as monitoring and statistics data.

6.4.8.1 Activity Display

The activity display feature provides a simple view of the operation activities of the voice mailbox.
To use activity display service, go to the **[VOICE MAIL > Status Screen]** menu.

6.4.8.2 System Report

This feature shows the usage activities of the voice mail system. To view the system report, go to the **[VOICE MAIL > View System Report]** menu.

Application Report

This shows call activities for each application.

- Report duration: This shows the beginning and the end of the reporting period.
- Φ Count: This is the total number of calls serviced by the application.
- Total time connected (min): This is the total time (minutes) callers were connected to the application.
- Total caller percentage (%): This is the percentage of the callers connected to the application out of the total number of callers.

System Report

This shows the call activities for the subscriber's extension number.

- Report Duration: This is the reporting period. This is begins on day the report counter was last deleted and ends with the current counter.
- Subs Calls: This is the total number of calls for the subscriber's extension number grouped by the process types (including completed, forwarded, and rejected).
- Subscriber calls: This shows how the incoming calls for the subscriber were processed.
- Tot Subs Calls: This is the total number of incoming calls for the subscriber's extension number.
- Caller Hold Time: This is the total time (minutes) for which the subscriber had been waiting without ending calls.

Message Status Report

This shows the call activities for external callers and mailbox subscribers.

- Report Duration: This is the reporting period. This is begins on day the report counter was last deleted and ends with today's date.
- Activity: This is the message activity by type. There are a few different categories.
- External: The first column is the total number of external callers connected to message activities of a specific type. The second column is the percentage of the total number of external callers out of the total number of callers (including subscriber callers) connected to message activities of a specific type.
- Subscriber: The first column is the total number of subscriber callers connected to message activities of a specific type. The second column is the percentage of the total number of subscriber callers out of the total number of callers (including external callers) connected to message activities of a specific type.
- Total: This is the total number of callers connected to message activities of a specific type.

Call Code Activity Report

This shows the system operation activities by call codes.

- Report Duration: This is the reporting period. This is begins on day the report counter was last deleted and ends with today's date.
- Port Utilization by Call Code: This is the list of call code types.
- Calls: This is the total number of calls recorded for each call code.
- Total calls (%): This is the percentage of calls for a specific call code.
- Total time connected (min): This is the total time connected (minutes) for all calls for a specific call code.

Daily Report

This shows call activities for each hour of a day.

- Report Duration: This is the reporting period. This is begins on day the report counter was last deleted and ends with today's date.
- Calls: This is the total number of incoming calls for a specific hour.
- Total calls (%): This is the percentage of incoming calls during a specific hour.
- Total time connected (min): This is the total time connected (minutes) for all calls during a specific hour.

Port Number Report

This shows call activities for each port.

- Report Duration: This is the reporting period. This is begins on day the report counter was last deleted and ends with today's date.
- Calls: This is the total number of incoming calls via a specific port.
- Total calls (%): This is the percentage of incoming calls via a specific port.
- Total time connected (min): This is the total time connected (minutes) for all calls via a specific port.

Weekly Report

This shows call activities for each day of a week.

- Report Duration: This is the reporting period. This is begins on day the report counter was last deleted and ends with today's date.
- Calls: This is the total number of incoming calls for a specific day of a week.
- Total calls (%): This is the percentage of incoming calls for a specific day of a week.
- Total time connected (min): This is the total time connected (minutes) for all calls for a specific day of a week.

6.4.9 Voice Studio

Voice Studio allows you to create, edit, or delete prompts and announcements used by the voice mail system. You can either dial from your phone to record your voice directly or select pre-recorded files for use.
To use your phone to record:

- Select Prompt/Announcement in the **[VOICE MAIL > Voice Studio]** menu.
- Enter your phone number in the window next to the Call button and click the Call button. The voice mail system will dial your phone. Answer your phone and click the Create button to enter a prompt/announcement number and create. On your phone, you will hear the prompt number entered. Press any digit on the phone to start recording. Press '#' to end recording. The prompt file will be created.
- Select a prompt/announcement and click the Change button to edit.
- Select a prompt/announcement and click the Delete button to delete.

6.4.10 VM/AA Config Data Backup

A back up feature for data such as VM prompt, recorded personal greetings, etc.

Backup
In the **[Manage > Manage Database > Database Backup]** menu, select "Now" as the backup type.
Select "Yes" in the voice mailbox data backup, and the backup will start.

In the **[Manage > Manage Database > Manage Database Backup Files]** menu, you can check the backup files. (2.0.0.15_xxxxxxxx_VPU_VM.tar or 2.0.0.15_xxxxxxxx_GWU_VM.tar)

Restore
In the **[Manage > Manage Database > Manage Database Backup Files]** menu, select the file to restore among the backup files, and click "Restore" to restore the file.

6.4.11 Voice Mail Data Backup

To backup VM audio files, you must access VPU or GWU through FTP and backup the data.

In the **[Configuration > Cabinet/Slot > Internal Service Control]** menu, set FTP in the slot where VPU (GWU) is installed to On and save, and you can access VPU through FTP from your PC.

Audio file save path: /scm_data/ums/userdata/voicemail/

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

6.5 Call Recording

Since IPX-S300B V2.0, new recording feature is supported. Unlike recording supported in the previous VM, it consolidate and manage recorded files, and it is easy to listen, download, and backup recorded files. To use this feature, a VPU card must be installed.

6.5.1 Recording Method

IPX-S300B has two call recording methods, and the user can choose any.

Conference Recording

It is a recording feature with a 3 way conference function including the caller, receiver and 3rd party with the recording VPU. It can be used in all kinds of telephones such as analog telephones (FXS) and SIP telephones.

Phone Recording

It is supported in some of our SIP telephones (i60xx, i5243, etc.). It is a recording feature that transmit the call content to the VPU.

Recording Method Setting

In the **[Configuration > User Group > Service Group > Call Recording Method]** menu, the recording method can be selected. In the **[Configuration > User > Single Phone User > Service > Call Recording Method]** menu, the method per user can be set.

6.5.2 Recording Specifications

VPU Number

Two VPU's can be installed: one for VM/Conf and one for Recording. Two VPU's cannot be installed for a single purpose.

Recording Channel Number

A maximum of 10 for conference recording and a maximum of 20 for telephone recording are supported. When both are used, the available number decreases in the ratio of one conference recording to two telephone recording. In addition, this can be limited by the license entered.

Recording Time

A maximum of 500 hours of recording files can be saved to the VPU. However, it can continue to record, overwriting old files or deleting old files after backup. The maximum recording time per call is limited to 24 hours.

Recording File List

The recording file list can store up to 100,000 and oldest ones will be deleted after exceeding 100,000.

Recording File Type

Recorded files are saved to the VPU as G.711 8000 Hz Mono 64 kbps, and also transmitted to the backup device in this format. However, when playing and downloading in the SCM Administrator, it is downloaded after converted to WAV.

> **NOTE**
>
> If a conference member wants conference recording, the VPU(REC) channel is connected to the conference. In that case, some services are limited including transfer service. So the phones supporting RTP mixing should use phone recording not conference recording.

6.5.3 Slot/Application Setting

Slot Setting

Set the VPU installed in **the [Configuration > Cabinet/Slot > Cabinet Configuration > Slot Configuration]** menu to VPU (Recording).

Application Setting

In the **[Configuration > Application > VM/CR Server]** menu, activate UG1.VPU.REC. Then, select UG1.VPU.REC as the recording server in the **[Configuration > Application > Application Server Group]** menu.

6.5.4 Recording Backup

The recorded contents are saved to the VPU, and can be backed up automatically or manually. An FTP server must be arranged for backup

Backup Setting

In **the [Configuration > Service > Call Recording > Call Recording Configuration]** menu, set the items required for recording file backup.

Item	Description
FTP Server IP	The IP address of the external FTP server to backup recording files
FTP Server Port	The port number connected to the FTP server: 21 for FTP and 22 for

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
	SFTP in general.
User ID	The user ID to access the FTP server
User Password	The password to access the FTP server
FTP/SFTP	Select the protocol to use when connecting to the FTP server FTP (File Transfer Protocol) and SFTP (Secure File Transfer Protocol) are supported. SFTP is one of the additional features of SSH and is a different protocol from FTPS (FTP-SSL).
Passive Mode	Access in the passive mode when connecting to the FTP server
FTP Server Directory	Select the directory to save backup files; If there is no setting, save to the root directory
Automatic Backup	Back up automatically after checking recording files periodically

Manual Backup

In the **[Configuration > Service > Call Recording > Call Recording Backup**] menu, recording files, which have not bee backed up, can be backed up manually.
Depending on the number of files to back up, it may take a long time, and an alert showing the result will appear after the backup is completed.

6.5.5 Recording File Management Setting

To manage recording files, the following features are supported.

Item	Description
Delete After Backup	Delete original files after transferring files to the external backup device
Disk Full	Choose to stop recording or to continue to record by overwriting old files when the VPU disk is full. If overwriting is chosen, no VPU disk alert will appear.
User Maintenance	Choose whether to use daily sync operation, comparing recording file records saved in the DB with recording files within the disk.
Maintenance Time	Select the time for daily management: the default is 00:00.

6.5.6 Recording File Search/Play/Download/Delete

In the **[Configuration > Service > Call Recording > Call Recording List]** menu, you can search recording files by telephone numbers, time, etc., and select, play, download, and delete files.

Call Recording File List

The following information can be identified in the recording file list, and you can search files with this information.

Item	Description
Call ID	There can be a number of files with the same identifier and number used to identify each call.
Sequence	The file order in a single call. A file records up to 1 hour: if the call is longer, the number of files will increase.
File Name	The file name. A file is named as recording start time_extension number_the other party's number_call identifier_order_serial number.
Extension	The extension number recording a call
Opponent	The other party's number, communicating with the extension number
Start Time	The time of starting recording
Stop Time	The time of stopping recording
Length	The file length (hour:minute:second)
File Backup	If the recording file is transfered to the external backup device or not
File Exist	If the recording file exists in the disk or not

Recording File Play/Download/Delete

A file can be selected, played, downloaded, and deleted from the recording file list.
When playing or downloading, the file is played or downloaded to the PC after changed to WAV. If the file is big, it may take longer for format change.
If the file is not in the VPU, it cannot be played or downloaded.

6.5.7 VPU Status Check

Possible Recording Time

In the **[Configuration > Service > Call Recording > Call Recording Status]** menu, the time currently available for recording can be checked. However, you can continue to record, overwriting or deleting old files according to the setting.

LED Status

When using the VPU as recording, RUN, MC, and MEM LED, located on the front side, operate as follows.

Item	Description
RUN	- Off: VPU not operating - On: VPU booting - Blink slowly: VPU normally operating after registration
MC	Blink while reading/writing to SSD
MEM	- Off: up to 79 % of SSD in use - On: 80-94 % of SSD in use - Blink: more than 95% of SSD in use

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

6.5.8 Conference Recording Restrictions

The conference recording service provided by IPX-S300B has the following limitations.
If both caller and callee are set auto recording, the caller only can be serviced auto recording.
In case of meet me conference call, auto record service is not allowed.
The call connected to a voice mail does not allow auto record service.
The recording call though conference server (including VPU conference) are limited some services like transfer and so on. IP phone should use phone recording service.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

6.6 External Application

6.6.1 PC Attendant

An attendant service is phone to use when exchange operators connect received calls to other users, a PC attendant service is a software functioning as PC attendant service in linkage with IPX-S300B.

For detailed function description and usage, refer to 'IP attendant service manual'.

6.6.1.1 PC Attendant Service Registration

If a PC attendant service is to be linked with IPX-S300B, generate user information as shown below, and register the PC attendant service to IPX-S300B.

1) Click the Generate button and enter corresponding values in **[Configuration > User > Single Phone User]** menu. It is the same as generating normal users. Specifies Phone Type to Samsung-PC-Attendant.
2) Click the Generate button and enter a corresponding value in **[Configuration > User > Attendant Console]** menu.
   - User Group: Select a user group to link with the PC attendant service.
   - Name: Specify suitable PC attendant service names.
   - Phone > All: Displays all normal users specified as Samsung-PC-Attendant.
   - Phone > Select: Select one PC attendant service to use for the user group among other attendant services shown to all, and assign it.
3) Click the Generate button and enter a corresponding value in **[Configuration > User > Attendant Console]** menu.

6.6.1.2 PC Attendant Service BLF Specification

A PC attendant services the BLF (Busy Lamp Field) function to display user status.

You can specify the users to display the status as follows
Click the Find button in **[CONFIGURATION > User > Attendant Console BLF Setting]** menu Select one among the attendant console list, click the Change button, and enter a corresponding value.

- Display Name: Specify a suitable name for the button.
- Key: Specify as BLF. It can only be specified as BLF.
- Extension Number: Enter the extension number of a user that you want to know the status.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

6.6.2 Communicator

The Samsung Communicator is Unified Communications Client software running on a client PC that takes the functionality commonly used and understood on our telephones and puts it at your finger tips and Screens on your PC. The Samsung Communicator can be run in three different Device Modes. The Samsung Communicator can be a Stand-alone device when in the Soft Phone Mode, when in UC Phone Mode it can work in connection to a UC Phone (SMT-i Series) device. When in Desk Phone Mode, it can work in connection to IPX-S300B directly via CSTA I/F for call control of UC Phone (SMT-i Series)

NOTE

Refer to the 'Samsung Communicator User Guide' for detailed information.

6.6.3 Messenger

6.6.3.1 Messenger Server Registration

The Messenger Server is a server program which provides the communicators' presence, instant messages, organization maps, and buddy functions, and is included in SAS (Samsung Application Suite). SAS is a Window-based server product which will be installed separately from IPX-S300B system.

In order to register the Messenger Server, click the Generate button and enter corresponding values in **[CONFIGURATION > Application > Messenger Server]** menu.

Item	Description
User Group	Select a user group to interwork with the Messenger Server (The Messenger Server supports only one user group).
Name	Specify a suitable server attendant service name.
Application Type	Select External Messenger.
IP address	Enter the IP address of the Messenger Server.
Port	Enter the port number of the Messenger Server. The default value is 5070.
Public IP Address	Specify the Public IP of the Messenger Server. It is when the Messenger Server is within the NAT, the Communicator outside the NAT tries to connect to the Messenger Server.
Public Port	Specify the Public Port of the Messenger Server. It is when the Messenger Server is within the NAT, the Communicator outside the NAT tries to connect to the Messenger Server.

Set Public IP address and Public Port in **[CONFIGURATION> Miscellaneous > System Option]** menu, when System Under NAT item is set to Enable and the Communicator outside the NAT tries to connect to the Messenger Server.

6.6.3.2 Interworking with User Information

User Information is managed in SCM Administrator, and should be set to synchronize with the User Information of IPX-S300B on the screen of SAS Administrator. User Information to be interworked with SAS includes user, department and position.

NOTE

Refer to the 'Samsung Communicator User Guide' for detailed information.

6.6.3.3 User Information field settings

Some of User Information only used in messengers is not specified as a user field just after the installation. So, in order to use that information, you must specify application user fields as shown below.
Specify a name by index in **[Environment Settings > User Group > User Profile]** menu. For the name field, you should enter suitable terms as that field is displayed in application user generation/change screens, and should use indexes as suitable for their purpose.

Index	Description
1	Company phone number (Default)
2	Mobile phone number (Default)
3	Fax number (Default)
4	Email (Default)
5	User detailed description
6	Home phone number
7	Mobile extension number
8	Address
9	Nickname
10	Responsibilities
11	Image file name
12	PIN CODE (Resident registration number)

NOTE

Refer to the 'Samsung Application Suite Administrator's Guide' for detailed information.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

6.6.4 ACD Agent Program

ACD Agent is a call center agent that enables call center agents to operate the IPX-S300B system effectively and enables managing call center agents.
Agent provides all functions that can be performed by call center agents for operating the call center.

ACD Agent Desktop is the Windows-based Client Application that resides on the agent's computer. With Agent ACD, you can control calls with your computer.
The ACD Agent allows you to:

Call Control

- Make Calls, Answer Calls, Hang-up Call
- Hold Call, Retrieve Call
- Transfer Calls, Blind Transfer Call
- Conference Calls

Favorites

Agents, Tel Number, ACD Group

History Call Log

ACD Incoming Call, Non-ACD Call

Monitoring

Provides the real time status information of the ACD Group where the Agent belongs.

- Number of wait calls: Currently waiting for the call distribution.
- Number of Logged-in Agents
- Longest ACD wait time (current): Waiting time of the call that has waited for the longest time among the calls currently waiting
- Number of IDLE Agents: Agents that the call distribution service has available.

> **NOTE**
> Refer to the 'ACD Agent User Guide' for detailed information.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

6.6.5 SCplus Server

SCplus is a CTI middleware solution to build up a contact center by interworking with IPX-S300B. Broadly, it consists of an inbound module and an outbound module.
The inbound module has features such as connecting very quickly a customer to one of the most suitable consultants when he or she requests a consultant (skill based routing, designated consultant, reception scheduling, VIP customer/consultant, network ICD, virtual contact center routing, blending, etc.), and managing calls and consultants. The outbound module has features such as automatically dialing for a customer with a consultant having to use the phone directly when a consultant is required to do so in order to improve default outbound businesses and maximize their efficiency.

NOTE

Refer to the 'SCplus (M-Server) Operator Manual' for detailed information.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

6.7 Solution Partner Application

6.7.1 External Ringback Tone Server

A coloring server is i a device to provide a ringback tone. If interworked with a coloring server, a caller can hear a coloring sound until the call is connected to a receiver.

License Key Registration

In order to interwork with a coloring server, an additional license key for external applications are needed. A license key can be registered in **[CONFIGURATION > Resource > Activation Key] menu**.

Interworking with a coloring server can be performed within the number of SIP Application Channel, so you should check the number of interworking channels for SIP auxiliary devices after entering a license key.

CAUTION	The number of interworking channels for SIP auxiliary devices include the number of interworking channels for other application servers. If other applications use a number of channels, the number of channels available for a coloring server is reduced.

Coloring Server Configuration Method

In order to configure a coloring server, enter the fields as shown below in **[CONFIGURATION > Application > Other Application Server]** menu.

In default, IPX-S300B checks the connection with an application server through SIP OPTIONS.

Index	Description
User Group	Select a user group.
Name	Set a name for the application server.
Application Type	Select External Ringback Tone.
Location	Select one among configured location information.
Access Number	Enter a number to represent the application server This number should be different from other extension numbers, Access numbers, conference channel numbers and system shortcut dial numbers.
SIP URI	Enter in a form of email address. e.g. aa @ IP address
Registration mode	Select None.
Proxy server	Enter the IP address of the application server.
Port	Set to 5060. If the application server does not use other ports, enter the port number being used.
URI Type	Select SIP.
Protocol	Select UDP. In order to interwork with the application server via

Index	Description
	different protocols, select TCP or TLS.
Register Expire (sec)	Do not enter.
User name	Do not enter.
Password	Do not enter.
Keep Alive	Select Enable.
Keep Alive interval (sec)	Keep the default value of 30.
Maximum number of Keep Alive Retry	Keep the default value of 1.
Maximum Register Retry.	Do not enter.
Register Retry Interval (sec)	Do not enter.
Register Pause time (sec)	Keep the default value of 60
Keep Alive Retry Interval (sec)	Keep the default value of 35.
Maximum Call	Specify the number of channels available for the coloring server. It can be specified in the number of interworking channels for SIP Application Channel in **[CONFIGURATION > Resource > Activation Key]** menu.
Route Name	Do not enter.
Check Registration Protocol	Do not enter.
Default Code	Do not enter.
Authentication	Do not enter.
A-A Primary Node	Do not enter.
A-A Dual Registration	Do not enter.
A-A Dual Operation	Do not enter.

The conditions to play a coloring ringback tone can be specified by user or extension group. Below is the procedure for configuration.

In order to specify by user, select users in **[CONFIGURATION > User > Single Phone User]** menu and then push the Change button. In order to specify by hunt group, select a hunt group in **[CONFIGURATION > Service > Group Service > Hunt Group]** menu and then push the Change button.

1) Select either Internal, External or Both in External Ringback Tone Use.
   - Internal: Coloring is provided when a call originates from extension lines.
   - External: Coloring is provided when a call originates from external lines.
   - Both: Coloring is provided irrespective of the origination of a call.

2) Specify the server generated in Application Server menu in External Ringback Tone Server Specification.
If the server is not specified, The default coloring server will operate. If there is no server specified as the default coloring server, any coloring server may operate.

6.7.2 Voice Mail System Server

6.7.2.1 License Key Registration

In order to interwork with a VMS server, an additional license key for external applications are needed.
A license key can be registered in **[CONFIGURATION > Resource > Activation Key] menu**.
Interworking with a VMS server can be performed within the number of interworking SIP Application Channel, so you should check the number of interworking channels for SIP auxiliary devices after entering a license key.

> **CAUTION**
> The number of interworking channels for SIP auxiliary devices include the number of interworking channels for other application servers. If other applications use a number of channels, the number of channels available for a coloring server is reduced.

VMS Server Configuration Method

In order to configure a VMS server, enter the fields as shown below in **[CONFIGURATION > Application > VM/AA Server]** menu. In default, IPX-S300B checks the connection with an application server through SIP OPTIONS.

Index	Description
User Group	Select a user group.
Name	Configures the name of a server.
Application Type	Select solution partner VMS.
Location	Select one among configured location information.
Access Number	Enter a number to represent the application server This number should be different from other extension numbers, call numbers, conference channel numbers and system shortcut dial numbers.
SIP URI	Enter in a form of email address. e.g. aa @ IP address
Register Type	Select 'None'.
Proxy server	Enter the IP address of the application server.
Port	Set to 5060. If the application server does not use other ports, enter the port number being used.
URI Type	Select SIP.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Index	Description
Protocol	Select UDP. In order to interwork with the application server via different protocols, select TCP or TLS.
Register Expire (sec)	Do not enter.
User name	Do not enter.
Password	Do not enter.
Keep Alive	Select Enable.
Keep Alive interval (sec)	Keep the default value of 30.
Maximum Keep Alive Retry	Keep the default value of 1.
Maximum Rgister Retry	Do not enter.
Register Retry Interval (sec)	Do not enter.
Register Pause time (sec)	Keep the default value of 60
Keep Alive Retry Interval (sec)	Keep the default value of 35.
Maximum Call	Specify the maximum number of channels available for the voice mail server. If channel values are not given, all calls from a server not set to Internal IMS will be rejected.
Check Registration Protocol	Select whether to check protocols (UDP, TCP, TLS) used in registration.
Authentication	Select whether to allow a voice mail server to use the authentication method for user groups.
A-A Primary Node	Do not enter.
A-A Dual Registration	Do not enter.
A-A Dual Operation	Do not enter.
Service List	Several voice mail servers can be generated. The three services as shown below can be specified for each server. However, one service can be specified for one server. - Voice Mail - Call Recording - Auto Attendant

6.7.3 Recording Server

Currently, a recording server using IP packet mirroring can be interworked with IPX-S300B. In order to use a recording server using IP packet mirroring, additional settings are required for IP network devices, but not for IPX-S300B.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

CHAPTER 7. System Management

This chapter describes the features required for system management.

7.1 Managing System Access

7.1.1 Access Permission

You can use SCM Administrator to control access for operators.
Use the **[MANAGEMENT > Access Permission > Account List]** menu to manage operators' ID, password, class, login timeout, password duration, forced password change, etc.
Class is identified with ENGINEERING, TECHNICIAN, and CUSTOMER.
Login timeout supports function which auto logout after time set if nobody use administrator.
Password duration supports how many days login without password confirmation.
If duration is over, it should display password change dialog box.
Forced password change supports even password duration is over, administrator can operate without password change. If set to 'No', operator just close password change dialog and continuously work. But if set to 'Yes', close password change dialog, it should logout SCM Administrator
Password is following these rules:

Status	Description
Password Duration	Default (30), Min (7), Max (999) days
Use same password with previous	Not allowed
Password length	Min (6), Max (40)
Simple Password	Not allowed only characters or digits (ex: abcdef, 12345)
Consecutive use	Not allowed more than 4 consecutive use of a same character or digit (ex: aaaa12, 1111pass)
Same as ID	Not allowed same as ID or reserved ID

Use the **[MANAGEMENT > Access Permission > Terminal Control]** menu to control access by entering user terminal information.
Use the **[MANAGEMENT > Access Permission > Login Status]** menu to view status of current users. To log out a user, select the user and click the Log Out button.

7.1.2 Access Control List (ACL)

IPX-S300B provides ACL function to control access from outer side.
Administrator can configure the five types of ACL services to allow or deny accesses.

ACL Options

Administrator can enable/disable ACL function per services like SNMP Trap ACL, ICMP ACL, Administrator can configure the ACL options (like Port, Policy, and Level) for each service in **[Management > Access Control List (ACL) > ACL Options]** menu.

ICMP ACL

Administrator can configure policy (drop/allow) for incoming ICMP request by specifying IP Address and Net mask.

Unauthorized SIP ACL

This option is able to protect the fraudulent sip call use of IPX-S300B. By setting it enable, you can prevent a unauthorized SIP call from going through IPX-S300B via SIP trunk or SIP peering. In addition, IPX-S300B blocks the unauthorized IP address, port and protocol for specified period.
The allowed IP lists are the following.

- Endpoint IP
- Registered sip station IP
- Application Server IP
- Miscellaneous ACL IP

SIP Storming

This option is able to protect the SIP burst packets from same IP address. By setting it enable, you can prevent attacks that using abnormal sip packets from same IP address. IPX-S300B block the detected IP address for specified period using decision time and threshold value.

Management Port ACL

This options allows administrator control the default ports of FTP, SSH, Telnet, WEB, and SNMP.
Administrator can configure the trusted IP Address and Net mask. If this option is enabled in ACL option menu, only trusted configured IP Address/Net mask can access this system.

Miscellaneous ACL

This options allows administrator control any protocol (None, TCP, and UDP), source IP Address/Net mask, source Port, destination IP Address/Net Mask, destination port, Policy (drop, allow) and Direction (Inbound, outbound).
Administrator can configure his/her own ACL rule by making combination of those above. It works when the Miscellaneous ACL option is set to 'Enable' in ACL menu like other ACL services.

SNMP Trap ACL

Administrator can configure policy (drop/allow) for incoming SNMP Trap by specifying IP Address, Net mask and Port.

7.2 Process Management

IPX-S300B is a complex ecosystem of many processes (programs). Therefore, such processes are managed by Process Manager, which performs the following functions:

- Start/stop SCM
- Process management
- Start/stop individual processes by administrator request
- View process version information

7.2.1 IPX-S300B Start/Stop

Automatic IPX-S300B Start

After installing IPX-S300B, if you execute the database initialization and IP configuration, the IPX-S300B system automatically restarts. Also when the IPX-S300B server is turned on, IPX-S300B starts automatically.

Manual IPX-S300B Stop/Start

You can manually stop or start IPX-S300B by executing desktop icons on Linux screen. They are ‘Start SCM’ and ‘Stop SCM’ icons.

7.2.2 Process Management

Process Monitoring

IPX-S300B’s Process Manager is constantly monitoring status of all processes. When a process halts, it is automatically restarted.
If the restarted process is terminated abnormally again a number of times, Process Manager will not restart it any more. This is because Process Manager determines that the process will be terminated abnormally again even if it is restarted.

Process Status

A process in IPX-S300B can be in any of the status listed below. NORMAL indicates that the process is normally running.

Status	Description
NORMAL	The process is running normally.
ALIVE	The process is running but unable to exchange IPC messages with other processes.
WAIT	The process is started but has not yet exchanged the initial heartbeat message.
FAIL	The process is not running.

You can view the process status in the **[Performance > System Management > Process Management]** menu.

Process Level (Importance)

IPX-S300B classifies processes into Critical, Major, or Normal levels depending on their effect on the system. The table below shows management policy for each level.

Level	Policy
CRITICAL	The process cannot be terminated by the system administrator.
	If the process is terminated abnormally, it is restarted.
MAJOR	The process can be terminated by the system administrator.
	If the process is terminated abnormally, it is restarted.
NORMAL	The process can be terminated by the system administrator.
	If the process is terminated abnormally, it is restarted.

Individual Process Start/Stop

The SCM Administrator can stop running processes or start stopped processes.
You can stop or start processes in the **[Performance > System Management > Process Management]** menu.

- Click the **[Activate]** button and select a stopped process and click the **[Activate]** button to start it.
- Click the **[Deactivate]** button and select a running process and click the **[Deactivate]** button to stop it.

Viewing Process Version Information

For every process in IPX-S300B there is information on the version, date created and time created.
You can view this information in the **[Performance > System Management > Process Version]** menu.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

7.3 System Operation

7.3.1 Managing Configuration

To view or change the system configuration, log into SCM Administrator and click the **[CONFIGURATION]** icon on the main menu.

The Configuration menu contains various sub-level menu items including Location, User Group, User, Trunk Routing, Cabinet/Slot, Time Schedule, Service, Wireless Enterprise, Application, Phone Setting, Announcement, Network, Miscellaneous, Resource and Gateway.

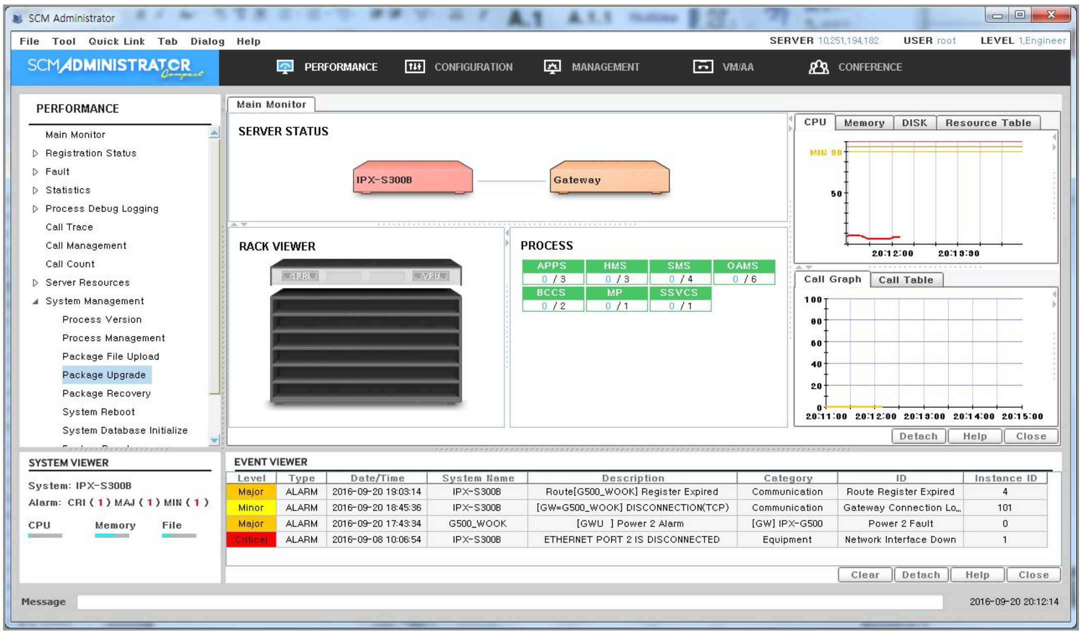


Figure 20. Managing Configuration

7.3.2 Managing Performance

You can check the current IPX-S300B system performance status including CPU, memory, and disk utilization on the right side of the main monitor. You can also view the system and process resource usage activity using the **[PERFORMANCE > Server Resources]** menu.

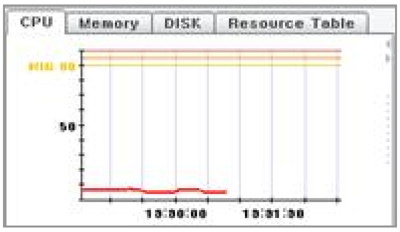


Figure 21. Managing Performance

System Resource Monitoring

IPX-S300B monitors the system resources in five second intervals and displays the information in SCM Administrator. Also, when a specific resource's usage increases, alarms are generated in the order of Minor > Major > Critical to notify the administrator of any system problems.
The system resources monitored by IPX-S300B include CPU, memory, hard disk drives, and network cards.

Viewing System Resource Information

You can view the changes in CPU, memory, and hard disk usage in the **[PERFORMANCE > Main Monitor]** menu. You can also view CPU, memory and hard disk usage in the System Viewer screen in the bottom left corner.
In addition, you can view detailed CPU, memory and hard disk usage as well as network card status in the **[PERFORMANCE > Server Resources > System]** menu.
If a network card stops working, it generates an alarm, which is cleared when the problem is rectified.

Viewing Resources by Processes

You can view memory and CPU usage by processes in the **[PERFORMANCE > Server Resources > Process]** menu.
Also, when a specific process's CPU usage increases, alarms are generated in the order of Minor > Major > Critical to notify the administrator of any system problems.

7.3.3 Software Upgrade

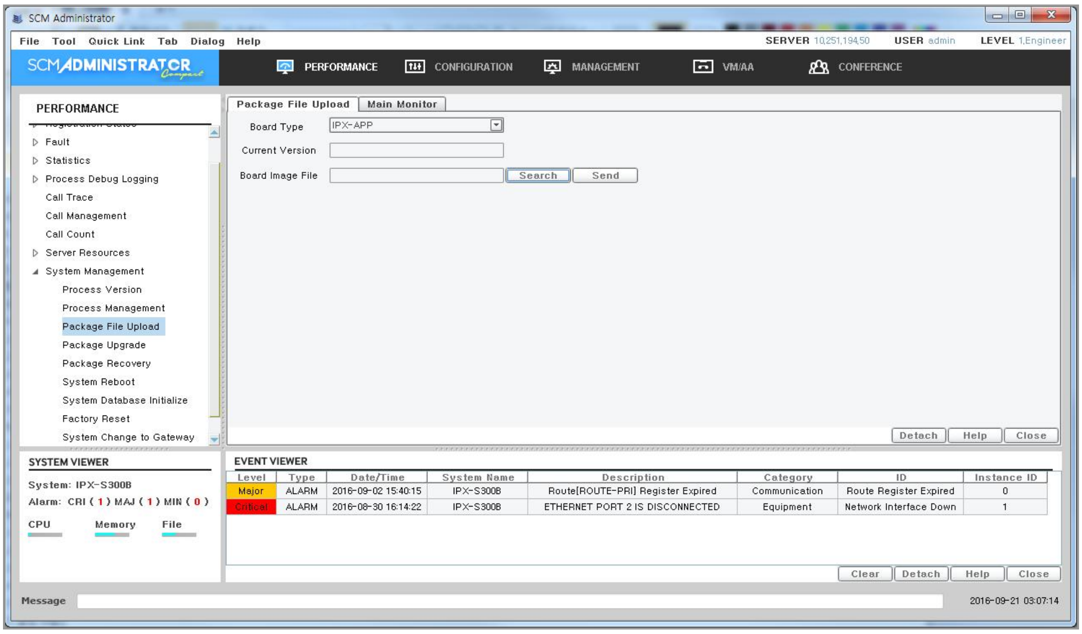
Store a package to upgrade in a specific directory of a PC.
Then, connect to SCM Administrator and perform the package upgrade as follows.

7.3.3.1 Description for Software package

1) ipx-app_x.x.x.x.tgz: Package for IPX-APP (CMU) board
2) ipx-sys_x.x.x.x.tgz: Package for IPX-SYS (CMU Nand and GWU) board
3) ipx-vpu_x.x.x.x.tgz: Package for IPX-VPU (VPU) board
4) ipx-pri_x.x.x.x.tgz: Package for IPX-PRI (1PRI, 2PRI) board
5) ipx-ecu_x.x.x.x.tgz: Package for IPX-ECU (VPU) board

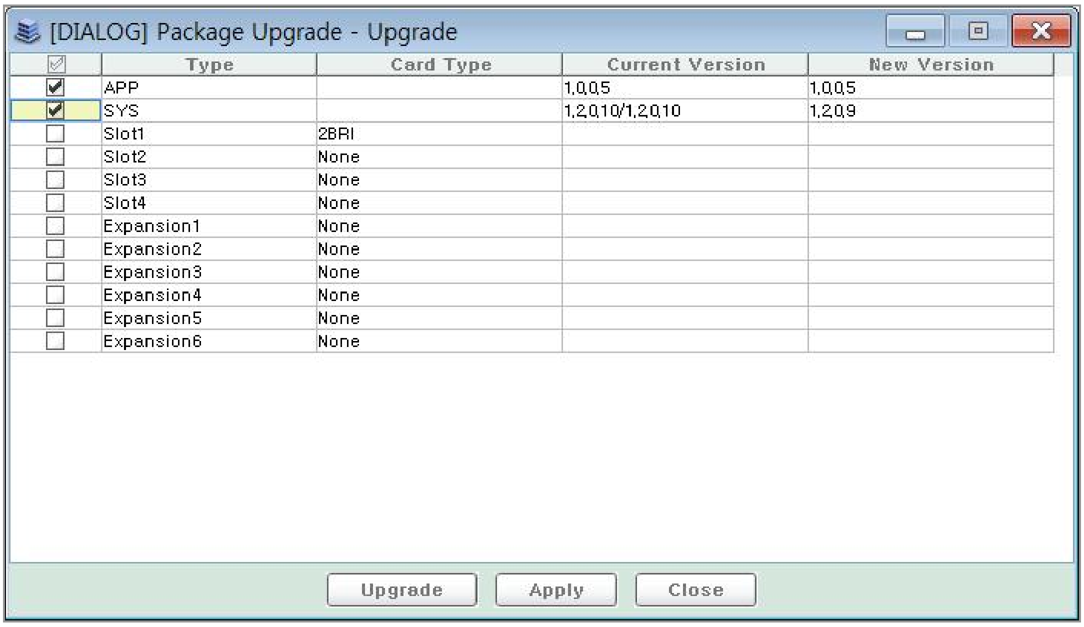
7.3.3.2 File Upload

Upload the software package to update in Statistics/Performance-System Management-Package File Upload.
Push the Find button, move to the fold where the package exists, selects the file according to board type, and push the Send button.



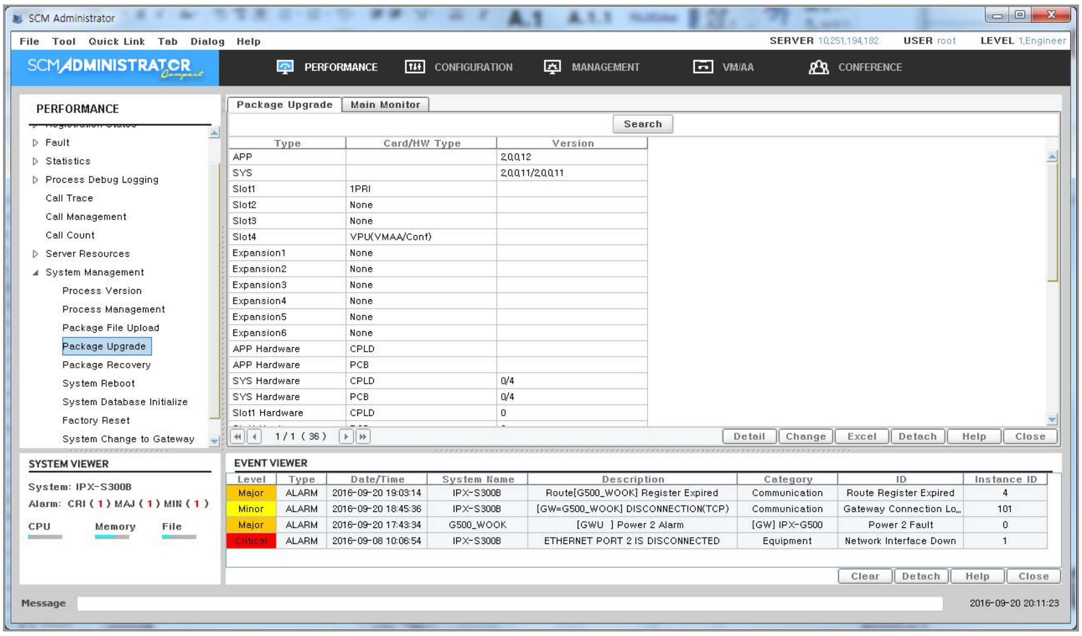
7.3.3.3 Upgrade Execution

- Choose the type to upgrade in Statistics/Performance-System Management-Package File Upload, and then push the Upgrade button and push ‘OK’ button in the popup window which will appear.
- After the upgrade is automatically performed and completed the package whose type has been selected, then reboot.
- When upgrading APP or SYS type packages, connection with SCM Administrator may be disconnected. Then after the upgrade is complete, you can re-connect with SCM Administrator.



7.3.3.4 Checking Version

After the upgrade is completed, you can check the versions according to type in Statistics/Performance-System Management-Package File Upload.



7.3.4 Managing Announcement

IPX-S300B's built-in sound source system can play voice announcements and system tones when necessary.

Error Announcements

This service plays voice announcements when calls are not processed normally due to errors, etc.

Click the **[CONFIGURATION > Announcement > Release Announcement]** menu to view the list of announcements.

Select an announcement and click the **[Play]** button to have the selected announcement play through the PC's sound device.

Select an announcement and click the **[Change]** button to change some of the information.

Level	Policy
Announcement ID	This Announcement ID is used for identifying the announcement sound source.
Description	This describes the condition for which the announcement is played. This can be changed.
Announcement	Specify whether to play the announcement for the given condition.
Interval (100 ms)	Specify the interval between announcements if the selected announcement is

Level	Policy
	played multiple times.
Repeat time	The announcement can be played multiple times. Specify the number of times to play.
File Name	This is the name of the announcement sound source file.

Service Announcements

Voice announcements can be played when using call processing services.
Click the **[CONFIGURATION > Announcement > Service Announcement]** menu to view the list of announcements.
Select an announcement and click the **[Play]** button to have the selected announcement play through the PC's sound device.
Select an announcement and click the **[Change]** button to change some of the information.

Level	Policy
ID	This ID is used for identifying the announcement sound source.
Description	This describes the condition for which the announcement is played. This can be changed.
File Name	This is the name of the announcement sound source file. Since this cannot be changed, make sure to upload a sound source file with the same name.

Music On Hold (MOH)

You can manage the system tones for the music and tones played when calls are put on hold or forwarded. You can also register different sound sources required for the site and service them.
Click the **[CONFIGURATION > Announcement > MOH]** menu to view the list of MOH.
Select a MOH and click the **[Play]** button to have the selected MOH play through the PC's sound device.
Select a MOH and click the **[Change]** button to change some of the information.

Level	Policy
ID	This ID is used for identifying the MOH sound source.
Description	This describes the MOH. This can be changed.
File Name	This is the name of the MOH sound source file. Since this cannot be changed, make sure to upload a sound source file with the same name.

Language Settings

IPX-S300B supports announcements in multiple languages. However, due to the complexity of settings to configure different conditions for different languages, only one language is serviced at a time.

The current language can be selected in the **[CONFIGURATION > Announcement > Language]** menu.

Name	Description
Announcement Language	This Announcement Language is used for this User Group - Korea-Korean - English-America - English-British - German - Turkish - Italian - Russian

7.3.5 Managing Individual Call

IPX-S300B provides a feature for viewing the information of currently processed calls.
It also allows the administrator to terminate currently processed calls by different criteria such as unusually long calls or illegitimate calls. IPX-S300B also provides the trace feature which allows the administrator to trace calls or protocol messages.

7.3.5.1 Call Management

You can view the currently processed calls in the **[PERFORMANCE > Call Management]** menu. Click the **[Search]** button to view the list of currently processed calls.
You can filter the call list displayed by entering advanced conditions such as caller numbers, called party numbers and call durations.
Select a call from the list and click the **[Delete]** button to terminate the selected call.

7.3.5.2 Signaling Trace

IPX-S300B supports a protocol tracing feature (SIP signaling trace) for calls.
Create a protocol trace item in the **[PERFORMANCE > Call Trace]** menu to view protocol messages by call stages in Job Monitor.
Following three types of protocol tracing is supported based on the call type.

Call Trace

All messages are traced per call basis. All messages for the calls from initialization to termination are traced.
It can be filtered by Calling Number or Called Number.
When the Real Time option is set to ‘Real Time’, the messages will be displayed at the monitor windows in real time. ‘Call Base’ option displays the messages after termination of the calls.

Trunk Call Trace

All messages are traced per Route basis. The messages for the specific Route from initialization to termination are traced.
It can be filtered by User Group, Route Name and/or Direction.

IP Address Call Trace

All messages are traced per IP Address basis. All messages for the specific IP Address from initialization to termination are traced.
It can be filtered by IP Address.

7.3.6 Managing Database

IPX-S300B provides a feature for backing up the database during operation.
When upgrading IPX-S300B to a newer version, you can back up the database, upgrade the version, and then restore the database for use.

Database Usage

IPX-S300B provides a feature for displaying the current size of the database.
The maximum allowed database size, the current database size, and the free database size are shown in KB and percentage.
You can view the database usage in the **[MANAGEMENT > Database > Database Space]** menu.

Database Backup

You can configure periodical database backup to backup the database periodically or perform an immediate database backup.
IPX-S300B also can send backuped-file to FTP server.
You can back up the database in the **[MANAGEMENT > Database > Database Backup]** menu.

Level	Policy
Back-up Type	Specify the method for backing up the database. - DAY: Back up at a specified time every day. - WEEK: Back up at a specified time on a specified day of the week. - 15DAYS: Back up at a specified time every 15 days. - MONTH: Back up at a specified time on the first day of every month. - NOW: Back up now.
Weekday	This is enabled when the backup type is Week. Specify the day of the week to perform backup.
Back-up Time	Specify the time to perform backup on the selected day.
Max Current Version File Count	Select the max backup File count for same version
VM Data Backup	Choose whether to save the VM Data
Send To Backup Server (FTP)	Choose whether to send backup File to FTP Server
Server Directory	Directory Path of FTP Server
Server IP Address	IP Address of FTP Server
Server Port	Port Number of FTP Server
Login ID	Login ID of FTP Server
Login Password	Login Password of FTP Server
Secure Mode	Secure Mode (sFTP) of FTP Server

Backup File Management

IPX-S300B provides a feature for managing the database backup file. you can download the backup file, upload the backup file, restore the backupfile and delete the backup file. You can manage the backup file in the **[MANAGEMENT > Database > Backup File Management]** menu.
Restriction: version of backup file and version of IPX-S300B must be same.

Name	Description
Download	Backup file in system is downloaded to your PC
Upload	Backup File in your PC is uploaded to system.
Restore	system restore database as the backup file selected.
Delete	Backup File in system is deleted.

Database Initialize

IPX-S300B provides a feature for initializing the database of system.
IP address of system remains.
You can initialize the database in the **[MANAGEMENT > Database > Database Initialize]** menu.

7.3.7 Managing Individual User

IPX-S300B provides a feature for managing IPX-S300B users including phones, endpoints, gateways, and applications.

Extension User Management

IPX-S300B manages extension users by adding, deleting or changing extension users in the database. Extension user information can be either device information with physical properties or user information with logical properties.
For more information on extension user management, see the ‘2.4. Adding Individual User’ section.

Trunk User Management

IPX-S300B manages trunk users by adding, deleting or changing trunk users in the database. Trunk user information can be either endpoint information with physical properties or route information with logical properties.
For more information on trunk user management, see the ‘2.5. Adding Individual Trunk’ and ‘4.1.3 Least Cost Route (LCR)’.

Registration Management

IPX-S300B manages the registration status of the users-including phones, endpoints, gateways, and applications-which provide services by performing SIP registration with IPX-S300B and displays their current status.
For more information on registration management, see the ‘6.8 Registration Management’.

Authentication Management

IPX-S300B can authenticate registration of the users-including phones, endpoints, gateways, and applications-which provide services by performing SIP registration with IPX-S300B. Also, when an extension subscriber makes a call, IPX-S300B provides a service for allowing the call to be made after obtaining an external server’s authentication.
For more information on authentication management, see the ‘4.1.37 User Authentication’.

Security Management

IPX-S300B can provide the feature for encrypting signaling and voice data for calls.
For more information on registration management, see the ‘4.1.35 VoIP Security’.

Service Allowance Management

IPX-S300B can allow each individual extension user to use different sets of features by assigning them to service classes.
For more information on service allowance management, see the ‘4.1.36 Feature Service’.

7.3.8 Managing Access Permission

IPX-S300B provides a feature for managing SCM Administrator.

Account List

IPX-S300B manages account list of SCM Administrator. Each account is different in grade due to account class.

Account Class

SCM Administrator basically has three account classes, Engineer, Technician and Customer. New account classes can be created by Engineer user. When creating a new account class, you can permit or restrict View, Create, Change and Delete operation by menu.
View permission limit menu tree and Create, Change and Delete permission activate or deactivate operation buttons. And you can limit User Group.

Administrator Control

You can allow or drop connection from a specific IP range for Engineer users. Basically all Engineer users are allowed.

Terminal Control

You can allow connection from a specific IP range for non-Engineer users. Basically all non-Engineer users are not allowed.

Login Status

It displays account list in log-in state. Engineer user can force non-Engineer users to log-out.

Operating History

IPX-S300B provides management history information for the administrator.
You can be able to account for each and see the details and content of the summary and classification of the command that was run at a specific time, to confirm the history of login/logout of each account.

Login History

IPX-S300B provides a history of information sessions for each administrator account.
Login time, logout time, can check the status and current IP of the terminal is connected.

7.3.9 Managing Maximum Calls

On an IP-based PBX, it is not possible to limit the number of phones physically connected or the number of calls made simultaneously. However, since system resources are limited, a service is required to limit the maximum number of calls at any one time.
IPX-S300B provides the Call Admission Control (CAC) feature which limits the maximum number of calls allowed. The CAC service provided by IPX-S300B includes CAC by call counts, CAC by location bandwidth, and CAC by system resources.
For more information on CAC, see the ‘5.1.2 Call Admission Control’.

7.4 Call Detail Records (CDR)

Account information includes Call Detail Records (CDR) and Station Message Detail Records (SMDR). Whenever a call starts or ends, IPX-S300B records the call information according to the account data recording method defined for each user group.
Account data can be recorded by Local, FTP, RADIUS, TCP or TCP_SMDR. Names of the files saved and their directory names are determined by the recording method used.
You can configure the account data recording method for each user group in 'CDR Storage Options' under the **[CONFIGURATION > User Group >Change User Group > Information]** menu.

- None: No CDR data is generated.
- Local: CDR data is recorded in the IPX-S300B hard disk.
- File Transfer Protocol (FTP): CDR data is saved as files in IPX-S300B and transferred to a specified FTP server at a specified interval.
- Transmission Control Protocol (TCP): Each time the CDR data is generated, it is transferred to the CDR server connected by TCP.

You must configure the required settings for each method of saving CDR files generated.

7.4.1 Saving Account Information in IPX-S300B

The CDR files generated are backed up and saved in the IPX-S300B hard disk without interoperating with any external account systems.
When saving CDR files in IPX-S300B, you can configure the required information in the **[MANAGEMENT > CDR Storage Options > Local Store]** menu.

Name	Description
CDR Local Backup Interval	When configuring CDR files to be backed up in the hard disk, specify the backup interval. All CDR files generated are moved to the backup directory and the files in the local directory are deleted at this interval. Only the CDR files not saved in the backup directory will be left in the local directory.
CDR Local Backup Lifetime	When configuring CDR files to be backed up in the hard disk, specify the number of days for which the backed up CDR files will be kept in the hard disk. At midnight every day, the system automatically deletes any backed up CDR files which have passed the specified date.
CDR Local Backup Used	Specify whether to back up the generated CDR files in the hard disk. If enabled, the files are backed up in the /DI/CM/data/cdr/local/Backup directory.
CDR Local Create Interval	Specify the interval in minutes at which the CDR data files will be generated. New CDR files are generated at this interval. No CDR file will be generated if there is no CDR information for this period. The CDR files generated are saved in the /DI/CM/data/cdr/local directory.

7.4.2 FTP Interoperation for Accounting System

The CDR files generated are transferred to the external accounting system interoperating by FTP protocol.
When interoperating with the external accounting system over FTP, you can configure the required information in the **[MANAGEMENT > CDR Storage Options > FTP Send]** menu.

Name	Description
CDR FTP Backup Lifetime	When interoperating with the accounting system over FTP, the CDR files generated can be backed up in IPX-S300B even after they have been transferred by FTP. Specify the number of days for which the backed up CDR files will be kept. At midnight every day, the system automatically deletes any backed up CDR files which have passed the specified date.
CDR FTP Backup Used	When interoperating with the accounting system over FTP, the CDR files generated can be backed up in IPX-S300B even after they have been transferred by FTP. If enabled, the CDR files are backed up in the /DI/CM/data/cdr/ftp/Backup directory in IPX-S300B.
CDR FTP Create Interval	When interoperating with the accounting system over FTP, specify the interval at which the CDR files are generated. New CDR files are generated at this interval. No CDR file will be generated if there is no CDR information for this period. The CDR files generated are saved in the /DI/CM/data/cdr/ftp directory.
CDR FTP Link 1/2/3/4 Directory	Specify the name of the directory in the FTP server where the files will be saved when transferring CDR files over FTP.
CDR FTP Link 1/2/3/4 IP	Specify the IP address of the FTP server when transferring CDR files over FTP.
CDR FTP Link 1/2/3/4 Login Name	Specify the login name of the FTP server when transferring CDR files over FTP.
CDR FTP Link 1/2/3/4 Password	Specify the password of the FTP server when transferring CDR files over FTP.
CDR FTP Link 1/2/3/4 Port	Specify the port number of the FTP server when transferring CDR files over FTP.
CDR FTP Link 1/2/3/4 Secure	Specify whether to use Secure-FTP when transferring CDR files over FTP.
CDR FTP Link 1/2/3/4 Used	Specify whether to use FTP Server Link.
CDR FTP Transfer Interval	When interoperating with the accounting system over FTP, specify the interval (in minutes) at which the CDR files are transferred. All CDR files generated are transferred over FTP and the successfully transferred files are moved to the backup directory at this interval. Only the CDR files not transferred over FTP will be left in the local directory.

7.4.3 TCP Interoperation for Accounting System

IPX-S300B interoperates with the external accounting system over a native TCP method. Whenever CDR data is generated, the CDR data is transferred to the TCP server. CDR files are also backed up in IPX-S300B.

When interoperating with the external accounting system over TCP, you can configure the required information in the **[MANAGEMENT > CDR Storage Options > TCP Send]** menu.

Name	Description
CDR TCP Backup Lifetime (day)	When interoperating with the accounting system over TCP and backing up CDR files in IPX-S300B, the number of days for which the backed up CDR files will be kept in IPX-S300B. At midnight every day, the system automatically deletes any backed up CDR files which have passed the specified date.
CDR TCP Create Interval (minute)	When interoperating with the accounting system over TCP and backing up CDR files in IPX-S300B, specify the interval (in minutes) at which the CDR files to be backed up are generated. New CDR files are generated at this interval. No CDR file will be generated if there is no CDR information for this period. The CDR files generated are saved in the /DI/CM/data/cdr/tcp directory.
CDR TCP Link1 IP	Specify the IP address of the first of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link1 Used	Specify whether to transfer the CDR data to the first of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link2 IP	Specify the IP address of the second of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link2 Used	Specify whether to transfer the CDR data to the second of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link3 IP	Specify the IP address of the third of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link3 Used	Specify whether to transfer the CDR data to the third of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link4 IP	Specify the IP address of the fourth of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link4 Used	Specify whether to transfer the CDR data to the fourth of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.

7.4.4 Billing Output by Call Types

This is a function that the CDR data is created by call types. To use this function, you should set the item whose 'use billing output' is 'ENABLE' in the **[MANAGEMENT > CDR Storage Options > CDR Option]** menu. And you should set items in the **[MANAGEMENT > CDR Storage Options > Set Billing Output]** menu.

Name	Description
App Server Call Billing Output [App Server]	The CDR is created if Calling type (4) is application and Called Type is application (4)
App Server Call Billing Output [Service]	The CDR is created if Calling Type (4) is Application and Called Type is Service (2)
App Server Call Billing Output [Subscriber]	The CDR is created if Calling Type (4) is Application and Called Type is Subscriber (1)
All Call Billing Output Set	The CDR is created about all calls
Incoming Call Billing Output [Normal]	The CDR is created if the trunk is a normal type and call is a outgoing call
Incoming Call Billing Output [TIE]	The CDR is created if the trunk is a TIE type and is call is a outgoing call
Outgoing Call Billing Output [Normal]	The CDR is created if the trunk is a normal type and call is a incoming call
Outgoing Call Billing Output [TIE]	The CDR is created if the trunk is a TIE type and call is a incoming call
Service Call Billing Output [App Server]	The CDR is created if Calling Type is Service (2) and Called Type is application (4)
Service Call Billing Output [Service]	The CDR is created if Calling Type is Service (2) and Called Type is Service (2)
Service Call Billing Output [Subscriber]	The CDR is created if Calling Type is Service (2) and Called Type is Subscriber (1)
Subscriber Call Billing Output [App Server]	The CDR is created if Calling Type is Service (1) and Called Type is application (4)
Subscriber Call Billing Output [Service]	The CDR is created if Calling Type is Service (1) and Called Type is Service (2)
Subscriber Call Billing Output [Subscriber]	The CDR is created if Calling Type is Subscriber (1) and Called Type is Subscriber (1)

7.4.5 Billing Delete Length

This is a function that deletes access code of trunk in the 'connect number' of CDR.
To use this function, you should set 'Length of Bill Delete' in the **[Configuration > Trunk Routing > Route]** menu

7.5 Statistics Reports

IPX-S300B provides statistical information for calls and alarms generated in the system by hours, dates and months.
You can view the statistical information in the **[PERFORMANCE > Statistics]** menu.
The statistical information is kept in the database for the duration specified by 'Statistic DB Keep Up Lifetime' in the **[CONFIGURATION > Miscellaneous > System Options]** menu.
The duration for which to query the statistical information must be entered with following conditions in the **[PERFORMANCE > Statistics]** menu. Hourly statistics cannot exceed 7 days, daily statistics cannot exceed 90 days, and monthly statistics cannot exceed 365 days. Also, hourly statistics older than 30 days or daily statistics older than 365 days cannot be queried.

7.5.1 Call Traffic Reports

IPX-S300B provides statistics on Inbound calls and Outbound calls for individual Users. This feature is not supported by IPX-S300B Enterprise.
You can view statistics on Inbound calls and Outbound calls for individual Users in the **[PERFORMANCE > Statistics > User > User-Outgoing]** menu and in the **[PERFORMANCE > Statistics > User > User-Incoming]** menu. Only hourly statistics is available for Inbound calls and Outbound calls for individual Users.

7.5.1.1 System Statistics

Using the **[PERFORMANCE > Statistics > System]** menu, you can view the following statistical information by menu items:

Internal Calls

This shows the statistical information collected when calls were attempted between internal Users.

Item	Description
Date	The start time of statistics measurement
Total	Total number of attempted calls.
Answer	Number of answered calls
Answer %	Success ratio of calls
Management Release	Number of calls released by an operator command
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as

Item	Description
	route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy)
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

Outbound Calls

This shows the statistical information collected when internal Users attempted to call external Users.

Item	Description
Date	The start time of statistics measurement
Total	Total number of attempted calls.
Answer	Number of answered calls
Answer %	Success ratio of calls
Management Release	Number of calls released by an operator command
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times

Item	Description
	of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

Inbound Calls

This shows the statistical information collected when external Users attempted to call internal Users.

Item	Description
Date	The start time of statistics measurement
Total	Total number of attempted calls.
Answer	Number of answered calls
Answer %	Success ratio of calls
Management Release	Number of calls released by an operator command
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).

Item	Description
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

Tandem Calls

This shows the statistical information collected when external Users attempted to call other external Users through IPX-S300B.

Item	Description
Date	The start time of statistics measurement
Total	Total number of attempted calls.
Answer	Number of answered calls
Answer %	Success ratio of calls
Management Release	Number of calls released by an operator command
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times

Item	Description
	of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

Call Failures

This shows the statistical information for failed Inbound and Outbound authentications.

Item	Description
Date	The start time of statistics measurement
Outgoing-Unknown	Number of calls which an extension line User tries to an unknown number
Inbound-Unknown	Number of calls which an external User tries to an unknown number
Dialing-Unknown	Number of calls which are attempted to an unknown number
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.

All Calls

This shows the statistical information for all calls.

Item	Description
Date	The start time of statistics measurement
Total	Total number of attempted calls.
Internal Calls	Number of calls between extension line Users
Internal Call (%)	Percentage of calls between extension line Users
Outbound Calls	Number of calls which an extension line User tries to an external User
Outbound Calls (%)	Percentage of calls which an extension line User tries to an external User

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Inbound Calls	Number of calls which an external User tries to an extension line User
Inbound Calls (%)	Percentage of calls which an external User tries to an extension line User
Tandem Calls	Number of calls which an external User tries to an external User
Tandem Calls (%)	Percentage of calls which an external User tries to an extension line User
Outgoing-Unknown	Number of calls which an extension line User tries to an unknown number
Outgoing-Unknown (%)	Percentage of calls which an extension line User tries to an unknown number
Inbound-Unknown	Number of calls which an external User tries to an unknown number
Inbound-Unknown (%)	Number of calls which an external User tries to an unknown number
Dialing-Unknown	Number of calls which are attempted to an unknown number
Dialing-Unknown (%)	Number of calls which are attempted to an unknown number

Failure Reason

This shows the statistical information for all failed calls.

Item	Description
Date	The start time of statistics measurement
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons:

Item	Description
	That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.

7.5.1.2 User Group Statistics

Using the **[PERFORMANCE > Statistics > UserGroup]** menu, you can view the following statistical information by menu items:

User Group-Outgoing

This shows the statistical information collected when Users of the user group attempted to make Outbound calls.

Item	Description
Date	The start time of statistics measurement
User Group	Name of user group for which statistics is measured
Total	Total number of attempted calls
Total Internal	Number of calls which are attempted to an extension number
Internal Answer	Number of calls which receive an answer from an extension number
Total Outbound	Total number of Outbound calls

Item	Description
Outbound Answer	Number of Outbound calls which receive an answer
Answer %	Success ratio of calls
Management Release	Number of calls which are released by an operator command
Failure	Number of calls which are failed
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.

Item	Description
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

User Group-Incoming

This shows the statistical information collected when Inbound calls were attempted for Users of the user group.

Item	Description
Date	The start time of statistics measurement
User Group	Name of user group for which statistics is measured
Total	Total number of attempted calls
Total Inbound	Number of calls which an external User tries to an extension line User
Inbound Answer	Number of times when an extension line User answers a call which an external User tries to the extension line User
Total Internal	Number of calls which an extension line tries to an extension line User
Internal Answer	Number of times when an extension line User answers a call which an extension line User tries to an extension line User
Answer %	Success ratio of calls
Management Release	Number of calls which are released by an operator command
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
	even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

7.5.1.3 Service Group Statistics

Using the **[PERFORMANCE > Statistics > ServiceGroup]** menu, you can view the following statistical information by menu items:

Service Group-Outgoing

This shows the statistical information collected when Users of the service group attempted to make Outbound calls.

Item	Description
Date	The start time of statistics measurement
Service Group	Name of service group for which statistics is measured
User Group	Name of a user group where a service group is belonged
Total	Total number of attempted calls
Total Internal	Number of calls which are attempted to an extension number
Internal Answer	Number of calls which receive an answer from an extension number
Total Outbound	Total number of Outbound calls
Outbound Answer	Number of Outbound calls which receive an answer
Answer %	Success ratio of calls
Management Release	Number of calls which are released by an operator command
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons:

Item	Description
	That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

Service Group-Incoming

This shows the statistical information collected when Inbound calls were attempted for Users of the service group.

Item	Description
Date	The start time of statistics measurement
Service Group	Name of service group for which statistics is measured
User Group	Name of a user group
Total	Total number of attempted calls
Total Inbound	Number of calls which an external User tries to an extension line User
Inbound Answer	Number of times when an extension line User answers a call which an external User tries to the extension line User
Total Internal	Number of calls which are received to an extension number
Internal Answer	Number of calls answered from an extension number

Item	Description
Answer %	Success ratio of calls
Management Release	Number of calls which are released by an operator command
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not

Item	Description
	support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Management Release	Number of calls which are failed because there is no route or wrong number format.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

7.5.1.4 Route Statistics

Using the **[PERFORMANCE > Statistics > Route]** menu, you can view the following statistical information by menu items:

Route-Inbound

This shows the statistical information collected when Inbound calls were attempted for internal Users through the route.

Item	Description
Date	The start time of statistics measurement
Route	Route name for which statistics is measured
User Group	User group where a route is belonged
Total	Total number of call tries
Answer	Number of answers
Answer %	Success ratio of calls
Management Release	Number of call releases by an operator
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)

Item	Description
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

Route-Outbound

This shows the statistical information collected when internal Users attempted to call external Users through the route.

Item	Description
Date	The start time of statistics measurement
Route	Route name for which statistics is measured
User Group	Name of a user group where a route is belonged
Total	Total number of call tries
Answer	Number of answers
Answer %	Success ratio of calls
Management Release	Number of call releases by an operator
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Route Unavailable	Number of times when a trunk cannot be used because of the following reasons:

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
	- End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

7.5.1.5 Hunt Group Statistics

Using the **[PERFORMANCE > Statistics > HuntGroup]** menu, you can view the following statistical information by menu items:

Hunt Group-Incoming

This shows the statistical information collected when Inbound calls were attempted for Users of the hunt group.

Item	Description
Date	The start time of statistics measurement
Hunt Group	Name of hunt group for which statistics is measured
User Group	Name of a user group
Total	Total number of attempted calls
Total Inbound	Number of calls which an external User tries to an extension line User
Inbound Answer	Number of times when an extension line User answers a call which an external User tries to the extension line User
Total Internal	Number of calls which are attempted to an extension number
Internal Answer	Number of calls which receive an answer from an extension number
Answer %	Success ratio of calls
Management Release	Number of calls which are released by an operator command
Failure	Number of failed calls

Item	Description
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.

Item	Description
Etc	Number of times when a call is failed because of other reasons.
Management Release	Number of calls which are failed because there is no route or wrong number format.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

7.5.1.6 User Statistics

Using the **[PERFORMANCE > Statistics > User]** menu, you can view the following statistical information by menu items:
IPX-S300B package provides User-outgoing and User-Incoming statistics but IPX-S300B Enterprise package does not.

User-Outgoing

This shows the statistical information collected when the User attempted to make an outbound calls.

Item	Description
Date	The start time of statistics measurement
Subscriber	User's extension number for which statistics is measured
Service Group	Name of a service group where a user is belonged
User Group	Name of a user group where a service group is belonged
Total	Total number of attempted calls
Total Internal	Number of calls which are attempted to an extension number
Internal Answer	Number of calls which receive an answer from an extension number
Total Outbound	Total number of Outbound calls
Outbound Answer	Number of Outbound calls which receive an answer
Answer %	Success ratio of calls
Management Release	Number of calls which are released by an operator command
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number

Item	Description
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

User-Incoming

This shows the statistical information collected when Inbound calls were attempted for the User.

Item	Description
Date	The start time of statistics measurement
Subscriber	User's extension number for which statistics is measured
Service Group	Name of a service group where a user is belonged
User Group	Name of a user group
Total	Total number of attempted calls
Total Inbound	Number of calls which an external User tries to an extension line User
Inbound Answer	Number of times when an extension line User answers a call which an external User tries to the extension line User
Total Internal	Number of calls which are attempted to an extension number
Internal Answer	Number of calls which receive an answer from an extension number
Answer %	Success ratio of calls
Management Release	Number of calls which are released by an operator command
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service

Item	Description
	configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Route Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

7.5.1.7 Statistics Group

To use the statistics group, you need the following items are set.

Using the **[CONFIGURATION > Service > Group Service > Statistics Group]** menu, you need to create a statistics group, set the following data.

Item	Description
User Group	Select the user group
Group Name	Input the name of statistics group
Group Type	Select the group type **[Extension/Route]** of statistics group

Using the **[CONFIGURATION > Service > Group Service > Extension Statistics List]** menu, you need to assign statistics group to extension, set the following data.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Subscriber Statistics Include	Select whether to display the statistics in **[User Statistics]** menus
Statistics Group Name	Assign statistics group to extension

Using the **[CONFIGURATION > Service > Group Service > Route Statistics List]** menu, you need to assign statistics group to route, set the following data.

Item	Description
Statistics Group Name	Assign statistics group to route

7.5.1.8 Extension Statistics Group

Using the **[PERFORMANCE > Statistics > Extension Statistics Group]** menu, you can view the following statistical information by menu items:

Extension Statistics Groups-Outgoing

This shows the statistical information collected when the user in extension statistics group attempted to make an outbound calls.

Item	Description
Date	The start time of statistics measurement
Statistics Group	Name of a statistics group where a user is belonged
Service Group	Name of a service group where a user is belonged
User Group	Name of a user group where a service group is belonged
Total	Total number of attempted calls
Total Internal	Number of calls which are attempted to an extension number
Internal Answer	Number of calls which receive an answer from an extension number
Total Outbound	Total number of Outbound calls
Outbound Answer	Number of Outbound calls which receive an answer
Answer %	Success ratio of calls
Management Release	Number of calls which are released by an operator command
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
	- Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Extension Statistics Groups-Incoming

This shows the statistical information collected when Inbound calls were attempted for the user in extension statistics group.

Item	Description
Date	The start time of statistics measurement
Statistics Group	Name of a statistics group where a user is belonged
Service Group	Name of a service group where a user is belonged
User Group	Name of a user group
Total	Total number of attempted calls
Total Inbound	Number of calls which an external User tries to an extension line User
Inbound Answer	Number of times when an extension line User answers a call which an external User tries to the extension line User
Total Internal	Number of calls which are attempted to an extension number
Internal Answer	Number of calls which receive an answer from an extension number
Answer %	Success ratio of calls
Management Release	Number of calls which are released by an operator command
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service

Item	Description
	configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Route Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

7.5.1.9 Route Statistics Groups

Using the **[PERFORMANCE > Statistics > Route Statistics Groups]** menu, you can view the following statistical information by menu items:

Route Statistics Groups-Inbound

This shows the statistical information collected when Inbound calls were attempted for internal Users through the route in route statistics group.

Item	Description
Date	The start time of statistics measurement
Statistics Group	Name of a statistics group where a user is belonged
User Group	User group where a route is belonged
Total	Total number of call tries
Answer	Number of answers

Item	Description
Answer %	Success ratio of calls
Management Release	Number of call releases by an operator
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.

Item	Description
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

Route Statistics Groups-Outbound

This shows the statistical information collected when internal Users attempted to call external Users through the route in route statistics group.

Item	Description
Date	The start time of statistics measurement
Statistics Group	Name of a statistics group where a user is belonged
User Group	Name of a user group where a route is belonged
Total	Total number of call tries
Answer	Number of answers
Answer %	Success ratio of calls
Management Release	Number of call releases by an operator
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction

Item	Description
	by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Route Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

7.5.1.10 System Service Statistics

Using the **[PERFORMANCE > Statistics > System Service]** menu, you can view the following statistical information by menu items:

Mobile Remote Dial

This shows the statistical information collected when the System attempted to make an mobile remote dial.

Item	Description
Date	The start time of statistics measurement
Total	Total number of attempted calls

Item	Description
Success	Number of success calls
Success %	Success ratio of calls
User Release	Number of calls which are released by an user
Failure	Number of failed calls
Wrong Number	Number of calls which are failed because there is no route or wrong number format.
Resource Unavailable	Number of calls which are failed because of the following reasons: - Failed because it cannot find a route during routing process such as route partition, special route sequence, or route sequence after number interpretation - Failed because of restriction configuration - Failed because too much resource is required
Wrong Prefix	Number of calls which are failed because of wrong prefix of a called number
User Error	Number of calls which are failed because they do not receive a call signal (Setup Timeout)
Called Number Busy	Number of calls which are failed because a counterpart is busy (Busy).
No Answer	Number of calls which are failed because a counterpart does not answer even though there is a ring (No Answer)
Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout
Trunk Unavailable	Number of times when a trunk cannot be used because of the following reasons: - End Point Not Registration - Number of times when all the trunks are busy - All the trunks are QoS and failed for a long time even with many times of retries - When the connection between a gateway and TCP is lost
User Unavailable	Number of times when there is no User.
Service Deny	Number of times when a call is released because the ITSP Gatekeeper disconnects the connection to a gateway (e.g. deleting

Item	Description
	the endpoint of a gateway) during call.
Signaling Fault	Number of times when a call is released because a phone does not support a specific function. - A call is released because there is no Empty TCS function. - A call is released because there is no function which processes the request_mode of an endpoint during fax operation.
Etc	Number of times when a call is failed because of other reasons.
Total Talk Time	Total talk time.
Average Talk Time	Average talk time.

Smart Routing

This shows the statistical information collected when the System attempted to make all smart routing.

Item	Description
Date	The start time of statistics measurement
Total	Total number of attempted routings
Success	Number of success routings
Success %	Success ratio of routings
Failure	Number of failed routing
Smart Routing Call Reject	When smart routing is rejected
User Unregister	When unregistered
User No Response	When does not answer

Smart Routing (Internal)

This shows the statistical information collected when the System attempted to make internal smart routing.

Item	Description
Date	The start time of statistics measurement
Total	Total number of attempted routings
Success	Number of success routings
Success %	Success ratio of routings
Failure	Number of failed routing
Smart Routing Call Reject	When smart routing is rejected
User Unregister	When unregistered
User No Response	When does not answer

Smart Handover

This shows the statistical information collected when the System attempted to make smart handover.

Item		Description
Date		The start time of statistics measurement
Handover-out	Total	Number of attempted handover-out
	Success	Number of success handover-out
	Success %	Success ratio of handover-out
	Failure	Number of failed handover-out
	Invalid Call	When handover-out is unavailable
	Route Unavailable	When routing for handover-out is unavailable
	User Cancel	When user canceled handover-out
Handover-in	Total	Number of attempted handover-in
	Success	Number of success handover-in
	Success %	Success ratio of handover-in
	Failure	Number of failed handover-in
	Invalid Call	When handover-in is unavailable
	Service Unavailable	When service type is not Handover Both
	User Cancel	When user canceled handover-in

Mobile Call

This shows the statistical information collected when Samsung mobile phone attempted the make calls.

Item		Description
Date		The start time of statistics measurement
Internal Wifi	Total	Total number of attempted calls for Internal wifi network
	Success	Number of success calls for Internal wifi network
	Failure	Number of failed calls for Internal wifi network
External Wifi	Total	Total number of attempted calls for External wifi network
	Success	Number of success calls for External wifi network
	Failure	Number of failed calls for External wifi network
Mobile	Total	Total number of attempted calls for mobile network

Item		Description
	Success	Number of success calls for mobile network
	Failure	Number of failed calls for mobile network
Unknown	Total	Total number of attempted calls for Unknown network
	Success	Number of success calls for Unknown network
	Failure	Number of failed calls for Unknown network
Internal Wifi	Success %	Success ratio of calls for Internal wifi network
External Wifi	Success %	Success ratio of calls for External wifi network
Mobile	Success %	Success ratio of calls for mobile network
Unknown	Success %	Success ratio of calls for Unknown network
Internal Wifi External Wifi Mobile Unknown	User Invalid	Number of calls which are failed because of the following reasons: - No Route or Wrong Number Format - Wrong prefix of a called number - Do not receive a call signal (Setup Timeout) - Called Number Busy - No Answer
	Authentication Release	Number of calls which are failed because a call is not allowed due to Outbound blocking, Inbound blocking, calling CRU (Call Restriction by User), CRO (Call Restriction by Operator), authentication failure, authentication time expiration, or Call Screening, etc.
	Service Release	Number of calls which are failed because of the following reasons: That is, number of calls which are failed because of service configuration. - Service Release Call DND - Service Release Call ABS - Service Release Call (Service Error) - Service privilege failure (User Deactivation) - Service Timeout

7.5.1.11 Phone Usage by Type Statistics

Using the **[PERFORMANCE > Statistics > Phone usage by Type]** menu, you can view the following statistical information by menu items:

Internal-Outgoing Usage

This shows the statistical information collected when outbound calls were attempted between internal Users

Item		Description
Date		The start time of statistics measurement
Total		Total number of attempted calls.
Samsung SIP Phones		Number of calls which are attempted to Samsung SIP Phones
Samsung Soft Phones		Number of calls which are attempted to Samsung Soft Phones
Samsung Mobile Phones	Internal Wifi	Number of calls which are attempted to Samsung Mobile Phones for Internal WIFI network
	External Wifi	Number of calls which are attempted to Samsung Mobile Phones for External WIFI network
	Mobile	Number of calls which are attempted to Samsung Mobile Phones for Mobile network
	Unknown	Number of calls which are attempted to Samsung Mobile Phones for unknown network
Analog FXS Phones		Number of calls which are attempted to Analog FXS Phones
3rd Party SIP Phones		Number of calls which are attempted to 3rd Party SIP Phones
Samsung PC Attendants		Number of calls which are attempted to Samsung PC Attendants
FMS Phones		Number of calls which are attempted to FMS Phones
Samsung SIP Phones %		Usage ratio of Samsung SIP Phone
Samsung Soft Phones %		Usage ratio of Samsung Soft Phone
Samsung Mobile Phones %	Internal Wifi	Usage ratio of Samsung Mobile Phone for Internal wifi network
	External Wifi	Usage ratio of Samsung Mobile Phone for External wifi network
	Mobile	Usage ratio of Samsung Mobile Phone for mobile network
	Unknown	Usage ratio of Samsung Mobile Phone for unknown network

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Analog FXS Phones %	Usage ratio of Analog FXS Phone
3rd Party SIP Phones %	Usage ratio of 3rd Party SIP Phone
Samsung PC Attendants %	Usage ratio of Samsung PC Attendant
FMS Phones %	Usage ratio of FMS Phone

Internal-Incoming Usage

This shows the statistical information collected when Inbound calls were attempted between Internal Users

Item		Description
Date		The start time of statistics measurement
Total		Total number of attempted calls.
Samsung SIP Phones		Number of calls which are attempted to Samsung SIP Phones
Samsung Soft Phones		Number of calls which are attempted to Samsung Soft Phones
Samsung Mobile Phones	Internal Wifi	Number of calls which are attempted to Samsung Mobile Phones for Internal WIFI network
	External Wifi	Number of calls which are attempted to Samsung Mobile Phones for External WIFI network
	Mobile	Number of calls which are attempted to Samsung Mobile Phones for Mobile network
	Unknown	Number of calls which are attempted to Samsung Mobile Phones for unknown network
Analog FXS Phones		Number of calls which are attempted to Analog FXS Phones
3rd Party SIP Phones		Number of calls which are attempted to 3rd Party SIP Phones
Samsung PC Attendants		Number of calls which are attempted to Samsung PC Attendants
FMS Phones		Number of calls which are attempted to FMS Phones
Samsung SIP Phones %		Usage ratio of Samsung SIP Phone
Samsung Soft Phones %		Usage ratio of Samsung Soft Phone
Samsung Mobile Phones %	Internal Wifi	Usage ratio of Samsung Mobile Phone for Internal wifi network
	External Wifi	Usage ratio of Samsung Mobile Phone for External wifi network
	Mobile	Usage ratio of Samsung Mobile Phone for mobile

Item		Description
		network
	Unknown	Usage ratio of Samsung Mobile Phone for unknown network
Analog FXS Phones %		Usage ratio of Analog FXS Phone
3rd Party SIP Phones %		Usage ratio of 3rd Party SIP Phone
Samsung PC Attendants %		Usage ratio of Samsung PC Attendant
FMS Phones %		Usage ratio of FMS Phone

Outbound Usage

This shows the statistical information collected when internal Users attempted to call external Users

Item		Description
Date		The start time of statistics measurement
Total		Total number of attempted calls.
Samsung SIP Phones		Number of calls which are attempted to Samsung SIP Phones
Samsung Soft Phones		Number of calls which are attempted to Samsung Soft Phones
Samsung Mobile Phones	Internal Wifi	Number of calls which are attempted to Samsung Mobile Phones for Internal WIFI network
	External Wifi	Number of calls which are attempted to Samsung Mobile Phones for External WIFI network
	Mobile	Number of calls which are attempted to Samsung Mobile Phones for Mobile network
	Unknown	Number of calls which are attempted to Samsung Mobile Phones for unknown network
Analog FXS Phones		Number of calls which are attempted to Analog FXS Phones
3rd Party SIP Phones		Number of calls which are attempted to 3rd Party SIP Phones
Samsung PC Attendants		Number of calls which are attempted to Samsung PC Attendants
FMS Phones		Number of calls which are attempted to FMS Phones
Samsung SIP Phones %		Usage ratio of Samsung SIP Phone
Samsung Soft Phones %		Usage ratio of Samsung Soft Phone
Samsung Mobile Phones %	Internal Wifi	Usage ratio of Samsung Mobile Phone for Internal wifi network

Item		Description
	External Wifi	Usage ratio of Samsung Mobile Phone for External wifi network
	Mobile	Usage ratio of Samsung Mobile Phone for mobile network
	Unknown	Usage ratio of Samsung Mobile Phone for unknown network
Analog FXS Phones %		Usage ratio of Analog FXS Phone
3rd Party SIP Phones %		Usage ratio of 3rd Party SIP Phone
Samsung PC Attendants %		Usage ratio of Samsung PC Attendant
FMS Phones %		Usage ratio of FMS Phone

Inbound Usage

This shows the statistical information collected when external Users attempted to call internal Users.

Item		Description
Date		The start time of statistics measurement
Total		Total number of attempted calls.
Samsung SIP Phones		Number of calls which are attempted to Samsung SIP Phones
Samsung Soft Phones		Number of calls which are attempted to Samsung Soft Phones
Samsung Mobile Phones	Internal Wifi	Number of calls which are attempted to Samsung Mobile Phones for Internal WIFI network
	External Wifi	Number of calls which are attempted to Samsung Mobile Phones for External WIFI network
	Mobile	Number of calls which are attempted to Samsung Mobile Phones for Mobile network
	Unknown	Number of calls which are attempted to Samsung Mobile Phones for unknown network
Analog FXS Phones		Number of calls which are attempted to Analog FXS Phones
3rd Party SIP Phones		Number of calls which are attempted to 3rd Party SIP Phones
Samsung PC Attendants		Number of calls which are attempted to Samsung PC Attendants
FMS Phones		Number of calls which are attempted to FMS Phones
Samsung SIP Phones %		Usage ratio of Samsung SIP Phone

Item		Description
Samsung Soft Phones %		Usage ratio of Samsung Soft Phone
Samsung Mobile Phones %	Internal Wifi	Usage ratio of Samsung Mobile Phone for Internal wifi network
	External Wifi	Usage ratio of Samsung Mobile Phone for External wifi network
	Mobile	Usage ratio of Samsung Mobile Phone for mobile network
	Unknown	Usage ratio of Samsung Mobile Phone for unknown network
Analog FXS Phones %		Usage ratio of Analog FXS Phone
3rd Party SIP Phones %		Usage ratio of 3rd Party SIP Phone
Samsung PC Attendants %		Usage ratio of Samsung PC Attendant
FMS Phones %		Usage ratio of FMS Phone

Total Usage

This shows the statistical information for all calls.

Item		Description
Date		The start time of statistics measurement
Total		Total number of attempted calls.
Samsung SIP Phones		Number of calls which are attempted to Samsung SIP Phones
Samsung Soft Phones		Number of calls which are attempted to Samsung Soft Phones
Samsung Mobile Phones	Internal Wifi	Number of calls which are attempted to Samsung Mobile Phones for Internal WIFI network
	External Wifi	Number of calls which are attempted to Samsung Mobile Phones for External WIFI network
	Mobile	Number of calls which are attempted to Samsung Mobile Phones for Mobile network
	Unknown	Number of calls which are attempted to Samsung Mobile Phones for unknown network
Analog FXS Phones		Number of calls which are attempted to Analog FXS Phones
3rd Party SIP Phones		Number of calls which are attempted to 3rd Party SIP Phones
Samsung PC Attendants		Number of calls which are attempted to Samsung PC Attendants
FMS Phones		Number of calls which are attempted to FMS Phones

Item		Description
Samsung SIP Phones %		Usage ratio of Samsung SIP Phone
Samsung Soft Phones %		Usage ratio of Samsung Soft Phone
Samsung Mobile Phones %	Internal Wifi	Usage ratio of Samsung Mobile Phone for Internal wifi network
	External Wifi	Usage ratio of Samsung Mobile Phone for External wifi network
	Mobile	Usage ratio of Samsung Mobile Phone for mobile network
	Unknown	Usage ratio of Samsung Mobile Phone for unknown network
Analog FXS Phones %		Usage ratio of Analog FXS Phone
3rd Party SIP Phones %		Usage ratio of 3rd Party SIP Phone
Samsung PC Attendants %		Usage ratio of Samsung PC Attendant
FMS Phones %		Usage ratio of FMS Phone

7.5.1.12 Phone Registration by Type Statistics

Using the **[PERFORMANCE > Statistics > Phone Registration by Type]** menu, you can view the following statistical information by menu items:

To Reg

This shows the statistical information for state change from Unreg to reg.

Item		Description
Date		The start time of statistics measurement
Total		Total number of changed events
Samsung SIP Phones		Number of events which are changed for Samsung SIP Phones
Samsung Soft Phones		Number of events which are changed for Samsung Soft Phones
Samsung Mobile Phones	Internal Wifi	Number of events which are changed for Samsung Mobile Phones for Internal WIFI network
	External Wifi	Number of events which are changed for Samsung Mobile Phones for External WIFI network
	Mobile	Number of events which are changed for Samsung Mobile Phones for Mobile network
	Unknown	Number of events which are changed for Samsung Mobile Phones for unknown network
Samsung PC Attendants		Number of events which are changed for Samsung PC Attendants

Item		Description
3rd Party SIP Phones		Number of events which are changed for 3rd Party SIP Phones
Analog FXS Phones		Number of events which are changed for Analog FXS Phones
FMS Phones		Number of events which are changed for FMS Phones
Samsung SIP Phones %		Usage ratio of Samsung SIP Phone
Samsung Soft Phones %		Usage ratio of Samsung Soft Phone
Samsung Mobile Phones %	Internal Wifi	Usage ratio of Samsung Mobile Phone for Internal wifi network
	External Wifi	Usage ratio of Samsung Mobile Phone for External wifi network
	Mobile	Usage ratio of Samsung Mobile Phone for mobile network
	Unknown	Usage ratio of Samsung Mobile Phone for unknown network
Analog FXS Phones %		Usage ratio of Analog FXS Phone
3rd Party SIP Phones %		Usage ratio of 3rd Party SIP Phone
Samsung PC Attendants %		Usage ratio of Samsung PC Attendant
FMS Phones %		Usage ratio of FMS Phone

To Unreg

This shows the statistical information for state change from reg to unreg.

Item		Description
Date		The start time of statistics measurement
Total		Total number of changed events
Samsung SIP Phones		Number of events which are changed for Samsung SIP Phones
Samsung Soft Phones		Number of events which are changed for Samsung Soft Phones
Samsung Mobile Phones	Internal Wifi	Number of events which are changed for Samsung Mobile Phones for Internal WIFI network
	External Wifi	Number of events which are changed for Samsung Mobile Phones for External WIFI network
	Mobile	Number of events which are changed for Samsung Mobile Phones for Mobile network
	Unknown	Number of events which are changed for Samsung Mobile Phones for unknown network

Item		Description
Samsung PC Attendants		Number of events which are changed for Samsung PC Attendants
3rd Party SIP Phones		Number of events which are changed for 3rd Party SIP Phones
Analog FXS Phones		Number of events which are changed for Analog FXS Phones
FMS Phones		Number of events which are changed for FMS Phones
Samsung SIP Phones %		Usage ratio of Samsung SIP Phone
Samsung Soft Phones %		Usage ratio of Samsung Soft Phone
Samsung Mobile Phones %	Internal Wifi	Usage ratio of Samsung Mobile Phone for Internal wifi network
	External Wifi	Usage ratio of Samsung Mobile Phone for External wifi network
	Mobile	Usage ratio of Samsung Mobile Phone for mobile network
	Unknown	Usage ratio of Samsung Mobile Phone for unknown network
Analog FXS Phones %		Usage ratio of Analog FXS Phone
3rd Party SIP Phones %		Usage ratio of 3rd Party SIP Phone
Samsung PC Attendants %		Usage ratio of Samsung PC Attendant
FMS Phones %		Usage ratio of FMS Phone

Registration Average

This shows the statistical information for registration average.

Item		Description
Date		The start time of statistics measurement
Total Average		Total average of registration
Samsung SIP Phones Average		Average which are registered for Samsung SIP Phones
Samsung Soft Phones Average		Average which are registered for Samsung Soft Phones
Samsung Mobile Phones	Internal Wifi Average	Average which are registered for Samsung Mobile Phones for Internal WIFI network
	External Wifi Average	Average which are registered for Samsung Mobile Phones for External WIFI network
	Mobile Average	Average which are registered for Samsung Mobile Phones for Mobile network

Item		Description
	Unknown Average	Average which are registered for Samsung Mobile Phones for unknown network
Samsung PC Attendants Average		Average which are registered for Samsung PC Attendants
3rd Party SIP Phones Average		Average which are registered for 3rd Party SIP Phones
Analog FXS Phones Average		Average which are registered for Analog FXS Phones
FMS Phones Average		Average which are registered for FMS Phones
Samsung SIP Phones %		Registration ratio of Samsung SIP Phone
Samsung Soft Phones %		Registration ratio of Samsung Soft Phone
Samsung Mobile Phones %	Internal Wifi	Registration ratio of Samsung Mobile Phone for Internal wifi network
	External Wifi	Registration ratio of Samsung Mobile Phone for External wifi network
	Mobile	Registration ratio of Samsung Mobile Phone for mobile network
	Unknown	Registration ratio of Samsung Mobile Phone for unknown network
Analog FXS Phones %		Registration ratio of Analog FXS Phone
3rd Party SIP Phones %		Registration ratio of 3rd Party SIP Phone
Samsung PC Attendants %		Registration ratio of Samsung PC Attendant
FMS Phones %		Registration ratio of FMS Phone

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

7.5.2 ACD Reports

7.5.2.1 ACD Group Statistics

Using the **[PERFORMANCE > Statistics > ACD Group]** menu, you can view the following statistical information by menu items:

System Summary

This shows the statistical information for all system-wide ACD calls.

Item		Description
Date		The start time of statistics measurement
ACD Incoming	Total	Total number of ACD calls
	Answer	Number of ACD answered calls
	Answer %	Success ratio of ACD calls
	Failure	Total number of failed ACD calls
	Average Wait Time	Average wait time of ACD
	Average Talk Time	Average talk time of ACD
Normal Incoming	Total	Total number of Inbound calls except ACD
	Internal	Number of Inbound internal calls except ACD
	Inbound	Number of Inbound external calls except ACD
	Total Talk Time	Total talk time of Inbound calls except ACD
	Average Talk Time	Average talk time of Inbound calls except ACD
Normal Outgoing	Total	Total number of Outbound calls except ACD
	Internal	Number of Outbound internal calls except ACD
	Outbound	Number of Outbound external calls except ACD
	Total Talk Time	Total talk time of Outbound calls except ACD
	Average Talk Time	Average talk time of Outbound calls except ACD

ACD-Group Summary Statistics

This shows the statistical information for all calls for the ACD group.

Item		Description
Date		The start time of statistics measurement
ACD Group		Name of an ACD group
User Group		Name of a user group
Queuing	Average Time	Average ring time of ACD Queuing
	Longest Time	Longest ring time of ACD Queuing
Overflow	In	Number of ACD Overflow In

Item		Description
	Out	Number of ACD Overflow Out
Incoming Calls	Total	Total number of ACD calls
	Answer	Number of answered ACD calls
	Answer %	Success ratio of ACD calls
	Failure	Total number of failed ACD calls
	Total Wait time	Total ring time of ACD
	Average Wait Time	Average ring time of ACD
	Longest Wait Time	Longest ring time of ACD
	Total Talk Time	Total talk time of ACD
	Average Talk Time	Average talk time of ACD
	Longest Talk Time	Longest talk time of ACD

ACD-Group Overflow Statistics

This shows the statistical information collected when overflow occurred for ACD group.

Item	Description
Date	The start time of statistics measurement
ACD Group	Name of an ACD group
User Group	Name of a user group
Overflow In	Number of ACD Overflow In
Overflow Out	Number of ACD Overflow Out

7.5.2.2 ACD Agent Statistics

Using the **[PERFORMANCE > Statistics > ACD Agents]** menu, you can view the following statistical information by menu items:

Summary

This shows the statistical information for all agents' calls.

Item		Description
Date		The start time of statistics measurement
Agent ID		Agent ID
User Group		Name of a user group
ACD Incoming	Total	Total number of ACD calls
	Answer	Number of answered ACD calls
	Abandon	Number of abandoned ACD calls
	No Answer	Number of no-answer ACD calls

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item		Description
	Answer %	Success ratio of ACD calls
	Failure	Total number of failed ACD calls
	Total Wait Time	Total ring time of ACD
	Average Wait Time	Average ring time of ACD
	Longest Wait Time	Longest ring time of ACD
	Total Talk Time	Total talk time of ACD
	Average Talk Time	Average talk time of ACD
	Longest Talk Time	Longest talk time of ACD
Normal Incoming	Total	Total number of Inbound calls except ACD
	Internal	Number of Inbound internal calls except ACD
	Outbound	Number of Inbound external calls except ACD
	Total Talk Time	Total talk time of Inbound calls except ACD
	Average Talk Time	Average talk time of Inbound calls except ACD
Normal Outgoing	Total	Total number of Outbound calls except ACD
	Internal	Number of Outbound internal calls except ACD
	Outbound	Number of Outbound external calls except ACD
	Total Talk Time	Total talk time of Outbound calls except ACD
	Average Talk Time	Average talk time of Outbound calls except ACD

Utilization

This shows the statistical information for all agents' level of contribution.

Item	Description
Date	The start time of statistics measurement
Agent ID	Agent ID
User Group	Name of a user group
Avail %	Actual working ratio except break time
Total Logged Time	Displays the total log time
Total Break Time	Total break time

Activity

This shows the statistical information for all agents' activities.

Item	Description
Date	The start time of statistics measurement
Agent ID	Agent ID
ACD Group	Name of an ACD group

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
User Group	Name of a user group
Activity	Activity of an agent (Login, Logout, Break-in, Break-out)

7.5.3 Resource Statistics

Using the **[PERFORMANCE > Statistics > Resource]** menu, you can view the following statistical information by menu items:

CPU

This shows the statistical information for CPU.

Item	Description
Date	The start time of statistics measurement
System Name	System Name
Average User Use	Average User Use
Average System Use	Average System Use
Average Wait Use	Average Wait Use
Average Idle Use	Average Idle Use

Memory

This shows the statistical information for memory.

Item	Description
Date	The start time of statistics measurement
System Name	System Name
Total Memory	Total Memory
Free Memory	Free Memory
Used Swap	Used Swap
Free Swap	Free Swap
Average Usage	Average Usage

Disk

This shows the statistical information for disk.

Item	Description
Date	The start time of statistics measurement
System Name	System Name

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Total Disk	Total Memory
Used Disk	Used Disk
Available Disk	Available Disk
Average Usage	Average Usage

7.5.4 Alarm Statistics

Using the **[PERFORMANCE > Statistics > System Alarms]** menu, you can view the following statistical information by menu items:
This shows the statistical information for alarms, faults and status.

Item	Description
Date	The start time of statistics measurement
System	Name of a system
Level	Alarm levels (Critical, Major, Minor, Normal)
Alarm	Alarm name
Count	Number of alarms

7.5.5 Emergency Log

Using the **[PERFORMANCE > Statistics > Emergency Log]** menu, you can view the following log information by menu items:
This shows the log information for emergency call.

Item	Description
Date	The start time of statistics measurement
Extension	Extension Number
Start Time	Start Time of call
End Time	End Time of call
Call State	Call State
Call ID	Call ID
Callee Number	Callee Number
Manager	Manager for extension

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

7.6 Fault Management

This section describes various settings and methods for handling system events. Events are generated as alarms, faults or status whenever there is a problem with the system or a specific status changes. You can configure the profile for such events. Alarms, faults and status serviced by IPX-S300B are categorized in the following way:

- Event: It provides real-time monitoring of alarms, faults, status, and alarm clear information in the IPX-S300B system
- Alarm: A critical problem such as network failure or process termination has occurred and it can be cleared.
- Fault: A critical problem such as database backup failure has occurred and it cannot be cleared.
- Status: A status change, such as redundancy status change, which is not an alarm or a fault, has occurred.
- Alarm Cleared: It provides real-time monitoring of alarms generated and then cleared in the IPX-S300B system.

In the bottom left corner of the SCM Administrator screen is the System Viewer window which displays the number of alarms generated in the IPX-S300B system. When an alarm is generated in IPX-S300B, the System Viewer window displays the number of alarms generated for each level.

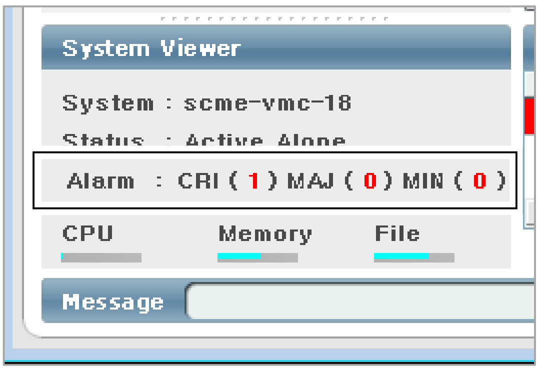


Figure 22. System Viewer window

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

7.6.1 Viewers

Event Viewer

Event Viewer provides real-time monitoring of alarms, faults, status, and alarm clear information in the IPX-S300B system.
You can view Event Viewer in the **[PERFORMANCE > Fault > Viewer > Viewer-Event]** menu.
Event Viewer displays alarm, fault, status, and alarm clear information simultaneously in the order of alarms > faults > status > alarm clear from the top.
Using the **[Clear]** button, you can force clear an alarm item.

Alarm Viewer

Alarm Viewer provides real-time monitoring of alarms generated in the IPX-S300B system.
You can view Alarm Viewer in the **[PERFORMANCE > Fault > Viewer > Viewer-Alarm]** menu.
Alarms are highlighted in different colors depending on the levels: red for critical, orange for major, yellow for minor, and green for normal.

Fault Viewer

Fault Viewer provides real-time monitoring of faults generated in the IPX-S300B system.
You can view Fault Viewer in the **[PERFORMANCE > Fault > Viewer > Viewer-Fault]** menu.
Using the **[Clear]** button, you can force clear a fault item.

Status Viewer

Status Viewer provides real-time monitoring of status generated in the IPX-S300B system.
You can view Status Viewer in the **[PERFORMANCE > Fault > Viewer > Viewer-Status]** menu.
Using the **[Clear]** button, you can force clear a status item.

Alarm Clear Viewer

Alarm Viewer provides real-time monitoring of alarms generated and then cleared in the IPX-S300B system.
You can view Alarm Clear Viewer in the **[PERFORMANCE > Fault > Viewer > Alarm Cleared]** menu.
Alarms are highlighted in different colors depending on the levels: red for critical, orange for major, yellow for minor, and green for normal. When an alarm is cleared, it is highlighted in green for normal level.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

7.6.2 Alarm Settings

The Alarm Settings include detailed information on alarms, faults and status occurring in IPX-S300B. Using this information, an operator can set how to display information on alarms, faults and status in SCM Administrator.

Alarm Profile (Alarm)

Alarms are located in **[Performance > Fault > Settings]** menu, and they display detailed information for alarm types and detailed information on each of them occurring in IPX-S300B.
The table below displays information on alarm profiles. For alarms which requires default values, alarms levels can be changed according to default values. For such alarms, therefore, default level values (CRITICAL, MAJOR or MINOR) do not exist.

Alarm name	Category	Level	Critical (%)	Major (%)	Minor (%)
Abnormal Block State	Processing Error	Critical	-	-	-
Abnormal 3rd Party Application State	Processing Error	Critical	-	-	-
CPU Over Used	Resource	-	100	95	80
Hard-Disk Over Used	Resource	-	100	95	80
Memory Over Used	Resource	-	100	95	80
CPU Over Used by Process	Resource	-	90	80	70
Network Interface Down	Equipment	Critical	-	-	-
Maximum Call	QoS	-	100	95	80
Maximum User	QoS	-	100	95	-
Gateway Connection Lost	Communication	Minor	-	-	-
Location Bandwidth Used	System Management	-	95	90	85
Resource Based CAC	System Management	Major	-	-	-

Alarm profiles can be checked in **[Performance > Fault > Settings > Settings-Alarm]** menu. Push the Find button to display alarm profiles information, and push the Change button to change alarm profiles.

Item	Default	Description
Name	-	Specifies the alarm name.
Level	-	Sets the default level of the alarm - NONE: Alarm with no alarm level. - Critical: Alarm with Critical level. - Major: Alarm with Major level. - Minor: Alarm with Minor level.

Item	Default	Description
Enable Flag	-	Sets whether to notice the alarm by alarm messages when the alarm occurs.
Pop-up Notice	-	Sets whether to notice by pop-up windows when the alarm occurs.
Audible Flag	-	Sets whether to notice by alarm sounds when the alarm occurs.
E-mail Flag	Disable	Sets whether to notice by e-mails when the alarm occurs.
SMS Notify	Disable	Sets whether to notice by SMS when the alarm occurs.
Critical Threshold	-	Reference value to be generated at a Critical level.
Major Threshold	-	Reference value to be generated at a Major level.
Minor Threshold	-	Reference value to be generated at a Minor level.

Fault Profile (Fault)

Fault profiles display fault types and detailed information on each of them occurring in IPX-S300B. The table below displays profiles information.

Fault Name	Category	Alarm Notice	E-mail notice
Authentication Fail	Communication	Enable	Disable
CDR FTP Send Fail	Processing Error	Enable	Disable
Illegal Call Try	Communication	Enable	Disable
DB Backup Fail	Processing Error	Enable	Disable
Unknown Registration	Communication	Disable	Disable
License Expired	System Management	Enable	Disable
Unsupported Codec Request	Communication	Enable	Disable

Fault profiles can be checked in **[Statistics/Performance > Fault Management > Alarms Settings > Setting-Fault]** menu. Push the Find button to display faults information, and push the Change button to change faults.

Item	Default	Description
Name	-	Specifies the fault name.
Enable Flag	-	Sets whether to notice by alarm messages when the fault occurs.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Default	Description
E-mail Flag	Disable	Sets whether to notice by e-mail when the fault occurs.
SMS Notify	Disable	Sets whether to notice by SMS when the alarm occurs.

Status Profile

Status profiles display the types of status profiles and detailed information on each of them occurring in IPX-S300B. The table below display status profiles information.

Status Name	Category	Alarm Notice	E-mail Notice
Block State Change	Equipment	Enable	Disable
HA Mode Change	Equipment	Enable	Disable
Wakeup Service	Communication	Enable	Disable
DB Backup	Communication	Enable	Disable
Data Sync	Communication	Enable	Disable
Long Duration Call	Communication	Enable	Disable
Malicious Call Claim	Communication	Enable	Disable
Database Over Used	Resource	Enable	Disable
CDR FTP Send Complete	System Management	Enable	Disable
Phone Update Notified	System Management	Enable	Disable
Switch Over by CPU Overload	Process Error	Enable	Disable
UMS Maintenance	Equipment	Enable	Disable
Stopping Port	Communication	Enable	Disable

Status profiles can be checked in **[Performance > Fault > Settings > Settings-Status]** menu. Push the Find button to display faults information, and push the Change button to change faults.

Item	Default	Description
Name	-	Specifies the status name.
Enable Flag	-	Sets whether to notice by alarm messages when the status occurs.
E-mail Flag	Disable	Sets whether to notice by e-mails when the status occurs.
SMS Notify	Disable	Sets whether to notice by SMS when the status occurs.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

7.6.3 Alarm History

This feature allows you to manage the history of faults and status information generated in IPX-S300B.

Alarm History

You can view the alarm history in the **[PERFORMANCE > Fault > History> History-Alarm]** menu. Select alarm history and click the **[Clear]** button to delete alarm history.

Fault History

You can view the fault history in the **[PERFORMANCE > Fault > History> History-Fault]** menu.
Select fault history and click the **[Clear]** button to delete fault history.

Status History

You can view the status history in the **[PERFORMANCE > Fault > History> History-Status]** menu. Select status history and click the **[Clear]** button to delete status history.

7.6.4 Email Notification

The alarm, fault, and status information generated in IPX-S300B can be notified to the administrator by email. To allow this, the email field in the alarm profile, fault profile or status profile must be set to Enable and the required settings for the email server must be configured in email settings.
You can configure your email settings in the **[PERFORMANCE > Fault > E-mail Notification Setup]** menu.

Item	Default Value	Description
SMTP Server: HostID	-	Specify the IP address or the host name of the SMTP server.
SMTP Server: Port	25	Specify the port number of the SMTP server.
SMTP Server: Domain	-	Specify the domain name of the SMTP server.
Auth Login: User ID	-	Specify the user login required for user authentication by the SMTP server.
Auth Login: Password	-	Specify the user password required for user authentication by the SMTP server.
Address: From	-	Specify the sender’s email address.
Address: To	-	Specify the recipient’s email address.

7.7 License Management

Evaluation License

Select CONFIGURATION > Resource > Evaluation License

The built-in Evaluation License Key is valid for 30 days after installation.

Item	Description
License Key Type	Shows system type. - S300BSCM Compact-Users - S300BSCM Compact-Embedded Application - S300BSCM Compact-External Application
License Key	Enter the license key obtained from the Mac address.
MAC Address	Display a MAC address which is interpreted from a license key.
License Status	Display license operation status.
Samsung SIP Phones	Display max. number of Samsung SIP Phones allowed in the license.
Samsung Soft Phones	Display max. number of Samsung Soft Phones allowed in the license.
Samsung Mobile Phones	Display max. number of Samsung Mobile Phones allowed in the license.
Samsung PC Attendants	Display max. number of Samsung PC Attendants allowed in the license.
3rd Party SIP Phones	Display max. number of 3rd Party SIP Phones allowed in the license.
Basic VM Channels	Display max. number of UMS Channels allowed in the license.
Total CSTA Applications	Display max. number of Total CSTA Applications allowed in the license.
Embedded ACD Agent Links	Display max. number of Embedded ACD Agent Links allowed in the license.
Communicators (Desktop)	Display max. number of Communicators (Desktop) allowed in the license.
Other CSTA Applications	Display max. number of Other CSTA Applications allowed in the license.
SIP Application Channels	Display max. number of SIP Application Channels allowed in the license.

Activation Key

Select CONFIGURATION > Resource > Activation Key

Activation key must be the MAC address of system eth0 or eth1. An operator can use the system by entering a activation key which is obtained in this page.

The built-in Sample Activation Key is valid while both license key and activation is not inputted.

The priority order of license is Activation Key > Evaluation License > Sample Activation Key.

License for Samsung Soft Phone for Windows, Samsung FMC Phones, Samsung PC Attendant and 3rd-Party SIP Phone can be used for Samsung SIP Phone.

Item	Description
Package Name	Shows package name of activation key.
Activation Key	Enter the activation key obtained from the Mac address.
MAC Address	Display a MAC address which is interpreted from a license key.
Status	Display license operation status.
Samsung SIP Phones	Display max. number of Samsung Mobile Phones allowed in the license.
Samsung Soft Phones for Windows	Display max. number of Samsung Soft Phones for Windows allowed in the license.
Samsung FMC for Android	Display max. number of Samsung FMC for Android allowed in the license. This is switchable for Samsung FMC for iPhone
Samsung FMC for iPhone	Display max. number of Samsung FMC for iPhone allowed in the license. This is switchable for Samsung FMC for Android
Samsung PC Attendants	Display max. number of Samsung PC Attendants allowed in the license.
3rd Party SIP Phones	Display max. number of 3rd Party SIP Phones allowed in the license.
VM Channels	Display max. number of VM Channels allowed in the license.
SIP Application Channels	Display max. number of SIP Application Channels allowed in the license. SIP auxiliary devices requiring licenses are as follows. - External UMS server - External IVR server - External Conference server - External Coloring server
SIP Trunk Channel	Displays the maximum number, allowed by the license, of SIP trunk channels for third companies
Embedded ACD Agent	Display max. number of Embedded ACD Agents allowed in the license.
Communicators Deskphone	Display max. number of Communicators Deskphone allowed in the license.
CSTA Link	Display max. number of CSTA Links allowed in the license.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Item	Description
Embedded Wall Board	Display max. number of Embedded Wall Boards allowed in the license.
Directory Service	Display max. number of Directory Service allowed in the license.
Smart WLAN Link	Display max. number of Smart WLAN Link allowed in the license.
VPU VM Channel	When using a VPU card, displays the maximum number of VM channels allowed by the license.
VPU Meet-Me conference channel	When using a VPU card, displays the maximum number of scheduled conference channels allowed by the license.
SBC Lite Call Connection	Displays the number of maximum SBC Lite call connections.
NAT Traversal Call Connection Number	Display the number of calls going through the NAT in the LAN Only mode

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

7.8 LDAP Data Import

IPX-S300B provides importing data from external LDAP server. The following data can be imported:

Item	Description
Department	Department list in **[CONFIGURATION > User Group > Department]**
Position	Position list in **[CONFIGURATION > User Group > Position]**
User Profile	User Profile data in **[CONFIGURATION > User > User Profile]** which defined in **[CONFIGURATION > User Group > User Profile Field Name]**

Using the **[Management > LDAP Data Import]**, you can set the following LDAP information by menu items:

LDAP Server

You can input external LDAP server information in **[Management > LDAP Data Import > LDAP Server]**.

Item	Description
LDAP Server IP	Specify the LDAP Server's IP.
LDAP Server Port	Specify the LDAP Server's Port. The default value is 389.
LDAP Bind DN	Specify the LDAP Server's Bind DN for login.
LDAP Bind Password	Specify the LDAP Server's Bind Password for login.

LDAP Data Mapping

You can make mapping data of IPX-S300B and data of LDAP server in **[Management > LDAP Data Import > LDAP Data Mapping]** menu.

Item	Description
User Group	Specify the user group for imported data.
Base DN	Specify the LDAP Server's Base DN.
Department Filter	Specify the entity filter for department.
Department Attribute	Specify the attribute for department.
Department Parent Attribute	Specify the attribute for department parent.
Position Filter	Specify the entity filter for position.
Position Attribute	Specify the attribute for position.
User Filter	Specify the entity filter for user profile.
User ID Attribute	Specify the attribute for user ID.

Item	Description
User Name Attribute	Specify the attribute for user Name.
User Department Attribute	Specify the attribute for user Department.
User Position Attribute	Specify the attribute for user Position.
User Profile Field Name	Specify the attribute for user Profile Field Name.

LDAP Schedule

You can import data periodically or immediately in **[Management > LDAP Data Import > LDAP Schedule]** menu.

Item	Description
Schedule Type	Specify the schedule type. - Day: import the data in time daily - Week: import the data in day weekly - 15 Days: import the data in 15th monthly - Month: import the data in 1th monthly - Now: import the data immediately
Weekday	Specify the weekday.
Time	Specify the time.

7.9 LDAP Server Service

7.9.1 Overview of the LDAP Server Service

LDAP server service provides that external devices linked with IPX-S300B can take user information of IPX-S300B. The external devices can query user information registered in IPX-S300B like as: Extension Number, Department, Position and so on.
IPX-S300B provides search function only, not changing or adding via LDAP server.

7.9.2 LDAP Server Service

Searching Information

LDAP server connects to the internal database of IPX-S300B and provides data to LDAP client.

LDAP server provides the following data:

- Department Information
- Position List
- User Group List
- User Information
- LDAP Manager Information

For more information on each, please refer to the **[IPX-S300B LDAP Server Interoperability Guide]** document.

LDAP Server access

LDAP server divides the user level into LDAP Manager and LDAP User for network management and system load. And LDAP User has some limitation when accesses to LDAP server.

- LDAP Manager
  User with LDAP Manager level can query all information that LDAP server provides and there is no restriction.
- LDAP User
  User with LDAP User level can not get the user password.

You can set the level in **[CONFIGURATION > User > User Profile]** menu.

Items	Description
LDAP Server Manager	- No: LDAP User (default) - Yes: LDAP Manager

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Basic information for LDAP server service

Basic information for connecting to LDAP Server is as follows:

Items	Description
LDAP Server port	389
Login ID	When you connect to the server with manager level - Login ID: cn='User ID', ou=manager, dc=scm, dc=samsung When you connect to the server with user level - Login ID: cn='User ID', ou=subscriber, dc=scm, dc=samsung 'User ID' is **[User ID]** in **[CONFIGURATION > User > User Profile]** menu.
LoginPassword	Application User Password (need to MD5 encryption.) **[Password]** in **[CONFIGURATION > User > User Profile]**
Base DN	dc=scm, dc=samsung

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

7.10 Directory Service for Mobile User

7.10.1 Overview of the Directory Service

Directory service provides the user information of IPX-S300B to Mobile User.
Directory service provides the following data:

- Directory Service
- Presence Service
- Location Service

7.10.2 Directory Service

IPX-S300B basically provides the user information data in **[CONFIGURATION > User > User Profile]** menu to Mobile User.

Items	Description
Application User ID	Input a ID of a application user.
Password	Input a password of a application user.
Name	Input a name of a application user.
User Group	Select a user group.
Extension	Select a extension number.
Department	Select a department.
Position	Select a position.
A-A Primary Node	Specify Primary Node when A-A Configuration
LDAP Server Manager	If set YES, it can use all LDAP functionality
Presence	Select a Presence of Phone. Phone can select a Presence also.
System Photo File Name	Select a photo file name of user.
Director	Select whether user is Director in user's Department.
Priority in Same Position	Input a priority level in same position. - range: 0~10000 (0 is the highest priority)
Top Department in Directory Service	Select a Top Department displayed on Phone in Directory Service.
NoticeBoard Service Authority	Select a authority on NoticeBoard Service.
Office Phone Number	Specifies in the **[Specify Application User Field]** menu, and it is allowed to change.
Cellphone Number	Specifies in the **[Specify Application User Field]** menu, and it is allowed to change.
FAX Number	Specifies in the **[Specify Application User Field]** menu, and it is allowed to change.
E-mail	Specifies in the **[Specify Application User Field]** menu, and it is allowed to change.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

For additional information of User, configure the data in **[CONFIGURATION > User Group > User Profile Field Name]** menu.

Items	Description
Language	Select Korean or English.
Index	Select index 1~20.
Name	Specify name of each index. This name is displayed in **[CONFIGURATION > User > User Profile]** page, so input appropriate name.
Type	Specify attibute of 'Name' field.
Display on Phone	Specify whether a phone display the data in directory service. - For Job Search on Mobile Phone, select 'Job' Type.

For Department information of User Group, configure the data in **[CONFIGURATION > User Group > Department]** menu.

Items	Description
User Group	Select A user group for which Department will be configured.
Parent Department	Specify a Parent Department. If created a user group, automatically a department is created. Select a created department for parent department.
Department Name	Specify a Department Name. This department is used in Single Phone User or Multi-Phone User or User Profile pages.
Search Parent in Directory Service.	Select whether user can search parent department in Directory Service on Phone.
Description	Specify a Description of department.

For Position information of User Group, configure the data in **[CONFIGURATION > User Group > Position]** menu.

Items	Description
User Group	Select A user group for which Position will be configured.
Position Name	Specify a Position Name. This position is used in Single Phone User or Multi-Phone User or User Profile pages.
Description	Specify a Description of position.

7.10.3 Presence Service

IPX-S300B provides the presence information of other user to Mobile User.
User can configure the presence data in **[CONFIGURATION > User > User Profile]** menu to Mobile User.

Items	Description
Presence	Select a Presence of Phone. Phone can select a Presence also.

Administrator can configure the additional options of Presence in **[CONFIGURATION > User Group > Presence Field Name]** menu.

Items	Description
Language	Select Korean or English.
Index	Select index 1~7.
Name	Specify name of each index. This name is displayed in Phone, so input appropriate name.
Icon Color	Specify color of presence name on Mobile Phone Screen.

7.10.4 Location Service

IPX-S300B provides the location information of other user to Mobile User.

7.10.5 Activation of Directory Service for Mobile User.

Administrator can configure the activation of Directory Service for Mobile User in **[CONFIGURATION > Wireless Enterprise > Mobile Phone Profile]** menu.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

7.11 Photo File Management

7.11.1 Overview of Photo file Management

Photo File Management is provided to handle photo files used in the Directory service and the Premium CID service. This feature supports the upload and download of batch file and gives tools to compress, delete, adjust and sync. As well as the size and the count of photo files which are stored in IPX-S300B can be checked. Photo File Management is limited file processing. If you want to enter the mapping information of a photo file and an user, use **[Tool > Customer Data Import/Export]**.

7.11.2 Photo File Management Feature

Searching Information

The count and the size of photo files which are stored in IPX-S300B can be displayed.

Photo File Management Tools

IPX-S300B provides several management tools for photo files.

- Command: request the file processing for photo files.
  - Compress All: Before download of photo files in IPX-S300B, execute this command to compress files. Each compressed file is divided to 1 Gbytes. This processing is a time-consuming job.
  - Delete All: Delete all photo files in IPX-S300B.
  - Adjust: Delete unused photo files. IPX-S300B check DB to check whether each file is used of not.
  - AS Sync: When AS Sync is needed, use this manual A/S Sync. IPX-S300B clears the photo file directory of Standby System and get all files of Active System.
- Upload: IPX-S300B provides Batch job for photo files. Each photo file size is limited to 1 Gbytes. The file format for batch file is TAR without compression when A/S Sync for uploaded batch file is provided. A/A Sync is not supported, so separate upload is need for AA System configuration. Single Photo file can be changed in the **[CONFIGURATION > User > User Profile]**, Personal Web Page and in the FMC phone.
- Download: IPX-S300B provides download function for the photo files that was uploaded and used in the system. It is required that all photo files should be compressed by executing the **[Command-Compress All]** before downloading it. With selection to the download command, several batch files separated by 1 Gbyte size can be downloaded at the same time.

Limitations

- File name written by Korean is not supported.
- Size of batch file for Upload is up to 1Gbyte at a time. This limitation is based on the guarantee for Call Server stability. Because photo file processing is a burden to Call Processing System, this must not be done within work time or during data sync time.
- When single photo file is changed, IPX-S300B supports A/S Hot Sync and A/A Hot Sync. In case of upload of batch file, A/A Sync is not supported but A/S sync is provided. Therefore batch file upload has to be done in the Master System and Slave System both.

CHAPTER 8. Troubleshooting Guide

This chapter describes troubleshooting.

8.1 IPX-S300B Server

The troubleshooting lists are following:

Type	Troubleshooting List
IPX-S300B Hardware	[1001] Server Fails to Boot (Before Video Output) [1002] Connecting with External Devices [1003] Problem with Video Output [1004] Problem with a USB Device
Linux	[1101] The Initial System Login Account Has Been Forgotten [1102] An Error Window is Displayed After Quitting the scmWizard. [1103] The Linux System User Password Has Been Forgotten [1104] The Linux System Root Password Has Been Forgotten
IPX-S300B Software Base	[1201] IPX-S300B Block's Abnormal Termination Alarm Has Occurred (Automatic Restart) [1202] IPX-S300B Block's Abnormal Termination Alarm Has Occurred (Terminated) [1203] A CPU Alarm Has Occurred and is Not Cleared [1204] A Hard Disk Alarm Has Occurred [1205] A Memory Alarm Has Occurred
SCM Administrator	[1301] Administrator can't access from a Web Browser [1302] Administrator can't access through NAT [1303] Administrator Does Not Work Correctly [1304] Cannot Install Administrator [1305] Administrator Does Not Run in a Web Browser [1306] Cannot Log Into PWP with SSO [1307] Administrator Password Has Been Forgotten [1308] Alarms That Have Occurred Are Not Displayed in Administrator [1309] No Email is Sent When an Alarm Occurs [1310] Querying the System Version [1311] Learning More About Process Diagram [1312] Turning the Alarm Sound Off [1313] Force Quitting SSO Agent

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Type	Troubleshooting List
	[1314] The Run Button is Disabled and Cannot be Pressed [1315] Cannot Add Subscriber [1316] Cannot Create Non-Subscriber Data [1317] The 'Evaluation License Expire' Alarm Has Occurred [1318] Where data except for users are not generated. [1319] 'Evaluation License Expire' Where the alarm occurs.

8.1.1 IPX-S300B Software Base

8.1.1.1 [1201] IPX-S300B Block's Abnormal Termination Alarm Has Occurred (Automatic Restart)

Symptoms

'Abnormal Block State' Alarm is found in EVENT VIEWER or ALARM HISTORY of SCM Administrator window. However, the corresponding process or subsystem containing the process is appeared in red.

Possible Causes

A process has malfunctioned.

Solutions

When a process is abnormally terminated, the Process Manager of the IPX-S300B automatically restarts the process. However, you should send the log and alarm information at the time of the error to the development team in order to for them to trace the root of the problem cause and prevent the same problem happening.

8.1.1.1.1 How to download the system log to local PC

1) Connect a keyboard, a mouse, and a monitor to the device.

2) Log into the system using the root account.

3) Right click on the desktop and select 'Open Terminal'.

4) Compress the log for the active system and the standby system and move it to the IPX-S300B account as shown below.

[Log File]

```
[admin@IPX-S30]B ~]# cd /scm_data/
[admin@IPX-S300B:/mnt/mmc/scm_data]# tar -zcvf
YYYYMMDD_scmname_LOG.tar.gz /mnt/nand1/SMB/log /scm_data/LOG/SMB/log
(YYYYMMDD: date, scmname: name of the device)
```

[Core File]

```
ls -al /scm_data/LOG/corefiles
mv  /scm_data/LOG/corefiles/core.xxx.exe.xxx /scm_data/
```

5) Connect PC to the switch or hub which is connected to the IPX-S300B device.

6) Set the IP address of the PC as the same network as the system IP address of the IPX-S300B.
   For example, if the IP address of the IPX-S300B system is 20.20.20.XXX, set the IP address of the PC to 20.20.20.123.

7) From the PC, establish a FTP connection to the IPX-S300B system, and then download the compressed log file. (The FTP login ID is ‘scm’ and the password is ‘samsung*#’.)

8.1.1.2 [1202] IPX-S300B Block’s Abnormal Termination Alarm Has Occurred (Terminated)

Symptoms

‘Abnormal Block State’ Alarm is found in EVENT VIEWER or ALARM HISTORY of SCM Administrator window. And, the corresponding process or subsystem containing the process is highlighted in red.

Possible Causes

A process has malfunctioned and has been restarted, but the process kept malfunctioning over 4 times in a row and is no longer restarted.

Solutions

In SCM Administrator, activate the corresponding process.

1) In the **[PERFORMANCE > System Management > Process Management]** menu, click the **[Act]** button, select the abnormally terminated process, and click the **[Act]** button to load the process.
2) Send the log and alarm information to the development team.

How to download the system log to local PC

1) Connect a keyboard, a mouse, and a monitor to the device.

2) Log into the system using the root account.

3) Right-click on the desktop and select ‘Open Terminal’.

4) Compress the log for the active system and the standby system and move it to the IPX-S300B account as shown below:

[Log File]

```
[admin@IPX-S30]B ~]# cd /scm_data/
[admin@IPX-S300B:/mnt/mmc/scm_data]# tar -zcvf
YYYYMMDD_scmname_LOG.tar.gz /mnt/nand1/SMB/log /scm_data/LOG/SMB/log
(YYYYMMDD: date, scmname: name of the device)
```

[Core File]

```
ls -al /scm_data/LOG/corefiles
mv  /scm_data/LOG/corefiles/core.xxx.exe.xxx /scm_data/
```

5) Connect PC to the switch or hub which is connected to the IPX-S300B device.

6) Set the IP address of the PC as the same network as the system IP address of the IPX-S300B.
For example, if the IP address of the IPX-S300B system is 20.20.20.XXX, set the IP address of the PC to 20.20.20.123.

7) From the PC, establish a FTP connection to the IPX-S300B system, and then download the compressed log file. (The FTP login ID is ‘scm’ and the password is ‘samsung*#’.)

8.1.1.3 [1203] A CPU Alarm Has Occurred and is Not Cleared

Symptoms

EVENT VIEWER or ALARM HISTORY in the SCM Administrator screen displays ‘CPU OverLoad (%)’ or the ‘CPU Over Used by Process’ alarm, and the alarm remained for a long time.

Possible Causes

A process in the Linux system or the IPX-S300B has malfunctioned and an alarm has occurred.

Solutions

1) In the **[PERFORMANCE > Server Resources > Process]** menu, check which process is hogging the CPU.
2) If a process is hogging the CPU, select the **[PERFORMANCE > System Management > Process Management]** menu in SCM Administrator and then deactivate and reactivate the process.
3) If the process hogging the CPU is not visible in SCM Administrator, or the problem is not resolved by the procedures above, use MINICLI to restart the IPX-S300B of the system with high CPU usage. (If you are restarting the IPX-S300B of the active system, all calls attempted during the switchover will fail.)

4) If there is no particular process hogging the CPU, you can establish a telnet connection to the device and execute the 'top' command to check which command is hogging the CPU.

8.1.1.4 [1204] A Hard Disk Alarm Has Occurred

Symptoms

EVENT VIEWER or ALARM HISTORY in the SCM Administrator screen shows 'Hard-Disk Over Used (%)' alarm has occurred and the alarm still remains.

Possible Causes

The hard disk usage is high and an alarm has occurred.

Solutions

If you have uploaded unnecessary files to the IPX-S300B system, delete the files.
If there are core files in the system, download them to a PC and send them to the development team. Carry out the following procedures to download the core files:

1) Log into the Linux console. Right-click on the desktop and select 'Open Terminal' to create a new window.

2) Execute the command below to log into the root account:

```
su -
```

3) Execute the command below to check whether the core files exist:

```
ls -al /DI/etc/corefiles
```

4) If the core files are listed, move to the /home/IPX-S300B directory in order to download the files to the PC:

```
mv /DI/etc/corefiles/* /scm_data
```

5) Establish an FTP connection from the PC to the IPX-S300B server and download the core files.

Core files contain information about the process status and memory usage at the time of abnormal termination of a process. These files could be very large depending on the processes and are one of the main reasons for excessive disk usage.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.1.1.5 [1205] A Memory Alarm Has Occurred

Symptoms

EVENT VIEWER or ALARM HISTORY in the SCM Administrator screen shows 'Memory Over Used (%)' alarm has occurred and the alarm still remains.

Possible Causes

This alarm occurs if the server system's memory usage reaches up to a certain level.

Solutions

In SCM Administrator, select the **[PERFORMANCE > Server Resources > Process]** menu to check which processes are hogging the memory; and use the **[PERFORMANCE > System Management > Process Management]** menu to deactivate the memory-hogging processes and reactivate them.
If there are core files in the system, download them to a PC and send them to the development team. Carry out the following procedures to download the core files.

1) Log into the Linux console. Right-click on the desktop and select 'Open Terminal' to create a new window.

2) Execute the command below to log into the root account:

```
su -
```

3) Execute the command below to check whether the core files exist:

```
ls -al /DI/etc/corefiles
```

4) If the core files are listed, move to the /home/IPX-S300B directory in order to download the files to the PC:

```
mv /DI/etc/corefiles/* /scm_data
```

5) Establish an FTP connection from the PC to the IPX-S300B server and download the core files.

Core files contain information about the process status and memory usage at the time of abnormal termination of a process. These files could be very large depending on the processes and are one of the main reasons for excessive disk usage.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.1.2 SCM Administrator

8.1.2.1 [1303] Administrator Does Not Work Correctly

Symptoms

The system has been upgraded with a new package, but the new features or items are not shown.

Possible Causes

The newly downloaded content has not been reflected in the system.

Solutions

Change the Java cache options in the Control Panel to allow new downloads.
Select **[Start > Control Panel > Java > Temporary Internet Files > Settings… > Delete Files]**.

8.1.2.2 [1304] Cannot Install Administrator

Symptoms

You have accessed the IPX-S300B from a web browser but a SCM Administrator does not start.

Possible Causes

Java is not installed. To verify that Java has been installed, check that the Java icon exists in **[Start > Control Panel]**.

Solutions

Download and install Java SE Runtime Environment (JRE) from Java web site as following.
(http://java.sun.com > downloads > Java SE-Java Platform > JRE)

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.1.2.3 [1305] Administrator Does Not Run in a Web Browser

Symptoms

You have accessed the IPX-S300B from a web browser (Firefox or Chrome) but a SCM Administrator does not start immediately.

Possible Causes

Each browser starts Java Web starts in a different way, and your browser may require additional steps.

Solutions

Administrator starts automatically in Internet Explorer.
For Firefox, select **[Open File Dialog > Open (Java Web Start Launcher) > OK]**.
For Chrome, admin.jnlp will be shown in the downloads list at the bottom of the screen. When the download is complete, click the file to start it.

8.1.2.4 [1306] Cannot Log Into Personal Assistant with SSO

Symptoms

Cannot log into Personal Assistant Webpage with Single Sign On (SSO).

Possible Causes

The SSO module is not supported by browsers other than Internet Explorer.

Solutions

Use Internet Explorer as your browser.

8.1.2.5 [1307] Administrator Password Has Been Forgotten

Symptoms

You have forgotten the password for SCM Administrator login.

Possible Causes

- The password has been incorrectly entered by user mistake (Caps lock is on or the Shift key is pressed).
- The user does not remember the correct password.

Solutions

- Check whether Caps lock is on or the Shift key is pressed down, and then enter the password again.
- Log in with another administrator account and reset the password.
- Request technical support.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.1.2.6 [1308] Alarms That Have Occurred Are Not Displayed in Administrator

Symptoms

When an alarm occurs in SCM Administrator, it is not displayed in Event Viewer.

Possible Causes

The alarm was not reported to SCM Administrator.

Solutions

In **[PERFORMANCE > Fault > History > Alarm/Fault/Status]**, confirm if there is alarm/fault/status history.
Request technical support.

8.1.2.7 [1309] No Email is Sent When an Alarm Occurs

Symptoms

When an alarm occurs in SCM Administrator, no email is sent.

Possible Causes

The email sending option is turned off in alarm settings, or the email settings are incorrect.

Solutions

1) In **[PERFORMANCE > Fault > Setting > Alarm/Fault/Status]**, enable E-mail Flag for desired items.
2) In **[PERFORMANCE > Fault > E-mail Notification Setup]**, enter information of SMTP Server, Auth Login, and Address and save.

8.1.2.8 [1310] Querying the System Version

Symptoms

You wish to query the system version information of the IPX-S300B installed with SCM Administrator.

Possible Causes

Not Applicable

Solutions

See the information in **[Help > About]**.
See the version information in **[PERFORMANCE > System Management > Process Version]**.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.1.2.9 [1311] Learning More About Process Diagram

Symptoms

One or more processes are highlighted in red on the main monitor of SCM Administrator, and you wish to know more about them.

Possible Causes

Not Applicable

Solutions

If a process is highlighted in red, it means that the process has a problem. Move your mouse pointer to the highlight to view more information in a new window.
If you wish to view more detailed information, view it in **[PERFORMANCE > System Management > Process Management]**.

8.1.2.10 [1312] Turning the Alarm Sound Off

Symptoms

The alarm sound keeps on playing even after the issue is already notified.

Possible Causes

The alarm sound is set for continuous playback.

Solutions

Select **[Main Menu > File > Sound]** and select alarm count. Enter a number of times to play or select disable.

8.1.2.11 [1313] Force Quitting SSO Agent

Symptoms

SSO Agent is no longer needed but keeps running and you do not know how to quit it.

Possible Causes

Not Applicable

Solutions

Look for SSO Agent in the windows tray and right-click on it to bring up a menu.
Select 'Quit' on the menu to quit SSO Agent.

8.1.2.12 [1315] Cannot Add Subscriber

Symptoms

No more subscribers can be added.

Possible Causes

- The maximum number specified in the license has been exceeded.
- Multiple devices are set when the single device setting is used.
- Mandatory values are not entered.

Solutions

- Check your license. Re-purchase your license for the additional number of subscribers not covered by your current license.
- Check the device settings.
- Check that the mandatory values (the items highlighted in blue on the screen) are entered.

8.1.2.13 [1316] Cannot Create Non-Subscriber Data

Symptoms

No data can be created.

Possible Causes

- Key data is entered in duplicate.
- Mandatory values are not entered.

Solutions

- Check that no duplicate data is entered.
- Check that the mandatory values (the items highlighted in blue on the screen) are entered.

8.1.2.14 [1317] The ‘Evaluation License Expire’ Alarm Has Occurred

Symptoms

The Evaluation License Expire alarm is displayed on the screen.

Possible Causes

The evaluation license initially installed has expired.

Solutions

Provide your vendor with the MAC address information of the device installed with IPX-S300B, obtain a new license and enter the new license in **[CONFIGURATION > Resource > Activation Key]**.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.2 Call Manager Features

The troubleshooting lists are following:

Type	Troubleshooting List
Call Routing	[2001] Cannot Make Outbound Trunk Call [2002] Cannot Receive Inbound Trunk Calls To the Number Specified as CLI or DID [2003] Inbound Trunk Calls Are Rejected [2004] Cannot Connect Inbound Trunk Calls For Some Subscribers [2005] Cannot Receive Inbound Trunk Calls [2006] Outbound Trunk Number is Different From the dialed number [2007] External Trunk Call Forward Fails For Inbound Trunk Call
Call Features	[2101] Status Lamp For a Phone Does Not Change Correctly [2102] Cannot Make Calls by Pressing the Keys on the Phone [2103] Secretary Function: Status of Other Phones is Not Shown Correctly [2104] Secretary Function: Shared Call Retrieve is Not Visible [2105] Multiline is Not Correctly Shown on the Phone [2106] Multi-ring is Not Played Correctly. [2107] Cannot Call Hunt Group Numbers
Voice Path Connection	[2201] Cannot Hear On-Hold Sound Source [2202] Announcement Language is Different to the Language Selected [2203] Cannot Hear Announcement for Trunk Calls
Security (TLS/sRTP)	[2301] Cannot Register Phones When Using TLS [2302] Cannot Interoperate with the Service Provider's SIP Server When Using TLS [2303] Cannot Establish Calls Between Phones Using sRTP [2304] Cannot Establish Calls Between Phone and Gateway Using sRTP [2305] Cannot Establish Calls Between Phones and the Service Provider's SIP Server Using sRTP
SMDR (CDR)	[2401] Cannot Create CDR Data [2402] Cannot Send CDR Data via FTP [2403] Cannot Send CDR Data via TCP

8.2.1 Call Routing

8.2.1.1 [2001] Cannot Make Outbound Trunk Call

Symptoms

Outbound trunk calls fail.

Possible Causes & Solutions

Cause	Solution
Access code is not set.	Check that the code for trunk selection is set under **[CONFIGURATION > Trunk Routing > Access Code]**. Select a type according to the method of using the trunk code.
Location Based Routing is not set.	Check whether call route type setting for the location is missing under **[CONFIGURATION > Routing > Location Based Routing]**. Location Based Routing must be set for all locations.
Route sequence is not set.	Check that the route sequence is set under **[CONFIGURATION > Trunk Routing > Priority Routing]** or **[CONFIGURATION > Trunk Routing > Load Balance Routing]**.
Trunk route lock is enabled for the trunk route.	If trunk Route Lock is enabled for the route disable it at **[CONFIGURATION > Trunk Routing > Route]**.
The route is not created or not registered.	If the Register Type for the route is not None, check whether the registration information for the route exists under **[PERFORMANCE > Registration Status]**. If it does not exist, check the registration information of the route again.

8.2.1.2 [2002] Cannot Receive Inbound Trunk Calls To the Number Specified as CLI or DID

Symptoms

Inbound trunk calls are not connected to the number specified as CLI Routing or DID Routing.

Possible Causes & Solutions

Cause	Solution
Call rejection is set for the CLI.	Check Call Reject value of CLI Routing under **[CONFIGURATION > Trunk Routing > CLI Routing]**. This value should be Disable for call connection.
Inbound CLI is Anonymous	Change Anonymous Call Reject to Disable under **[CONFIGURATION > Trunk Routing > Route].**
Called number translation is set for the DID.	Check the settings for deletion length and insertion digits under **[CONFIGURATION > Trunk Routing > DID Routing]**.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Cause	Solution
Call forwarding service is set for the called user.	Check whether call forwarding service is set for the called user under **[CONFIGURATION > Service > Feature Service > Service Activation]** and disable the service.
Called Number length is over 20 digits	If Called Number Length over 20 digits, IPX-S300B reject this number.

8.2.1.3 [2003] Inbound Trunk Calls Are Rejected

Symptoms

Inbound trunk calls are rejected.

Possible Causes

A call restriction policy is set.

Solutions

Check whether a call restriction policy is applied to the called user. If no call restriction policy is applied to the user, check whether a call restriction policy is applied to the user's service group or user group. Call restriction policies are applied in the priority order of user's call restriction policy > service group's call restriction policy > user group's call restriction policy.

Following items are activated in RingPlan. After you check Ring Plan, Please Check the top-priority call restriction policy under **[CONFIGURATION > Trunk Routing > Toll Restriction Policy]** and **[CONFIGURATION > Trunk Routing > Toll Restriction Policy]**.
If a restriction policy is applied for the prefix of the inbound trunk call, delete it or create a new call restriction policy and apply it to the user so that the inbound trunk calls are not rejected.

8.2.1.4 [2004] Cannot Connect Inbound Trunk Calls for Some Users

Symptoms

Inbound trunk calls are not connected for some of the users.

Possible Causes & Solutions

Cause	Solution
The called user does not exist or is not registered.	Check whether the registration information for the user is visible under **[PERFORMANCE > Registration Status]**. If it is not visible, check the registration information of the user again. Check that the registration information for the user is created under **[CONFIGURATION > User > Single Phone User/Multi-Phone**

Cause	Solution
	User]. Check that usage restriction for the user is set to 'None' under **[CONFIGURATION > User > Single Phone User/Multi-Phone User]**.
A call restriction policy is set for the user.	Check whether a call restriction policy is applied to the user. If no call restriction policy is applied to the user, check whether a call restriction policy is applied to the user's service group or user group. Call restriction policies are applied in the priority order of user's call restriction policy > service group's call restriction policy > user group's call restriction policy. Check the top-priority call restriction policy under **[CONFIGURATION > Trunk Routing > Toll Restriction List/Toll Restriction Policy]**. If a restriction policy is applied for the prefix of the inbound trunk call, delete it or create a new call restriction policy and apply it to the user so that the inbound trunk calls are not rejected.

8.2.1.5 [2005] Cannot Receive Inbound Trunk Calls

Symptoms

Inbound trunk calls cannot be received.

Possible Causes & Solutions

Cause	Solution
The Route is not created or registered.	Check that the Route Lock is set under **[CONFIGURATION > Trunk Routing > Route]**. If the Route Lock is not a 'Outbound Locked', check whether the registration information for the Route is visible under **[PERFORMANCE > Registration Status]**. If it is not visible, check a configuration of the Route again.
The Route Lock is set to 'Outbound Locked' under **[CONFIGURATION > Trunk Routing > Route]**.	Configure the Access Code for the outbound trunk calls under **[CONFIGURATION > Trunk Routing > Access Code]**.
If the Route Lock is set to 'Outbound Locked', change to 'None'.	
The Access Code is not set.	

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.2.1.6 [2006] Outbound Trunk Number is Different From the dialed number

Symptoms

Outbound trunk number is different from the dialed number.

Possible Causes & Solutions

Cause	Solution
Access code type is set to Normal.	When 'Normal' is selected for the number type under **[CONFIGURATION > Trunk Routing > Access Code]**, the trunk code is removed from the called number. If you want to use the dialed number as the actual called number, set the number type to Internal.
Number translation is set for the call route setting.	An OutBound MCN is set under **[CONFIGURATION > Trunk Routing > Priority Routing]** or **[CONFIGURATION > Trunk Routing > Load Balance Routing]**. You can view the number translation setting information under **[CONFIGURATION > Trunk Routing > OutBound MCN (Outbound Called Number)]**.

8.2.1.7 [2007] External Trunk Call Forward Fails For Inbound Trunk Call

Symptoms

External call forward fails for inbound trunk call.

Possible Causes & Solutions

Cause	Solution
The trunk type for the endpoint is set to Normal	Change this type to Tie for inbound trunk. If the trunk type is set to Normal under **[CONFIGURATION > Trunk Routing > TIE Trunk]**, calls from the trunk are not allowed to go out to the trunk again.
A user with the same trunk number already exists.	Check that the registration information for the user is created under if Single phone user, **[CONFIGURATION > User > Single Phone User]** or if Multi-Extension phone, **[CONFIGURATION > User > Multi-Extension Phone User]** or if Multi phone User, **[CONFIGURATION > User > Multi-Phone User]**. Delete the user. User numbers and trunk numbers must be different.

8.2.2 Call Features

8.2.2.1 [2101] Status Lamp For a Phone Does Not Change Correctly

Symptoms

The status lamp for a phone does not change even when it is in use.

Possible Causes

- There is an error with device button assignment.
- Phone configuration information is not received.

Solutions

Check that the lamp number is correctly set under **[CONFIGURATION > User > Phone Key Programming]**.
Create User Profile under **[CONFIGURATION > Phone Setting > Upgrade Software]** to receive the configuration information again.

8.2.2.2 [2102] Cannot Make Calls by Pressing the Keys on the Phone

Symptoms

No calls are made when the keys on the phone are pressed.

Possible Causes

The phone is not registered under **[PERFORMANCE > Registration Status]**.

Solutions

Check that the phone is correctly registered under **[PERFORMANCE > Registration Status]**.
If it is not registered, register the phone again. If it is a Samsung phone, press the **[MUTE]** button.

8.2.2.3 [2103] Secretary Function: Status of Other Phones is Not Shown Correctly

Symptoms

The status of other phones is not shown correctly.

Possible Causes

- The phone is not registered under **[PERFORMANCE > Registration Status]**.
- Call Appearance is not set under **[CONFIGURATION > User > Multi-Phone User]**.
- Call Appearance is not set to SCA under **[CONFIGURATION > User > Multi-Phone User]**.
- Configuration information of the phone is not received.

Solutions

Set Call Appearance to SCA under **[CONFIGURATION > User > Multi-Phone User]**.

8.2.2.4 [2104] Secretary Function: Shared Call Retrieve is Not Visible

Symptoms

Shared Call Retrieve does not work.

Possible Causes

- Feature code is not set for Shared Call Retrieve under **[CONFIGURATION > Service > Feature Service > Feature Code]**.
- Call Appearance is not set under **[CONFIGURATION > User > Multi-Phone User]**.
- Call Appearance is not set to SCA under **[CONFIGURATION > User > Multi-Phone User]**.
- Configuration information of the phone is not received.

Solutions

Add the feature code for Shared Call Retrieve under **[CONFIGURATION > Service > Feature Service > Feature Code]**.
Set Call Appearance to SCA under **[CONFIGURATION > User > Multi-Phone User]**.

8.2.2.5 [2105] Multiline is Not Correctly Shown on the Phone

Symptoms

Multiline is not correctly shown on the phone

Possible Causes

Only one user is allocated under **[CONFIGURATION > User > Multi-Extension Phone]**. Configuration information of the phone is not received.

Solutions

Allocate multiple users under **[CONFIGURATION > User > Multi-Extension Phone]**. Create User Profile under **[CONFIGURATION > Phone Setting > Phone Profile Information]** to receive the configuration information again.

8.2.2.6 [2106] Multi-ring is Not Played Correctly.

Symptoms

Multi-ring does not play correctly.

Possible Causes

User type is not set to Multi Device under **[CONFIGURATION > User > Multi-Phone User]**. Configuration information of the phone is not received.

Solutions

Set the user type to Multi Device under **[CONFIGURATION > User > Multi-Phone User]**. Create User Profile under **[CONFIGURATION > Phone Setting > Phone Profile Information]** to receive the configuration information again.

8.2.2.7 [2107] Cannot Call Hunt Group Numbers

Symptoms

Calls cannot be made to hunt group numbers.

Possible Causes

- The hunt group is not created under **[CONFIGURATION > Service > Group Service > Hunt Group]**.
- Group members for the hunt group are not assigned under **[CONFIGURATION > Service > Group Service > Hunt Group]**.

Solutions

Create a hunt group and assign the group members under **[CONFIGURATION > Service > Group Service > Hunt Group]**.

8.2.3 Voice Path Connection

8.2.3.1 [2201] Cannot Hear On-Hold Sound Source

Symptoms

The on-hold sound source cannot be heard.

Possible Causes

Check that MOH use is enabled for the user group.

Solutions

Select the user group and enable MOH use under **[CONFIGURATION > User Group > Change User Group > Information]**.

8.2.3.2 [2202] Announcement Language is Different to the Language Selected

Symptoms

The announcement language is different to the language of your country.

Possible Causes

Check that the announcement language setting is correct.

Solutions

Check the language setting under **[CONFIGURATION > Announcement > Language]**.

8.2.3.3 [2203] Cannot Hear Announcement for Trunk Calls

Symptoms

The announcement cannot be heard for trunk calls.

Possible Causes & Solutions

Cause	Solution
No announcement is played for incoming trunk calls.	Enable announcement for incoming trunk calls under **[CONFIGURATION > Trunk Routing > Route]**.
No announcement is played for outgoing trunk calls.	Enable announcement for outgoing trunk calls under **[CONFIGURATION > Trunk Routing > Route]**.

8.2.4 Security (TLS/sRTP)

8.2.4.1 [2301] Cannot Register Phones When Using TLS

Symptoms

Phones cannot be registered when using TLS.

Possible Causes

You need to check the device protocol settings.

Solutions

Change the protocol list of the subscriber's device to TLS under **[CONFIGURATION > User > Single Phone User/Muti-Extention Phone]**.

8.2.4.2 [2302] Cannot Interoperate with the Service Provider's SIP Server When Using TLS

Symptoms

You cannot interoperate with the service provider's SIP server when using TLS.

Possible Causes

You need to check the protocol settings in endpoint configuration.

Solutions

Under **[CONFIGURATION > Trunk Routing > Route]**, select the endpoint for the SIP server and change the protocol to TLS.

8.2.4.3 [2303] Cannot Establish Calls Between Phones Using sRTP

Symptoms

Calls cannot be established between phones using sRTP.

Possible Causes

You need to check the device media type settings.

Solutions

Change the media type of the subscriber's device to sRTP under **[CONFIGURATION > User > Single Phone User/Multi-Extension Phone]**.

8.2.4.4 [2304] Cannot Establish Calls Between Phone and Gateway Using sRTP

Symptoms

Calls cannot be established between a phone and the gateway using sRTP.

[Possible Causes & Solution]

Cause	Solution
You need to check the device media type settings.	Change the media type of the subscriber's device to sRTP under **[CONFIGURATION > User > Single Phone User/Multi-Extension Phone]**.
You need to check the sRTP setting for the gateway.	Enable the sRTP function for the gateway.

8.2.4.5 [2305] Cannot Establish Calls Between Phones and the Service Provider's SIP Server Using sRTP

Symptoms

Calls cannot be established between a phone and the service provider's SIP server using sRTP.

Possible Causes

You need to check the device media type settings.

Solutions

Change the media type of the subscriber's device to sRTP under **[CONFIGURATION > User > Single Phone User/Multi-Extension Phone]**.

8.2.5 SMDR (CDR)

8.2.5.1 [2401] Cannot Create CDR Data

Symptoms

CDR data is not created.

Possible Causes

Option 'None' is selected for the billing data creation method.

Solutions

Change the billing data creation method for the user group from 'None' to another option.

8.2.5.2 [2402] Cannot Send CDR Data via FTP

Symptoms

CDR is created for FTP transmission, but the CDR file cannot be sent to the FTP charging server.

Possible Causes

The CDR FTP IP address or the password in the charging FTP configuration is incorrect.

Solutions

Enter the CDR FTP IP address or the password in the charging FTP configuration again and check the network connection with the FTP charging server.

8.2.5.3 [2403] Cannot Send CDR Data Via TCP

Symptoms

CDR is created for TCP transmission, but the CDR file cannot be sent to the TCP charging server.

Possible Causes

The charging TCP link IP address in the TCP charging configuration is incorrect.

Solutions

Check the CDR TCP link IP address in the TCP charging configuration. If the address is correct, check the network connection with the TCP server.

8.3 Application Features

The troubleshooting lists are following:

Type	Troubleshooting List
ACD	[3001] Cannot Call ACD Group [3002] No ACD Group Announcement is Heard [3003] Overflowing Does Not Work For ACD Group [3004] Agent Cannot Login [3005] Agents Cannot Hear Ring [3006] Break is Not Set for Agent [3007] Wrap-up is Not Set for Agent After a Call
Conference	[3101] Conference Function Does Not Work [3102] Some Conference Function Does Not Work [3103] Meet-Me Conference Function Does Not Work [3104] Only the Meet-Me Conference Function Works [3105] Cannot Send Meet-Me Conference Reservation Mails [3106] Cannot Enter Meet-Me Conference Before Start Time [3107] Conference-Related Feature Codes Do Not Work When Changed
Voice Mail	[3201] Cannot Access UMS When Logged Into Mailbox [3202] AA Announcement is Played When Logged Into Mailbox [3203] Cannot Register in Outlook [3204] The Mailbox Password Has Been Forgotten [3205] New Voice Mail is Not Emailed to the Email Account [3206] Call Record Does Not Work [3207] Park & Overhead Paging Function Does Not Work [3208] When Listening to a Voice File Recorded in Voice Studio, I Hear Nothing or a Lot of Noise [3209] Recording of Personal Greetings is Not Allowed From the Menu
Interoperates with CSTA Applications	[3301] Cannot Connect CSTA Application [3302] Cannot Register CSTA Monitor [3303] CSTA Events Are Not Generated [3304] CSTA Commands Do Not Work

8.3.1 ACD

8.3.1.1 [3001] Cannot Call ACD Group

Symptoms

When attempting to call an ACD group number, you do not hear a ringback tone or an ACD group announcement but you get an error.

Possible Causes

- The ACD group number is not registered.
- The ACD group number is not registered in the DID table.
- In case of a call between user groups, the inter user group number is not registered.
- All of the agents of the ACD group are logged out and 'Next Destination When All Log-out' is not registered.
- All of the agents of the ACD group are busy or unavailable for answering calls and 'Next Destination When All Busy' is not registered.

Solutions

1) In SCM Administrator, check that the group number is correctly registered under **[CONFIGURATION > Application > ACD > ACD Group]**.
2) In case of an incoming trunk call, check that the ACD group number is registered under **[CONFIGURATION > Trunk Routing > DID Routing]**.
3) If calls within a user group work but calls between different user groups do not work, check that the inter user group number is correctly registered under **[CONFIGURATION > Application > ACD > ACD Group]**.
4) If all of the agents of the ACD group are logged out, check that a phone number is specified for 'Next Destination When All Log-out' under **[CONFIGURATION > Application > ACD > ACD Group]**.
5) If all of the agents of the ACD group are busy, check that a phone number is specified for 'Next Destination When All Busy' under **[CONFIGURATION > Application > ACD > ACD Group]**.

8.3.1.2 [3002] No ACD Group Announcement is Heard

Symptoms

When calling an ACD group, you hear a ringback tone but not the ACD announcement.

Possible Causes

- The MOH service of the system is not functioning correctly.
- Announcement settings for the ACD group are incorrect.
- The voice file is not functioning correctly.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

Solutions

1) Check that the MOH of the system is functioning correctly. Press the **[Hold]** button on the phone while on a call with another extension subscriber. Check that the MOH is heard on the other person's phone. If not, check that the status for MRAB, MRMB, and MOHB blocks are correct under **[CONFIGURATION > System Management > Process Management]**.
2) In SCM Administrator, check that 'First Greet Message', 'First Wait MOH', 'Second Greet Message', 'Second Wait MOH', 'Greet Iteration', 'Wait MOH Duration', etc. are correctly entered under **[CONFIGURATION > Application > ACD > ACD Group]**.
3) In SCM Administrator, check that the voice file is correctly played under **[CONFIGURATION > Announcement > Service Announcement]**.

8.3.1.3 [3003] Overflowing Does Not Work For ACD Group

Symptoms

Calls should be forwarded to a specified number when the call forward conditions are met for the ACD group, but calls are not forwarded and you get an error.

Possible Causes

- The call forward phone number is not registered for the ACD group.
- The call forward phone number is invalid.
- The service is restricted for the maximum number of calls forwarded.

Solutions

1) If you wish to have incoming ACD group calls forwarded when they are unanswered after ringing for a specified period of time, check that 'Next Destination Overflow Time' and 'Next Destination' are entered correctly under **[CONFIGURATION > Application > ACD > ACD Group]**.
2) If you wish to have calls forwarded when all agents are logged out, check that the login status of the agents is Log-out under **[CONFIGURATION > Application > ACD > ACD Group Status]** and check that 'Next Destination When All Log-out' is entered correctly under **[CONFIGURATION > Application > ACD > ACD Group]**.
3) If you wish to have calls forwarded when all agents are busy, check that 'Next Destination When All Busy' is entered correctly under **[CONFIGURATION > Application > ACD > ACD Group]**.
4) If the call forwarding phone number is in another ACD group, the maximum number of calls forwarded is restricted by 'Maximum Overflow Call Count' under **[CONFIGURATION > Application > ACD > ACD Group]**. Check that the maximum number of calls is not exceeded.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.3.1.4 [3004] Agent Cannot Login

Symptoms

Login fails when an agent attempts to log into an ACD group using the feature code or the function key.

Possible Causes

- The feature code is invalid.
- The Agent ID is not registered.
- The agent password is incorrect.
- The feature code, agent ID and password have been entered in an incorrect sequence.
- The ACD group number is incorrect or the agent is not a member of the ACD group.
- You are using a multi-device or multi-line phone to login.

Solutions

1) Check that the 'ACD Agent-Login' feature code is correctly registered under **[CONFIGURATION > Service > Feature Service > Feature Code]**.
2) Check that the agent ID and password are correctly registered under **[CONFIGURATION > Application > ACD > ACD Agent]**.
3) Check that the numbers are entered in the sequence of: feature code + agent ID + password + ACD group number. The ACD group number is optional. When the numbers are dialed as feature code + agent ID + password, and without an ACD group number, the agent will be logged into all of their ACD groups. When an ACD group number is entered, the agent will only be logged into that particular ACD group.
4) Agents cannot login using multi-device or multi-line phones. Check that the subscriber type of the phone used for login is Single Device under **[CONFIGURATION > user > Single Phone User]**.

8.3.1.5 [3005] Agents Cannot Hear Ring

Symptoms

When calling an ACD group, the caller hears the ringback tone and the ACD announcement, but the agent cannot hear the ring even when agents are available.

Possible Causes

Agent status is invalid.

Solutions

1) Check the login status of the agent concerned under **[CONFIGURATION > Application > ACD > ACD Group Status]**.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

2) Check that the status of the agent is normal under **[CONFIGURATION > Application > ACD > ACD Agent]**. The Wrap-up status must be Reset and the Break status must be Reset in order to receive calls. If any of them is Set, use the feature codes or function keys on the phone to register as Reset.
3) If the agent's phone status is Idle, check whether the agent's phone is engaged in any call under **[PERFORMANCE > Call Management]**. If the agent's phone is Idle and you see its phone number on this list, you must delete the call to return the phone to the Idle status.

8.3.1.6 [3006] Break is Not Set for Agent

Symptoms

When an agent attempts to use the Break function, you get an error.

Possible Causes

- The agent is logged out.
- The feature code is not registered in the system.

Solutions

1) Check the login status of the agent concerned under **[CONFIGURATION > Application > ACD > ACD Group Status]**.
2) Check that the 'ACD Agent Break Status-Set' feature code is correctly registered under **[CONFIGURATION > Service > Feature Service > Feature Code]**.

8.3.1.7 [3007] Wrap-up is Not Set for Agent After a Call

Symptoms

When an agent has finished responding to an incoming ACD group call, or when an agent does not answer a call, the wrap-up status should be turned on, but it is not.

Possible Causes

Release wrap-up time is incorrectly registered.

Solutions

1) In the **[CONFIGURATION > Application > ACD > ACD Group]** menu, check that the release wrap-up time is correctly registered.
2) When in wrap-up status, the agent can use the feature code or function key to extend the wrap-up period. To enable this, check that the 'ACD Agent Wrap-Up Status-Set' feature code is correctly registered under **[CONFIGURATION > Service > Feature Service > Feature Code]**. When the agent extends the wrap-up period, the system does not automatically release this and the agent must use the feature code to release it manually.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.3.2 Conference

8.3.2.1 [3101] Conference Function Does Not Work

Symptoms

Conference does not start when calling using the conference feature code or when pressing the conference button during a call.

Possible Causes

- The conference server has not been configured.
- The application server service group has not been configured.
- The feature code is changed after the conference server is created.

Solutions

In the **[CONFIGURATION > Application > Conference Server]** menu, select the user group with the problem and click the **[Search]** button.
Check whether Internal Conference or Gateway Conference exist in the search result. If not, create a conference server for the user group.
If a conference server has already been created, confirm the conference server assignment form the **[CONFIGURATION > Application > Application Server Service Group]** menu.
If everything is OK, select the conference server and click the **[Change]** button. Verify the services are assigned properly in the Service List. Click the **[Change]** button again to refresh the conference server information.

8.3.2.2 [3102] Some Conference Function Does Not Work

Symptoms

Among Add-on, One-Step, Meet-me and Paging, specific type(s) of conferences does not work.

Possible Causes

- When you create a Conference Server, you didn’t select those services from the Service List.
- The conference service(s) is not enabled for the user group.
- The conference service(s) is not assigned in the Application Server Service Group.

Solutions

1) In the **[CONFIGURATION > Application > Conference Server]** menu, click the **[Search]** button to display the user group list and select the Conference Server with the problem
2) Click the **[Change]** button to check whether the required services are selected.
   - Paging: Station Paging, Paging On Answer
   - Meet-Me Conference: Meet-Me
   - One-Step Conference: Predefined Conference, Progressive Conference, Intercom Conference, Dispatch Conference
   - Add-On Conference: Add-On Conference, Conference On Answer
3) In the **[CONFIGURATION > User Group > Change User Group > Information]** menu, click the **[Search]** button to display the user group list and select the user group with the problem.
4) Click the **[Detail]** button to check whether each conference service is enabled.
   - Add-On Conference: Add-On Conference, Conference On Answer, Predefined Conference, Progressive Conference, Intercom Conference, Dispatch Conference
   - Meet-Me Conference: Meet-Me
   - Station Paging: Station Paging
   - Paging On Answer: Paging On Answer
5) In the **[CONFIGURATION > Application > Application Server Service Group]** menu, verify that each conference server configured properly.

8.3.2.3 [3103] Meet-Me Conference Function Does Not Work

Symptoms

The number of attendees available for invitation is shown as 0.

Possible Causes

The license for Meet Me conference is not entered.

Solutions

In the **[CONFERENCE > System configuration]** menu, open the Misc Setting tab and check the number of voice channels for Meet Me conferences.
If this value is 0, it means you have not registered the license for Meet Me conference. You need to purchase a new Meet Me Conference License and register it.
If you have already purchased a license, enter it in the **[CONFIGURATION > Resource > Activation Key]** menu.

8.3.2.4 [3104] Only the Meet-Me Conference Function Works

Symptoms

Conference functions other than Meet Me conference do not work.
In the **[CONFERENCE > System configuration]** menu, the number of voice channels for Meet Me conferences in the Misc Setting tab is shown as 32 (for IPX-S300B).

Possible Causes

All the voice conference channels are licensed for Meet-Me conferences.

Solutions

In IPX-S300B, among the total of 32 voice conference channels, the number of channels specified by your Meet-Me conference license are reserved for Meet-Me conferences. If you have purchased and entered the Meet-Me conference license for 32 channels, which is the total number of available channels, no other conference functions except Meet-Me conferences will work. If you are not going to use all of the voice conference channels for Meet-Me conferences, you should purchase and enter your license again, taking into consideration the ratio between Meet-Me conferences and other conferences.

8.3.2.5 [3105] Cannot Send Meet-Me Conference Reservation Mails

Symptoms

When you attempt to send invitation mails after confirming your Meet Me conference reservation, a send failure error is displayed.

Possible Causes

Your e-mail settings are incorrect.

Solutions

In the **[Performance > Fault > E-mail Notification Setup]** menu, check that the values for SMTP Server and Auth Login fields are correct.

8.3.2.6 [3106] Cannot Enter Meet-Me Conference Before Start Time

Symptoms

You cannot enter a Meet Me conference, even though you have selected the advanced entrance option at the time of reservation.

Possible Causes

- Time allowed for pre-conference entrance is set to None.
- There are no more available Meet Me conference voice channels.

Solutions

In the **[CONFERENCE > System configuration]** menu, open the Misc Setting tab and check the value for the time allowed for pre-conference entrance setting. If set to None, change it to an appropriate value.
If the setting is correct, but you still cannot enter in advance, all of the Meet Me conference voice channels are being used at the time you are trying to enter. Although you may have made the correct settings to enter in advance, you can enter in advance only if there are available Meet Me conference channels.

8.3.2.7 [3107] Conference-Related Feature Codes Do Not Work When Changed

Symptoms

When you change conference-related function keys after creating a conference system server, new conference feature codes do not work.

Possible Causes

The changes are not relayed to the conference system.

Solutions

In the **[CONFIGURATION > Application > Conference Server]** menu, select the user group with the problem and click the **[Search]** button.
Select Internal Conference from the searched list and click the **[Change]** button to open the Conference Server-Change dialog. Click the **[Change]** button again to change the conference system information.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.3.3 VM/AA

8.3.3.1 [3201] Cannot Access VM When Logged Into Mailbox

Symptoms

When you press the VM feature code on the phone, you hear the announcement 'Service is unavailable'.

Possible Causes

This problem may occur if Internal UMS is not registered in the application server.

Solutions

1) Under **[CONFIGURATION > Application > Other Application Server]**, select a user group and click the **[Search]** button to check that Internal UMS is registered.
2) If Internal UMS is not registered, click the **[Create]** button to register Internal UMS.

8.3.3.2 [3202] AA Announcement is played When Logged Into Mailbox

Symptoms

When you press the VM access feature code on the phone, you do not hear the announcement 'Please enter your password'.

Possible Causes

This problem occurs when a mailbox is not created when creating the subscriber.

Solutions

1) Query **[VOICE MAIL > Open Block Table > Extension]** and check whether the extension number exists.
2) Query **[VOICE MAIL > Open Block Table > MailBox]** and check whether the mailbox exists.
3) If not, click the **[Create]** button to create the extension number and mailbox.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.3.3.3 [3203] Cannot Register in Outlook

Symptoms

- When you attempt to register the Outlook Add-in after installing it, you get the 'Failed to register' message.
- When you attempt to register the Outlook Add-in after installing it, you get the 'IP-UMS is unavailable for this user' message.
- When you attempt to register the Outlook Add-in after installing it, you get the 'Cannot access the email server' message.

Possible Causes

- The server IP address is incorrectly entered.
- You attempted to register for a subscriber who does not exist, or your application ID and password are incorrect.
- The SMTP or IMAP IP address is incorrectly entered or the email password is incorrect.
- If you wish to enter domain names for SMTP and IMAP instead of IP addresses, they must be registered in the DNS.

Solutions

1) Make sure the server IP addresses are entered correctly and try again.
2) Make sure the ID and the password are entered correctly and try again.
3) Make sure the SMTP and IMAP information is entered correctly and try again.

8.3.3.4 [3204] The Mailbox Password Has Been Forgotten

Symptoms

You attempted to login to your UMS by pressing the VM access feature code on the phone, but cannot login because you have forgotten the password.

Possible Causes

User negligence.

Solutions

In the **[VOICE MAIL > Open Block Table > MailBox]** menu, select the extension number and press the Reset User Password button in mailbox control to reset the password to the default password.
(The user's default password uses the default customer password registered as the system password under the system variable menu.)

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.3.3.5 [3205] New Voice Mail is Not Emailed to the Email Account

Symptoms

You have got a new voice mail in your mailbox but an email attachment of the voice mail is not sent to the email account you specified.

Possible Causes

Email server information is not set.

Solutions

In the **[VOICE MAIL > Open Block Table > Mailbox Class]** menu, select the currently used block (usually Standard) and entered Email Server IP address in the HOST ID field of SMTP Server tab.

8.3.3.6 [3206] Call Record Does Not Work

Symptoms

When you attempt to record while on a call with another extension number or with a trunk number, recording does not start.

Possible Causes

- Call recording settings are incorrect.
- There are no available channels in the conference system.

Solutions

1) In the **[CONFIGURATION > Application > Other Application Server]** menu, select a user group and click the **[Search]** button to check that Internal Conference is created.
   If not, click the Create button to create Internal Conference.
2) In the **[CONFIGURATION > Application > Other Application Server]** menu, select a user group and click the **[Search]** button to check that Internal UMS is created.
   If not, click the Create button to create Internal UMS.
3) In the **[CONFIGURATION > Service > Feature Service > Feature Code]** menu, select a user group and click the **[Search]** button to check that the 'Adhoc Conference', 'Call Record-Start', and 'UMS' feature codes are registered. If not, create them.
4) If there are no available channels in the conference system, you must wait until a channel becomes available before you can use the call recording function.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.3.3.7 [3207] Park & Overhead Paging Function Does Not Work

Symptoms

An incoming trunk call requested paging but the paging function does not work.

Possible Causes

- Park & overhead paging function settings are incorrect.
- There are no available channels in the conference system.

Solutions

1) In the **[CONFIGURATION > Application > Other Application Server]** menu, select a user group and click the **[Search]** button to check that Internal Conference is created.
If not, click the **[Create]** button to create Internal Conference.
2) In the **[CONFIGURATION > Application > Other Application Server]** menu, select a user group and click the **[Search]** button to check that Internal UMS is created. If not, click the **[Create]** button to create Internal UMS.
3) In the **[CONFIGURATION > Service > Feature Service > Feature Code]** menu, select a user group and click the **[Search]** button to check that the 'Adhoc Conference', 'Paging', 'UMS', and 'Parked Call Pickup' feature codes are registered. If not, create them.
4) If there are no available channels in the conference system, you must wait until a channel becomes available before you can use the call recording function.

8.3.3.8 [3208] When Listening to a Voice File Recorded in Voice Studio, I Hear Nothing or a Lot of Noise

Symptoms

You have recorded a prompt file in Voice Studio but when listening to it on the phone, you cannot hear anything or a lot of noise.

Possible Causes

This problem occurs if a wave file of incorrect format has been registered.

Solutions

The voice mail system supports wave files in the 16 bit/8 khz/mono format. Please change the file format and register it again.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.3.3.9 [3209] Recording of Personal Greetings is Not Allowed From the Menu

Symptoms

When you log onto UMS on the phone and access the '5' Personal Greeting Recording menus, you do not get the greeting recording menu (busy greeting, call block greeting, night personal greeting, etc.).

Possible Causes

Greeting recording is not allowed in Administrator.

Solutions

1) In the **[VOICE MAIL > Open Block Table > Extension]** menu, select the extension number concerned.
2) In the Caller Options Processor tab, change the greeting from 'No Greeting' to 'Basic' and save.
3) To record a busy greeting, go to the **[VOICE MAIL > Open Block Table > Extension]** menu. In the Authorization tab, set Busy greeting allowed to 'Yes' and save.

8.3.4 Interoperation with CSTA Applications

8.3.4.1 [3301] Cannot Connect CSTA Application

Symptoms

You cannot access the system from the CSTA application.

Possible Causes

- There is a problem with the IP communication between the system and the application PC.
- You have attempted to connect to an incorrect CSTA port.
- You have exceeded the CSTA license.

Solutions

1) Solve the problem by carrying out the following procedures:
2) Try ping the system from the application PC.
3) Check that the CSTA license is correctly entered and the number is adequate under **[CONFIGURATION > Miscellaneous > License]** in SCM Administrator.
4) Log into the system with the root account, execute 'pkill-SIGUSR1CALLSTSB.exe' and check that the license maximum in the '/mnt/nand1/SMB/log/CALLSTSB0.log' file has not been reached.

```
=============================================================
 Resource            Maximum   Current
----------------------------------------------------------------------
--------------------------------------
 SkTCPConn (server)            00002     00002
 CstaLinkInfo        00110     00000
 CstaMonitorInfo     00100     00000
 CstaFilterInfo      00300     00000
 License             00060     00000 ← CSTA License
```

8.3.4.2 [3302] Cannot Register CSTA Monitor

Symptoms

The CSTA monitor cannot be registered.

Possible Causes

- You have attempted to monitor an invalid number.
- You have reached the maximum monitor number or maximum filter number allowed in the system.

Solutions

1) Solve the problem by carrying out the following procedures:
2) Check that the number to monitor is valid.
3) Log into the system with the root account, execute ‘pkill-SIGUSR1CALLSTSB.exe’ and check that the maximum monitor number and maximum filter number in the ‘/mnt/nand1/SMB/log/CALLSTSB0.log’ file have not been reached.

```
=============================================================
 Resource           Maximum   Current
----------------------------------------------------------------------
--------------------------------------
 SkTCPConn (server)          00002     00002
 CstaLinkInfo       00110     00000
 CstaMonitorInfo    00100     00000 ← Number of Monitor
 CstaFilterInfo     00300     00000 ← Number of Filter
 License            00060     00000
```

8.3.4.3 [3303] CSTA Events Are Not Generated

Symptoms

The phone does not send the event messages according to its call status to the CSTA application, even though the CSTA application is connected correctly and a phone number is correctly registered to be monitor.

Possible Causes

The phone status is invalid and its events are not sent to the CSTA application.

Solutions

The CSTA application sends the SnapShot Device command to the phone to check the phone status. If the phone returns an invalid SnapShot Device Response, the system regards the phone status as invalid.
If the phone status is Idle, check whether any call is engaged for the phone number under **[PERFORMANCE > Call Management]**. If the phone is Idle and you see its phone number on this list, you must delete the call to return the phone to the Idle status.

8.3.4.4 [3304] CSTA Commands Do Not Work

Symptoms

When a phone executes a CSTA command, the command does not function correctly even though the CSTA application is connected correctly and a phone number is correctly registered to be monitored.

Possible Causes

The phone's status is invalid or it is unable to process the CSTA command.

Solutions

The system responds to any CSTA command generated by the CSTA application. You must analyze the cause value of this response.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.4 Interoperate with Phones and Gateways

The troubleshooting lists are following:

Type	Troubleshooting List
Interoperate with Phones	[4001] Cannot Connect to Network [4002] Cannot Register with Server [4003] Cannot Register with Server When Using PNP [4004] Registration is Done with Previous Server [4005] Cannot Download Profile From Server [4006] Cannot Upgrade [4007] Call is Not Established and Noise is Heard [4008] Cannot Make Calls [4009] Connection is Lost While on a Call [4010] Time Displayed is Incorrect [4011] Service Menu is Incorrect [4012] Supplementary Service Function Does Not Work [4013] UC Function Does Not Work [4014] Cannot Set Functions in Menu [4015] Cannot Use AOM [4016] Fonts Are Garbled
Interoperate with Gateways	[4101] Cannot Register Gateway [4102] Gateway Does Not Fetch Profile [4103] FXS Phone on the Gateway is Inactive

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.4.1 Interoperation with Phones

8.4.1.1 [4001] Cannot Connect to Network

Symptoms

You cannot connect to the network from the phone.

Possible Causes

- Poor LAN cable connection
- Incorrect IP settings
- VLAN information error
- 802.1x port information not set

Solutions

1) Check that the phone's LAN cable is connected correctly.
2) Check that the phone's IP address is correctly set under the phone's **[Menu > Settings > Network Information]** menu.
3) If you fail to obtain a valid IP address when using DHCP, contact the system administrator.
4) If the IP address is correctly set, your network connection may be affected by VLAN or 802.1x settings. Contact the system administrator.

8.4.1.2 [4002] Cannot Register with Server

Symptoms

You cannot register the phone with the IPX-S300B server.

Possible Causes

- Network error
- Profile download failure
- Subscriber information error
- Certificate error when using TLS protocol
- IPX-S300B server error

Solutions

1) Check that the network is functioning correctly.
2) Check that the profile is correctly downloaded.
   ① On an SMT-i5243 phone, you can check this with the ↑ icon on the desktop and the **[Menu > Settings > Network Information > Boot Log]** menu.
   ② Establish a telnet connection to the phone and check the downloaded profile in the /tmp/Provision/Profiles folder.

③ If the profile is not correctly downloaded, contact the system administrator.
④ If the profile is correctly downloaded, use Ethereal to identify the cause of registration failure. (Contact the system administrator.)
⑤ In case of an SIP 404 Not Found error, the subscriber information has not been entered in the server.
⑥ When using TLS protocol, registration may not be possible due to certificate authentication failure.

3) Registration may not be possible if the IPX-S300B server is in abnormal status. (Contact the system administrator.)

8.4.1.3 [4003] Cannot Register with Server When Using PNP

Symptoms

Server registration is not performed correctly even though using PNP.

Possible Causes

- Configuration server information error
- IP address acquisition failure
- Profile download failure
- Subscriber information error
- Certificate error when using TLS protocol
- IPX-S300B server error

Solutions

1) Configuration information must exist in the options information of DHCP in order to use PNP correctly. (Contact the system administrator.)
2) Use Ethereal to check that configuration information exists in the option number 43, which is a DHCP option field.
   ① Check that the DHCP IP address is correctly acquired.
   ② Check that the IP address is correctly set under the phone's **[Menu > Settings > Network Information]** menu.
3) Check that the profile is correctly downloaded.
   ① On an SMT-i5243 phone, you can check this with the ↑ icon on the desktop and the **[Menu > Settings > Network Information > Boot Log]** menu.
   ② Establish a telnet connection to the phone and check the downloaded profile in the /tmp/Provision/Profiles folder.
   ③ If the profile is not correctly downloaded, contact the system administrator.
   ④ If the profile is correctly downloaded, use Ethereal to identify the cause of registration failure. (Contact the system administrator.)
   ⑤ In case of a 404 Not Found error, the subscriber information has not been entered in the server.
4) When using TLS protocol, registration may not be possible due to certificate authentication failure.
5) Registration may not be possible if the IPX-S300B server is in abnormal status. (Contact the system administrator.)

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.4.1.4 [4004] Registration is Done with Previous Server

Symptoms

Registration information has been changed but the phone is registered with the previous information.

Possible Causes

- MAC profile exists in the server.
- Provisioning failure

Solutions

1) Even when the user changes the phone's login information, if the MAC profile exists in the sever, provisioning is done with the login information in the MAC profile. Therefore, you should check whether the MAC profile exists in the server.
2) When profile downloading fails, the phone uses the last valid registration information for initialization. Therefore, you should check that the profile is correctly downloaded. (See the **[Menu > Settings > Network Information > Boot Log]** menu)

8.4.1.5 [4005] Cannot Download Profile from Server

Symptoms

The profile is not correctly downloaded from the IPX-S300B server.

Possible Causes

- Network error
- Configuration server information error
- IPX-S300B server error
- Subscriber information error
- User mode setting

Solutions

1) Check that the network is functioning correctly.
2) Check that the configuration server information is correctly set.
   ① When in PNP mode, contact the system administrator.
   ② Check the configuration information of Easy Install.
3) If the profile is correctly downloaded, use Ethereal to identify the cause of registration failure. (Contact the system administrator.)
   ① Check the TFTP/HTTP file request and file transmission process.
   ② Check the server information in the boot profile.
4) In case of a HTTP authentication failure, check the IPX-S300B subscriber information.
5) When using the settings in the user mode (server type: standard), settings are not performed by downloading the profile but the user sets the server and the registration information, and initializes the phone. Check this setting mode.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.4.1.6 [4006] Cannot Upgrade

Symptoms

The system administrator has performed a software upgrade but the phone is still using the old software.

Possible Causes

- Network status (system administrator)
- Upgrade server address error (system administrator)
- Upgrade server's software package path setting error (system administrator)
- Upgrade server's software version is the same as the current phone (user/system administrator)

Solutions

1) In case of an automatic upgrade, contact the system administrator.
2) If you are the system administrator, check the causes above in sequence.

8.4.1.7 [4007] Call is Not Established and Noise is Heard

Symptoms

When a call is connected, you hear nothing but noise.

Possible Causes

- Connect negotiation failure
- sRTP processing error
- Multiple sources are sending RTP to the phone's RTP port.
- Faulty device

Solutions

1) Connect the phone and a PC to the hub to allow Ethereal.
2) Check the SIP message that negotiation for the codec information is correctly performed.
3) Convert the RTP packets to audio and check whether the source voice is noisy. (A separate tool is required for sRTP).
4) Check whether there are multiple IP addresses sending RTP to the phone's RTP port.
5) Check whether you get the same problem with the speaker and with the handset.

8.4.1.8 [4008] Cannot Make Calls

Symptoms

- The call screen is not displayed.
- When you attempt to make a call, the call is actually not made and is terminated.

Possible Causes

- Network error
- Server is not registered.
- IPX-S300B server error

Solutions

1) Check that the registration status is normal from the phone's idle screen.
2) Check that the network status is normal. (Ping the phone from a PC.)
3) Press the Mute key to register the phone again.
4) Try to make a call from another phone to check whether the IPX-S300B is correctly running the service.
5) Use Ethereal to perform a packet analysis. (Contact the system administrator.)
6) Check that the SIP INVITE message is correctly sent to the server.

8.4.1.9 [4009] Connection is Lost While on a Call

Symptoms

Calls are terminated even though you did not end the call.

Possible Causes

- Session timer is active.
- The BYE message is received from the opposite party or the server.

Solutions

1) Connect the phone and a PC to the hub and use Ethereal to analyze the packets.
2) Check whether the INVITE, 200 OK message includes parameters related to session timer.
3) If session timer is in use, check whether re-invite or update messages are sent or received before the session timer expires.
4) When the problem occurs, check whether messages such as BYE are received or sent by the phone.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.4.1.10 [4010] Time Displayed is Incorrect

Symptoms

Time shown by the phone is incorrect.

Possible Causes

- Network error
- The system's time zone is incorrect.
- The phone's time zone is incorrect.

Solutions

1) Check **[Menu > Settings > Application Setting > Time Zone > Time Update]**.
2) If set to default, contact the system administrator.
3) If set to user, check your time zone.

8.4.1.11 [4011] Service Menu is Incorrect

Symptoms

- All or part of the service menu which is called up by pressing the service key is not displayed.
- The font for the service menu is garbled.

Possible Causes

- SoftMenu profile download failure
- Class of service (COS) is not allowed for the phone number.
- Access code for performing service does not exist.
- Service is unavailable.
- Text encoding in the SoftMenu profile and text encoding in the actual display data are different.

Solutions

1) Check that SoftMenu and Service profiles are correctly downloaded during the provisioning process.
2) If the menu for a specific service is not displayed, check whether the service is activated and check its access code. (Contact the system administrator.)
3) Check the text encoding of the SoftMenu profile and the text encoding of the display data.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.4.1.12 [4012] Supplementary Service Function Does Not Work

Symptoms

When you attempt to perform a supplementary service function, it does not work.

Possible Causes

- Service is unavailable for this particular phone number.
- Class of service (COS) is not allowed.
- Access code for performing the service does not exist.
- Service is unavailable.
- Processing error by the IPX-S300B server

Solutions

1) Check that provisioning is correctly performed.
2) Check the error message displayed on the screen when performing the service.
3) Check COS, activation of supplementary services, and access code for the phone number performing the service.
4) Perform an Ethereal packet analysis to identify the location of the problem.

8.4.1.13 [4014] Cannot Set Functions in Menu

Symptoms

- Phone function settings error
- System function settings error

Possible Causes

1) Phone function settings error
   ① Settings cannot be saved in the phone’s current status.
   ② Settings are already in place and cannot be saved.
2) System settings error
   ① Network error
   ② No privilege
   ③ Provisioning failure

Solutions

1) Phone function settings error
   ① Exit the menu and set again.
   ② Try setting with another value first and then change it to a desired value.
2) System function settings error
   ① Check the Network status.
   ② Contact the system administrator.

8.4.1.14 [4015] Cannot Use AOM

Symptoms

- AOM is not registered with the phone in PnP mode.
- Nothing happens when the AOM button is pressed.

Possible Causes

- The option number 43 in the DHCP server is not set or the AOM information is not set in the phone.
- No functions are assigned to the AOM keys.

Solutions

1) Check the option number 43 in the DHCP server (system administrator). Register the MAC of the AOM in the phone (system administrator).
2) Assign a function to each key of the AOM (system administrator).

8.4.1.15 [4016] Fonts Are Garbled

Symptoms

- The font displayed on the phone's menu is garbled.
- The text displayed by the system is garbled. (Service menu, temporary service menu, multi-purpose button information, AOM information, busy information, etc.)

Possible Causes

1) The font displayed on the phone's menu is garbled.
   ① The phone's software temporarily failed to load the font table.
   ② The database file is damaged or corrupted.
2) The text displayed by the system is garbled.
   ① System text encoding type error.
   ② System database setting error.

Solutions

1) The font displayed on the phone's menu is garbled.
   ① Restart the phone.
   ② Clear or delete the functions with the garbled text and reset them (Schedule, Address book, etc.).
2) The text displayed by the system is garbled.
   ① Contact the system administrator.

8.4.2 Interoperation with Gateways

8.4.2.1 [4101] Cannot Register Gateway

Symptoms

Registration is not performed because the registration message from the gateway and the registration message stored in the IPX-S300B are different.

Possible Causes

- Registration method for endpoint in routing is incorrect.
- User name, primary proxy server IP address, etc. of endpoint in the routing are incorrect.

Solutions

- Check that the registration method for endpoint in the routing is Receive REGISTER.
- Check the user name and the primary proxy server IP address.

8.4.2.2 [4102] Gateway Does Not Fetch Profile

Symptoms

The gateway does not fetch the profile to be used in survival mode or FXS.

Possible Causes

If you do not create a gateway, you can proceed with registration but the profile will not be fetched.

Solutions

Create a gateway to use under **[CONFIGURATION > Gateway]**.

8.4.2.3 [4103] FXS Phone on the Gateway is Inactive

Symptoms

A FXS phone on the gateway is not correctly registered and cannot be used.

Possible Causes

The IPX-S300B does not change the FXS subscriber information to FXS.

Solutions

In the subscriber information, change the phone type to FXS-Phone.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.5 IPX-G500B Gateway

The troubleshooting lists are following:

Type	Troubleshooting List
Gateway Installation	[7001] Cannot Turn On Gateway [7002] LAN LED on the system is Off
IPX-S300B Interoperation Mode	[7101] Cannot Register Analog Phones with SCM [7102] Cannot Register Analog Trunk Lines with SCM [7103] Cannot Dial or Receive Calls on Analog Phones [7104] Cannot Dial or Receive Calls on Analog Trunk Lines
Survival Mode	[7201] Cannot Dial or Receive Calls on Analog Phones [7202] Cannot Dial or Receive Calls on SIP Phones [7203] Cannot Dial or Receive Calls onl Analog Trunk Lines [7204] Cannot Hear Voice When on Calls with Analog Phones [7205] Cannot Hear Voice When on Calls with SIP Phones [7206] Cannot Hear Voice When on Calls with Analog Trunk Lines

8.5.1 Gateway Installation

8.5.1.1 [7001] Cannot Turn On Gateway

Symptoms

The IPX-G500B gateway cannot be switched on.

Possible Causes

- The power cable is not connected.
- The power switch is in the off position.
- The IPX-G500B gateway's power supply is faulty.

Solutions

- Check the power cable and its connections.
- Check that the power switch of the system is in the on position.
- Contact your IPX-G500B gateway vendor for service.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.5.1.2 [7002] LAN LED on the system is Off

Symptoms

The LAN LED on the system is off.

Possible Causes

- The LAN cable is disconnected from the gateway.
- The LAN cable connected to the gateway is disconnected from the switch.
- There is a problem in the LAN cable.

Solutions

- Check that the LAN cable is connected correctly to the gateway.
- Check that the LAN cable connected to the gateway is connected correctly to the switch.
- Replace the LAN cable and connect a new cable.

8.5.2 IPX-S300B Interoperation Mode

8.5.2.1 [7101] Cannot Register Analog Phones with SCM

Symptoms

Analog phones cannot be registered with the SCM.

Possible Causes

- The IPX-G500B gateway is disconnected from the network.
- The IPX-S300B or IPX-G500B settings are incorrect.

Solutions

- Perform a ping test on the gateway system and check whether the network is normal. If it is not normal, check the network status and the firewall settings.
- Check the gateway IP setting on the **[Configuration > Gateway > IPX Setting > Configuration]**.
- Check the phone user setting on the **[Configuration > User > Single Phone User]**.
- Connect to the console port of IPX-G500B and run “cli”. Then check the IPX-S300B IP address setting that is connected.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.5.2.2 [7102] Cannot Register Analog Trunk Lines with SCM

Symptoms

Analog trunk lines cannot be registered with the SCM.

Possible Causes

- The gateway is disconnected from the network.
- The IPX-S300B or gateway system settings are incorrect.

Solutions

- Perform a ping test on the gateway system and check whether the network is normal. If it is not normal, check the network status and the firewall settings.
- Check the gateway IP setting on the **[Configuration > Gateway > IPX Setting > Configuration]**.
- Check the phone user setting on the **[Configuration > User > Single Phone User]**.
- Connect to the console port of IPX-G500B and run “cli”. Then check the IPX-S300B IP address setting that is connected.

8.5.2.3 [7103] Cannot Dial or Receive Calls on Analog Phones

Symptoms

Calls cannot be dialed or received by analog phones.

Possible Causes

- The phone is not registered with the SCM.
- The analog phone or the phone line has a problem.
- If the phone is connected to the expansion, the expansion is disconnected from the network.

Solutions

- Check whether the analog phone is registered with the IPX-S300B under **[Performance > Registration Status > Registration Status]**. If not, resolve the problem by referring to [7101].
- If you cannot hear a dial tone when you pick up the handset, check the RUN LED of the system that the phone is connected. If the system is IPX-G500B, check 2nd RUN LED is normal. In case of IPX-G520S/G540, check the network and RUN LED of it.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.5.2.4 [7104] Cannot Dial or Receive Calls on Analog Trunk Lines

Symptoms

Calls cannot be dialed or received by analog trunk lines.

Possible Causes

- The analog trunk lines is not registered with the SCM.
- The configuration for analog trunk is incorrect.

Solutions

- Check whether the analog trunk lines are registered with the IPX-S300B under **[Performance > Registration Status > Registration Status]**. If not, resolve the problem by referring to [7102].
- Check the DID is entered on **[Configuration > Trunk Route > DID Routing]**.
- Check the 'Ring Destination Number' is set on **[Configuration > Gateway > IPX Setting > Analog Trunk]**.

8.5.3 Survival Mode

8.5.3.1 [7201] Cannot Dial or Receive Calls on Analog Phones

Symptoms

Calls cannot be dialed or received by analog phones in survival mode.

Possible Causes

- The analog phone or phone line has a problem.
- The gateway system operation is not normal.

Solutions

- If you cannot hear a dial tone when you pick up the handset, replace the analog phone or the phone lines and check again.
- Check the RUN LED of the system that the phone is connected is normal. If it's connected to the IPX-G500B, check 2nd RUN LED.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.5.3.2 [7202] Cannot Dial or Receive Calls on SIP Phones

Symptoms

Calls cannot be dialed or received by SIP phones in survival mode

Possible Causes

The SIP phone is not registered with the gateway.

Solutions

- Perform a ping test on the OS7000 system and check whether the network is normal. If it is not normal, check the network status and the firewall settings.
- Select the gateway under **[Configuration > Gateway > Gateway Link Setting]** and check that the SIP phone is selected
- Registration may not be possible if the IP address used by the SIP phone on the network is used by another system. Perform a ping test on the IP address of the SIP phone or establish a connection to the web screen of the SIP phone. If there is a problem, change the IP address of the other system or change the IP address of the SIP phone.
- When switching from IPX-S300B interoperation mode to survival mode, the SIP phone should register itself automatically with the gateway, but your SIP phone may not support this feature. In this case, replace the SIP phone or you have to manually change the registration information of the SIP phone to register it with the gateway.
- If the SIP phone keeps showing the same error screen, it means that it has not sent out the registration request to the gateway. Reboot the SIP phone.

8.5.3.3 [7203] Cannot Dial or Receive Calls on Analog Trunk Lines

Symptoms

Calls cannot be dialed or received by analog trunk lines in survival mode

Possible Causes

- The phone line or FXO card has a problem.
- The gateway system operation is not normal.

Solutions

- Replace the phone line or FXO card and test again.
- Check the RUN LED of the system that the phone is connected is normal. If it’s connected to the IPX-G500B, check 2nd RUN LED.

Ошибка! Используйте вкладку "Главная" для применения 제목 1 к тексту, который должен здесь отображаться.

8.5.3.4 [7204] Cannot Hear Voice When on Calls with Analog Phones

Symptoms

Voice cannot be heard when on a call with a analog phone in survival mode

Possible Causes

- The analog phone or phone line has a problem.
- The gateway system operation is not normal.

Solutions

- Replace the analog phone or phone line and test again.
- Check the RUN LED of the system that the phone is connected is normal. If it's connected to the IPX-G500B, check 2nd RUN LED.

8.5.3.5 [7205] Cannot Hear Voice When on Calls with SIP Phones

Symptoms

Voice cannot be heard when on a call with a SIP phone in survival mode

Possible Causes

The gateway system operation is not normal.

Solutions

- Check the RUN LED of the system that the phone is connected is normal. If it's connected to the IPX-G500B, check 2nd RUN LED.
- You may not be able to hear the voice if the IP address used by the gateway system on the network is used by another system. Perform a ping test on the IP address of the gateway. If there is a problem, change the IP address of the other system or change the IP address of the gateway.

8.5.3.6 [7206] Cannot Hear Voice When on Calls with Analog Trunk Lines

Symptoms

Voice cannot be heard when on trunk calls with analog phones or SIP phones in survival mode

Possible Causes

The gateway system operation is not normal.

Solutions

- Check the RUN LED of the system that the phone is connected is normal. If it's connected to the IPX-G500B, check 2nd RUN LED.
- You may not be able to hear the voice if the IP address used by the gateway system on the network is used by another system. Perform a ping test on the IP address of the gateway. If there is a problem, change the IP address of the other system or change the IP address of the gateway.

8.6 SCM Administrator

8.6.1 Execute SCM Administrator

8.6.1.1 When display a popup of “Your security settings have blocked downloading file”

Symptoms

Blocked downloading an execute file of SCM Administrator

Possible Causes

Security setting of browser

Solutions

1) Should allow pop-up in **[Tools]** of browser.
2) In **[Tools > Internet Options < Security Setting tab]** of browser, set the security level to medium-high (default) and enable “File Download” in **[Custom Level > Downloads]**.
3) If the log-in dialog of SCM Administrator is not displayed, input the URL as follows:
   - <http://(System IP Address)> or <http://(System IP Address)/admin>

8.6.1.2 When display a popup of “Your security settings have blocked a signed application from running” after launched SCM Administrator

Symptoms

Blocked a self-signed SCM Administrator

Possible Causes

Modified a java secure execution environment by user or java upgrade.

Solutions

1) In Java Control Panel > Advanced > Secure Execution Environment, check the “Allow user to grant permissions to signed content”.
2) [Java7 and higher] In Java Control Panel > Security, change the security level to “Medium”-or-Add the site in “Exception Site List”.

ABBREVIATION

A

AA	Auto Attendant
AAR	Automatic Alternative Routing
ACD	Automatic Call Distribution
ACL	Access Control List
AME	Answering Machine Emulation
APC	Access Point Controller
AR	Alternative Route

B

BHCA	Busy Hour Call Attempt
BLF	Busy Lamp Field

C

CAC	Call Admission Control
CDR	Call Detailed Record
CID	Caller Information Data
CLI	Calling Line Identification
CLIR	Calling Line Identification Restriction
COA	Change of Address
COS	Class of Service
CPS	Call Per Second
CSTA	Computer Supported Telephony Application
CTI	Computer Telephony Interface

D

DID	Direct Inward Dial
DISA	Direct Inward System Access
DN	Directory Number
DND	Do Not Disturb
DOD	Direct Outward Dial
DR	Direct Route
DTMF	Dual Tone Multi-Frequency
DTS	Direct Trunk Select

F

FMS	Fixed Mobile Substitution
FTP	File Transfer Protocol
FXO	Foreign Exchange Official
FXS	Foreign Exchange Station

G

GW	Gateway

I

ICMP	Internet control message protocol
ITSP	Internet Telephony Service Provider
IVR	Interactive Voice Response

H

HTTP	Hyper Text Transport Protocol

L

LDAP	Lightweight Directory Access Protocol

M

MCN	Modification Calling & Called Number
MCS	Multimedia Conference System
MOBEX	Mobile Extension
MOH	Music On Hold
MP	Main Processor
MPEG	Moving Picture Experts Group
MWI	Message Waiting Indication

N

NFC	Near Field Communication
NMS	Network Management System

P

PBX	Private Branch eXchange
PSTN	Public Switched Telephone Network

R

RADIUS	Remote Authentication Dial In User Service
RFC	Request For Comments
RST	Reset
RTP	Real Time Protocol

S

SBC	Single Board Computer
SCM	Samsung Communication Manager
SIO	Serial Input and Output
SIP	Session Initiation Protocol
SMDR	Station message detailed record
SMS	Short Message Service
SNMP	Simple Network Management Protocol

T

TCP	Transmission Control Protocol
TFTP	Trivial File Transfer Protocol
TLS	Transport Layer Security
ToS	Type of Service

U

UMS	Unified Messaging System

V

VM	Voice Mail
VMS	Voice Mailing System
VoIP	Voice over Internet Protocol
VQM	Voice Quality Monitoring

W

WE	Wireless Enterprise
WiFi	Wireless Fidelity

SCM Compact (IPX-S300B)

Operation Manual

SAMSUNG