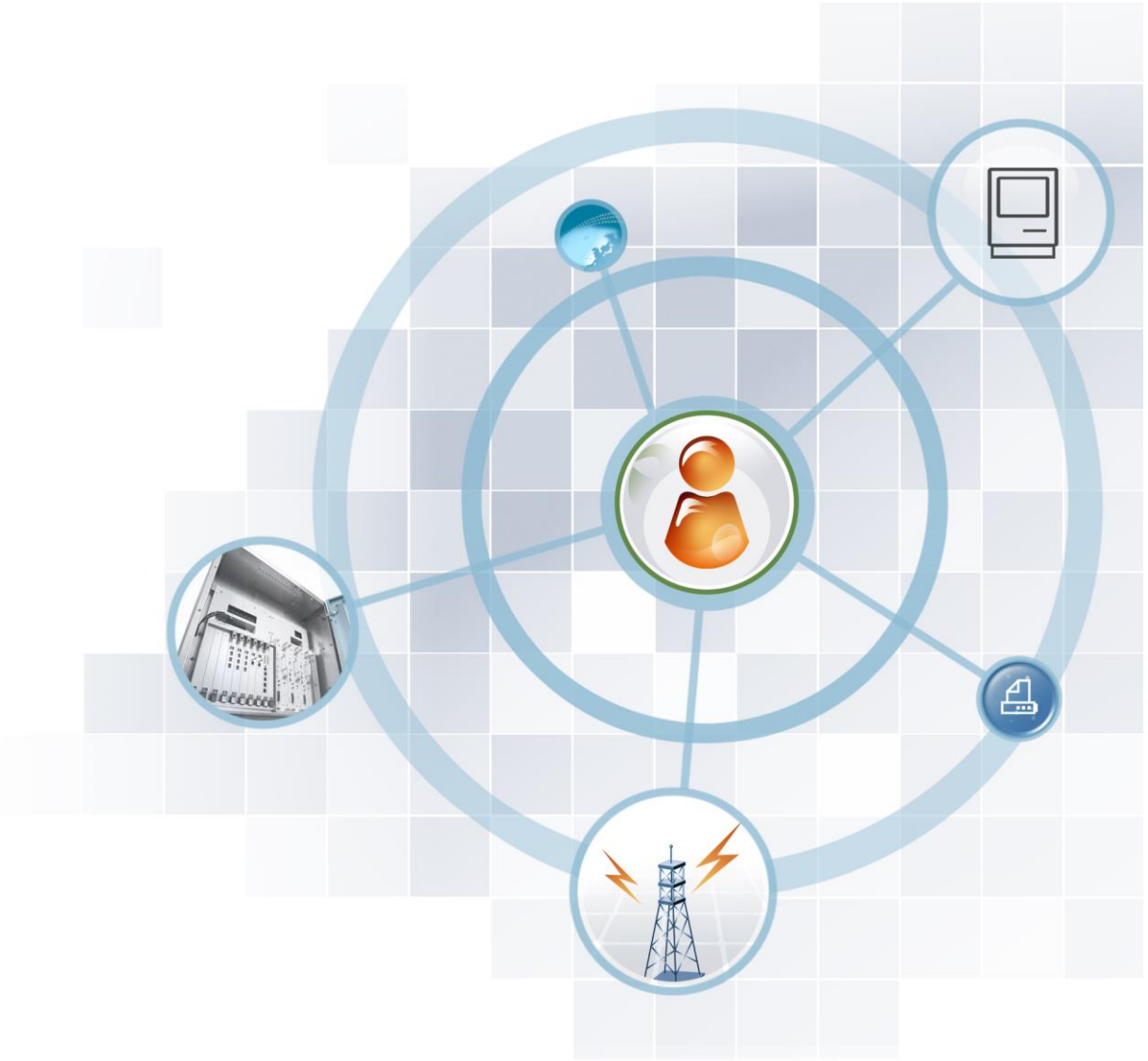
2600-009P2XGA2 Ver. 10.0

SCM Express

System Description



SAMSUNG ELECTRONICS

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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

General User Information

Radio Frequency Interference

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to FCC Part 15 Rules.
These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his/her own risk.

FCC Requirements

The SCM equipment complies with FCC Part 68 Rules and requirements adopted by Administrative Council for Terminal Attachment (ACTA).
The FCC Part 68 label is located on the bottom of the chassis. The label contains:
Product Identifier Number
FCC Registration Number
Ringer Equivalence Number (REN)

Telephone Company Interfaces

A plug and jack is used to connect this equipment to the premises wiring and telephone network must comply with the FCC Part 68 rules and requirements adopted by the ACTA. A compliant telephone cord and modular plug is provided with this product which is designed to connect to a compatible Standard Modular jack.
Connection to the telephone network should be made by using standard modular telephone jacks, type RJ-11C. The RJ-11C plug and/or jacks used must comply with the FCC Part 68 rules.

Unauthorized Modifications

Any change or modifications performed on this equipment that are not expressly approved in writing by SAMSUNG ELECTRONICS, CO., LTD. could cause non-compliance with the FCC rules and void the user’s authority to operate the equipment.

Ringer Equivalence Number

The REN is used to determine the number of devices to be connected to a telephone line. If the total allowable REN load is exceeded, the phone circuit may fail to ring. In most cases, the total REN for a telephone line should not exceed Five (5).
Contact Local Telephone Company, to be certain about the number of devices connected to a line, which is determined by the total REN.
For earlier products, the REN is separately shown on the label.

Incidence of Harm

The telephone company will notify you in advance about the temporary discontinuation of service, if this equipment is causing harm to the telephone network.
In case advance notification is not feasible, the telephone company will notify the customer as soon as possible and you will also be advised about your right to file a complaint with the FCC, if it is necessary.

Changes to Telephone Company Equipment or Facilities

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment.
If this happens, the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

Service Center

If you need assistance during troubleshooting, please contact out local office SAMSUNG ELECTRONICS, CO., LTD. for repair or warranty information.
If the trouble is causing harm to the telephone network, the telephone company may request you to remove the equipment from the network until the problem is resolved.

Field Repairs

Only technicians certified on the Ubigate iBG2016 are authorized by SAMSUNG ELECTRONICS, CO., LTD. to perform system repairs.
Certified technicians may replace modular parts of a system to repair or diagnose trouble.
Defective modular parts can be returned to SAMSUNG ELECTRONICS, CO., LTD. for repair.

General

Connection to party line service is subject to state tariffs. Contact the State Public Utility Commission, Public Service Commission or Corporation Commission for information.

Equal Access Requirements

Through the use of access codes, this equipment is capable of providing user's access to interstate providers of operator services. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operator Consumers Act of 1990.

Electrical Safety Advisory

Parties responsible for equipment requiring AC power should consider including an advisory notice in their customer information suggesting them to use a surge arrestor. Telephone companies report that electrical surges, typically lightning transients, are very destructive to customer terminal equipment connected to AC power sources. This has been identified as a major nationwide problem.

Music on Hold Warning

In accordance with US copyright laws, a license may be required from the American Society of Composers, Authors and Publishers (ASCAP) or other similar organizations if copyright music is transmitted through the Music on Hold feature

SAMSUNG ELECTRONICS, CO., LTD. hereby disclaims any liability arising out of failure to obtain such a license.

DISA Warnings

The lines used for the DISA feature must have the disconnected Supervision option provided by the telephone company.

As it is impossible to control who may access your DISA line it is suggested that you do not turn this feature on unless you intend to use it. If you do use this feature, it is good practice to frequently change pass codes and periodically review your telephone records for unauthorized use.

Safety Warnings

High touch current earth connection is essential before making telecommunication network connection.

Energy Hazard-careful treatment is needed.

Every wire for communication should be larger than 26 AWG.

Double pole/neutral fusing.

Underwriters Laboratories

The SCM has been tested to comply with Safety Standards in the United States and Canada. This system is listed with Underwriters Laboratories. The cUL Mark is separately shown on the label.

Installation Safety Guidelines and Warnings

Safety Recommendations

The Safety Warnings that appear in this document (such as the one below) indicate a procedure that can harm you if not done correctly.

Electric hazard exists. Verify the power is turned off. Do not work on energized equipment. Working on energized equipment can result in serious electrical shock.

To avoid electric shock, do not connect Safety Extra-Low Voltage (SELV) circuits (found in LAN ports) to Telephone-Network Voltage (TNV) circuits (found in WAN ports).

This equipment must be installed and maintained by properly trained service personnel. Make sure the proper electrical service is available before plugging the unit and turning it on. Disconnect the telecommunication lines before unplugging the main power connector.

Class 1 Laser Product

The SCM is equipped with Small Form Pluggable (SFP) laser transceiver on some ports.

Invisible laser radiation may be emitted from disconnected fibers or connectors.

Do not stare into beams or view directly with optical instruments.

Cover Panels

Do not operate the SCM with missing blank faceplates and cover panels. These covers prevent exposure to hazardous voltages and currents inside the chassis. They are important to maintaining proper air flow through the chassis. They also prevent electromagnetic interference (EMI) that might disrupt other equipment.

> Laser radiation and EMI are present when the router cover panel is open.

Electrostatic Discharge (ESD) Warning

Observe the following guidelines to minimize the potential for Electrostatic Discharge (ESD) damage, which can cause intermittent or complete component failures.

> When handling SCM or its components, wear grounding wrist straps to avoid ESD damage to the equipment. Do not directly touch the backplane with your hand or any metal tool, or you could shock yourself.

Always use an ESD wrist strap or ankle strap, and verify that it is in direct contact with your skin. Avoid contact between the component and your clothing as it causes ESD damage.

When handling any component that is removed from the chassis, verify that the equipment end of your ESD strap is attached to one of the ESD points on the chassis.

Use care when installing or uninstalling modules or interface cards. Tighten the captive installation screws to ensure a proper connection when inserting modules or interface cards.

When removing or installing a component, always place it component-side up on an antistatic surface, in an antistatic card rack. If you are returning a component, place it in an electrostatic bag before packing it.

INTRODUCTION

Purpose

This document describes the basic overview and features of the SCM.

Document Content and Organization

This manual consists of the following parts.

CHAPTER 1. SCM System Overview

This part describes the SCM network interface and the operation environments and capacity of the SCM.

CHAPTER 2. SCM Hardware

This part describes the characteristics of the system hardware.

CHAPTER 3. Call Manager Features

This part describes the characteristics of the call manager features.

CHAPTER 4. Application Features

This part describes the characteristics of the application features.

CHAPTER 5. System Reports

This part describes the characteristics of the system reports.

ANNEX A. Supported Specifications and RFC

ANNEX B. Open Source Announcement

ABBREVIATION

This part describes the acronyms used in this document.

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
10.0	12. 2015	Revised for SCM version 5.4.2
9.0	08. 2015	Revised for SCM version 5.4
8.0	05. 2015	Revised for SCM version 5.3
7.0	03. 2015	Revised for SCM version 5.2
6.0	09. 2014	Revised for SCM version 5.1
5.0	09. 2014	Revised for SCM version 5.0 Revised for SCM version 4,1
4.0	06. 2013	Revised for SCM version 4.0
3.0	01. 2013	Revised for SCM version 3.3
2.0	12. 2010	- Revised for SCM version 3.2.2.x - Added all component (gateway, switch and phone) * Manual Edition allocation method is changed. (Ed.01 → Ver.2.0)
00	02. 2010.	First Version

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CHAPTER 1. SCM System Overview

1.1 General Description

The SCM (Samsung Communication Manager) is an IP-based Private Branch Exchange (PBX) that provides Internet telephony by controlling voice gateways and IP phones on data networks. SCM can connect to the existing PSTN (Public Switched Telephone Network) through voice gateways.

The SCM is available as SCM Express or SCM Enterprise edition, depending on the system capacity. SCM Express is a small-scale system that embedded the required application servers.

The total SCM solution organizes SCM, gateway, switch, phone and application as shown following diagram.

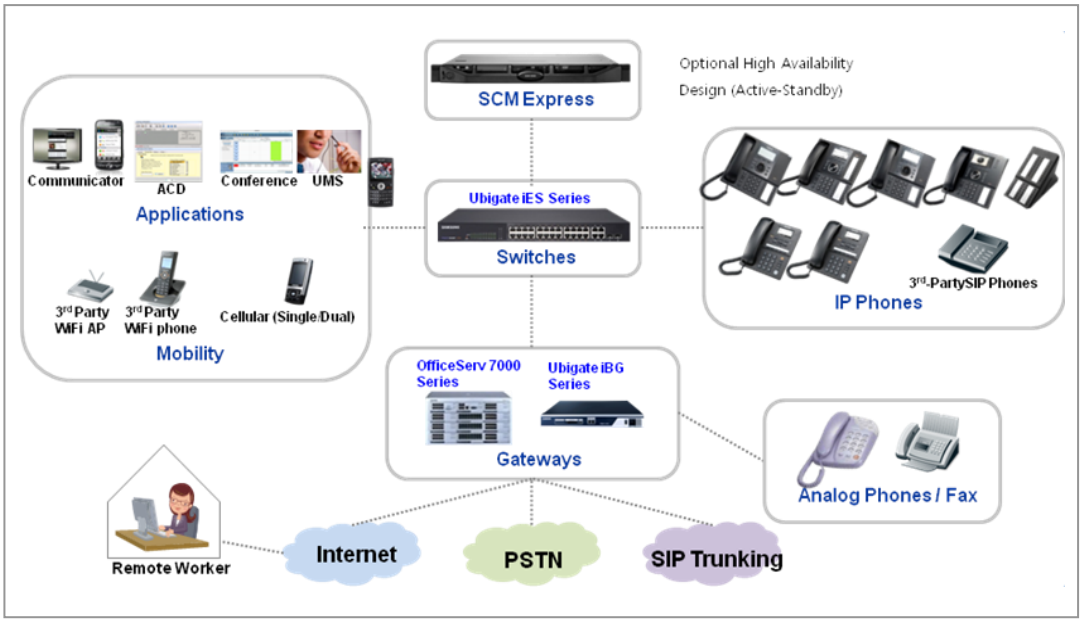


Figure 1. Organization of SCM Solution

1.2 System Architecture

The following diagram shows how SCM can be implemented in a voice-and-data network.

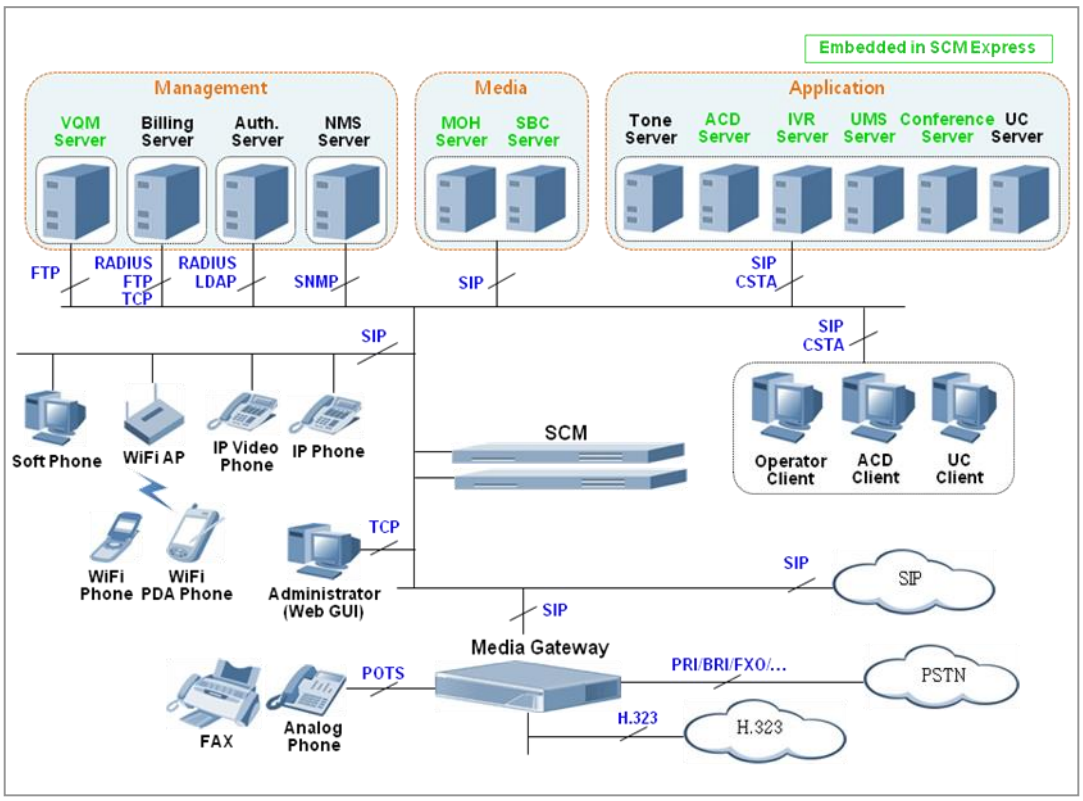


Figure 2. SCM Network Architecture

1.2.1 SCM

SCM performs call processing, communicating with phones, voice gateways, and other entities on the network using the SIP protocol. It also provides supplementary PBX services. SCM uses the SIP protocol for call processing. SCM performs call processing by interoperating with multiple voice gateways.

1.2.2 Voice Gateways

The voice gateway is responsible for connecting the existing telephone and networks.
It communicates with the existing telephone network (PBX or PSTN) through the T1, E1, and PRI interfaces as well as analog interfaces, including E & M and FXO.
The voice gateway acts as a media gateway, which performs the conversion between PCM data and packet data. It communicates with SCM over the standard SIP protocol.

1.2.3 IP Phones

IP Phone provides general telephone calls on the IP network. For providing basic calling and supplementary services, IP Phone communicates with SCM using the SIP protocol (Samsung SIP Extension) that is partially extended from the standard.
The following types of IP phones are in use, based on configuration.

IP Video Phone

Video Phone allows voice and video calls over an IP network.

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.

1.2.4 SCM Administrator

Administrator is web-based service with a graphical user interface for managing SCM. You can use a web browser on a normal PC to access SCM and execute various SCM commands.

1.2.5 Application Servers

Application Servers provide value added application services. Depending on the scope and capacity of the SCM, each application can run in the SCM as an embedded application or in a separate server as a stand-alone application. The following application servers are included in the SCM Express system. The large-scale SCM Enterprise system only includes a subset of the application servers.

MOH (Music On Hold) 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.

MPS (Media Proxy Server) Server

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. SCM Express includes the MPS server that performs the NAT traversal functions which is one of the services of SBC. This functionality is used for the tele-worker in the non-SBC environment and supported number of calls is restricted.

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 SCM 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.

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.
SCM includes a built-in ACD server that provides basic ACD features, such as basic call distribution, and aggregates statistics on the agents. The MOH server plays an announcement or a tone for calls standing by in the ACD server.

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 SCM Express includes the VM/AA server, but its use requires a separate license. The SCM Enterprise system requires an external VM/AA server.

1.2.6 External Application Servers

The following application servers are separated in the SCM Express system.

NMS (Network Management System) Server

NMS server provides the functions for management of the network and each node system that constitutes the network. The NMS communicates by using each node system and the SNMP protocol.

Authentication Server

The authentication server manages authentication information of users and provides LDAP or RADIUS interface. SCM communicates with the Authentication server for the purposes such as for user Authentication.

Billing Server

The billing server is a device that receives the billing information (CDR: Call Detailed Record) that are generated from various call handling systems, extracts the billing data needed for actual operation, and performs the tasks such as various billing related statistics and others. SCM sends CDR that is generated after the call handling to the Billing server by using the protocols such as RADIUS, TCP or FTP.

1.2.7 Interface between the Network Components

The table below describes the interface between each system component.

Area	Physical Connection	Protocol
SCM (Active) ↔ SCM (Standby)	LAN	SIP
SCM ↔ Voice Gateway	LAN	SIP
SCM ↔ Management	LAN	TCP
SCM ↔ 3rd Party Solution	LAN	SIP/CSTA
SCM ↔ IP Phone	LAN	SIP
SCM ↔ Wi-Fi Phone	LAN	SIP
SCM ↔ Soft Phone	LAN	SIP
SCM ↔ Media Gateway	T1/E1/PRI/BRI/FXO	T1/E1/PRI/BRI/Loop Start
Media Gateway ↔ Analog Phone	FXS	Samsung Proprietary

1.3 Network Configuration

In this section, we present a couple of examples about network configuration to provide VoIP services. Our examples may not cover all conditions and requirements which are requested by clients. However we provide representative cases to deploy a network environment which is appropriate for SCM VoIP services.

1.3.1 Single Site-Active SCM

Following figure shows an example of network configuration for a single site. There is a SCM Express server which is an active-alone mode. OfficeServ7400 Gateway is directly connected to PSTN. If needs, it can be connected to PSTN via a Legacy PBX.
We recommend the voice network to be separated from the data network. For security, NAT and Firewall is also recommended.

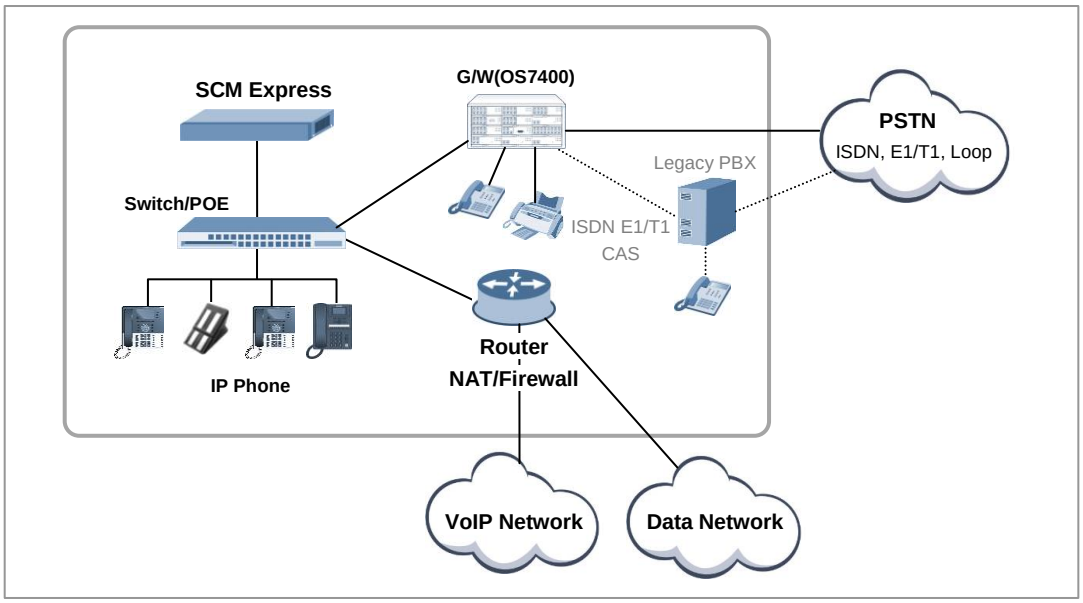


Figure 3. Active-Alone SCM Configuration Example

1.3.2 Single Site-Active/Standby SCM

Following figure shows an example of network configuration for an Active-Standby SCM Express. There two SCM servers where each SCM server is connected to both two switches. It is good to improve availability when one switch (network link) fails. OfficeServ7400 Gateways can be duplicated too.

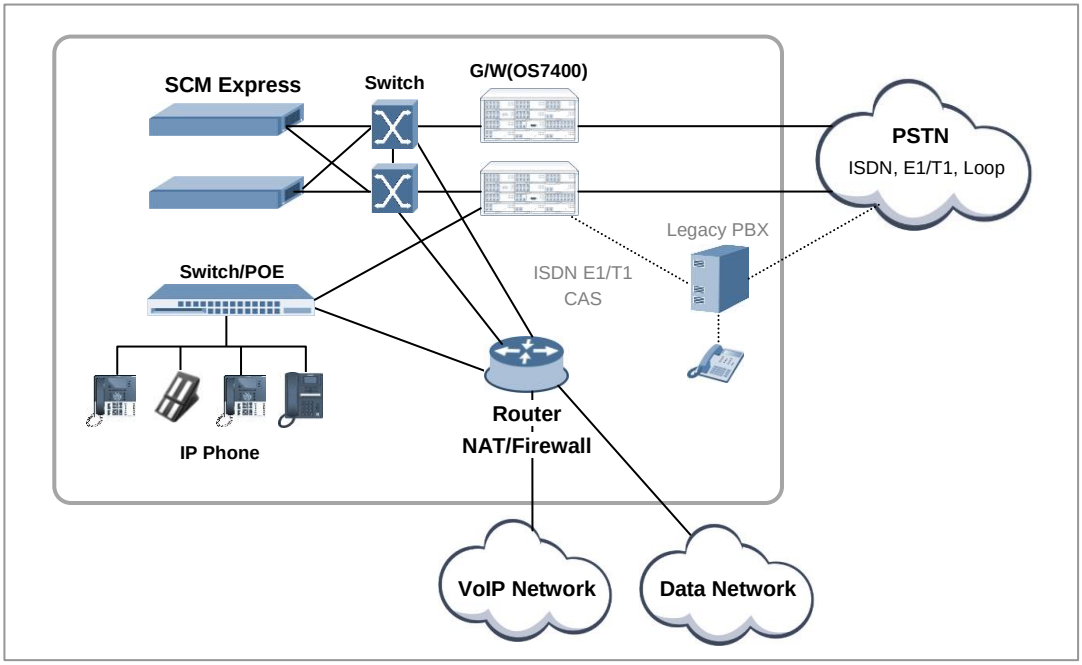


Figure 4. Active-Standby SCM Configuration Example

1.3.3 Multiple Site

Following figure shows networks between a headquarters and a branch. The headquarters and the branch may have a dedicated link such as an E1 VPN or a MSPP tunnel.
The iBG2016 gateway in the branch may have a backup line which is connected to PSTN. The backup line is optional and it is used when the Ubigate iBG2016 gateway loses a connection with SCM Express at the headquarters.

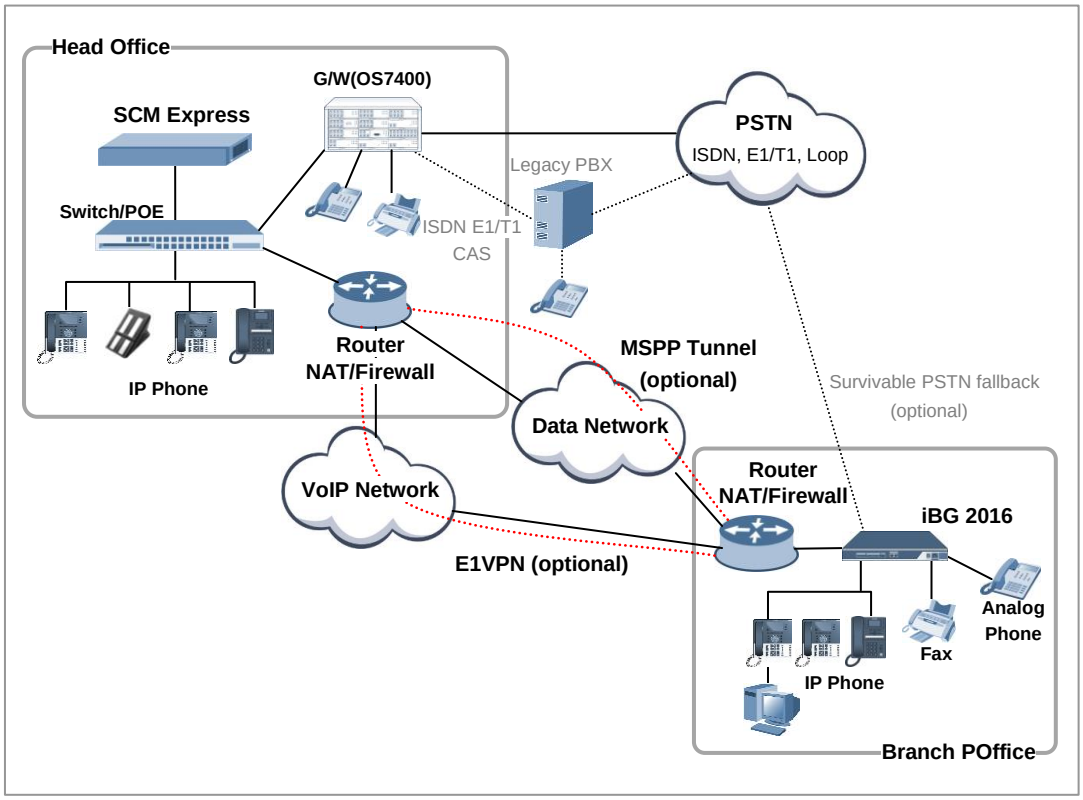


Figure 5. Headquarter and branch Configuration Example

In following figure, there are multiple branches. Each branch is recommended to use NAT and Firewall to improve security. If there is no network node which supports the function of NAT and Firewall, you can configure iBG Gateways to provide the function of NAT and Firewall as well as the function of routing. The gateways in branches may have a connection to Local PSTN alternatively.

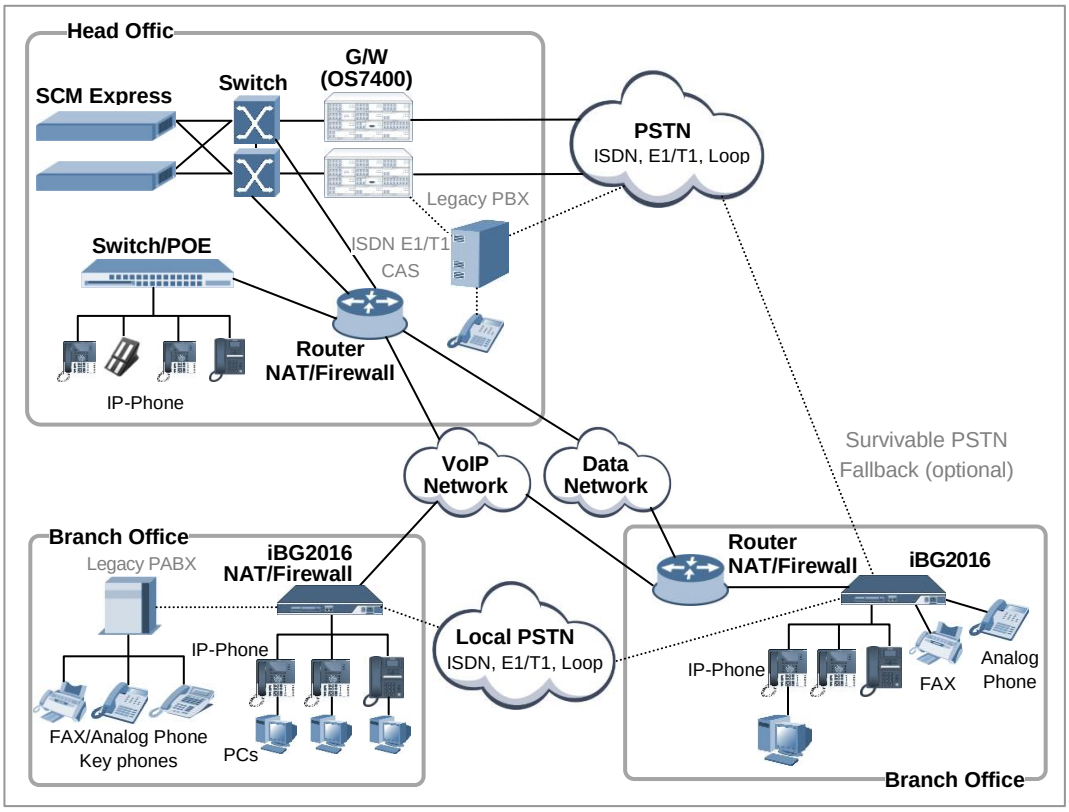


Figure 6. Headquarter and multiple branches Configuration Example

1.3.4 Active-Active Network Configuration

This configuration expands the active alone or active/standby configuration to active/standby-active/standby configuration to provide the geo-redundancy for disaster recovery.

Active-Active SCM Example

The SCM servers are configured as master node and slave node. Each node consists of an active alone SCM. The master node can update the system database. If one of the node fails, the other node takes over and provides services. If the network between the nodes fails, each node provides services separately.

The SCM Express can accommodate 3000 users. When the SCM is configured as active-active, the users increases up to 6000 users but the performance is same as that of the 3000 users configuration.

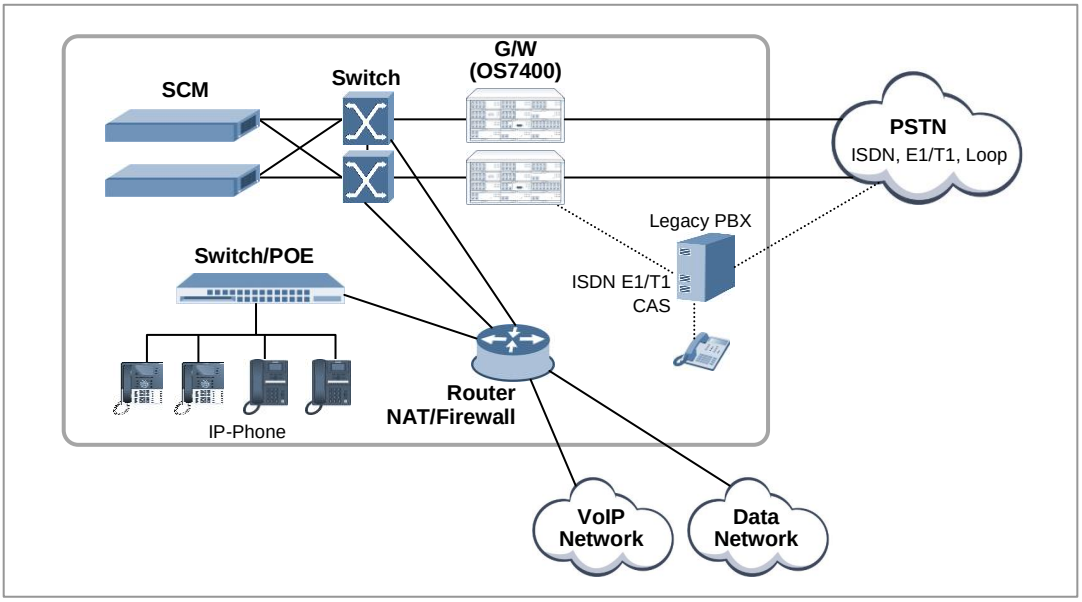


Figure 7. Active-Active Configuration Example

Active/Standby-Active/Standby SCM Example

Basically same as Active-Active SCM case, but each node consists of active/standby SCM. The network between the nodes must be routable which means any network components that separate networks such as NAT, SBC, or firewall must not be located between the nodes.

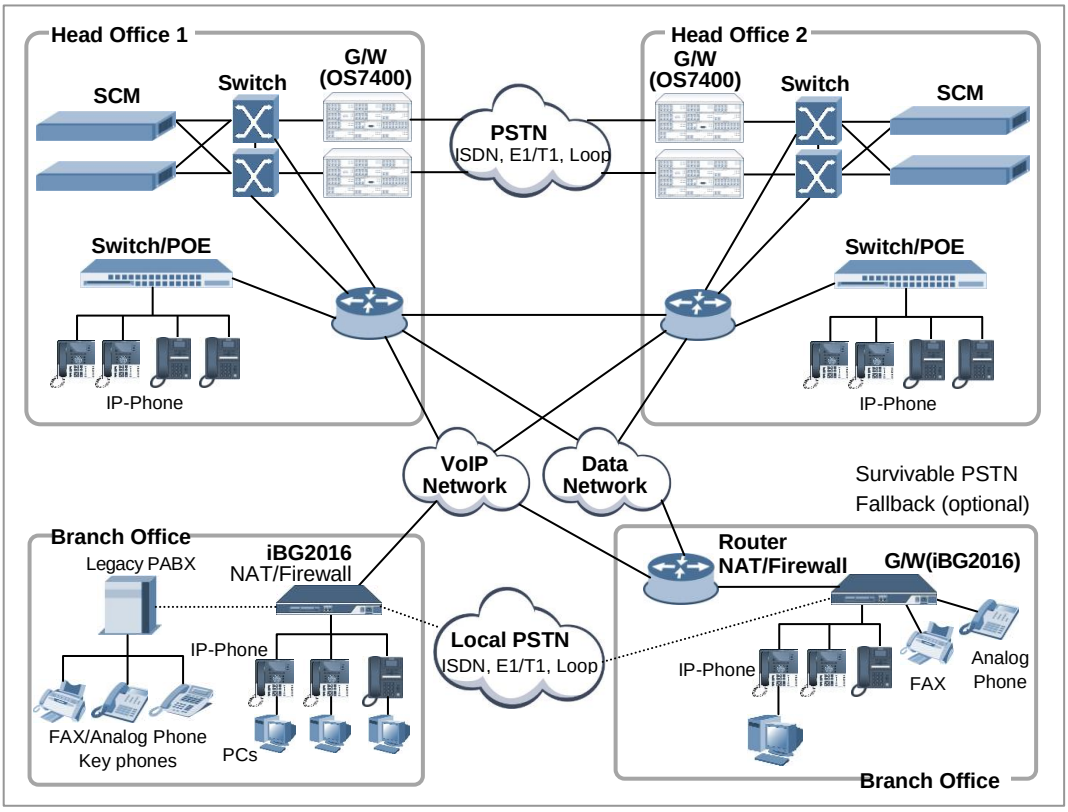


Figure 8. Active/Standby-Active/Standby Configuration Example

1.4 SCM Environment

1.4.1 Hardware Environment

The table below lists the minimum hardware requirements for installing the SCM.

Category	SCM Express	SCM Enterprise
Server	Commercial Linux Server (19” Cabinet Mountable, 1U)	Commercial Linux Server (19” Cabinet Mountable, 1U)
CPU	2.4 GHz or higher (Quad Core, at least 1 CPU)	2.4 GHz or higher (Quad Core, at least 1 CPU)
RAM	4 GB or more (DDR3, 1333 MHz or higher)	8 GB or more (DDR3, 1333 MHz or higher)
HDD	300 GB or more	300 GB or more
ODD	1 DVD-ROM	1DVD-ROM
Network Interface	3 Gigabit Ethernet	3 Gigabit Ethernet

1.4.2 Software Environment

The SCM requires the following software environments.

Category	Specifications
Operation System	CentOS 5.4
DBMS	PostgreSQL
SNMP	NET-SNMP

1.5 SCM Capacity

SCM provides services to 3,000 users but it can extend the number of users to 10,000 through database reinstallation.
There are two types of installation for SCM; one is for SCM Express that includes embedded applications, and the other is for SCM Enterprise which has a call manager function only.

1.5.1 Service Availability

Category	SCM Express	SCM Enterprise
VM/AA Service	Yes	No
Add-On Conference	Yes	Yes
Meet-Me Conference	Yes	Yes
ACD	Yes	Yes
MPS	Yes	Yes
MOH	Yes	Yes
Redundancy	Yes	Yes

1.5.2 System Capacity

Category	SCM Express	SCM Enterprise	Remarks
Max. SIP Phones	3,000 (6,000*)	10,000 (20,000*)	(*) active-active operation
Max. AOM Devices	100	500	-
Max. Gateways	512	512	-
VM/AA Channels	128	0	-
Meet-Me Conference Channels	64	64	Assigned from Total Conference Channels
Total Conference Channels	128	128	-
Paging Channels	128	128	Station Paging & Paging On Answer
MPS Calls	200	200	-
MOH Channels	1,024	1,024	-

1.5.3 Performance (UDP signaling case)

Category	SCM Express	SCM Enterprise	Remarks
Max. Registered Users	3,000 (6,000*)	10,000 (20,000*)	(*) active-active operation
Max. BLF	30,000 (60,000*)	100,000 (200,000*)	10 BLFs/phone
BHCC	60,000	150,000	60 sec/call
Max. Concurrent Calls	1,000	2,500	-
CPS (Normal, no BLF)	20	45	72 hours, 600 sec register expires
CPS (Normal, max BLF)	15	35	-

1.5.4 Performance Capacity (TLS signaling case)

Generally, in case of IP-PBX using TLS, can be 25 % level of performance for call processing as contrasted with UDP.

Category	SCM Express	SCM Enterprise	Remarks
Max. Registered Users	3,000 (3,000*)	10,000 (10,000*)	(*) active-active operation
Max. BLF	30,000	100,000	5 BLFs/phone
BHCC	60,000	150,000	60 s/call
Max. Concurrent Calls	1,000	2,500	-
CPS (Normal, no BLF)	20	45	72 hours, 600 sec register expires
CPS (Normal, max BLF)	15	35	-

1.5.5 Table Capacity

Category	SCM Express	SCM Enterprise	Remarks
Max. User Group	128	128	-
Max. Service Group	512	512	-
Max. Location	1,024	1,024	-
Max. End point	1,024	1,024	-
Max. DID Routing	5,000	5,000	-
Max. CLI Routing	5,000	5,000	-
Max. Account Code	3,000	3,000	-
Max. Authorization Code	3,000	3,000	-

Category	SCM Express	SCM Enterprise	Remarks
Max. DISA Pass CLI	3,000	3,000	-
Max. Hunt Group	512	1,024	-
Max. ACD Agent	500	500	-
Max. ACD Group	256	256	-
Max. ACD Group Members	500	500	-

1.5.6 VM/AA Capacity

Category	SCM Express	SCM Enterprise	Remarks
Max. Concurrent Port	128	-	-
Max. Hard Drive	30 GB	-	-
Max. Mailboxes	3000	-	-
Max. Email Gateway Users	3000	-	-
Sync with Exchange Server	Yes	-	

Message Storage Capacity

30 GB HDD: Approx. 500 Hours (30,000,000B/960,000B)/60 M = 520 hours

1.5.7 Mailbox Capacity

Category	Default	Range
Max. Subscriber Digits in Length	4 digits	Any length from 1~16 digits
Message Retention	9999days	0~9999 days
Total Messages per Mailbox	999	0~9999
Total Message Duration	600 seconds	0~9999 seconds-

1.6 Limitations of Active-Active Configurations

In Active-Active geo-redundancy configuration, each node consists of servers in Active-Standby.

1.6.1 List of services limited

- Last Number Redial
- Callback
- Call Park
- Call Monitoring
- Load Balance Routing
- VM/AA (VM Access, VM Administration, VM Memo, VM Message)
- CAC
- Alarm/Fault/Status
- Statistics and History

1.6.2 Details of limitation

Last Number Redial

Last number redial service is not retained between nodes. Each node stores its own last numbers and does not share with each other.

Callback

The callback service is invoked only when the user is at the same node as the user activated the callback service.

Call Park

The call park feature allows the user to park the current call so that it can be picked up on another phone by pressing the button or the feature code.

Call Monitoring

Calls can be monitored at the node connected to the monitoring server.

Load Balance Routing

Balance counting for the load balancing is not shared between nodes. Hence, the outbound route is decided at each node independently.

VM/AA

Balance counting for the load balancing is not shared between nodes. Hence, the outbound route is decided at each node independently.

CAC

Since CAC are calculated in each node independently, the CACs of a node are applied to the calls at that node independently of the other node.

Alarm/Fault/Status

The alarm, fault, and status events are independently managed by each node. The operator can view those events of a node by connecting the SCM Administrator to the node.

Statistics and History

The statistics and history are independently managed by each node. The operator can view those data of a node by connecting the SCM Administrator to the node.

1.7 SCM virtualization

Virtualized SCM can be mounted on a hypervisor. Virtualized SCM and SCM installed on a physical server provide the same service.

1.7.1 Required Hardware Resource for virtualized SCM

In order to install the virtualized SCM, allocate resources indicated below.

Category	SCM Express	SCM Enterprise
CPU	2.0 GHz 4 core or higher	2.0 GHz 4 core or higher
RAM	6 GB or more	10 GB or more
HDD	300 GB or more	300 GB or more
Network bandwidth	100 Mbps or more	100 Mbps or more
Network Interface	3 network adapter	3 network adapter

1.7.2 Restrictions for virtualized SCM

Hypervisor supports only the VMware ESXi.

CHAPTER 2. SCM Hardware

2.1 IPX-S500 SCM Servers

SCM software to install on the server to meet the minimum requirements for Dell's R210 Model IPX-S500 is supplied to the system.
For additional hardware specification of the IPX-S500, please refer to the 'IPX-S500 Hardware Owner's Manual'.

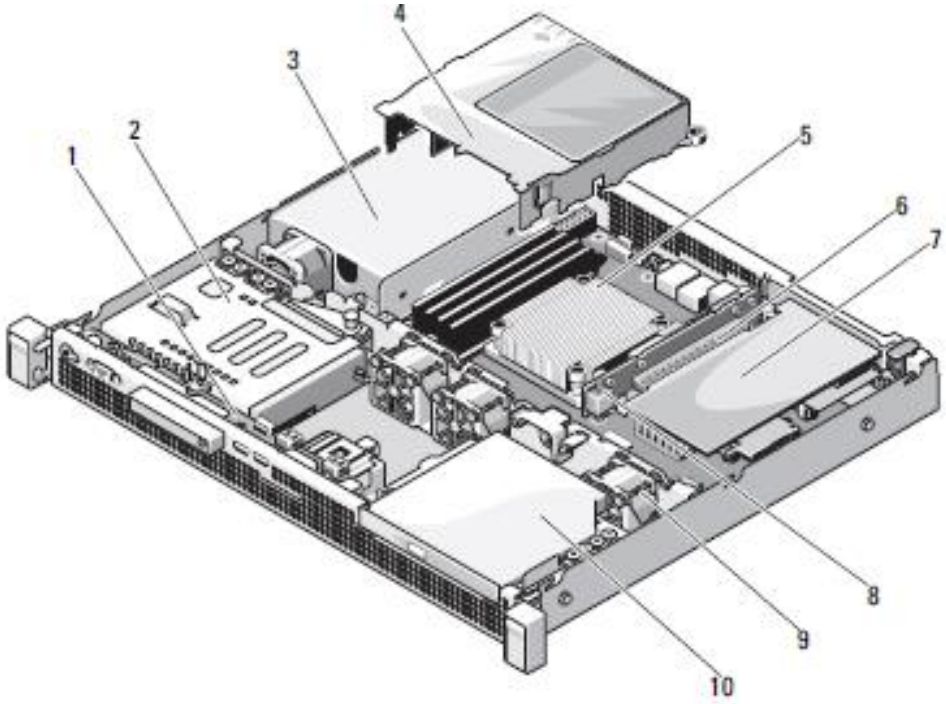


Figure 9. Inside the IPX-S500

No.	Name	No.	Name
①	control panel board	⑥	expansion-card riser
②	hard drive	⑦	expansion card
③	power supply	⑧	chassis intrusion switch
④	cooling shroud	⑨	system cooling fans
⑤	heat sink/processor	⑩	optical drive

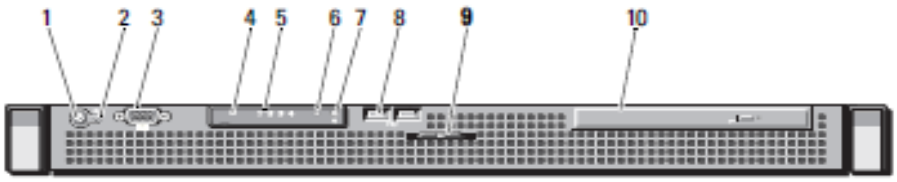


Figure 10. IPX-S500 Front-Panel Features and Indicators

	Indication, Button, or Connector	Description
1	Power-on indicator/power button	The power-on indicator lights when the system power is on. The power button controls the DC power supply output to the system. When the system bezel is installed, the power button is not accessible. Note: When powering on the system, the video monitor can take from several seconds to over 2 minutes to display an image, depending on the amount of memory installed in the system. Note: On ACPI-compliant operating systems, turning off the system using the power button causes the system to perform a graceful shutdown before power to the system is turned off.
2	NMI button	Used to troubleshoot software and device driver errors when using certain operating systems. This button can be pressed using the end of a paper clip. Use this button only if directed to do so by qualified support personnel or by the operating system's documentation.
3	Video connector	Connects a monitor to the system.
4	Hard drive activity indicator	Lights up when the hard drive is in use.
5	Diagnostic indicator lights (4)	The four diagnostic indicator lights display error codes during system startup.
6	System status indicator	Lights blue during normal system operation. Lights amber when the system needs attention due to a problem.
7	System identification button	The system identification buttons on the front and back panels can be used to locate a particular system within a rack. When one of the buttons is pushed, the system status indicators on the front and back panels light blue until one of the buttons is pushed again.
8	USB connectors (2)	Connects USB devices to the system. The ports are USB 2.0-compliant.
9	System identification panel	A slide-out panel for system information including the Express Service Tag, embedded NIC MAC address, and iDRAC6 Enterprise card MAC address. Space is provided for an additional label.
10	Optical drive (optional)	One optional slim-line SATA DVD-ROM drive or DVD+/-RW drive, or combination CD-RW/DVD drive (when available). Note: DVD devices are data only.

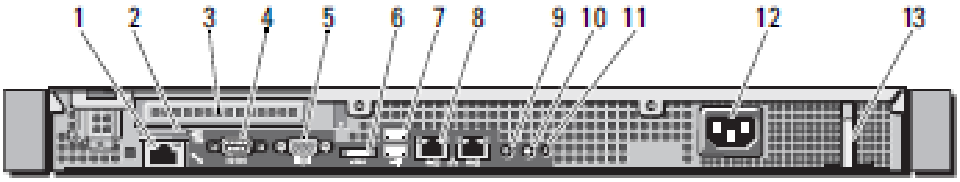


Figure 11. IPX-S500 Back-Panel Features and Indicators

	Indication, Button, or Connector	Description
1	iDRAC6 Enterprise port (optional)	Dedicated management port for the optional iDRAC6 Enterprise card.
2	VFlash media slot (optional)	Connects an external SD memory card for the optional iDRAC6 Enterprise card.
3	PCIe expansion card slot	Connects a PCI Express expansion card.
4	Serial connector	Connects a serial device to the system.
5	Video connector	Connects a VGA display to the system.
6	eSATA	Connects additional storage devices.
7	USB connectors (2)	Connects USB devices to the system. The ports are USB 2.0-compliant.
8	Ethernet connectors (2)	Embedded 10/100/1000 NIC connectors.
9	System status indicator	Lights blue during normal system operation. Lights amber when the system needs attention due to a problem.
10	System identification button	The system identification buttons on the front and back panels can be used to locate a particular system within a rack. When one of the buttons is pushed, the system status indicators on the front and back panels light blue until one of the buttons is pushed again.
11	System identification connector	Connects the optional system status indicator assembly through the optional cable management arm.
12	Power supply	250 W power supply.
13	Retention clip	Secures the power cable.

2.2 Ubigate iBG Gateways

2.2.1 iBG 3026

The Ubigate iBG3026 is enclosed in a rack-mount 2U enclosure, designed for installation in a standard 19-inch and 23-inch rack. The unit is approximately 17.5 in. wide, 16 in. deep and 3.5 in. high. The chassis is enclosed at the front, except for status indicators and an air intake grill. All modules and all cabling, including power cables, are installed from the rear.

iBG3026 Front Panel

The front panel of the iBG3026 provides status of the router's performance and operation by using LEDs. Proper LED status is shown as following.

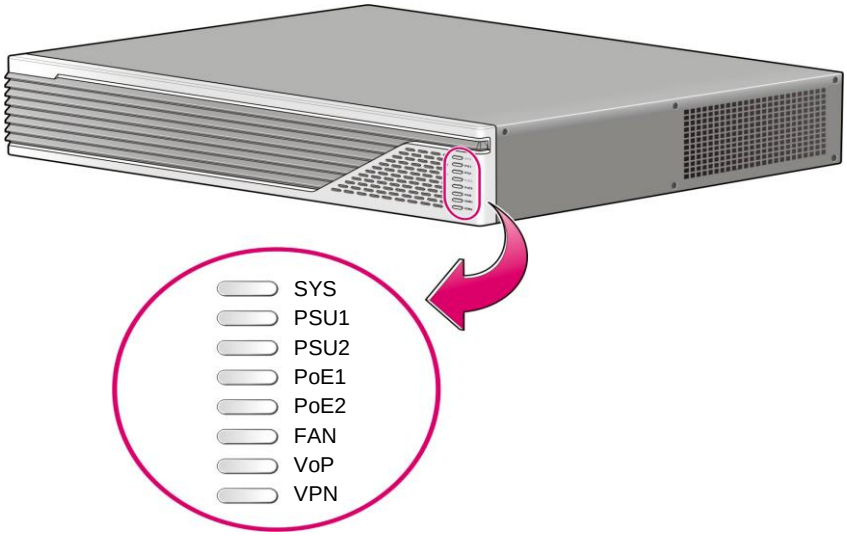


Figure 12. iBG3026 Front View

LED	Indication & Color	Description
SYS (Router Power)	Solid green	System is operating normally.
	Blinking green	Running ROM monitor with no errors detected.
	Amber	Router is receiving power but malfunctioning.
	Off	Router is not receiving power.
PSU1 (Power Supply 1)	Solid green	Power supply 1 installed and operating normally.
	Amber	Power supply 1 installed and powered off, or fault condition detected.
	Off	Power supply 1 not present.
PSU2 (Power Supply 2)	Solid green	Power supply 2 installed and operating normally.
	Amber	Power supply 2 installed and powered off, or fault condition detected.
	Off	Power supply 2 not present.

LED	Indication & Color	Description
PoE1 (PoE Power 1)	Solid green	-48 V power module 1 installed and operating normally.
	Amber	-48 V power module 1 installed and powered off, or fault condition detected.
	Off	-48 V power module 1 not present.
PoE2 (PoE Power 2)	Solid green	-48 V power module 2 installed and operating normally.
	Amber	-48 V power module 2 installed and powered off, or fault condition detected.
	Off	-48 V power module 2 not present.
FAN	Solid green	Fan is operating properly
	Amber	Fan present with failure
VoP	Solid green	VoP or IVM card present and enabled.
	Amber or Red	VoP or IVM card present with failure.
	Off	VoP and IVM card not present
VPN	Solid green	VPN card present and enabled.
	Amber or Red	VPN card present with failure.
	Off	VPN card not present

iBG3026 Rear Panel

The iBG3026 rear panel provides connections for power and networking. The iBG3026 rear panel provides connections for one management Ethernet port, one USB port, a Compact Flash slot, two 10/100/1000 Base-T Ethernet ports, two Gigabit SFP ports, one console port, and one auxiliary port. The rear panel has the following features.

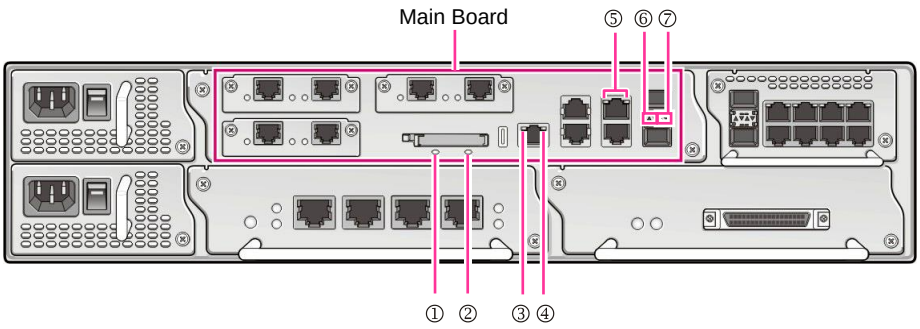


Figure 13. iBG3026 Main Board View

No.	LED	Indication & Color	Description
①	SYS RDY	Solid green	The module's power is normal status.

No.	LED	Indication & Color	Description
		Off	Power fail or removal status.
②	CF	Solid green	Compact Flash being accessed
		Off	Compact Flash not mounted or not being accessed.
③	Management port Left LED (Link)	Solid Green	The management Link connected.
		Green Blink	Link connect and transmit data
		Off	The management port is not connected.
④	Management port Right LED (Speed)	Solid Amber	The link Speed is 100 Mbps.
		Off	The link Speed is 10 Mbps.
⑤	UTP GbE Port LED	Solid Green	Link is established with speed 10/100 Mbps.
		Green Blink	Blinking green indicates transmit/receive activity with speed 10/100 Mbps.
		Solid Amber	Link is established with Speed 1 G bps.
		Amber Blink	Blinking amber indicates transmit/receive activity with speed 1 G bps.
		Off	Link fail or no connect.
⑥	GbE SFP Left LED	Solid Green	Link is established with speed 1 G bps.
		Off	Link fail or no connect.
⑦	GbE SFP Right LED	Blinking Amber	Blinking amber indicates transmit/receive activity with speed 1 G bps.
		Off	No activity

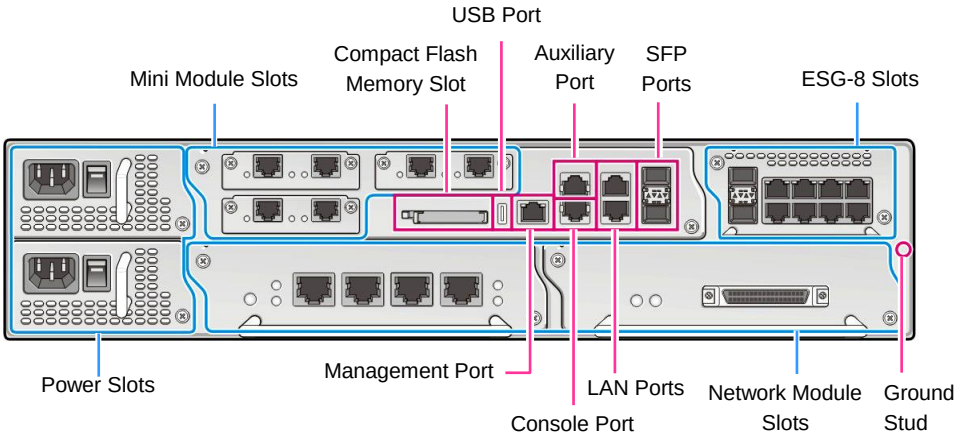


Figure 14. iBG3026 Rear View

Connector	Description
Power Slots	Power slots up to two power supplies for AC, AC with PoE, and DC with PoE.
Mini Module Slots	Up to three mini modules supporting a variety of interfaces

Connector	Description
Compact Flash Memory Slot	Slot for Compact Flash memory card
USB Port	Supports USB1.1 interface
Management Port	Supports Fast Ethernet interface
Auxiliary Port	Serial ports supporting remote monitoring
Console Port	Serial ports supporting local monitoring and configuring
LAN Ports	Supports triple speed Ethernet interface
SFP Ports	Support optional Gigabit SFP modules
ESG-8 Slots	Slot for ESG-8 Gigabit Ethernet option module
Network Module Slots	Up to two network modules supporting a variety of Interfaces.
Ground Stud	Screw holes for grounding lug

2.2.2 iBG 2016

The Ubigate iBG2016 is enclosed in a rack-mount 1U enclosure, designed for installation in a standard 19-inch and 23” rack. The unit is approximately 17.5” wide, 16.7” deep and 1.75” high. The chassis is enclosed at the front, except for status indicators and an air intake grill. All modules and all cabling, including power cables, are installed from the rear.

Ubigate iBG2016 Front Panel

The front panel of the Ubigate iBG2016 provides status of the router’s performance and operation by using LEDs. Proper LED status is shown as following.

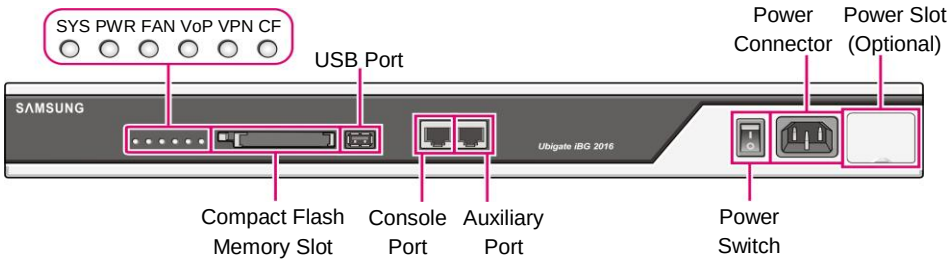


Figure 15. iBG2016 Front View

LED	Indication & Color	Description
SYS	Solid green	System is operating normally.
	Amber	Router is receiving power but malfunctioning.
	Off	Router is not receiving power.
PWR	Solid green	Power supply installed and operating normally.

LED	Indication & Color	Description
	Amber	Power supply installed and powered off, or fault condition detected.
	Off	Power supply not present.
FAN	Solid green	Fan is operating properly
	Amber	Fan present with failure
VoP	Solid green	VoP or IVM card present and enabled.
	Amber or Red	VoP or IVM card present with failure.
	Off	VoP and IVM card not present
VPN	Solid green	Internal option card present and enabled.
	Amber or Red	Internal option card present with failure.
	Off	Internal option card not present
CF	Solid green	Compact Flash memory being accessed
	Off	Compact Flash memory not mounted or not being accessed.

Connector	Description
Compact Flash Memory Slot	Slot for Compact Flash Memory Card
USB Port	Supports USB1.1 interface
Console Port	Serial port supporting local monitoring and configuring
Auxiliary Port	Serial port supporting remote monitoring
Power Switch	Power switch
Power Connector	AC power connector
Power Slot (Optional)	Optional external DC power supply inputs

Ubigate iBG2016 Rear Panel

The Ubigate iBG2016 rear panel provides connections for networking.
The Ubigate iBG2016 rear panel provides connections for one Fast Ethernet SFP port and four 10/100 Fast Ethernet ports.
The Rear Panel has the following features.

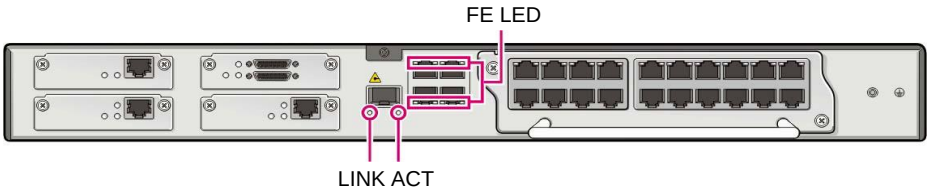


Figure 16. iBG2016 Rear View (LED)

LED	Indication & Color	Description
LINK (SFP Port)	Solid Green	Link is established with speed 100 Mbps.
	Off	Link fail or not connected.
ACT (SFP Port)	Blinking Amber	Blinking Amber indicates transmit/receive activity with speed 100 Mbps.
	Off	No activity.
UTP FE Port Left LED	Solid Green	Link is established with speed 10/100 Mbps.
	Off	Link fail or not connected
UTP FE Port Right LED	Amber Blink	Blinking Amber indicates transmit/receive activity with speed 10/100 Mbps.
	Off	No activity.

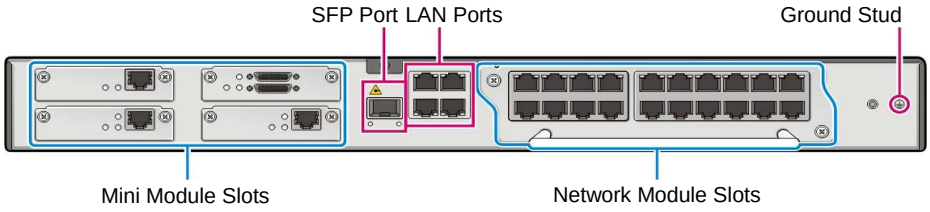


Figure 17. iBG2016 Rear View (Connector)

Connector	Description
Mini Module Slots	Up to three mini modules supporting a variety of interfaces and one mini module supporting for voice only
SFP Port	Supports optional FE SFP module
LAN Ports	Supports Fast speed Ethernet interface
Network Module Slots	Slot for network module supporting a variety of Interfaces.
Ground Stud	Screw holes for grounding lug

2.2.3 iBG 2006

Ubigate iBG2006 is enclosed in a rack-mount 1U enclosure, designed for installation in a standard 19-inch or 23-inch rack. The unit is approximately 17.38” wide, 13.8” deep and 1.75” high. The front panel has LED indicators, console port, and auxiliary port.
All modules and all cabling, including power cables, are installed from the rear.

Ubigate iBG2006 Front Panel

The front panel of Ubigate iBG2006 has LEDs in order to indicate the router’s performance and operation status as shown Figure 2.2. Proper LED status is shown as follows.

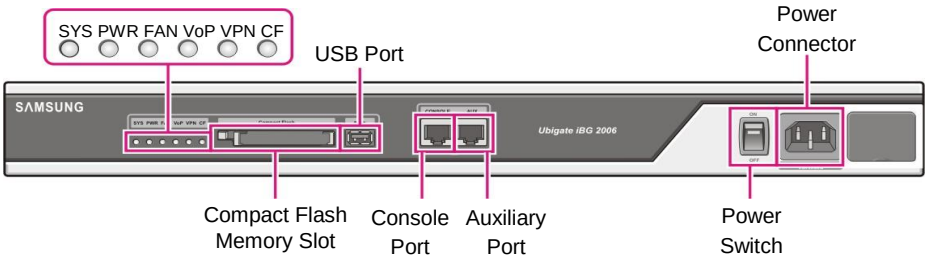


Figure 18. iBG2006 Front View

LED	Indication & Color	Description
SYS	Solid green	System is operating normally.
	Blinking green	Running ROM monitor with no errors detected.
	Amber	Router is receiving power but malfunctioning.
	Off	Router is not receiving power.
PWR	Solid green	Power supply installed and operating normally.
	Amber	Power supply installed but powered off or fault condition detected.
	Off	Power supply not present.
FAN	Solid green	Fan is operating properly.
	Amber	Fan present with failure.
VoP	Solid green	VoP or IVM card present and enabled.
	Amber or Red	VoP or IVM card present with failure.
	Off	VoP and IVM card not present.
VPN	Solid green	VPN card present and enabled.
	Amber or Red	VPN card present with failure.
	Off	VPN card not present.
CF	Solid green	Compact Flash memory being accessed.
	Off	Compact Flash memory not mounted or not being accessed.

The front panel also has the following connectors.

Connector	Description
Compact Flash Memory Slot	Slot for Compact Flash memory card
USB Port	Supports USB2.0 interface
Console Port	Serial port supporting local monitoring and configuring
Auxiliary Port	Serial port supporting remote monitoring
Power Switch	Power switch
Power Connector	AC power connector

Ubigate iBG2006 Rear Panel

Ubigate iBG2006 rear panel has LEDs, connectors and mini module slots as shown in figure. Ubigate iBG2006 rear panel has one Fast Ethernet SFP port and four 10/100 Fast Ethernet ports, all of which are built-in.
Here are the description of the LEDs, connectors and min-module slots.

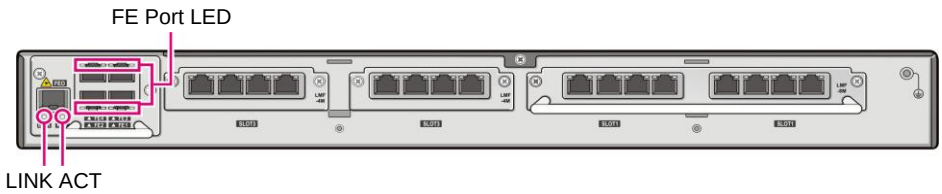


Figure 19. iBG2006 Rear LED

LED	Indication & Color	Description
FE SFP LINK LED	Solid Green	Link is established with speed 100 Mbps.
	Off	Link fail or not connected
FE SFP ACT LED	Blinking Amber	Blinking amber indicates transmit/receive activity with speed 100 Mbps.
	Off	No activity
FE Port Left LED	Solid Green	Link is established with speed 10/100 Mbps.
	Off	Link fail or not connected
FE Port Right LED	Blinking Amber	Blinking amber indicates transmit/receive activity with speed 10/100 Mbps.
	Off	No activity

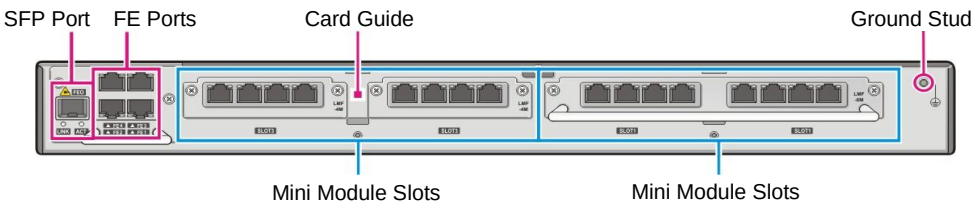


Figure 20. iBG2006 Rear Connector

Connector	Description
SFP Port	Support optional FE SFP modules
FE Ports	Supports Fast Ethernet interface
Mini Module Slots	A variety of data and interface cards can be plugged into mini module slots. Depending on the widths of modules, up to four modules can be plugged.
Card Guide	Guides inserting mini module (Remove guides when install double wide mini module LMF-8 M)
Ground Stud	Screw holes for grounding lug

2.2.4 iBG 1003

Ubigate iBG1003 should be installed on a desktop or 19-inch rack. The front side has LED indicators. The rear side has console port, T1/E1 ports, and Fast Ethernet UTP ports. All cabling, including power cable, are installed from the rear side.

iBG1003 Front Side

The front side of the Ubigate iBG1003 has LEDs in order to indicate the system's performance and operation status as shown in figure below.

Figure 21. iBG1003 Front View

Proper LED status is shown in the following table.

LED	Indication & Color	Description
⏻ (Power)	Solid blue	Power supply installed and operating normally.
	Amber	Power supply installed but power fault condition detected.
	Off	Power supply not present or Power supply malfunctioning.
SYS	Solid green	System is operating normally.
	Solid red	System is not operating normally.

LED	Indication & Color	Description
	Amber	System diagnostic mode.
	Off	Router is not receiving power.
IOM	Solid green	IOM (Internal Option Module) card present and operational.
	Solid red	IOM card present but not operational.
	Off	IOM card not present.
FAN	Solid green	Fan is operating properly.
	Solid red	Fan present but malfunctioning.
	Off	Fan has been stopped by user configuration.
P0~P3 (T1/E1)	Solid green	T1/E1 port is operating normally.
	Solid red	T1/E1 port cable is not connected properly or a critical alarm has been detected.
	Amber	User alarm detected.
	Off	System is not operating normally or port is disabled.
Link0, 1 (Ethernet)	Solid green	Ethernet port link is detected.
	Off	Ethernet port link is not detected.
Act0, 1 (Ethernet)	Blinking Amber	Blinking Amber indicates transmit/receive activity with speed 10/100 Mbps.
	Off	No activity.

iBG1003 Rear Side

Ubigate iBG1003 rear side has four T1/E1 ports, two Fast Ethernet UTP ports, two FXO ports, two USB ports, and one console port.
Depending on sub-models, iBG1003 has different voice connectors such as four RJ-11 ports for accommodating four FXS ports, one 50-pin champ connector for accommodating 8 or 16 FXS ports, or two RJ-45 ports for accommodating two T1/E1 PRI ports. A model shown in figure below has a 50-pin champ connector for accommodating 8 or 16 FXS ports. In addition, the rear side has one mini-module slot to accommodate an ADSL module optionally. And two USB ports and one console port are provided in the rear side.
Power connector, power switch, and ground stud are also located in the rear side.

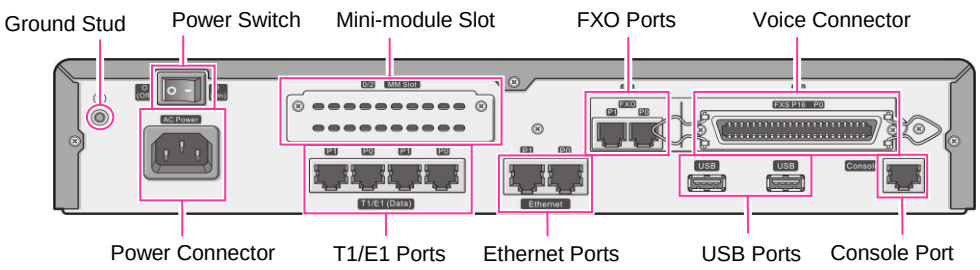


Figure 22. iBG1003 Rear View

Connector	Description
Ground stud	A screw hole for grounding lug
Power switch	Switch to turn on or off the power supply
Power connector	AC power connector
Mini module slot	An optional ADSL module can be plugged into this slot.
FXO ports	Analog voice channel port for connecting to PBX station lines or FXS/DID lines from a Central Office of the Public Switched Telephone Network (PSTN).
Voice connector	Depending on sub-models, one of the following voice connectors are provided. - 50-pin champ connector for accommodating 8 or 16 FXS ports - 4 RJ-11 ports for accommodating 4 FXS ports - 2 RJ-45 ports for accommodating 2 T1/E1 PRI ports
T1/E1 ports	T1/E1 WAN connection
Ethernet ports	Fast Ethernet LAN connection
USB ports	Supports USB2.0 interface
Console port	Serial port for local monitoring and configuring

2.2.5 iBG Voice Modules

VoP-32, VoP-64, VoP-128 (VoIP Option Cards)

The VoIP option card is the root of the voice subsystem and must be installed to have any voice features. It includes a VoIP Processing DSP and a TDM switch. It is an internal card that mounts above the main board and must be installed onto the main board before it is inserted into the chassis. VoP option boards are not hot-swappable.

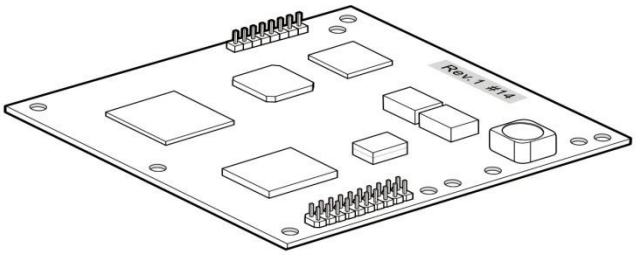


Figure 23. VoP-32, 64, 128 Internal Option Cards

T1E1-4 (4-Port T1/E1 Network Module)

This module provides four T1 ports, each running at 1.544 Mbps and supporting 24 data timeslots, or four E1 ports, each running at 2.048 Mbps and supporting 30 data timeslots.

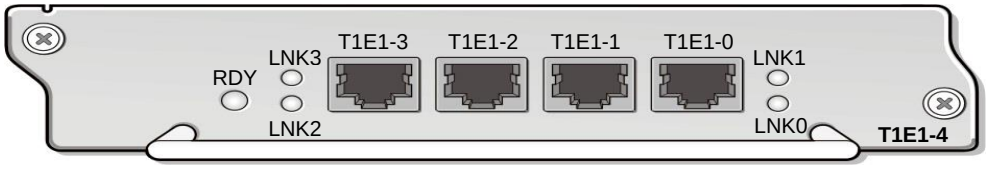


Figure 24. T1E1-4 Network Module

The following table explains the LEDs states in detail.

LED	Indication & Color	Description
RDY	Solid Green	Network module is operational and power on.
	Off	Indicates the port is power down.
LNK 0 LNK 1 LNK 2 LNK 3	Green	Indicates the port is enabled. (Carrier Detect)
	Amber	Indicates an alarm condition exists on the remote end of one of the T1/E1 ports. On Remote End, detected alarm condition: - RAI (Remote Alarm Indication) - AIS (Alarm Indication Signal).
	Red	Indicates an alarm condition exists locally on one of the T1/E1 ports. Locally, detected alarm condition: - LOS (Loss of signal) - LOF (Loss of frame)
	Off	Indicates no alarms detected on any port.

VCU-A (Voice Carrier Network Module)

This module installs as a network module and can hold any pair of mini modules. Thus, it can be used to add additional FXO, FXS/DID, E & M, T1/E1 or ISDN mini modules to the iBG3026. The maximum number of voice mini modules supported by an iBG3026 is seven (three mini module slots, two network modules with two voice mini modules each).

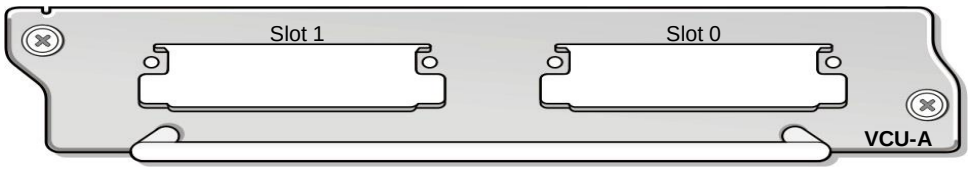


Figure 25. VCU-A Network Module

T1E1-2M (2-Port T1/E1 Mini Module)

This 2-Port T1/E1 mini module provides two T1 ports, each running at 1.544 Mbps and supporting 24 voice or data timeslots, or two E1 ports, each running at 2.048 Mbps and supporting 30 voice or data timeslots.
This module supports generic 1-port or 2-port T1 or E1 trunk interfaces for voice, data, and integrated voice and data applications. The mini module provides basic structured and unstructured service for T1 or E1 networks.
The mini module can be used as trunk interfaces for voice and data services, as fractional n x 64-kbps service for WANs (Frame Relay or leased line), or for time-division multiplexing (TDM) drop-and-insert (voice and data integration) services.

The T1/E1 mini modules provide voice and data access to the PSTN domain through TDM ports, and include an integrated Channel Service Unit/Data Service Unit (CSU/DSU).

Figure 26. T1E1-2M Mini Module

The following table explains the LEDs states in detail.

LED	Indication & Color	Description
RDY	Green	The mini module is operational and power on.
	Off	Indicates the port is power down.
LNK 0 LNK 1	Green	Indicates the port is enabled. (Carrier Detect)
	Amber	Indicates an alarm condition exists on the remote end of one of the T1/E1 ports. On Remote End, detected alarm condition: - RDI (Remote Detect Indication) - AIS (Alarm Indication Signal).
	Red	Indicates an alarm condition exists locally on one of the T1/E1 ports. Locally, detected alarm condition: - LOS (Loss of signal) - OOF (Out-of-Frame)
	Off	Indicates that the port is not enabled, and no alarms detected on any port.

FXO-4M (4-Port Analog FXO Mini Module)

This module provides four analog voice channel ports for connections to Plain Old Telephone Services (POTS) telephones. The voice channels connect to the voice subsystem.

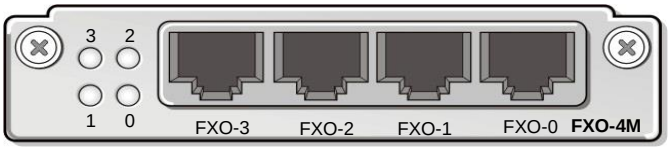


Figure 27. FXO-4M Mini Module

The following table explains the LEDs states in detail.

LED	Indication & Color	Description
0 1 2 3	Green	The port0/1/2/3 connection is active.
	Amber	The port0/1/2/3 connection is abnormal or initial state.
	Red	The port0/1/2/3 connection is in alarm status.
	Off	The port0/1/2/3 activity is occurring.

FXS-4M (4-Port Analog FXS/DID Mini Module)

This module provides four analog voice channel ports for connections to PBX station lines or FXS/DID lines from a central office of the Public Switched Telephone Network (PSTN). The voice channels connect to the voice subsystem.

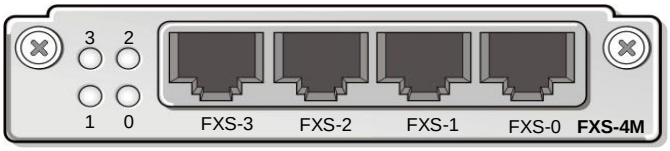


Figure 28. FXS-4M Mini Module

The following table explains the LEDs states in detail.

LED	Indication & Color	Description
0	Green	The port0/1/2/3 connection is active.
1 2 3	Amber	The port0/1/2/3 connection is abnormal or initial state.

E & M-2M (2-Port E & M Mini Module)

This module provides internetworking functions for POTS, and trunk interfaces.
The E & M (Ear and Mouse) voice mini module is a mini module that mates with the main board and/or the voice carrier module. The E & M interface module is a mini module that slide into the voice network module.
The E & M mini module used:
Mainly between PBXs or other network-to-network telephony switches.
There are four main parameters defining the different analog E & M implementations:
E & M Interface Types and Wiring Arrangement (Type I, II, III and V)
Audio Implementation (two-wire/four-wire)
Start Dial Supervision Signaling (immediate, wink and delay)
Address Signaling (pulse, DTMF)

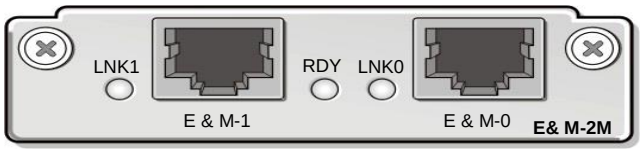


Figure 29. E & M-2M Mini Module

The following table explains the LEDs states in detail.

LED	Indication & Color	Description
RDY	GREEN	The module passed its self-test and is available to the router.
	OFF	Reset, power down, Removal state.
LNK0 LNK1	Green	The port0/1 connection is active.
	Amber	The port0/1 connection is abnormal or initial state.
	Red	The port0/1 connection is in alarm status.
	Off	No port0/1 activity is occurring.

FXS-24 (24-Port Analog FXS/DID Network Module)

This module provides 24 analog voice channel ports for connections to POTS telephones. The voice channels connect to the voice subsystem.
(Actually, FXS-24 provides only 12 ports, and It will be increased 24 ports by FXS-4E sub-board. One FXS-4E sub-board provide 4 ports)

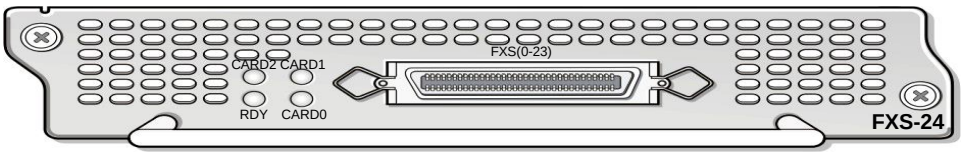


Figure 30. FXS-24 Network Module

The following table explains the LEDs states in detail.

LED	Indication & Color	Description
RDY	GREEN	All diagnostics pass, and the network module is operational.
	OFF	Reset, power down or removal status.
CARD0 CARD1 CARD2	Solid Green	Internal extension card present
	Off	Internal extension card not present

FXS-24 Extension Box

This module provides 24 port analog voice channel connector between FXS-24 module and telephones. It has 24 RJ-11 connector in the front side and 1 champ connector in the rear side.

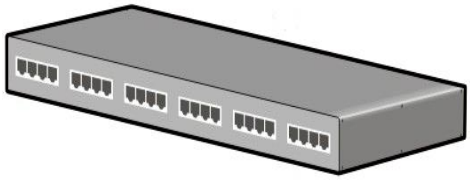


Figure 31. FXS-24 Extension Box

2.3 OfficeServ 7000 Gateways

2.3.1 OfficeServ 7400

This section introduces the hardware features, chassis configuration, and module functions and configuration of the OfficeServ 7400 system. In addition, this section describes terminals, wireless LAN equipment, and additional equipment available in the OfficeServ 7400 system.

Chassis Configuration

OfficeServ 7400 consists of 3 chassis (basic/expansion chassis) mounted on the 19-inch rack and a functional server that operates externally.

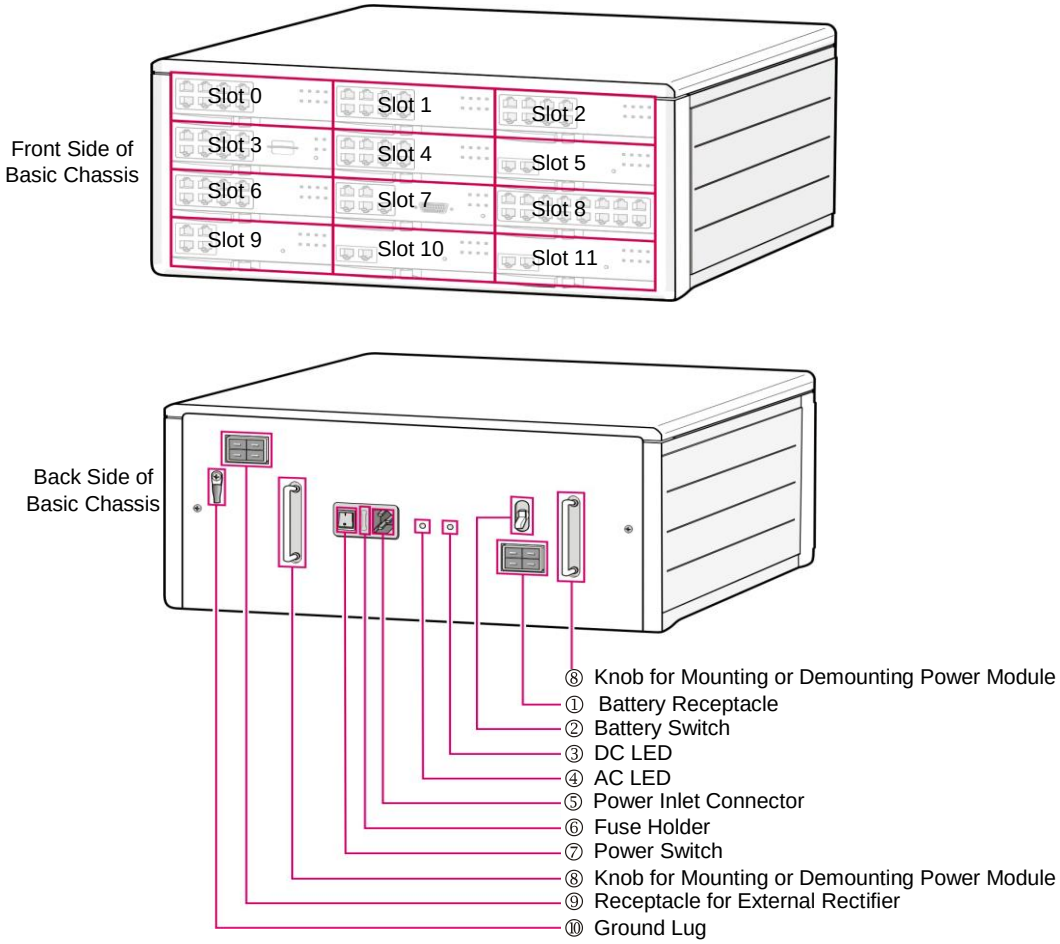


Figure 32. Configuration of OfficeServ 7400 Chassis

The Main Control Processor 40 (MP40) module, which is the main control part, is installed into the basic chassis to manage the entire OfficeServ 7400 and performs the switching, the signal processing and user station management functions. The LP40 which is the sub control part and is installed into the basic/expanded chassis, controls various line modules and sends/receives information to/from the MP40. The other components are modules, the power supply, and fans.

Part	Function
① Battery Receptacle	Receptacle for an external battery.
② Battery Switch	Switch to supply the power to OfficeServ 7400 or charge an battery
③ DC LED	The LED indicates the normal output of DC power.
④ AC LED	The LED indicates the input of AC power.
⑤ Power Inlet Connector	Connector for power cable
⑥ Fuse Holder	Fuse to protect AC input power
⑦ Power Switch	Power-on/off of OfficeServ 7400
⑧ Knob for Mounting or Demounting Power Module	Knob to mount or demount power module
⑨ Receptacle for External Rectifier	Receptacle to supply external DC power for Power over Ethernet (PoE)
⑩ Ground Lug	Lug for grounding system communication

Configuration of Slots

Each of the basic chassis and expansion chassis has 12 slots on which modules can be mounted. The modules below are mounted on the slots depending on the configuration type of the OfficeServ 7400:

Chassis	Slots	Mountable Modules
Basic Chassis (OfficeServ Access)	Slot 0	Special purpose for LP40
	Slot 3	Special purpose for MP40
	Slot 1 and 2 Slot 4~11	Modules except MP40 and LP40
Expansion Chassis (OfficeServ Expansion)	Slot 0	Special purpose for LP40
	Slot 3	Boards excluding LIM, PLIM, and PLIM2
	Slot 1, 2, 4~11	Modules except MP40 and LP40

For using total capacity of TEPRI2 and MGI64, basic chassis that supports 64 channels per slot is used.

MP40 (Main Control Module)

MP40 is the main control module that controls all functions of the OfficeServ 7400 and is mounted on Slot 3 of the basic chassis. The MP40 performs voice switching, signal processing, and user's station management functions.
The MP40 controls the entire system, performs system booting and data management functions. In addition, the MP40 recognizes/monitors/controls the modules mounted on the universal slot of the expansion chassis through IPC by connecting to the LP40 that is a control module in expansion chassis of OfficeServ 7400 or the LCP that is a control module in expansion chassis of OfficeServ 7200.
The MP40 connected to Ethernet network via the LAN Interface of the front panel to drive various applications. The flexibility of the system is improved by accommodating the VoIP function and the load of the system is balanced by using a control module for each chassis. IPC between chassis uses HDLC protocol to increase the reliability.
The front view of the MP40 main control part is as shown in the figure below.

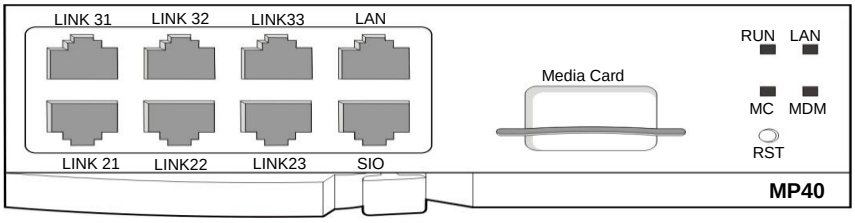


Figure 33. Front View of MP40

The components on the front panel of the MP40 module have the functions below:

Ports, LEDs	Functions
LINK21~LINK23	Connection port between MP40 and LP40 in the first expansion chassis
LINK31~LINK33	Connection port between MP40 and LP40 in the second expansion chassis
LAN	Port to connect 10/100 BASE-T LAN
SIO	UART Port (for test) and CPLD JTAG Port (for management)
RST	Button for MP40 module reset
Media Card	Auxiliary Memory Device (NAND Flash)
RUN LED	Status of MP40 operation - Off: No-power - On (Green): On Booting - Blink (Green): Normal Operation of Program The blink cycle is 500 ms while running S/W
LAN LED	Status of LAN operation - Off: No-power and no-connection of LAN port - On: The color of the LED shows the LAN transmission speed and connection speed. • On (Green): Good connection + Operation at 100 Mbps

Ports, LEDs	Functions
	- Blink (Green): Good connection + Operation at 10 Mbps
MC LED	Memory Card Access Status - Off: No-Memory Card - On: The color of the LED shows the mounting status and the normal operation status after access. • On (Green): Mounted + Normal Operation • On (Red): Non-mounted or Mounted + Abnormal Operation - Blink (Green): Memory Card is mounted and in access mode.
MDM LED	The mounting status and the operation status of the MODEM - Off: No-MODEM - On (Green): MODEM mounted - Blink (Green): On transmitting data

LP40 (Sub-Control Section Module)

LP40 is the minor control module to controls overall functions of OfficeServ 7400 and is mounted on the slot 0 in the basic and expansion chassis. LP40 manages user cards and terminals under the control of the main control module, MP40, and transfers various event signals generated in the user cards and terminals to MP 40.
3 optional modules are mounted on the LP40 module. It is available to mount optional modules selectively according to a function. For the functions of the optional modules, there are DTMF, R2, CID, Conference and MISC.
The front view of the LP40 minor control module is shown in the figure below:

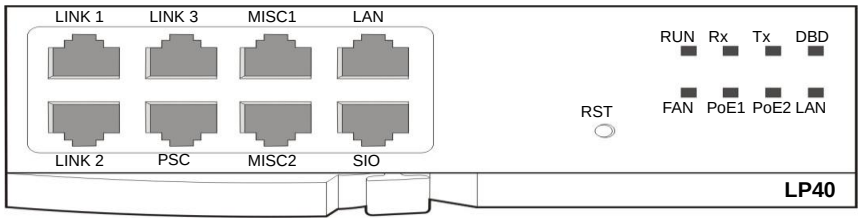


Figure 34. Front View of LP40

The components on the front view have the functions below:

Port, LED	Function Description
LINK1~LINK3	Ports for connecting MP40 and LP40
MISC1~MISC2	Ports for connecting external music, paging, loud bell, common bell and door bell
PSC	Connection port for PoE power status check
LAN	Port for connecting 10/100 BASE-T LAN (for tests)
SIO	UART port (for tests), CPLD JTAG port (for management)
RST	Button for resetting LP40 module

Port, LED	Function Description
RUN LED	Indicating the status of LP40. - Off: No power supplied or abnormal status - Orange Blink: On booting - Green Blink: Normal status
Rx LED	Indicating the status related to data receipt in communication with MP40. - Off: No signal - Green Blink: Data reception in progress
TX LED	Indicating the status related to data transmission in communication with MP40 - Off: No signal - Green Blink: Data transmission in progress
DBD LED	Indicating the daughter board mount. - Off: Daughter board dismounted - Green On: 1 daughter board or more mounted
FAN LED	Indicating the operation of FAN. - Green On: All fans normal - Red Blink: 1 FAN or more abnormal
PoE1 LED	Indicating the status of PoE1 power supply. - Off: PoE1 power supply dismounted - Green On: Normal - Red On: Abnormal
PoE2 LED	Indicating the status of PoE2 power supply. - Off: PoE2 power supply dismounted - Green On: Normal - Red On: Abnormal
LAN LED	Indicating the status of LAN. - Off: LAN disconnected - Green Blink: Operated in 10 Mbps - Green On: Operated in 100 Mbps

2.3.2 OfficeServ 7200

This section introduces the hardware features, cabinet configuration, and board functions and configuration of the OfficeServ 7200. Also, this section describes the terminals, wireless LAN equipment, and additional equipment available in the OfficeServ 7200.

Cabinet Configuration

The OfficeServ 7200 consists of two cabinets (basic/expansion cabinet) mounted on the 19-inch rack and a feature server that operates externally.
The MP20, which is the main control part and is installed into the basic cabinet, manages the entire OfficeServ 7200, performs switching, processes signals, and manages the user terminals. The LCP, which is the minor control part and is installed into the expansion cabinet, controls the line boards and sends/receives information to/from the MP20.
In addition, line boards, power, and cooling fans are in the OfficeServ 7200 cabinets.

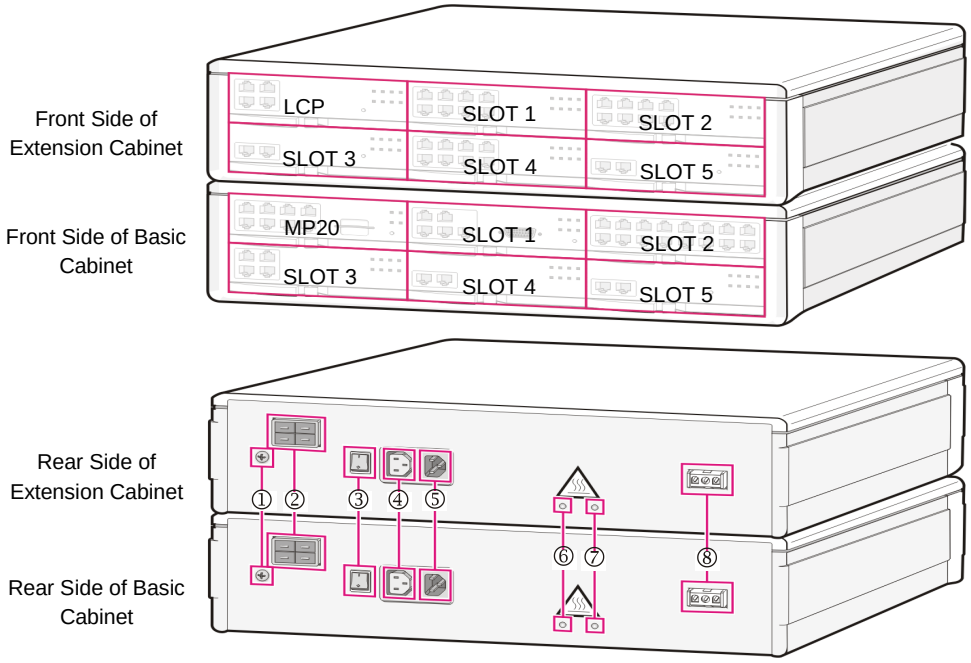


Figure 35. Cabinet Configuration of the OfficeServ 7200 with MP20

Rear Side Configuration of Cabinet	Function
① Ground Lug	Lug for grounding system communication
② External Rectifier Socket	External DC power supply socket for PoE.
③ Power Switch	Turn on/off the OfficeServ 7200 power.
④ Power Connection Connector between Basic Cabinet and Extension Cabinet	The connector to supply the supplied power to the external cabinet.
⑤ Power Input/Output Connector	The connector to connect power cable.
⑥ AC LED	LED is turned on when AC power is entered.
⑦ DC LED	LED is turned on when DC power is properly output.

Rear Side Configuration of Cabinet	Function
⑧ Battery Connection Socket	The socket to connect the external battery.

Slot Configuration

There are 6 slots that are available for mounting boards in each of the basic cabinet and extension cabinet. The following table shows mountable boards for each slot of the cabinet.

Cabinet	Slot	Mountable Boards
Basic Cabinet (OfficeServ Access)	Slot 0	MP20, MP20S
	Slot 1	All boards except for MP20, MP20S, LCP, TEPRIa, LIM, PLIM, PLIM2, GPLIMT, and GSIMT
	Slot 2	All boards except for MP20, MP20S, LCP, TEPRIa and WIM
	Slot 3, 4, 5	All boards except for MP20, MP20S, LCP, and WIM
Expansion Cabinet (OfficeServ Expansion)	Slot 0	LCP only
	Slot 1	All boards except for MP20, MP20S, LCP, TEPRIa LIM, PLIM, PLIM2, GPLIMT, and GSIMT
	Slot 2	All boards except for MP20, MP20S, LCP, TEPRIa and WIM
	Slot 3	All boards except for MP20, MP20S, LCP, and WIM
	Slot 4, 5	All boards except for MP20, MP20S, LCP, TEPRIa and WIM

MP20

The MP20 is the main control board that controls all the functions of the OfficeServ 7200 and is mounted on slot 0 of the basic cabinet. The MP20 performs voice switching, processes signals, and manages the user terminals.
The front view of the MP20 is shown in the figure below:

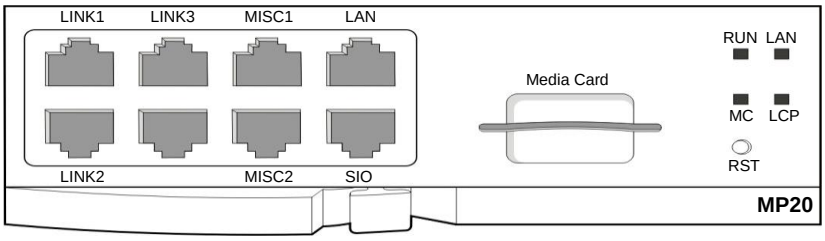


Figure 36. Front View of the MP20

The components on the front panel of the MP20 have the functions below:

Ports, LEDs	Functions
LINK1~3	Ports that connect the MP20 with the LCP.
MISC1~2	Ports that connect external music sources, paging device, loud bell, common bell, or door bell.
LAN	Port for establishing the 10/100 BASE-T Ethernet connection.
SIO	UART port (for tests).
Media Card	Port for installing the NAND-type flash memory.
RUN LED	This LED indicates the status of the MP20. - Off: Power is not connected. - On (Green): Booting. - Blink (Green): The Program is operating properly. - Blink (Red): Fan module failed. - Blink (orange): Reset button is pushed. - ON (orange): Flash Memory (Data base) clear
LAN LED	This LED indicates the status of the connection to LAN. - Off: MP20 is not connected to LAN. - On: MP20 is connected LAN. - Blink: MP20 is transmitting or receiving Data through LAN port.
MC LED	This LED indicates the status of the Smart Media/Media Card access. - Off: The SD Card is not installed. - On: The SD Card is installed, however is not accessed. - Blink: The SD Card is installed and is being accessed.
LCP LED	This LED indicates the status of signaling message processing. - Off: There's no message exchange between MP20 and LCP. - On: Messages are being sent/received to/from the LCP.
RST	Button for resetting the MP20 board. Button for Data base clear when pushed more than 7second.

LCP

The Local Control Processor (LCP) is the minor control board that interworks the MP20, which is the main control part of the basic cabinet, with the expansion cabinet.
The LCP controls a variety of line boards and sends/receives information to/from the MP20.
The front view of the LCP is shown in the figure below:

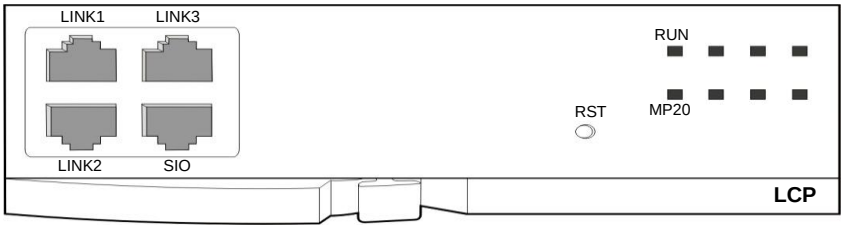


Figure 37. Front View of the LCP

The components on the front panel of the LCP have the functions below:

Ports, LEDs	Functions
LINK1~3	Ports that connect the MP20 with the LCP.
SIO	UART port (for tests).
RST	Button for resetting the LCP board.
RUN LED	This LED indicates the status of the LCP. - Off: Power is not connected. - On: Booting. - Blinking: Program in operation.
MP20 LED	This LED indicates the status of signaling message processing. - Off: There's no message exchange between MP20 and LCP. - On: Messages are being sent/received to/from the MP20.

2.3.3 OfficeServ 7100

This section introduces the hardware features, cabinet configuration, and board functions and configuration of OfficeServ 7100 system. In addition, this section describes terminals, wireless LAN equipment, and additional equipment available in OfficeServ 7100 system.

Cabinet Configuration

OfficeServ 7100 is configured of the main device with a basic cabinet and OfficeServ solution. The Main device cabinet is a one stage shelf composed of three slots and consists of a control part on the main slot and two user parts on the universal slot.

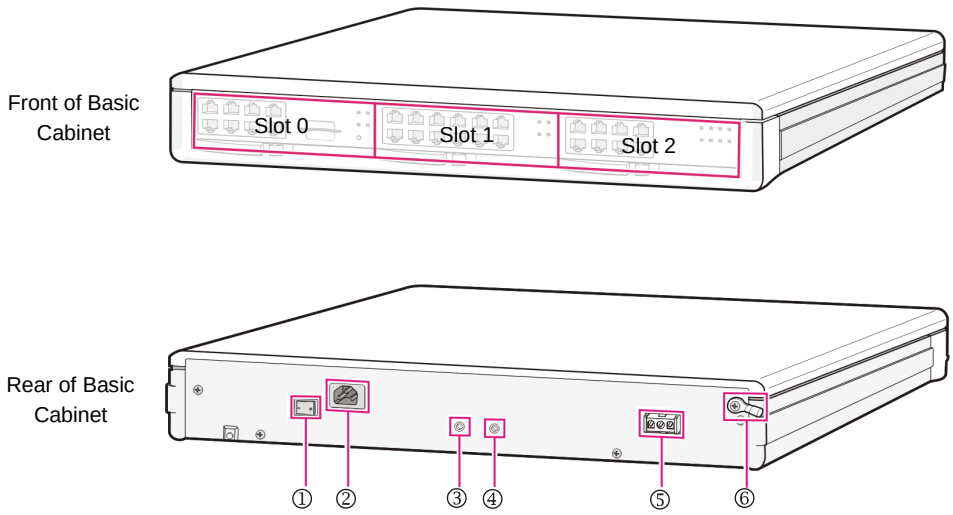


Figure 38. Configuration of OfficeServ 7100 Cabinet

Configuration	Function
① Power Switch	Power on/off OfficeServ 7100 system.
② Power I/O connector	Connector to connect the power cable
③ AC LED	The LED turns on while applying AC power.
④ DC LED	The LED turns on while the DC power normally comes out.
⑤ Battery Socket	Socket to connect an external battery
⑥ Ground Lug	Lug to ground the system communication

Configuration of Slots

OfficeServ 7100 has three board slots. These boards are equipped with the following boards depending on the configuration of OfficeServ:

Cabinets	Slots	Mountable Boards
Main Control Part	Slot 0	MP10, MP10a, MP11
User Part	Slot 1 and Slot 2	- OS 7100 Card: UNI board - OS 7200 Card: 8DLI/16DLI2, 8SLI/16SLI2, 8COMBO, 8TRK, TEPRIa, LIM, MGI16 - OS7400 Card: TEPRI2, MGI64

MP10a

This paragraph describes the configuration and the functions of MP10a board, which are the main control board that controls all functions of OfficeServ 7100.

MP10a board is a main control part board that controls all functions of OfficeServ 7100 and is mounted on slot 0 of the basic cabinet. It performs the voice switching function, signal processing function and PSS management function. MP10a board carries out the system booting function and data management function
If 4SWM, which is an option board, is not equipped, MP10a board is connected to LAN Interface Module (LIM) of the universal slot or an external switch via the LAN interface and starts various applications. If the 4SWM is mounted, LAN interface is connected automatically. MP10a board strengthens the flexibility of system and by applying the VoIP function and IPC between cabinets raises the reliability by using the HDLC protocol.
The front view of the MP10a main control board is as shown in the figure below.

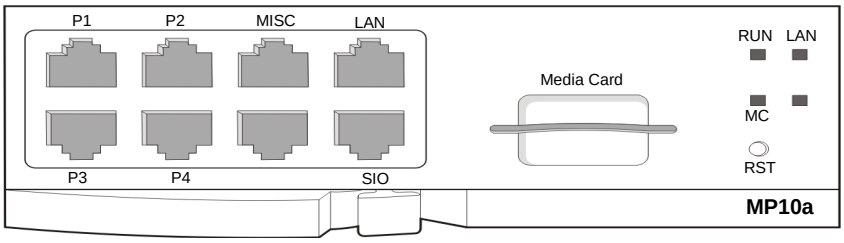


Figure 39. Front View of MP10a

The components on the front panel of the MP10a boards have the functions below:

Ports and LEDs	Functions
P1~P4	User connection port of 4DLMI/4SWM
MISC	Port to connect Ext PAGING, DRY CONTACT and Ext MOH
LAN	Port to connect 100 BASE-T LAN
WAN	Port to connect 100 BASE-T WAN (only MP11)
SIO	Port connect to serial I/O for development tool
Media Card	Port to insert an MMC + /SD card, which is a storage media
RUN LED	Status of Main CPU operation - Off: No-power - On (Green): On Booting, Reset - Blink (Green): Normal Operation of Program - Blink (Red): Fan module failed Operation of Program - Blink (Orange): Push the reset button under the 7 sec (MP10a, MP11) - On (Orange): Push the reset button over the 7 sec, DB clear (MP10a, MP11)
LAN LED	Status of LAN operation - Off: Link and no-connection of LAN port - On (Green): Link and LAN port connection - Blink (Green): Tx/Rx Data through LAN port.
WAN LED	Status of WAN operation (MP11) - Off: Link and no-connection of WAN port - On (Green): Link and WAN port connection - Blink (Green): Tx/Rx
MC LED	Status of MMC + /SD card operation - Off: Non-mounted MMC + /SD card - On (Green): Mounted MMC + /SD card - Blink (Green): In Tx/Rx of MMC + /SD card - On (Red): If the Multi Media card is installed but not detected
RST	Button for resetting board and DB clearing.

2.3.4 OfficeServ 7070

This section introduces the hardware features, cabinet configuration, and board functions and configuration of OfficeServ 7070 system. In addition, this section describes terminals, wireless LAN equipment, and additional equipment available in OfficeServ 7070 system.

Cabinet Configuration

The OfficeServ 7070 is installed on a wall. The system has a single control part (BMP). Various subscriber option boards are mounted onto the BMP (MAIN part), BMP (B8S part) andE8S.ThefollowingsectionsshowtheappearanceoftheOfficeServ7070systemcabinet.

OfficsServ 7070 top view

Figure 40. OfficeServ 7070 Top View

The descriptions about each part are listed in the table below.

Table 1. Parts on the top of OfficeServ 7070

Part	Function
① RUN LED	CPU operation status
② LAN LED	LAN operation status
③ MEM LED	CPU access status of Flash Memory
④ PRI LED	Port status

OfficeServ 7070 Side view

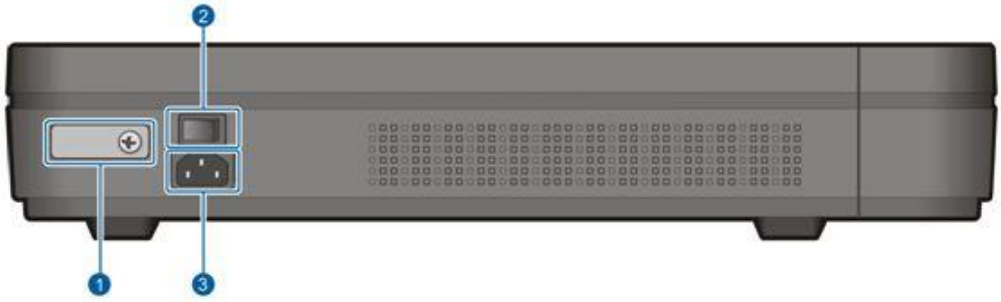


Figure 41. OfficeServ 7070 side view-1

The descriptions about each part are listed in the table below.

Table 2. Parts on the side of OfficeServ 7070-1

Part	Function
① Ground Lug	Ground lug for system communications
② Power Switch	Switch to turn the OfficeServ 7070 on/off
③ Power Connector	Connector to use when connecting to the power cable

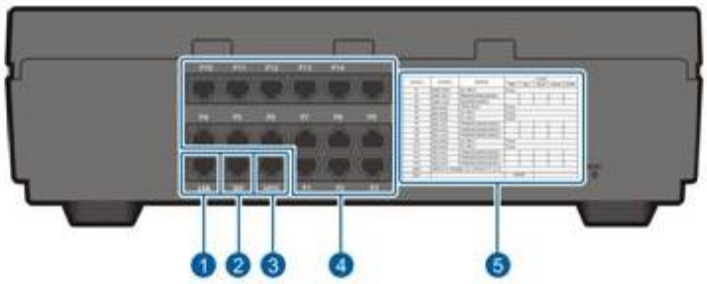


Figure 42. OfficeServ 7070 side view-2

The descriptions about each part are listed in the table below.

Table 3. Parts on the side of OfficeServ 7070-1

Part	Function
① Ground Lug	Ground lug for system communications
② Power Switch	Switch to turn the OfficeServ 7070 on/off
③ Power Connector	Connector to use when connecting to the power cable

Table 4. Parts on the side of OfficeServ 7070-2

Part	Function
① LAN	LAN port
② SIO	Debugging port
③ MISC	Connector used to connect to an external audio device, broadcasting device, or shared bell, etc.
④ P1~P14	General-purpose ports
⑤ Installation Record Label	Label to record the installed boards on.

Slot Configuration

The user can mount up to three (3) option boards on the BMP board (MAIN part), three (3) on the BMP board (B8S part), and three (3) on the E8S board. The option boards that can be mounted within the OfficeServ 7070 depending on its configuration are listed in the table below.

Table 5. Mountable Boards for Different Slots

Cabinet	Module	Slot	Mountable Board
Basic Cabinet	BMP (MAIN Part) -	LOC1	PRM, 4DLM, 4SL2
		LOC2	4DLM, 4SL2
		LOC3	Modem
	BMP (B8S Part) -	LOC1	4TRM, 4DLM, 4SL2, 2BRM
		LOC2	4TRM, 4DLM, 4SL2, 2BRM
		LOC3	4TRM, 4DLM, 4SL2, 2BRM
Optional Cabinet	E8S	LOC1	4TRM, 4DLM, 4SL2, 2BRM
		LOC2	4TRM, 4DLM, 4SL2, 2BRM
		LOC3	4TRM, 4DLM, 4SL2, 2BRM

2.3.5 ISDN Trunk Cards

TEPRIa

TEPRIa board provides the digital trunk line. TEPRIa board provides ISDN E1 (T1) PRI, and functions as the Q-SIG. This board transmits voice via the trunk line and a channel transmits the voice data of 64 Kbps.
The front view of TEPRIa voice trunk line board is shown in the figure below:

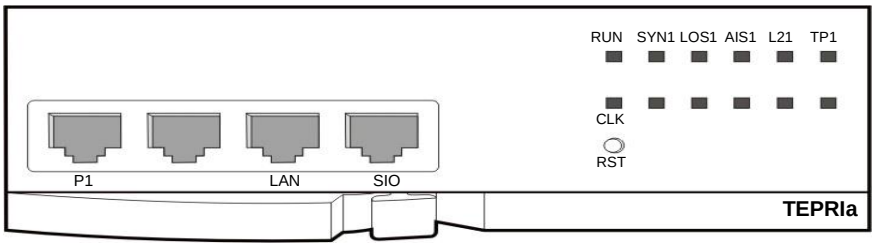


Figure 43. Front View of TEPRIa

The components on the front panel of TEPRIa have the functions below:

Port, LED	Function Description
P1	Ports to connect T1/E1/PRI cables
LAN	Port connected to Ethernet
SIO	UART port (for test)
TP1 LED	Program type on operating in port1 - On: Port1 operation in PRI - Off: Port1 operation in T1/E1
L21 LED	Layer 2 operation status - On: Normal operation of Layer 2 - Off: Abnormal operation of Layer 2
AIS1 LED	Reception status of alarm bit from the counterparty switch - On: Reception of alarm bit from the counterparty switch - Off: No-reception of alarm bit from the counterparty switch
LOS1 LED	Signal loss status (LOS) from the counterparty switch - On: Weak signal or signal loss from the counterparty switch - Off: Normal signal reception for the counterparty switch
SYN1 LED	Frame synchronization status with the counterparty switch - On: Out of synchronization with the counterparty switch - Off: Synchronization with the counterparty switch
RUN LED	- On (Green): Normal operation (blink at the interval of 200 ms) - On (Orange): Debug mode operation (blink at the interval of 200 ms)
CLK LED	On when the Reference clock is used as the system clock
RST	Button for resetting board

TEPRI2

TEPRI2 provides the digital trunk line. TEPRI2 board provides two ports for ISDN PRI respectively, and functions as the Q-SIG. This board transmits voice via the trunk line and a channel transmits the voice data of 64 Kbps.
The front view of TEPRI2 voice board is shown in the figure below:

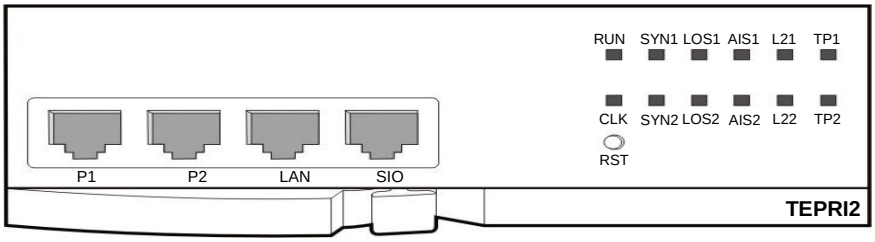


Figure 44. Front View of TEPRI2

The components on the front panel of TEPRI2 have the functions below:

Port, LED	Function Description
P1	Port 1 for connecting T1/E1/PRI cables
P2	Port 2 for connecting T1/E1/PRI cables
LAN	Port for connecting to Ethernet
SIO	UART port (for tests)
TP1 LED	Indicating the type of the program operated in Port 1 - On: Port 1 operation in PRI - Off: Port 1 operation in T1/E1
TP2 LED	Indicating the type of the program operated in Port 2 - On: Port 2 operation in PRI - Off: Port 2 operation in T1/E1
L21 LED L22 LED	Indicating the status of Layer 2 operation - On: Normal - Off: Abnormal
AIS1 LED AIS2 LED	Indicating the reception of the alarm bit of the counterparty switch - On: Alarm bit received - Off: Alarm bit not received
LOS1 LED LOS2 LED	Indicating the signal loss (LOS) of the counterparty switch - On: When signals are week or has been damaged - Off: When signals received properly
SYN1 LED SYN2 LED	Indicating the status of frame synchronization with the counterparty switch - On: No frame synchronized - Off: Frame synchronized
RUN LED	- On (Green): Normal operation(blink at the interval of 200 ms) - On (Orange): Debug mode operation (blink at the interval of 200 ms)
CLK LED	On when the reference clock is used as the system clock.
RST	Button for resetting board

4BRI

The 4BRI module provides 4 ports for BRI-T/S connection. It transmits voice via the trunk line and a channel transmits voice data at 64 kbps.
The front view of the 4BRI voice module is shown in the figure below:

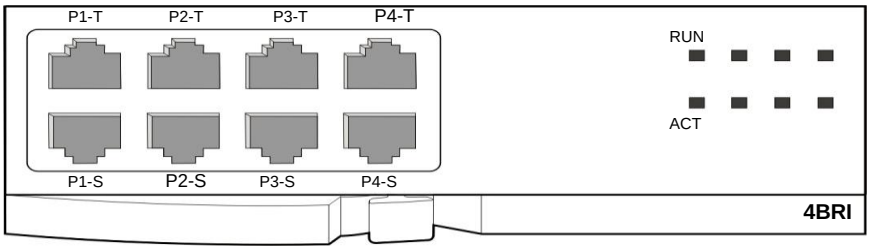


Figure 45. Front View of 4BRI

The components on the front panel of the 4BRI have the functions below:

Port, LED	Function Description
P1-T~P4-T	Office (trunk) Port
P1-S~P4-S	Internal (station) Port
RUN LED	4BRI Board Operation Status - Off: 4BRI board is abnormal or power is disconnected - Blink: 4BRI board is properly operated
ACT LED	4BRI Board LAYER1 SETUP Status - Off: LAYER1 SETUP Abnormal - On: LAYER1 SETUP Normal

2.3.6 Analog Trunk Cards

8TRK

8TRK board provides analog trunk line ports. One board has the CID path. In addition, the board provides voice through trunk lines and transmits the voice data of 64 kbps to each channel.

8TRK2/16TRK

The 8TRK2/16TRK board provides 8/16 ports of analog trunk line, and supports the PRS, CID paths. It also provides the voice though the trunk line; each channel supports 64 Kbps voice data transmission.

4HTRK

The 4HTRK (Hybrid Trunk) provides 4 ports of analog trunk line, and can support the DID, E & M and R/D paths within a single board. It provides the voice though the trunk line; each channel supports 64 Kbps voice data transmission.

2.3.7 Analog Phone Cards

8SLI

8SLI (Single Line Interface) board supports 8-port for analog stations.
It interworks with regular phones via the station to provide voice communication.

8SLI2/16MWSLI/16SLI2

The 8SLI2/16MWSLI/16SLI2 module has 8 port/16 ports for analog stations.
It interworks with regular phones via the station to provide voice communication.
16MWSLI module functions as message waiting.

2.3.8 UNI

UNI can be used as a voice trunk line board or voice user line board depending on the mounted option board. If 4TRM and 2BRM are mounted on UNI, it operates as a voice trunk line board. If 4SLM, 4SL2 and 4DLM are mounted, it operates as a voice user line board.
The front view of UNI is shown in the figure below:

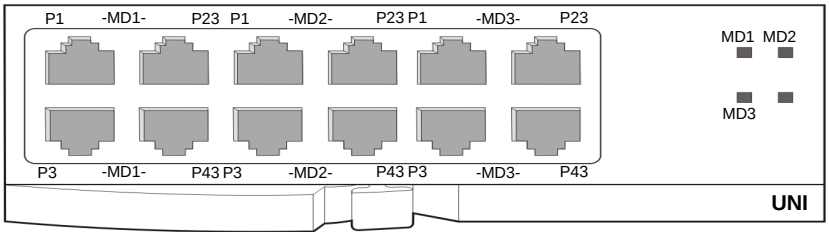


Figure 46. Front View of the UNI

The components on the front panel of UNI functions as follows:

Ports, LEDs	Functions
MD1 P1~4 MD2 P1~4 MD3 P1~4	Port support in accordance with the option boards mounted on the position of MD1, MD2 and MD3
MD1 LED MD2 LED MD3 LED	Module mounting status at the each MD position and user status - Off: No-module mounted - On (Red): 4DLM mounted - On (Green): 4TRM or 2BRM mounted - On (Orange = Green + Red): 4SLM or 4SL2 mounted

Mountable option boards for UNI are as follows.

Option board	MP20			Max
	LOC1	LOC2	LOC3	
2BRM	○	○	○	3
4TRM	○	○	○	3
4SLM	○	○	○	3
4SL2	○	○	○	3

If 2BRM and 4TRM is mounted, UNI operates as a voice trunk line board.

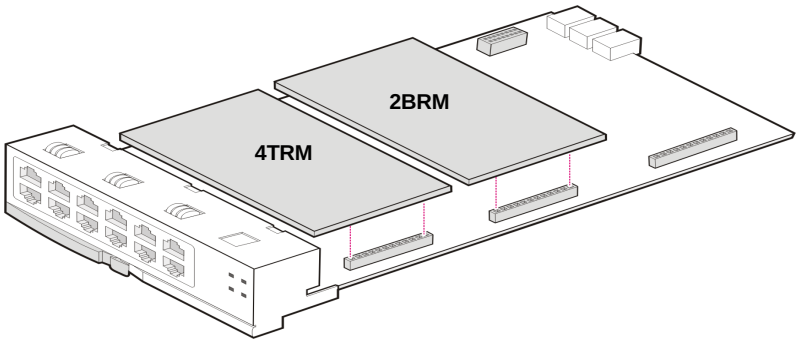


Figure 47. Option board of the UNI (Voice Trunk Line)

2BRM

The 2BRM (BRI Module) is mounted on UNI and provides two 2B+1D digital trunk ports. This module is connected to 4 channel ISDN trunk in S and T mode. In S mode do not support DC power feeding.

4TRM

The 4 Port Trunk Module (4TRM) is mounted on UNI and provides four (4) analog trunk ports. A 4TRM provides both of the PRS and CID paths.
The 4TRM can be mounted as a daughter board on the UNI.
There is no separate line connection part within the 4TRM. It is connected to an external line through the RJ-45 connector on the line connection part, located at the left side of the system.

4SL2

The 4 Port SLI Module 2 (4SL2) is mounted on UNI and is used to process regular phone connections.

4SLM

The 4 Port SLI Module (4SLM) is mounted on UNI and is used to process regular phone connections.

2.3.9 Media Gateway Cards

MGI64/MGI16

MGI64/MGI16 is a module that transmits and receives voice via data network after converting into data. Up to 64 channels and 16 channels are provided for MGI64 and MGI16, respectively. In addition, MGI64 and MGI16 decompress voices of G.729, G.723, G.726 and G.711. The MGI64/MGI16 provides the VoIP functions to serve as both of a client and server. The MGI64/MGI16 board converts voice to data and sends/receives the data via the data network. Up to 64 channels and 16 channels are provided for MGI64 and MGI16, respectively. In addition, MGI64 and MGI16 decompress voices of G.729, G.723, G.726 and G.711. It also provides a T.38 specification compliant Fax function.

The front view of the MGI64 voice application module is shown in the figure below:

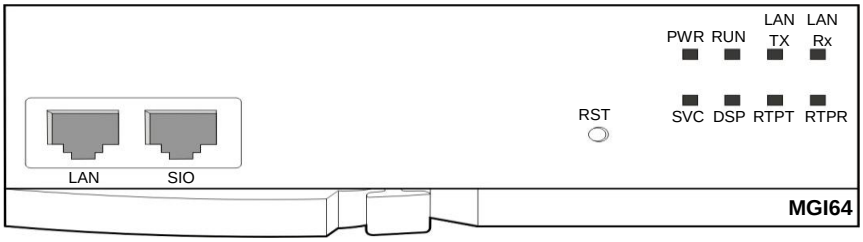


Figure 48. Front View of MGI64

The components on the front panel of MGI64/MGI16 are as follows:

Port, LED	Function Description
LAN	Port for connecting to Ethernet
SIO	UART port (for tests)
RST	Button for resetting MGI module
PWR LED	Power supply - Off: Power supply blocked - On: Power supplied
RUN LED	MGI64/MGI16 status - Off: Power supply blocked - On: On booting - Blink: RAM program in operation
LAN Tx LED	Ethernet Data Transmission - Off: No data - On or Blink: Data are being transmitted
LAN Rx LED	Links and Ethernet Data Reception - Off: No data or no link connected - On or blink: On data reception
SVC LED	Service

Port, LED	Function Description
	- LED blinks when the task service of the software is available
DSP LED	VoIP DSP operation - LED blinks when VoIP DSP is operated
RTPT LED	Voice packet transmission - LED turns on when transmitting voice packets
RTPR LED	Voice packet reception - LED turns on when receiving voice packets

OAS (OfficeServ Application Server)

The OAS converts the voice into data and transmits the data through the data network. It supports a maximum of 32 channels and provides the G.729, G.723, G.711 voice compression/decompression function. It also provides a T.38 specification compliant Fax function. OAS can be mounted on slots 1, 2, 3, 4, and 5 of the basic and expansion cabinet. If the OAS board is mounted on one of the slot 1 and 2, no board can be mounted on the other slot. If any board is mounted on the other slot, the board will not work.
The front view of the OAS is shown in the figure below:

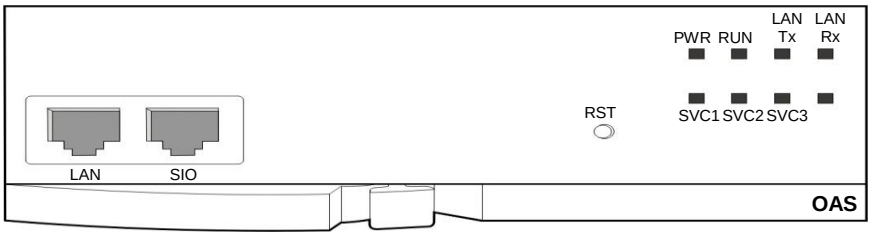


Figure 49. Front View of the OAS

The components on the front panel of the OAS have the functions below:

Ports, LEDs	Functions
LAN	Port that connects the Ethernet.
SIO	UART port (for tests).
RST	Button for resetting the OAS.
PWR LED	This LED indicates the power supply status. - Off: Power is not being supplied. - On: Power is being supplied properly.
RUN LED	This LED indicates OAS status. - Off: Power is not being supplied. - On: Booting. - Blink: The RAM program is operating.
LAN Tx LED	This LED indicates the status of the Ethernet data transmission. - Off: Data does not exist.

Ports, LEDs	Functions
	- On or blink: Data is being transmitted.
LAN Rx LED	This LED indicates the reception status of the link and Ethernet data. - Off: Data does not exist or the link is not connected. - On or blink: Data is being received.
SVC1 LED	This LED indicates if the MGI service is being offered. - This LED turns on when the MGI software task can be serviced.
SVC2 LED	This LED indicates if the MFR service is being offered. - This LED turns on when the MFR software task can be serviced.
SVC3 LED	This LED indicates if the MPS service is being offered. - This LED turns on when the MPS software task can be serviced.
SVC4 LED	This LED indicates if the LINK is connected to MP. - This LED turns on when the LINK is connected to MP.

2.4 IPX-G500B Gateway

2.4.1 Exterior

2.4.1.1 IPX-G500B

IPX-G500B can be mounted on 19-inch (482.6 mm) wide rack and its height is 1.75-inch (1U). The appearance is like the following.

Front

Front (Dual Power Supply)

Figure 50. IPX-G500B exterior

2.4.1.2 IPX-G520S

Front

Front (Dual Power Supply)

Figure 51. IPX-G520S exterior

2.4.1.3 IPX-G540S

Front

Figure 52. IPX-G540S exterior

2.4.2 Hardware Structure

2.4.2.1 IPX-G500B

IPX-G500B hardware consists of NPU board, GWU board, optional modules, power supply and 3 fans.

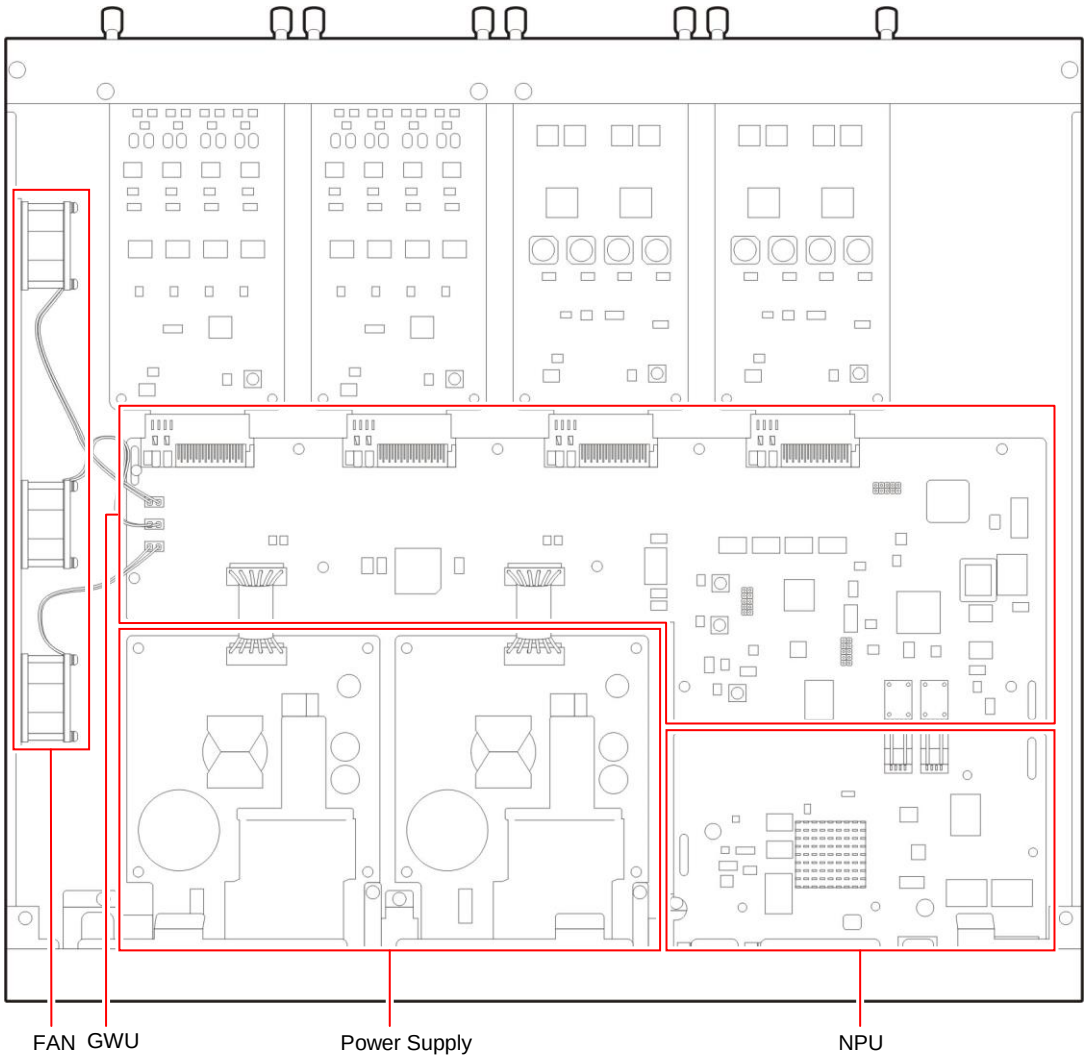


Figure 53. IPX-G500B interior

NPU Board

The NPU board of IPX-G500B is like the following.

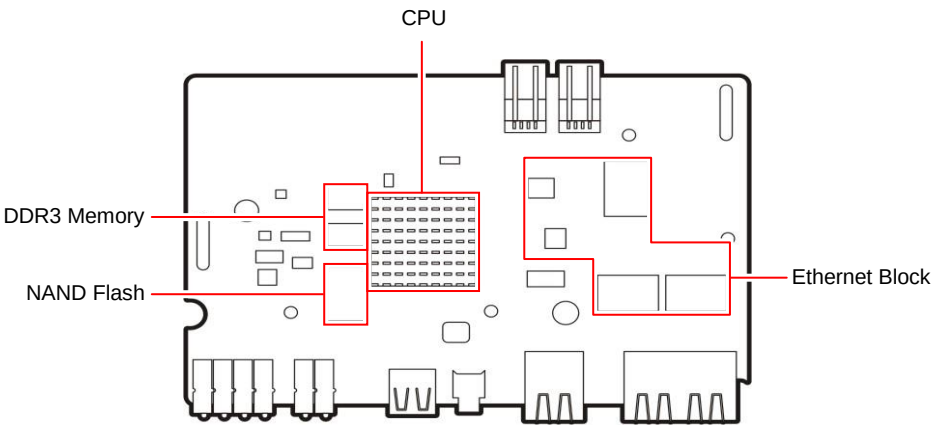


Figure 54. NPU Board

GWU Board

The GWU board of IPX-G500B is like the following.

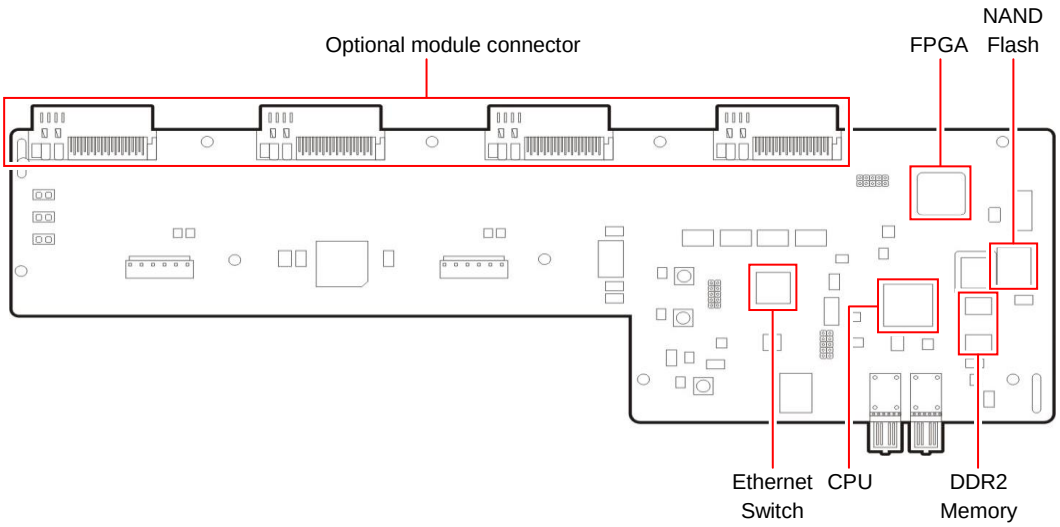


Figure 55. GWU Board

Optional Modules

There are the following optional modules for IPX-G500B.

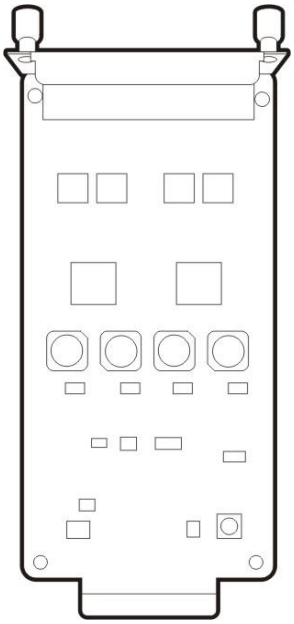


Figure 56. 4FXS optional module

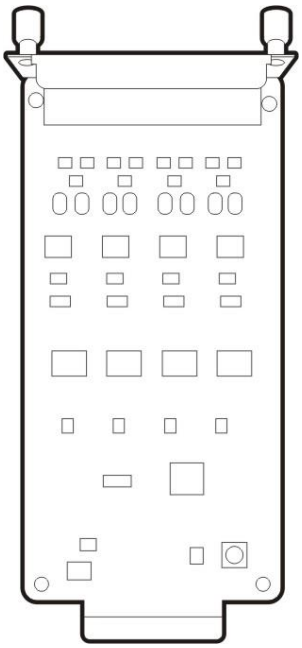


Figure 57. 4FXO optional module

Power Supply

Figure 58. IPX-G500B Power Input

Name	Description
① Power Input Connector	The connector to insert power cable
② Power Switch	The switch to turn on or off power

FAN

IPX-G500B provides 3 embedded fans (40 mm size) and it has holes for the fan on the left side of the system.

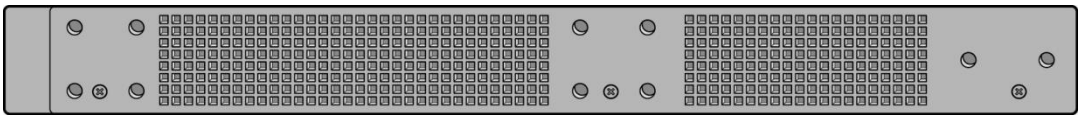


Figure 59. IPX-G500B FAN

2.4.2.2 IPX-G520S

IPX-G520S hardware consists of EDU board, 20FXS board, power supply and 3 fans.

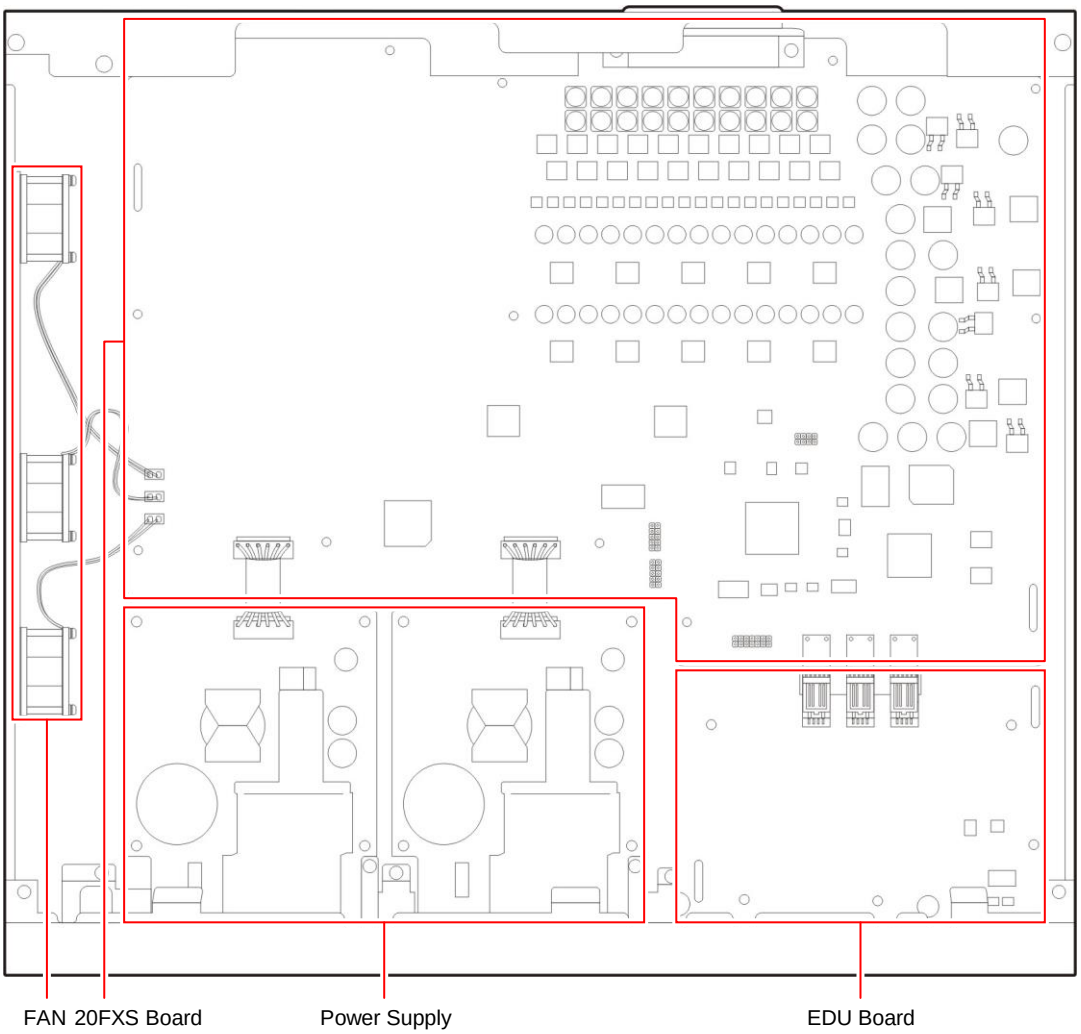


Figure 60. IPX-G520S Interior

EDU Board

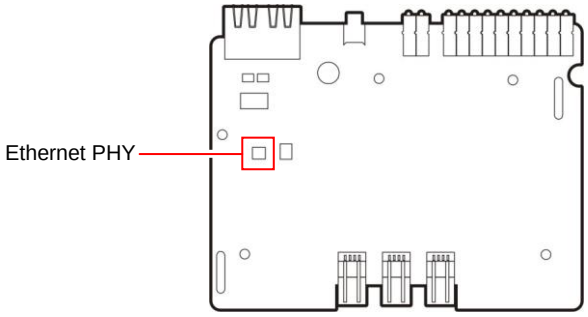


Figure 61. EDU Board

20FXS Board

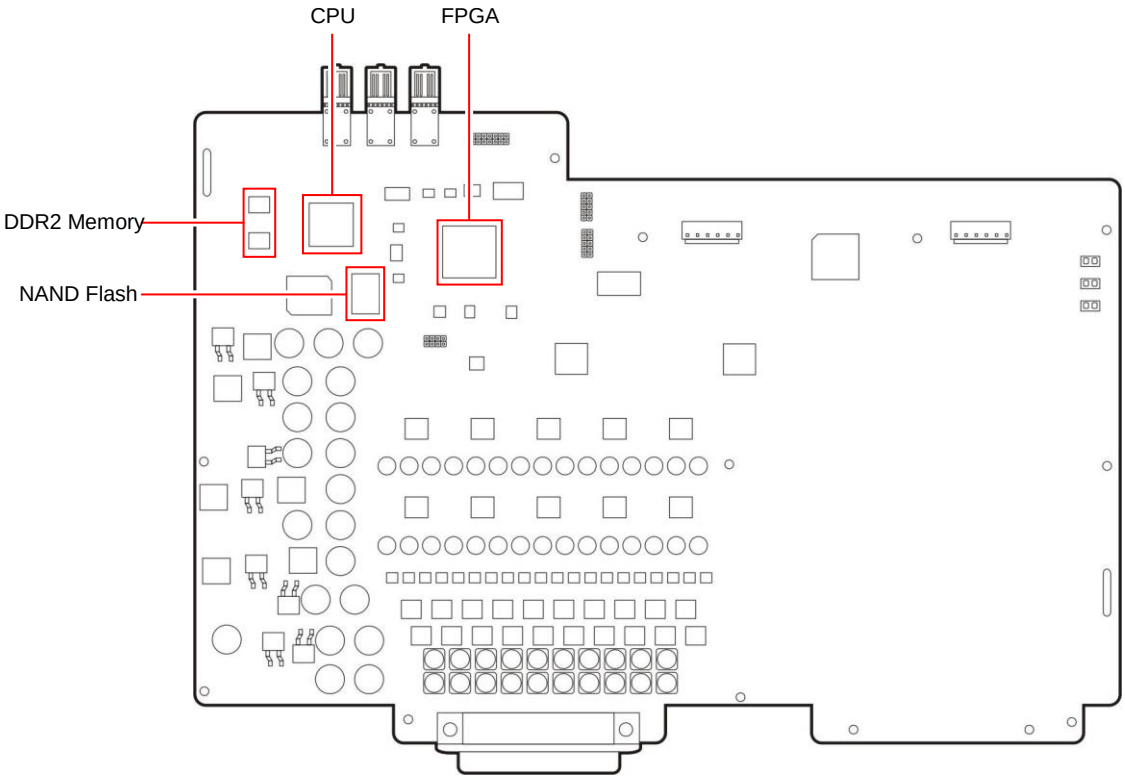


Figure 62. 20FXS Board

Power Supply

Figure 63. IPX-G520S Power Input

Name	Description
① Power Input Connector	The connector to insert power cable
② Power Switch	The switch to turn on or off power

FAN

IPX-G520S provides 3 embedded fans (40 mm size) and it has holes for the fan on the left side of the system.

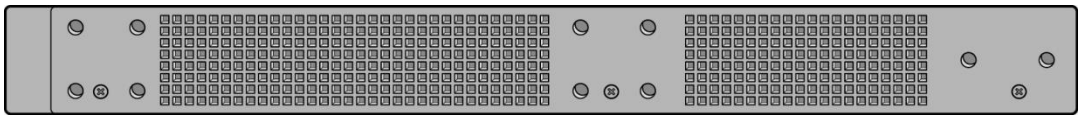


Figure 64. IPX-G520S FAN

2.4.2.3 IPX-G540S

IPX-G540S hardware consists of EDU board, 40FXS board, power supply and 3 fans.

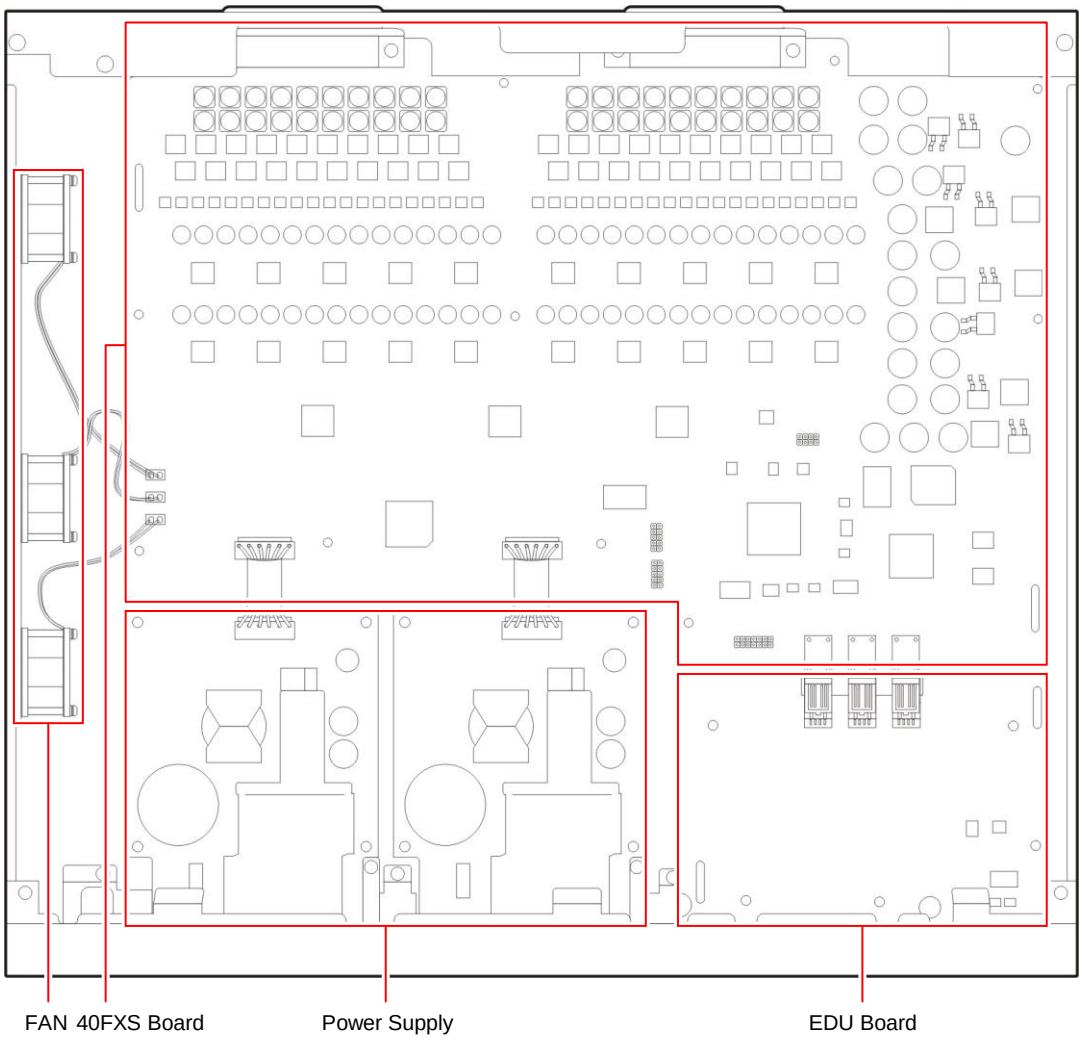


Figure 65. IPX-G540S Interior

EDU Board

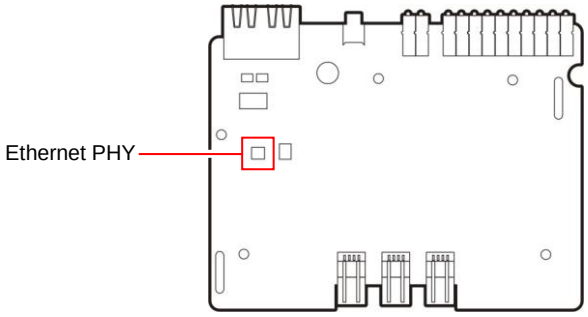


Figure 66. EDU Board

40FXS Board

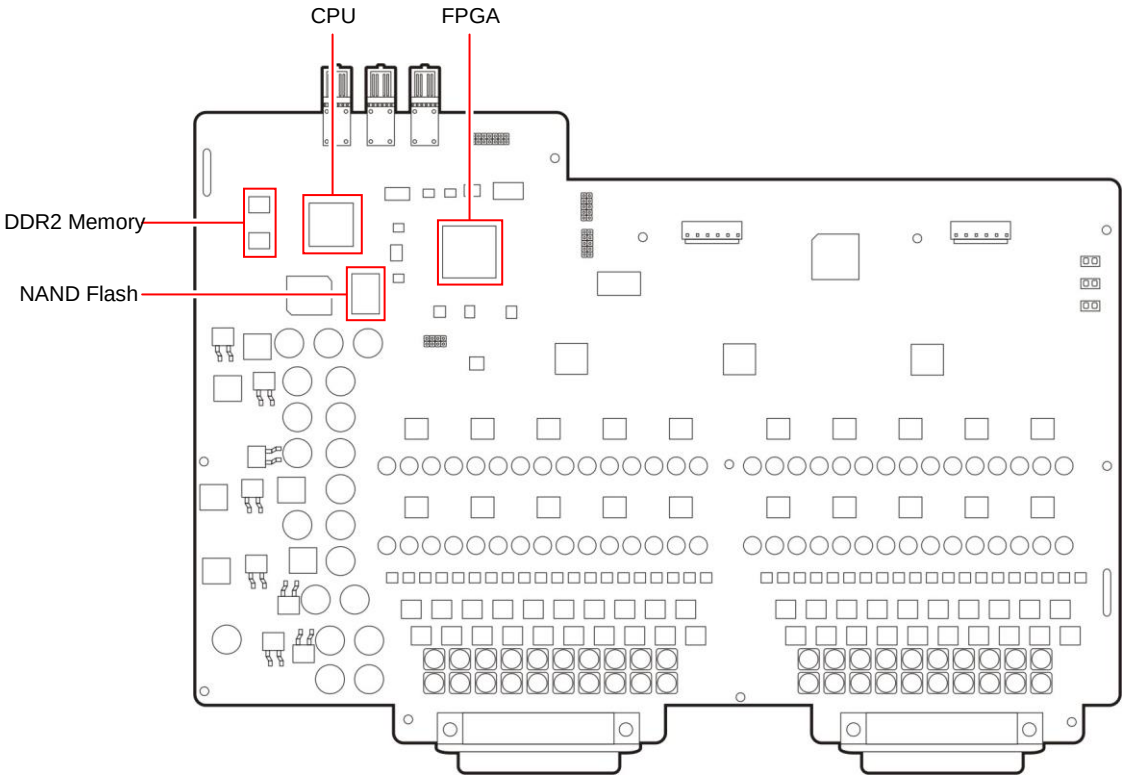


Figure 67. 40FXS Board

Power Supply

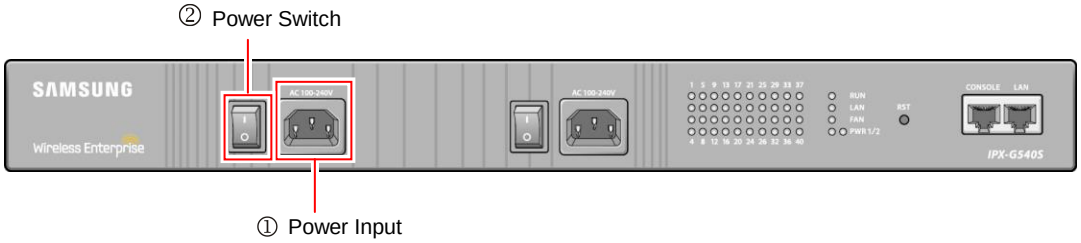


Figure 68. IPX-G540S Power Input

Name	Description
① Power Input Connector	The connector to insert power cable
② Power Switch	The switch to turn on or off power

FAN

IPX-G540S provides 3 embedded fans (40 mm size) and it has holes for the fan on the left side of the system.

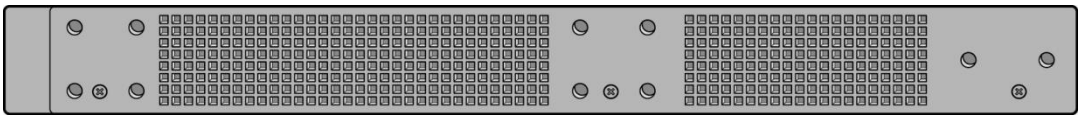


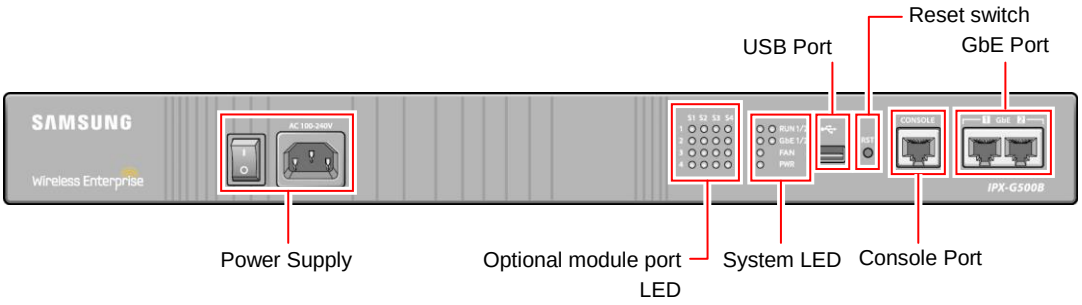
Figure 69. IPX-G540S FAN

2.4.3 External Interface

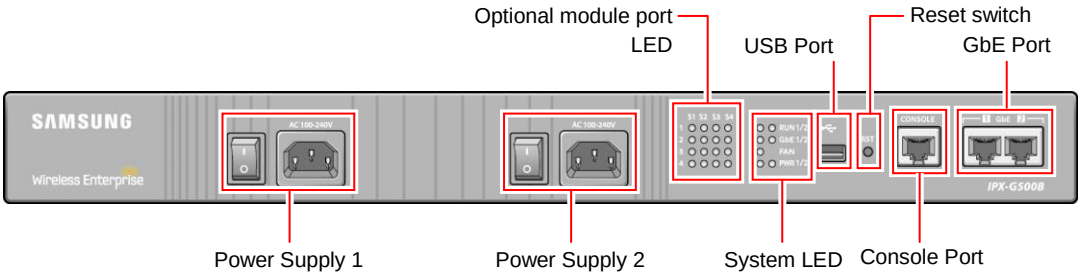
2.4.3.1 IPX-G500B

IPX-G500B provides external interfaces like the following.

- **IPX-G500B Front**



- **IPX-G500B (Dual Power Supply) Front**



- **IPX-G500B Rear**

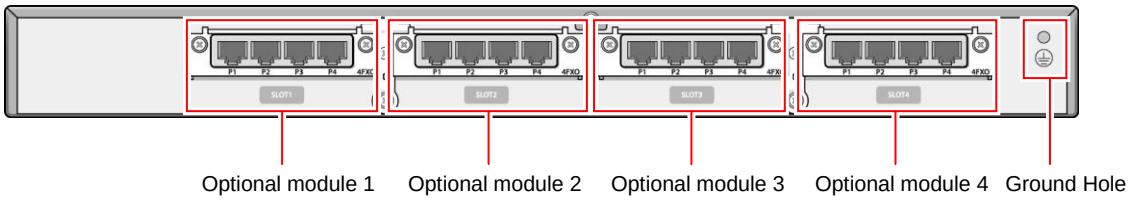
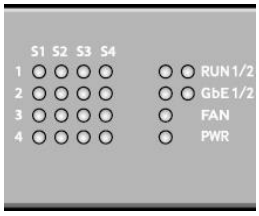


Figure 70. IPX-G500B Interface Front/Rear

System LED

It provides System LED to display various status of the system. Refer to the following for its meaning.

- **IPX-G500B**



- **IPX-G500B (Dual Power Supply)**

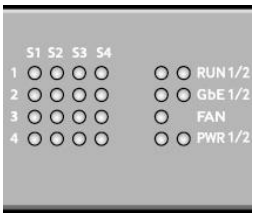


Figure 71. IPX-G500B LED

Table 6. IPX-G500B LED Description

LED		State	Description
Optional module port		Off	Module Not Exist or Ready to Service
		On	Operating
		Blink	Ringing state
System	RUN1/2	On	Preparing to boot
		Blink quickly	Booting
		Blink slowly	Normal operation
	GbE1/2	Off	LAN disconnected
		On	LAN connected
		Blink	TX/RX Data
	FAN	Off	FAN out of order
		On	Normal operation
	PWR1/2	Off	Power is off or out of order
		On	Power is on

Console Port (RS232C)

IPX-G500B provides console port to debug the system or to set basic configuration by cli. The baud rate is 38,400 bps.

GbE Port (1GE UTP)

To connect the network, it provides two 1000 BASE-T UTP ports.

USB Port (Host 2.0)

It provides USB port to update the software of IPX-G5X0 Series. It supports general USB memory stick. (This feature is not available yet.)

Reset Switch

It supports to reset the system by reset switch.

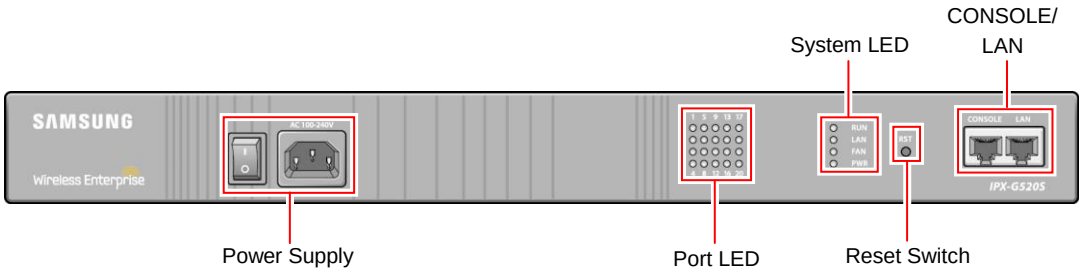
Optional Modules

IPX-G500B supports 4 optional modules. 4 ports FXO card or 4 ports FXS card.
At first, blank dummies are embedded, so remove them when you install optional modules.

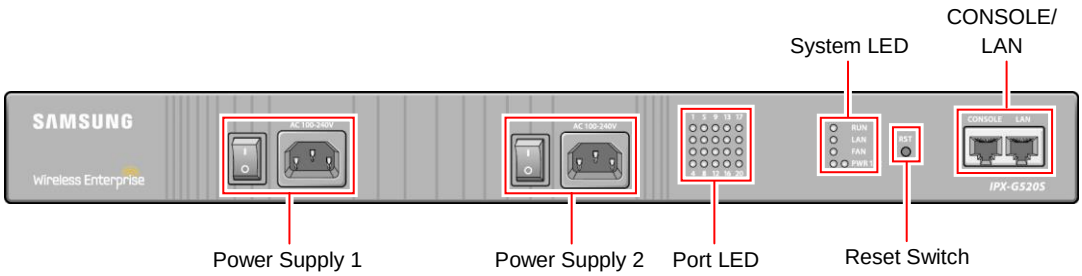
2.4.3.2 IPX-G520S/G540S

IPX-G520S/G540S provides external interfaces like the following.

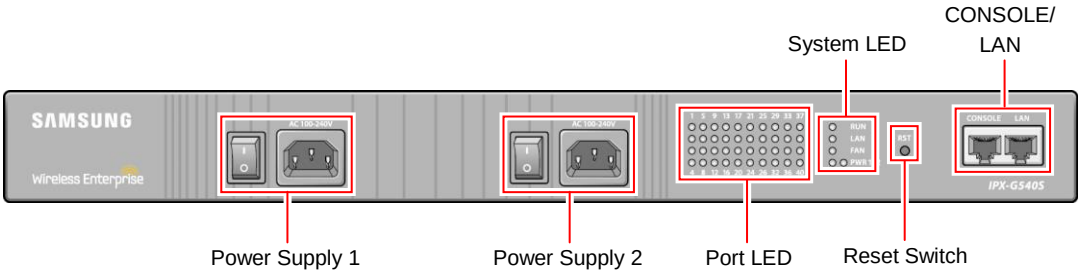
- **IPX-G520S Front**



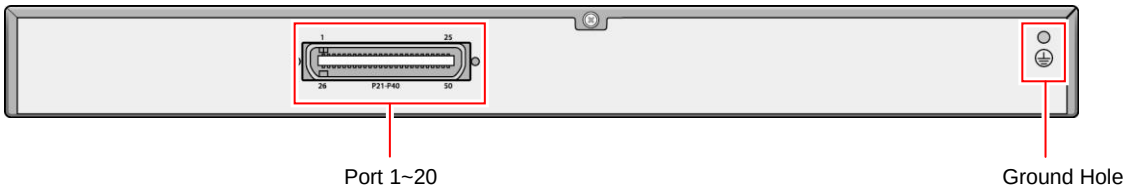
- **IPX-G520S (Dual Power Supply) Front**



- **IPX-G540S Front**



- **IPX-G520S Rear**



- **IPX-G540S Rear**

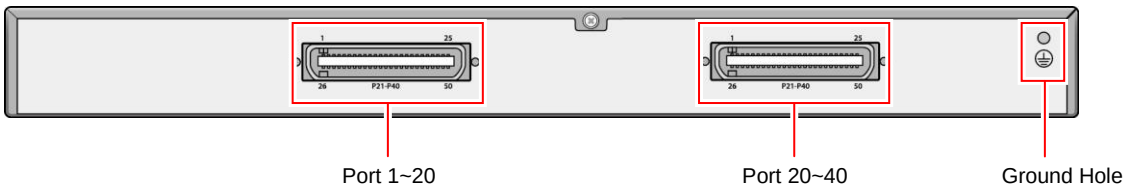
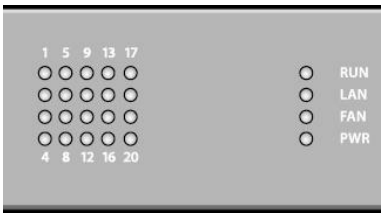


Figure 72. IPX-G520S/G540S Interface-Front/Rear

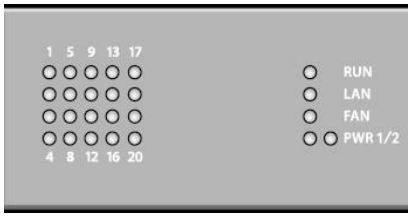
System LED

It provides System LED to display various status of the system. Refer to the following for its meaning.

- **IPX-G520S**



- **IPX-G520S (Dual Power Supply)**



- **IPX-G540S**

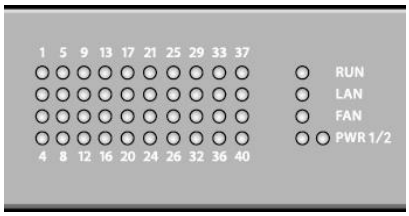


Figure 73. IPX-G520S/G540S LED

Table 7. IPX-G520S/G540S LED Description

LED		State	Description
Port		Off	Ready to Service
		On	Operating
		Blink	Ringing state
System	RUN1/2	On	Preparing to boot
		Blink quickly	Booting
		Blink slowly	Normal operation
	GbE1/2	Off	LAN disconnected
		On	LAN connected
		Blink	TX/RX Data
	FAN	Off	FAN out of order
		On	Normal operation
	PWR1/2	Off	Power is off or out of order
		On	Power is on

Console Port (RS232C)

IPX-G520S/G540S provides console port to debug the system or to set basic configuration by cli. The baud rate is 38,400 bps.

LAN Port

To connect the network, it provides 100 BASE-T UTP port.

Reset Switch

It supports to reset the system by reset switch.

FXS Port

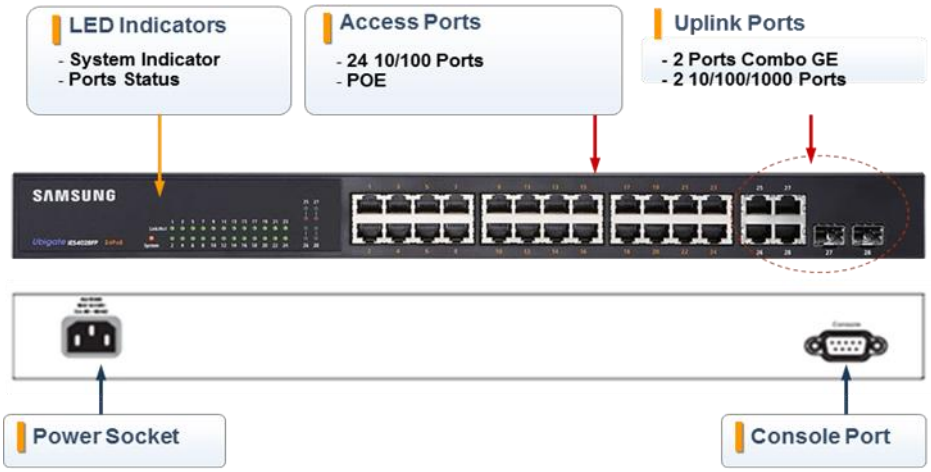
IPX-G520S provides 20 FXS ports and IPX-G540S provides 40 FXS ports.

2.5 Ubigate iES Switches

Samsung Ubigate iES Series Ethernet Switches provide reliability while reducing TCO and complexity in IP Telephony environment.
An integrated part of this switch family is the IP Clustering function. The Feature allows customers to group together up to 36 switches into a single IP Address. And also provide advanced QoS and Security features that can reduce the latency and threat through the network.

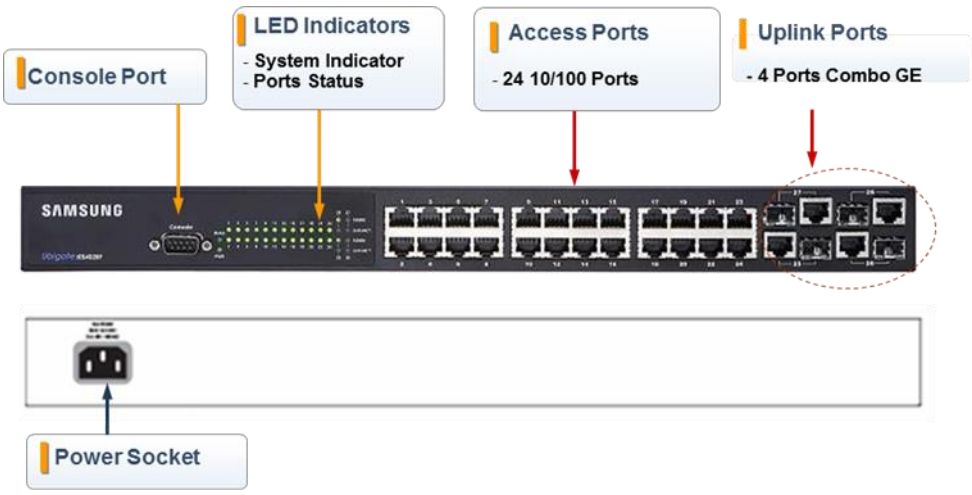
iES4028FP

iES4028FP supports 24 ports 10/100 Managed Layer 2 Switch with PoE and 4 GE ports.



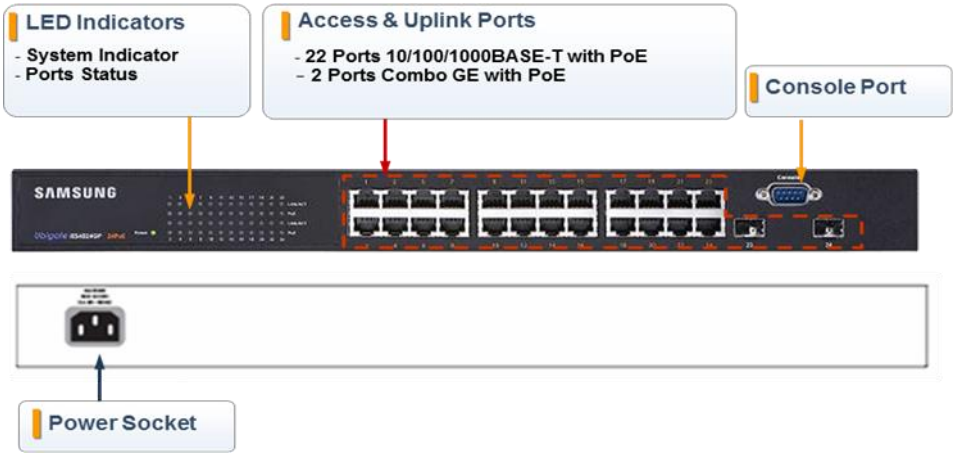
iES4028F

iES4028F supports 24 ports 10/100 Managed Layer 2 Switch and 4 GE ports.



iES4024GP

iES4024GP supports 22 ports 10/100/1000 Managed Layer 2 Switch with POE and 2 Combo ports.



2.5.1 Hardware Specification

	iES4028FP	iES4028F	iES4024GP
Ports	24 10/100 BASE-T ports with POE 2 10/100/1000 BASE-T ports 2 Gigabit combo ports 1 Console port	24 10/100 BASE-T ports 4 Gigabit combo ports 1 Console port	22 10/100/1000 BASE-T ports with POE 2 Gigabit combo ports with POE 1 Console port
SWITCHING FABRIC	12.8 Gbps 9.5 Mpps	12.8 Gbps 9.5 Mpps	48 Gbps 35.7 Mpps
SWITCHING DB	8 K MAC address entries	8 K MAC address entries	8 K MAC address entries
WEIGHT	4.13 kg 9.11 lbs	3 kg 6.7 lbs	4.33 kg 9.53 lbs
DIMENSIONS	4.3 × 44 × 33 cm 1.7 × 17.3 × 12.992 inch	4.3 × 44 × 17.2 cm 1.7 × 17.3 × 6.7 inch	4.3 × 44 × 32 cm 1.7 × 17.3 × 12.6inch
AC INPUT	100 to 240 V, 50-60 Hz	100 to 240 V, 50-60 Hz	100 to 240 V, 50-60 Hz
POWER CONSUMPTION	225 W (System 45 W, POE 180 W)	30W	225 W (System 45 W, POE 180 W)
POWER-OVER-ETHERNET	Maximum output power per port: 15.4 W Maximum output power per port: 7.5 W simultaneously	NA	Maximum output power per port: 15.4 W Maximum output power per port: 7.5 W simultaneously
NETWORK	10/100 BASE-T ports		N/A

	iES4028FP	iES4028F	iES4024GP
INTERFACE	10/100/1000 BASE-T ports SFP Transceiver slots supporting SX, LX and ZX SFP Multimode fiber cable; 62.5/125 or 50/125 microns Single mode fiber cable: 9/125 micron		

2.5.2 Software Specification

Features	Description
Flow Control	IEEE 802.3x for full duplex mode Back pressure flow control half duplex mode
Spanning tree	IEEE 802.1D STP IEEE 802.1w RSTP IEEE 802.1s MSTP Spanning Tree Fast Forwarding Auto Edge Loop Protection
VLAN	802.1Q Tag-based VLAN 802.1Q Port-based VLAN 802.1v Protocol-based VLAN 256 VLANs entries out of 4K VLAN IDs GVRP Voice VLAN
IGMP snooping	V1, v2, v3 Querier Immediate Leave Filtering and throttling
Link Aggregation	IEEE 802.3ad with LACP 8 aggregation groups up to 8 ports
MVR	Yes
Jumbo Frame	10K in gigabit ports
QnQ	Yes
Quality of Service	Priority queue Scheduling: Strict priority, WRR 4 queues per port DiffServ COS IEEE 802.1p, DSCP based COS Rate limiting (Per Port based) Ingress, Egress
Security	Storm Control Broadcast storm Multicast storm DLF (Destination Lookup Failure)

Features	Description
	MAC Address filtering Username/Password authentication Access control list (L2/L3/L4) AAA RADIUS, TACACS+ MAC based Authentication
Security	HTTPS/SSL SSHv1/v2 802.1x Port-Based Supplicant Support VLAN Assignment Guest VLAN Co-works with Radius, TACACS+ server Management Interface Access filtering SNMP, WEB, Telnet DHCP Snooping IP Source Guard
Management	Management method Web-based Telnet (4 sessions) Software download TFTP, Xmodem Dual Firmware Images Configuration file download TFTP SNMP v1/v2c/v3 RMON (group 1, 2, 3, 9) BOOTP Client DHCP Client Relay (Option82) Port mirroring (one-to-many) Event/Error Log Local, Syslog, SMTP Remote Ping SNTP NTP IEEE 802.1ab (LLDP) UPnP Banner Web authentication IP Clustering (36 members)

2.6 SIP Phones (SMT-I series)

Session Initiation Protocol (SIP) phones use IP addresses to send/receive voice and data. They use existing data network lines, so do not need normal phone lines, and can be connected to devices such as a switching hub.
The SMT-i5200 and SMT-i3100 series SIP phones are as follows:

2.6.1 SMT-i5343

- 4.3” Color TFT LCD
- HD Voice
- Smart Phone Interworking
- WiFi Backhaul
- NFC
- Bluetooth Headset
- USB Camera
- 5 soft keys
- 10 line keys
- 14 programmable key
- Navigation keys for easy use of Keyset functions
- Keyset Status Indicator
- Volume Up/Down keys for digital control of speaker, handset and ringer volume

2.6.2 SMT-i5243

- 4.3” Color LCD
- USB Camera
- 5 soft keys
- 5 line keys
- 14 programmable key
- Navigation keys for easy use of Keyset functions
- Keyset Status Indicator
- Volume Up/Down keys for digital control of speaker, handset and ringer volume

2.6.3 SMT-i6021

- 3.2” Mono LCD
- 4 soft keys
- 24 programmable keys
- Navigation keys for easy use of Keyset functions
- Keyset Status Indicator
- Volume Up/Down keys for digital control of speaker, handset and ringer volume
- HD Voice
- Smart Phone Interworking
- WiFi Backhaul
- Bluetooth Headset
- USB

2.6.4 SMT-i6020

- 3.2” Mono LCD
- 4 soft keys
- 24 programmable keys
- Navigation keys for easy use of Keyset functions
- Keyset Status Indicator
- Volume Up/Down keys for digital control of speaker, handset and ringer volume
- HD Voice
- Smart Phone Interworking
- USB

2.6.5 SMT-i6011

- 3.2” Mono LCD
- 4 soft keys
- 12 programmable keys
- Navigation keys for easy use of Keyset functions
- Keyset Status Indicator
- Volume Up/Down keys for digital control of speaker, handset and ringer volume
- Smart Phone Interworking
- WiFi Backhaul
- Bluetooth Headset
- USB

2.6.6 SMT-i6010

- 3.2” Mono LCD
- 4 soft keys
- 12 programmable keys
- Navigation keys for easy use of Keyset functions
- Keyset Status Indicator
- Volume Up/Down keys for digital control of speaker, handset and ringer volume
- Smart Phone Interworking
- USB

2.6.7 SMT-i5230

- 3.2” Mono LCD
- 4 soft keys
- 5 Desi-less LCD programmable keys
- Navigation keys for easy use of Keyset functions
- Keyset Status Indicator
- Volume Up/Down keys for digital control of speaker, handset and ringer volume

2.6.8 SMT-i5220

- 3.2” Mono LCD
- 4 soft keys
- 24 programmable keys
- Navigation keys for easy use of Keyset functions
- Keyset Status Indicator
- Volume Up/Down keys for digital control of speaker, handset and ringer volume

2.6.9 SMT-i5210/5210S

- 3.2” Mono LCD
- 3 soft keys
- 14 programmable keys
- Navigation keys for easy use of Keyset functions
- Keyset Status Indicator
- Volume Up/Down keys for digital control of speaker, handset and ringer volume

2.6.10 SMT-i3105

- 2.8” Mono LCD
- 3 soft keys
- 5 programmable keys (with LED)
- Keyset Status Indicator
- Volume Up/Down keys for digital control of speaker, handset and ringer volume

2.6.11 SMT-i3100

- 2.8” Mono LCD
- 3 soft keys
- 5 programmable keys (Without LED)
- Volume Up/Down keys for digital control of speaker, handset and ringer volume

2.6.12 SMT-i2205

- 3.2” Mono LCD
- 4 soft keys
- 10 programmable keys (Without LED)
- Volume Up/Down keys for digital control of speaker, handset and ringer volume

2.6.13 SMT-i5264

64 Buttons Add On Module

CHAPTER 3. Call Manager Features

This chapter describes the call processing services provided by SCM and how to configure them.
SCM provides the following three types of call processing services.

- System services: Determine the overall operation of the system. You can configure a system service for the entire system or a user group.
- User services: Configured for each user.
- Special services: System services and user services that require special descriptions.

3.1 System Features

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

3.1.1 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.

3.1.2 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 SCM 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 by Call Counts

CAC by call counts restricts calls when the maximum usage ratio set for the maximum number of calls supported by SCM is exceeded.
If the maximum usage ratio for CAC by call counts is 100, the maximum number of calls supported by SCM is allowed. The default maximum usage ratio for CAC by call counts is 100.

CAC by Call Per Second

CAC by call per second restricts calls when requested call is over CPS value supported by SCM is exceeded.
For check call per second value, see the 'Performance Capacity' in section of '3.1. System Features'.

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.

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.

CAC by Trunk Call Counts

CAC by trunk call counts restricts calls when the maximum trunk call count supported by SCM is exceeded. If you don't set the maximum inbound call or maximum outbound call value, CAC for the trunk call (Inbound call or outbound call) does not work.

3.1.3 Least Cost Route (LCR)

SCM performs 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/rate based LCR, and equally distributed LCR) for each location of the calling party.

Priority Routing

The route sequence 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.

Time-based Routing

The special route sequence feature allows each service group to use its own route sequence features based on its time and rate conditions.

Load-balanced Routing

The route set 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.

3.1.4 Call Restriction

SCM 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.

Route Lock

You can specify whether to use a route for external calls coming through the endpoint connected to the route.

Call 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.

3.1.5 Number Translation

SCM can translate numbers for inbound and outbound calls.
Number translation by multiple MCN allows to replace particular digits in a first position of the called number for outbound calls. Additionally, deleting digits or adding digits are possible for Inbound and outbound calling/called number.

3.1.6 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 ten call buttons, the phone can control up to ten calls simultaneously. If there is no call button on the phone, it is treated as having two call buttons.

3.1.7 Call Monitoring

SCM can provide call monitoring service, by interworking 3rd party system.
After the end of the specific subscriber call, operator can hear that call using 3rd party system.

3.1.8 Internal CLI Number

Default Internal CLI Number is extension number.
Extension number is unique number within user group, not allow duplicate.
It is used for call for calling/called number.
SCM also provide another Internal CLI Number for specific use.

3.1.9 Outbound Calling Line Identification (CLI)

The Outbound CLI service provides several ways to express outbound CLI such as phone CLI, user CLI, service group CLI, user group CLI, and virtual trunk CLI.

Outbound CLI Number

SCM provides several ways to present user CLI. Multi-Extension Phone can have Virtual Phone CLI to express same CLI for several users of the phone. Each user can have Send CLI Number for outbound calls. Additionally, the members in a group such as Service Group or User Group can use the CLI Number for the group if the users don't have Send CLI Number in User Configuration. CLI Prefix in Route Configuration provides CLI prefix for the users who aren't assigned any CLI Numbers.

Outbound CLI Name

If a SIP ISP requires some specific number for the trunk, Virtual Trunk CLI can be used.
If this option is enabled for outbound call, an original 'display name' in From header of INVITE message is replaced with the Virtual Trunk CLI for the Route.

3.1.10 Internal CLI Name

SCM provides Internal CLI Name, used for inbound trunk call.
It is used for subscriber or service group.

3.1.11 Premium CID

SCM provides supplementary information of opposite user to a FMC user. Basically photo is delivered and additional information are possible up to 6 according to a contifuration.

3.1.12 Calling Line Identification (CLI)

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.
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 *) 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.'

3.1.13 Direct Inward Dialing (DID)

The Direct Inward Dialing (DID) 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 *) to enter multiple numbers at a time. The system also performs translation of the DID number so that the 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.'

3.1.14 Directory Service

SCM provides Directory Service that user can search subscriber's name or number.
If User wants save result in phone-book, SCM provides that service.
If using name, at least 2 letter input must required.
If using number, at least 3 letter input must required.
Directory Service provides Single Phone User, Hunt Group Number, System Speed Dial Index.

3.1.15 Direct Trunk Selection

SCM supports Direct Trunk Selection to use a specific trunk, which provides services according to a status of the designated trunk. This is a service only for the FXO of a Gateway.

3.1.16 FMS (Fixed Mobile Substitution)

Fixed Mobile Substitution is a Zone service for a specific trunk. FMS users are mobile users in the specific zone. The trunk calls from or to FMS users are treated as if internal calls with a virtual number. Users can make a call with the virtual number or the mobile number to the FMS user.

3.1.17 FMS Smart Routing

For FMS service, SCM provides two kinds of Smart Routings.

Smart Routing from FMS zone to Normal Trunk

A call to FMS user is normally routed to the FMS trunk to reach FMS Zone. If a designated response is received, SCM re-routes the call to a normal trunk with the mobile phone number.

Smart Routing from normal trunk to FMS zone

SCM re-routes the normal trunk call to FMS Zone if the called number is a mobile phone number of a FMS User. This makes the call can avoid the trunk billing.

3.1.18 Emergency Group

If user dials predefined emergency group number in an emergency situation then it is dialed to access code of emergency type automatically. After connecting with the opponent, it is connected to configured managers of emergency group at the same time.

3.1.19 History Log

SCM provides history logging capability for the events like a SPAM call, call logging, paging on answer call, wakeup call, feature set, registration fail, dispatch call, notice board etc.
Some Logging Services are provided for specific user.

3.1.20 Home Worker Support

SCM 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 SCM is on the Public IP Network

If SCM is connected to the public IP network, services can be provided to home workers without 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 SCM's Media Proxy Server (MPS).

When SCM is on a Private IP Network

If SCM is connected to a private IP network on NAT, a separate SBC system is required. SCM Express performs some of the SBC features through a built-in feature called MPS. When a call is made between two home workers' phones connected to the public IP network outside SCM'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 SCM's NAT and a home worker's phone, the data is exchanged through the MPS.

3.1.21 Hotel Service

Please refer to the 'SCM Express PMS Interoperability Guide'

3.1.22 PMS Interface

Please refer to the 'SCM Express PMS Interoperability Guide'

3.1.23 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. A logout button is included in Samsung phones' soft buttons. Pressing this logs the phone out.
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.

3.1.24 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.

3.1.25 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 to whom the call is directed is on the line or unavailable, the call is 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.

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.

3.1.26 Location-based Codec Negotiation

Location

Codec negotiation takes place between two Internet phones when a call is made between them using the SIP protocol. SCM 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.
As you set Forced Audio Codec, the codec negotiation can be forced with the codec you want within or between locations.

Default Audio Codec

When SCM receives a sent message, SCM 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.

Forced Audio Codec

When SCM receives a sent message, SCM checks whether the calling phone supports the codec which set as Forced Audio Codec. SCM send the message to a called phone with only the codec if the calling phone supports that, or rejects the call.

Default Video Codec

When SCM receives a sent message, SCM 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

SCM 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 SCM 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, SCM 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.

3.1.27 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.

Multi-Device

The multi-device service assigns one user (phone number) to multiple devices (phones). SCM 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 SCM 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 10 phone numbers can be assigned per phone.

3.1.28 Music On Hold

When a call is put on hold, SCM can connect its built-in sound source and play a tone or music for the phone or the trunk.
It is necessary to enable or disable MOH for each user group because there are limited number of channels for SCM'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.

3.1.29 Missed Call Display

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

3.1.30 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.

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.

3.1.31 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.
SCM 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. SCM supports a total of 11 ring plans, including the 10 ring plans (ring plans 1 through 10) 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 10 ring plans is applied.

Calendar Exceptions

To use ring plans, SCM refers the calendar which is made when a user group is created for maximum of 20 years.
Each day assigns to a day type from Monday to Sunday, Weekday and Weekend as a default. For a site, the special day types can be assigned to some days to provide different services according to the day type. SCM provides four day type, Holiday1, Holiday2, User1 and User2.

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.

Holiday Ring Plan Schedule

A holiday ring plan schedule is a table containing data which specifies ring plans by a designated day type for the site.

Ring Plan Override

SCM 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.

3.1.32 Group Call Forward

This service is used for forwarding all incoming calls for all the phones in a group to another number according to the ring plan. If Group Call Forward is activated by Feature Code, it overrides the forward number by ring plan. Group Call Forward can follow the ring plan only when Forced Group Call Forward by Feature Code is deactivated.

3.1.33 Service Group Local Number

Extension Number is a base number to make a call. Service Group Local Number is a kind of shortcut number of the extension number in a specific Service Group. If the extension numbers in a service group starts with a same prefix, the number except the prefix can be used for shortcut number to dial.

3.1.34 System Call Forward

This service performs call forwarding based on the system settings. In case that user sets his Call Forward All and Preset Call Forward All service is set, Call Forward All based on the settings made by User is performed.

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 busy is not set by users.
When any call is placed on called user, 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 preset call forward all setting is ignored.

Preset Call Forward Busy

This service performs preset call forward busy based on the settings made by the system administrator even if preset call forward 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 preset call forward busy, the preset call forward busy setting is ignored.

Preset Call Forward No Answer

This service performs preset call forward no answer based on the settings made by the system administrator.
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 preset call forward no answer, the preset call forward no answer setting is ignored.

Preset call forward Unreachable

This service performs preset call forward unreachable based on the settings made by the system administrator.
If the call forward Unreachable service is 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.
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 unreachable setting is ignored.

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 not set by users.
When the called user set up DND service, this feature can be used to forward the call to designated Number by system administrator.

Announcement for Call Forward

When a call set up because of Call Forward, SCM sends announcement before the real ring back. This service makes the caller knows the call is forwarded. Default announcement is Announcement ID 1203.

Group Call Forward

SCM can forward calls for multiple users to one common number by date, day of the week or time of the day according to the system administrator settings. This is call group call forwarding.
Applying the dates, days of the week, time of the day for group call forwarding is done by ring plans. For more information on ring plans, see the [Ring Plans] section.

3.1.35 VoIP Security

SCM provides various security functions such as RADIUS, LDAP protocols, etc. that are used for the Transport Layer Security (TLS) and user authentication process. Users can use security functions provided by SCM to apply various security policies.
Security Policy can be categorized 2 parts; VoIP Call and User Authentication.
SCM provides the following methods respectively.
VoIP Call is categorized in VoIP Signaling Encryption and Media (RTP) Encryption of Voice.

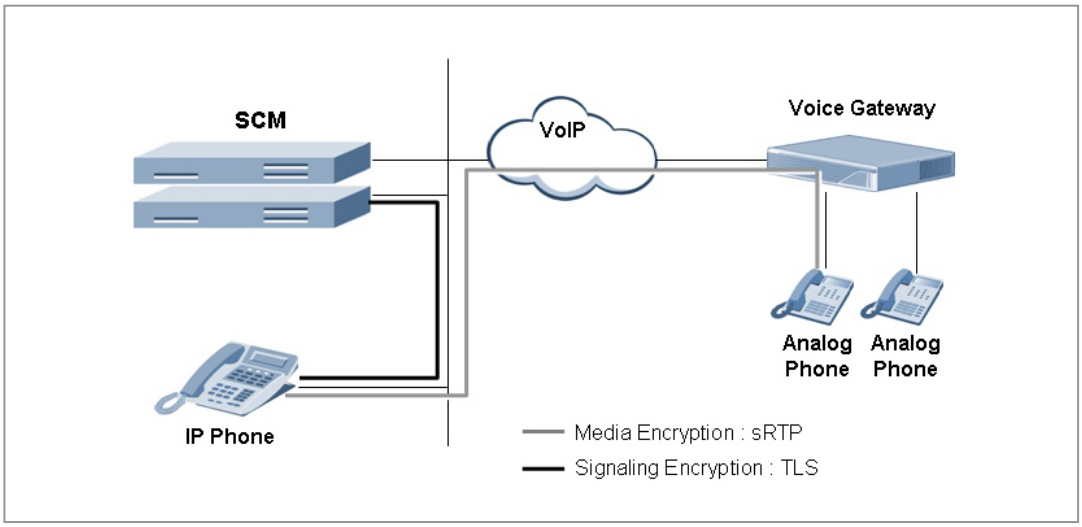


Figure 74. VoIP Call Security

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 SCM and SIP phones and between SCM and endpoints.
Specifications of TLS serviced by SCM are as follows:
Uses OpenSSL library and supports TLS v1.0.
AES and ARIA are supported as media encryption algorithms.
Key management method is RSA and key length is 1024 bits.
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 callee 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, SCM performs signaling for sRTP.
SCM supports media encryption for calls with phones, SCM's built-in conference system, SCM's built-in voice mail system, and endpoints, but not with SCM's built-in MOH system.
SCM 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.
Media encryption is not used for calls between phones or endpoints enabled with encryption and phones or endpoints not enabled with encryption.

3.1.36 Feature Services

Class Of Service (COS)

SCM allows the administrator to set privileges for each user. Since setting service privileges for all the users individually could be very complicated, Class of Service can be created with their own 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 Class of Service than the default Class of Service, you can create a new Class of Service and use it.
Each Class of Service 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.

Service List

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

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 SCM uses a standard protocol between phones and SCM, arbitrary feature buttons cannot be created. Therefore SCM uses feature codes when configuring features or when using features temporarily. If you press a feature button on a Samsung phone, SCM is also designed to process the feature code assigned to the feature button.
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.

3.1.37 User Authentication

SCM can limit its service to allow only those users who passed a series of authentication process to receive the service. For user authentication, SCM provides the HTTP Digest Authentication described in RFC3261 and RFC 2617, and Local Authentication and Remote Authentication methods are provided for the User Authentication function.

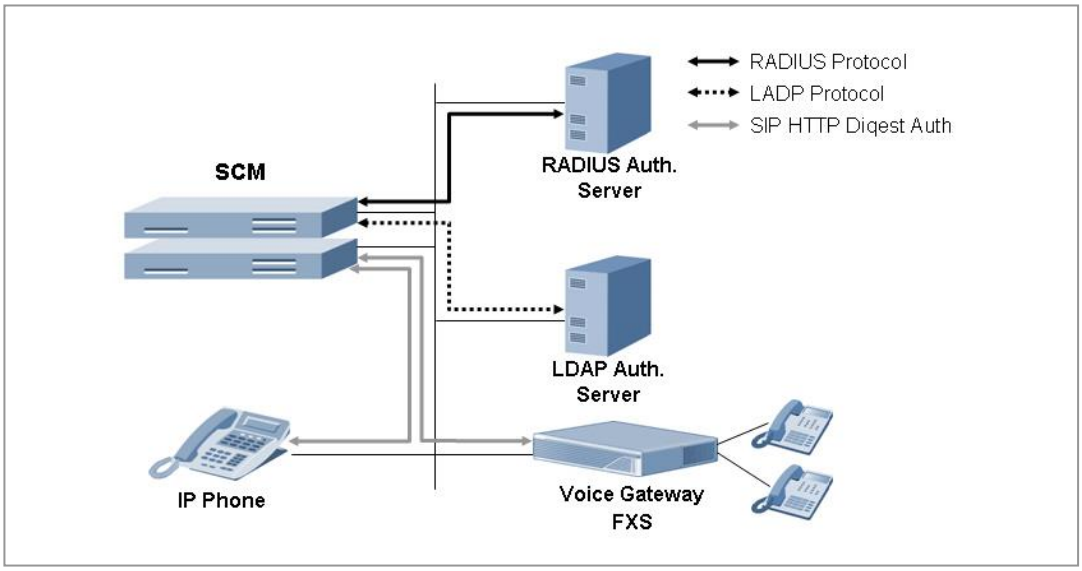


Figure 75. User Authentication

Local Authentication

SCM performs internal authentication in the following order:
The SIP phone transmits REGISTER without authentication header to SCM.
SCM transmits 401 Unauthorized with challenge information to the SIP phone.
SIP phone transmits REGISTER without authentication header to SCM.
After SCM executes Digest Authentication, it transmits 200 OK to the SIP phone.

RADIUS Authentication

SCM 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 SCM.
SCM acts as a relay between the user phone and the external RADIUS server.
Authentication is performed in the following order:
When SCM receives a REGISTER message from the user phone, it sends Access-Request to the RADIUS server.
When SCM receives Access-Accept or Access-Reject from the RADIUS server, it sends the authentication result to the user phone and finishes the authentication procedure.

LDAP Authentication

SCM acts as an LDAP client for remote LDAP authentication of users' phones. It provides LDAP and LDAPS (LDAP over SSL) for this task.
SCM 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:
When SCM receives a REGISTER message including a password from the user, it sends a Search-Request message to the LDAP server.
The user's password stored in the LDAP server is received through a Search-Result message.
The user phone's password received from the LDAP server is compared with the password received with the REGISTER message from the user phone. SCM sends the authentication result to the user phone and finishes the authentication procedure.

3.1.38 Boss/Secretary

The boss/secretary feature allows an 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 using multi-device and multi-line feature.

3.1.39 Busy Lamp Field (BLF)

SCM 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.

Line Monitoring

This can monitor other parties' status by mapping the extension of the person to monitor to the button.
When the status is Idle, the call is made to the person by pressing the button.
When the other party's call is ringing, Direct Pick can be performed by pressing the button.
When the other party is on the phone, can Barge In by pressing the button. (when having the Barge In authority)

Speed Dial

When pressing the button, a call is made to the registered dial number.

Line

Same to the right line button of a phone. When the button is pressed, the line that is pre-assigned is seized then waits for the user's dial input.

3.1.40 DTMF Detection Services

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-SCM 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.

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.

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 in advance before making the call, or you can put your current call on hold, press the account code button, enter an account code, and then reconnect the call on hold. The account code entered will be saved in the charging data record (CDR), which can be used for calculating call charges for the user.

When a trunk call is made on a phone set with forced account code input, actions are performed in the following order:

The user dials an access code and an external number.

If the user's phone is set with forced account code input, SCM connected the call to its built-in voice announcement system.

The voice announcement system plays an announcement for the user to enter a registered account code.

The user enters an account code as instructed. The voice announcement system verifies that a valid account code has been entered.

If the account code entered is valid, SCM uses the access code received in step (1) to select a route and makes a call for the external number specified.

If the account code is invalid, SCM plays an error announcement and terminates the call.

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.

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:

The user dials the number for the call authentication system.

SCM connects the call to the call authentication system.

The voice announcement system plays an announcement for the user to enter a registered call authentication code.

The user enters a call authentication code as instructed. The voice announcement system verifies that a valid call authentication code has been entered.

If the call authentication code entered is valid, SCM temporarily suspends the external call restriction set for the user.

The user can now dial an access code and an external number and the trunk call will be made.

The authentication code entered will be saved in the charging data record (CDR), which can be used for calculating call charges for the user.

DISA User Authentication

When using the Direct Inward System Access (DISA) feature, the user can call SCM 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, SCM connects the call to its built-in DISA user authentication announcement system and plays a voice announcement for the external caller.

3.1.41 System Speed Dial

This feature allows you to assign a two-digit or longer shortcut number to a phone number frequently dialed not by individual users but by all users of the system. If a lengthy phone number is mapped to a System Speed Dial number, users can access the destination just by dialing the shortcut number.

3.1.42 Local Based Number Translation

SCM provides Local Based Number Translation, between users' call belonging same location.
In same location, user can uses virtual extension number.

3.1.43 Basic Announcement

SCM provides basic announcement messages on users such as 'The dialed number you have dialed is not in service', and additional services such as notification for the user being unavailable to answer the call, Music On Hold, etc. by using the Built-In MOH function.

3.1.44 Computer Telephony Interface (CTI)

CTI provides call control and event reporting functions through SCM CSTA (Computer Supported Telephony Application) interface.

3.1.45 Digit Analysis and Numbering Plan

The digit analysis function is a process of deciding how to handle the incoming call by analyzing the digits dialed. When a call is generated, it first compares the digits dialed with DN (Directory Number) list of the user. If two numbers match, then the system recognizes the call as an internal call. And if not, the system compares again the dialed number with access code for call routing. If the two numbers match, the system considers it as a service call. And if not, the system compares again the dialed number with access code for call routing. If the dialed digits correspond to the call routing access code, then the call is handled according to the call routing procedures. If the dialed digits don't match any of mentioned numbers, then the system considers it as an incorrectly made call and rejects the call.

3.1.46 Direct Outward Dial (DOD)

DOD is the service that allows a user of a company's PBX or Packet Voice System (SCM) to directly connect to an external line, and it is provided by local phone service providers or local exchange carriers. When DOD is used, the individuals within a company can call outside directly without help from an operator or additional number input.

3.1.47 Fax over IP

SCM supports SIP signaling regarding T.38 Fax.

3.1.48 DTMF

SCM supports INFO method, NOTIFY method, and RFC2833 to either send or receive DTMF event to/from a user terminal or Voice Gateway system.

3.1.49 Privacy

Privacy operates as a defensive function against DND Override and Barge In services. When the user's Privacy level is higher than the Override level of the other users who attempt DND Override or Barge In, both DND Override and Barge In do not occur.

3.1.50 Survivable Telephony Support

SCM supports the Survivable Telephony function provided in Voice Gateway.

3.1.51 User Group

The User Group Function allows a separate grouping of the system users thus enabling various added services to be used independently among groups. Each User Group has a separate dial plan, and the User numbers allocated for User Groups can be overlapped.

Service Group

In concept, this is a Sub-Group of the User Group, and there are several Sub-Groups within one User Group to use for various added services. The functions using service group include Call Limiting between service groups, Route Occupation Limiting per service group, call statistics per service group, etc.

Inter Service Group Restriction

The operator may limit the calls between service groups by using this function. If a restriction has been set between service group A and B, a guidance message notifying that the call is restricted is played to a user of group A when he/she is calling a person in group B.

3.1.52 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.
To use this feature, bridge feature code and the extension number of bridged user should be entered in Hot Line service menu for the gateway FXS user.
If bridged user is not busy status, the gateway FXS user listens dial-tone and makes a call.

3.1.53 Wireless Enterprise Service

Wireless Enterprise Service is VoIP (Voice over IP) Service in FMC (Fixed Mobile Convergence) using WiFi of the smartphone user.

3.1.54 High Availability

Redundancy is the method used to guarantee high availability, and depending on the object of redundancy, this is divided into LAN Interface redundancy, system redundancy, and data redundancy.
For redundancy, SCM uses a Virtual IP address to allow connection from outside by using only one IP address. For the SCM that is configured in redundant structure, even when automatic switchover to standby system has occurred due to a fault occurred in Active system, terminals or voice gateways may process calls continuously without detecting the fault in SCM because the IP address from outside is the same.

Virtual IP Address

Virtual IP addresses can be categorized into two types of System Virtual IP and Component Virtual IP.

System Virtual IP Address
This is the virtual IP address used for LAN Card redundancy on one system.

Component Virtual IP Address
This is the virtual IP address to display as one system when the system has became redundant as Active-Standby.

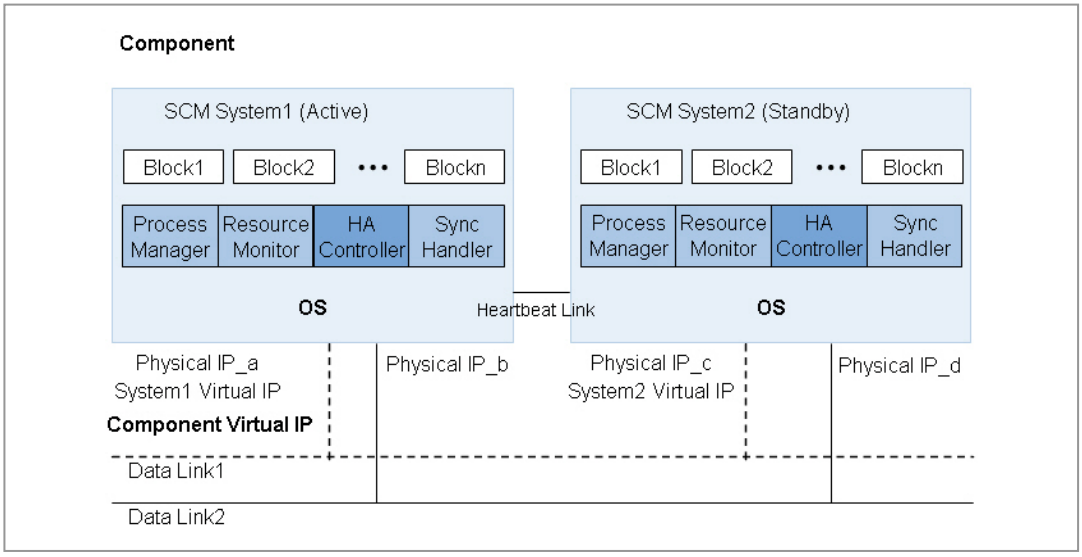


Figure 76. Virtual IP Address for High Availability

In the figure above, Physical IP_a, Physical IP_b, Physical IP_c, and Physical IP_d refer to the physical IP addresses of LAN. System1 Virtual IP, and System2 Virtual IP are the virtual IP address used for LAN Card redundancy, and Component Virtual IP is the virtual IP address used for System1 and System2 redundancy.

LAN Interface Redundancy

LAN Interface redundancy is a way to guarantee the system connection by using LA.
2 or more Physical LAN interface cards are installed to the system. And IP address (System Virtual IP address) that represents the system is mapped always to the interface card that is in normal state by constantly monitoring each interface card state. At this point, the LAN interface card should provide the method for monitoring the status. In a Linux environment, the status is checked through MII (Machine Independent Interface).

System Redundancy

System redundancy is a way to prevent the system service interruption due to a fault of a single system. The system consists of Active and Standby systems. Active system provides services by using the IP address (‘Component IP address’) that represents the service, and the Standby system monitors the status of the Active system by using Heartbeat. Since the other network components communicate with SCM by using Component IP address, the Standby system handles them as the Active while providing the service continuously when a fault has occurred in the Active system.

Data Redundancy

Data Redundancy is a way to prevent a data loss caused by a fault occurred in the system and to prevent the data integrity of having the same data even after the Switch-over has occurred by the system redundancy.

3.1.55 Limited non-encrypted calls

This function limits calls that are not encrypted. This can be set by the user group

3.1.56 Using Multi m-line in phone

This is the function that contains the 2 mline (AVP/SAVP) in the INVTIE message of sRTP Calls.

3.1.57 Nurse Call

SCM provides interoperability with Nurse Call System through GW (OS7500). SCM analyzes a nurse call to display ‘room and bed number’ as CLI.

3.1.58 Multiple SIP Account

Multiple SIP Account to one ISP is supported. Called number is used as a key to distinguish an incoming route.

3.1.59 Common Route

Common Route is a route which is shared between several user groups to connected to a ISP or a gateway.

3.1.60 Inter User Group Call Routing

Inter user group call is a call from a user in a user group to the other in another user group by dialing with trunk access code. Internally routing the call without trying through a trunk is supported.

3.1.61 Service Limitation

The service limit is the control function about the resource which the User Group's use commonly.
The resource of which the restricted service is possible is Internal UMS, Internal Conference Server, Common Type Trunk and Common Type Application Server.

3.1.62 Text To Speech

SCM can make a wave file to Support Broadcast Call Service using Text to Speech (TTS) Engine.
This feature is available only in Korea.

3.1.63 Minimum Call Limit

SCM 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 SCM, the call is denied.

3.1.64 WE VoIP Location CID

SCM provides the CLI Service for WE VoIP Registered Location.
Registered Location is Internal Wi-Fi, External Wi-Fi, mVoIP.

3.1.65 Display Mobile Phone Access Network

SCM provides the access network information display of the other mobile phone.
The access network information can be displayed Wi-Fi and LTE.

3.1.66 Tandem Representation Number

SCM provides the representation number about the tandem call to CDR data when used for incoming.
Also, it can be used for Outbound Diversion Number.

3.2 User Features

A user service is only available to the users authorized to that service. The authorization to a service is described in the format of Class of Service. The Class of Service can be assigned to a user and the user can use the services defined in it.
A user can set his or her services using SCM Personal Assistant.

3.2.1 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.

3.2.2 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. Auto Answer feature of a user can be ignored by Auto Answer Ignore option in the Route.

3.2.3 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.

3.2.4 Barge-In

This feature allows you to intrude into a user's current call for a three-way conference call. The Barge In feature is also known as call intrusion or Call Override.
The Barge In feature is provided with warning or without warning depending on whether the user is notified that the call has been intruded into. SCM services both types of the feature.
SCM uses its built-in conference system for three-way conference calls. Therefore, the basic settings for using the conference system must be configured.

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.

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.

3.2.5 Change Password

The change password feature is used when the user wishes to change his/her password for using function.

3.2.6 Call Forward

When there is an incoming call, this feature is used for forwarding the call to another number specified by the user.

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 'System Call Forward-Group Call Forward' section of '4.1. System Features.'

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 'System Call Forward-Preset call forward Busy' section of '4.1. System Features.'

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 'System Call Forward-Preset call forward No Answer' section of '4.1. System Features.'
You can use PWP 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.

Call Forward Unreachable

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.

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.

Selective Call Forward Allowance
If this feature is enabled, only the calls from specified numbers are forwarded and calls from unspecified numbers are not forwarded.

Selective Call Forward Restriction
If this feature is enabled, the calls from specified numbers are not forwarded and calls from unspecified numbers are forwarded.

No Response

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 respond to signaling.
If alternative route is enabled, all outgoing calls for external number are automatically re-routed to alternative route when direct route is not respond to signaling about outgoing call.

3.2.7 Call Hold

Call hold and Retrieve function allows a user to hold the call that is currently on line, attempt to make a new call and also reconnect to the original call. If the person on the other line is using specific services, (Ex.: call hold, conference, etc.) the call hold function cannot be used.

3.2.8 Call Park

The call park feature allows the user to park the current call so that it can be picked up 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.

Park Extension

The park extension 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.

Park Orbit

The 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. Therefore, this feature is useful for users who need to park many calls, such as operators.

Parked Call Pick-up

This feature allows a parked call to be reconnected on the phone for it had been parked or on another phone.

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.
In case of a trunk call, if the redirected call is not answered, the call is redirected to the operator. For more information, see the 'Operator Group-Operator Recall' section of '4.1. System Features.'

3.2.9 Call Pick-up

This feature allows the user to answer another user's incoming call. Call pick-up can be serviced in the following two ways.

Direct Call Pick-up

This feature allows you to pick-up another user's incoming call by specifying the user's number.
The user can press the direct call pick-up 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 Pick-up

If this feature is enabled, you can specify a call pick-up 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 pick-up an incoming call ringing for your own call pick-up group.
The user can press the group call pick-up feature code + the number of the call pick-up 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 pick-up feature code without a call pick-up group number to pick-up the current incoming call for his/her own call pick-up group.

3.2.10 Outbound Call Lock

The Outbound Call Lock feature allows a user to request for restriction of outbound trunk calls from his/her own number.

3.2.11 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.

Transfer Type

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 SCM 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 park button, call another number, and then transfer the call by pressing the transfer button 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.

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.
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 Group-Operator Recall' section of '4.1. System Features.'

3.2.12 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.

3.2.13 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.

3.2.14 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 pick-up 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.

3.2.15 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 number is redialed.

Callback Busy

The Callback Busy function is for connecting the calling and called users by generating a call to both parties when the called user for whom the SCM is scheduled becomes idle after the calling user has requested a callback to the called user whose line was busy in the first instance.

Callback No Answer

The Callback No Answer function is used for connecting the calling and the called users by generating a call to the both parties when the line status of the called user for whom the SCM is scheduled becomes busy then returns to idle after the calling user has requested a callback to the called user who did not answer in the first instance.

3.2.16 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.
When making a call, the user can dial the temporary CID restriction feature code + called party 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.

3.2.17 Do Not Disturb (DND)

When Do Not Disturb (DND) feature is enabled for a user, SCM 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.

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.

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.

3.2.18 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.

3.2.19 Individual Speed Dial

A user can register one-digit Individual Speed Dial IDs as shortcut numbers to frequently dialing numbers. By dialing these IDs, a user can dial to the destinations.

3.2.20 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.

3.2.21 Language Selection

This service allows the user to change the language displayed on their phone.
The user can use the [My Info] menu on SCM Personal Assistant to change his/her language.

3.2.22 Last Call Redial

The last call redial feature allows the user to redial the caller or the called party number of the most recent call.
Last Call Redial is independently operated by each node.

Last Outgoing Redial

The last outgoing redial service allows redialing the called number of the last outgoing call.

Last Incoming Redial

The last incoming redial service allows redialing the called number of the last incoming call.

3.2.23 No Ring

The no ring feature prevents the phone from ringing when there is an incoming call for the user. This service use 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.

3.2.24 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.
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.

- DISABLE: 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.
- ENABLE: 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.

3.2.25 Mobile Extension (MOBEX)

The mobile extension (MOBEX) feature allows incoming calls to be directed not only to the landlines and mobiles phones registered with SCM 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.

Call Pick-up on Desk Phone

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.
The user can dial the call pick-up 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 ‘MOBEX on Desk Pick up’ 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.

3.2.26 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.

3.2.27 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.

3.2.28 Voice Mail Integration

SCM'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.
Answering machine emulation and call recording services are provided as three-way conference calls. Since SCM utilizes its built-in conference system for establishing three-way conference calls, the basic settings for using the conference system must be configured.

Message Waiting Indication (MWI)

This is the function used to notify that there is an unread message by lighting the MWI lamp of the user's extension phone when the message is saved in the user's VM mail box.

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.

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. The AME Enable button is used to turn this feature On/Off.

AME Auto Start
This method allows the incoming calls to be connected to AME by configuring the AME feature in advance.
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.
When the user's phone rings, the user can press the 'AME-Manual 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-Manual Stop' button on the phone while AME is in action, the caller will be connected to the user and AME will stop.

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.
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.

Manual Call Record

This feature allows the call conversation to be recorded during a call by pressing the call record button.
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.

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.
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.

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.
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.

3.2.29 Personal SPAM Number

SCM denies incoming calls which calling number is registered in SPAM list.

3.2.30 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.
SCM 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

3.2.31 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.

3.2.32 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 Moved to the other multi-device member, if a user selects the menu.

3.2.33 NFC Service

SCM supports NFC services through the NFC phone (such as SMT-i5343) and the mobile installed SDM (A Samsung Mobile App). The call which is ringing or is on conversation can be moved by touching them.

3.2.34 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.

3.2.35 Mobile Remote Dial

A WE VoIP client can make a call through SCM, 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.

3.2.36 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 SCM let the DISA call be disconnected, SCM makes the 3G call for the WE VoIP. If the WE VoIP answers the call, SCM requests the client enter the destination number after listening announcement. And SCM makes a call to the destination which was entered. After the destination answer the call, SCM lets them be connected each other.

3.2.37 Manual Handover Service

If WE VoIP subscribers move out of WIFI network during a call, you can switch WIFI call to 3G call by manually before it goes out from the WIFI network.

3.2.38 Smart Handover Service

If WE VoIP subscribers move out of WIFI network during a call, WIFI call is switched to 3G or LTE call by automatically.
following is Smart handover Mode.

- Handover-Out by APC: WIFI call is switched to 3G call automatically.
- Handover-In by APC: 3G call is switched to WIFI call automatically. However, it is served only if call is on Smart Handover-Out.
- Handover by Mobile(Wi-Fi to 3G): WIFI call is switched to 3G call automatically by Mobile Phone.
- Handover by Mobile(Wi-Fi to LTE): WIFI call is switched to LTE call automatically by Mobile Phone.

3.2.39 CSTA Line Seize

Line seize status is considered to the CSTA event. Service Initiated Event is delivered when the receiver is hooked off and Connection Cleared Event is delivered when it hooked on.

3.3 Installation Features

This section describes the procedures for using the essential management tools (including the tools for installing phones, installing gateways, and batch importing or exporting large-volume information) for operating the SCM system.

3.3.1 Data File Export/Import

SCM Administrator's data file export/import feature allows exporting information from some of the 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 or delete lists. SCM can be updated with any changes made in the Excel spreadsheet.

3.3.2 Phone Settings

PNP Installation

This feature allows a phone to be registered with SCM when powered on so that it becomes available for service. If a MAC address is registered when creating the device, sec_{mac address}.xml is created in /tftpboot/sec_mac/. If a phone uses PNP, the phone reads this file to acquire the required information and automatically requests SCM for registration.
If PNP is not used, you can register the phone by manually entering the profile ID, password and the IP address of the SCM on the phone.
If PNP is used, it means that an IP address is automatically allocated to the phone by the DHCP server and the phone is automatically informed of the IP address of the profile server. Therefore, the IP address of the profile server must be included in Option 43 of the DHCP server. You may configure SCM to as a profile server without installing a separate profile server, but this is not recommended for SCM Enterprise.

Phone Profile Information
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.

Phone Update
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.

Manual Phone Update

Phones can be updated manually in the following ways:

Profile Update

This method sends an SIP Notify message which notifies profile changing to a phone. Although this message is sent automatically when changing profile, you can have force it to be sent when necessary.

The following profile types are possible.

- Service Profile
- Line Profile
- User Profile
- Phone Profile

Model Update

This method is used for upgrading the phone software.

'Model Update User Group' 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' 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

User Profile

You can configure the phone version, 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_boot.xml and sec_user_XXX.xml are changed, and a Notify message is sent to all phones.

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 SCM without waiting for the next number or timeout. SCM can set dial plans to be used by phones and send them to all phones.

Automatic Phone Update

SCM can automatically upgrade the phone software.
Load the package on a selected profile server and reboot the phone to have the phone upgraded automatically.
During automatic upgrade (if the version information in sec_boot.xml is different from the version on the phone), the phone software is automatically updated according to the <upgrade_server> IP address and the protocol (tftp or http) defined in sec_boot.xml.
When the update is complete, sec_boot.xml is loaded again for the rest of the procedure.
In a large-scale SCM system, a separate profile server should be installed to reduce load and upgrade time. (including sec_boot.xml)
The TFTP server and the upgrade server are included in SCM by default. If necessary, you can have separate servers for these functions

3.3.3 Gateway Settings

SCM provides the survivable telephony support service, whereby, when an IP phone is disconnected from SCM, the phone is connected to the gateway for minimum PBX features. The survival feature really works by the gateway and the IP phone and not by SCM. Generally, it is usually used not when there is a problem with SCM but when the IP network between the IP phone and SCM experiences a trouble, especially when the phone and SCM are in different locations.

3.4 Management Features

This section describes the features required for system management.

3.4.1 Access Control Management

Administrator can use SCM Administrator to control access for operators (Engineering, Technician and Customer-3 Levels supported).
Administrator can manage operators' IDs, passwords, levels, classes, login timeout password duration, forced password change, etc.
Administrator can use control access by entering user terminal information.
Administrator can use view status of current users. To log out a user, select the user and click the Logout button.

3.4.2 Process Management

SCM Express is a complex system of many processes (programs). Therefore, such processes are managed by Process Manager, which performs the following functions:

Process Management

Process Monitoring

SCM'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 SCM 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.
SYNC	When SCM is configured for redundancy, data has been synchronized between standby SCM and active SCM and the standby process has stopped.

Process Level

SCM Express classifies processes into Critical, Major, or Normal levels depending on their effect on the system. The table below shows management policy for each level.

Individual Process Start/Stop

The SCM administrator can stop running processes or start stopped processes.

Viewing Process Version Information

For every process in SCM there is information on the version, date created and time created.

3.4.3 Redundancy Management

SCM supports LAN port redundancy within one system. That is, if one of the two LAN ports becomes unavailable, the other one takes its place immediately.
Also, SCM itself can be configured redundant as active-standby. The active system is the one running currently. The standby system is one that can run in place of the active system when a critical fault occurs with it, such as LAN card failure, system halt, critical process termination, etc. Therefore, even if a fault occurs with the active system, services can be provided without interruption as the standby system takes over the place. You can also force the active and standby systems to be switched over.
In general, the SCM administrator can control the SCM features by accessing the active system. If necessary, the administrator can also access the standby system and can perform limited functions such as viewing information.
The following diagram shows the redundancy structure of SCM.

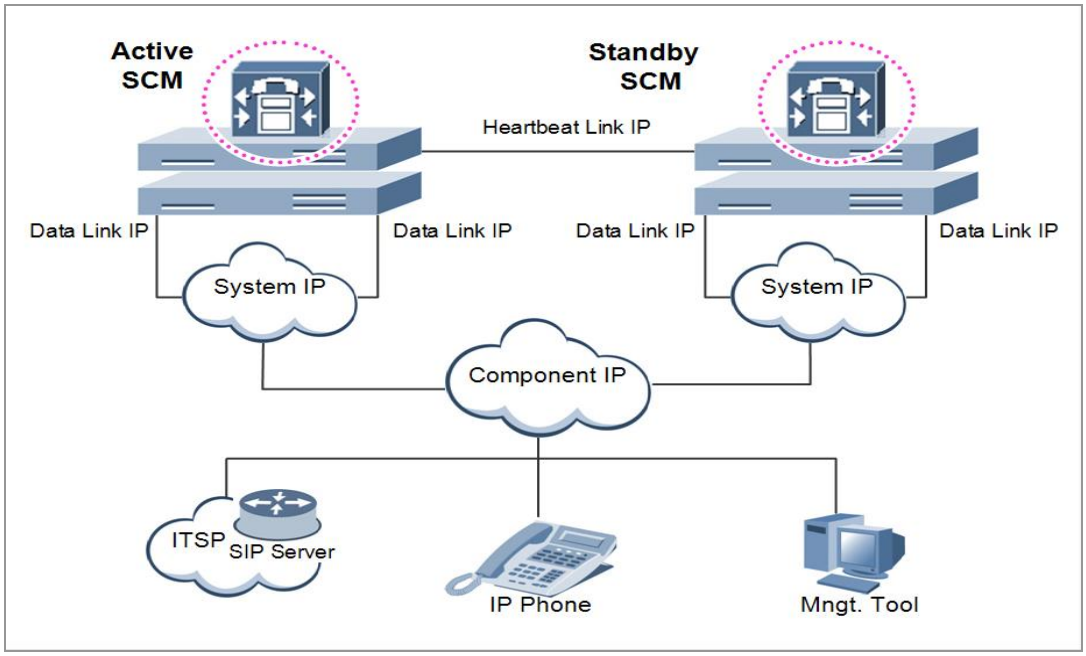


Figure 77. SCM IP Address for Redundancy

IP Address Configuration

The redundancy system uses a virtual IP address to allow external systems to connect to the SCM system through a single IP address. In the SCM system with redundant configuration, if a fault occurs in the active system and the standby system takes over, the system will continue to run with the same virtual IP address as seen from the outside. This allows external systems to continue interoperating with SCM without becoming aware of any fault.

IP Address Types

There are two types of virtual IP addresses: system virtual IP addresses (hereinafter referred to as system IP addresses) and component virtual IP addresses (hereinafter referred to as component IP addresses). Described below are the types of IP addresses (including the virtual IP addresses) which are managed by redundant SCM systems.

Redundancy Status

A redundant system can be in one of these four modes: active, standby, active-alone, and standby-alone.

Active Mode

The system is running and providing services.
In this mode, SCM sends and receives redundancy status information to and from the standby SCM system via the heartbeat link.

Standby Mode

The system is not providing services but standing by.
In this mode, SCM sends and receives redundancy status information to and from the active SCM system via the heartbeat link.

Active-Alone Mode

The system is running and providing services only as an active system.
In this mode, SCM does not send or receive redundancy status information to and from the standby system SCM via the heartbeat link.

Standby-Alone Mode

The system is not providing services but standing by.
In this mode, SCM does not send or receive redundancy status information to and from the active system SCM via the heartbeat link.

Unknown Mode

Redundancy status is unknown.
The system is currently loading and redundancy status has not yet been determined.

Redundancy Features

SCM redundancy supports LAN interface redundancy, system redundancy, and data synchronization.

LAN Interface Redundancy

LAN interface redundancy is a way to guarantee the system connection via LAN.
Two or more physical LAN interface cards are installed on the system. The system constantly monitors the status of the interface cards and ensures that the system's IP address (system virtual IP address) always stays mapped to a working interface card.
The LAN interface cards must provide ways to monitor their status. On Linux, their status can be checked by Machine Independent Interface (MII).
Therefore, even when the data link is fails due to LAN interface card failure or poor LAN cable connection, a standby data link resumes operation to provide uninterrupted services.

System Redundancy

System redundancy is a way to prevent system service interruptions by system faults.
The system consists of an active system and a standby system. The active system provides services using the system's representative IP address (component IP address).
The standby system uses the heartbeat to monitor the active system status. Since other network components communicate with SCM by using the component IP address, if there is a fault with the active system, the standby system switches over as the active system and continues to provide services using the component IP address.

Data Synchronization

If SCM is configured for redundancy, the standby system must keep the same data as the active system as to make provision for faults with the active system. Therefore, all active processes running in SCM communicate with the standby processes and perform data synchronization.

3.4.4 Operation Management

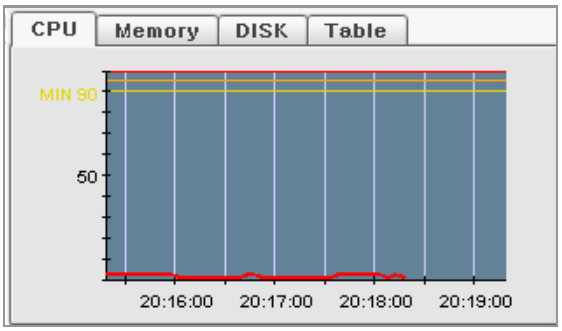
System Configuration

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, Time Schedule, Service, Application, Phone Setting, Announcement, Miscellaneous and gateway.

Figure 78. SCM Administrator

System Performance Management

To check the current SCM system performance status including CPU, memory, and disk utilization, watches the right side of the main monitor. Also, can check the system and process resource usage activity.
SCM Performance Monitor



System Resource Monitoring

SCM 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 SCM include CPU, memory and hard disk drives.

Viewing System Resource Information

The Chart of Right Side display CPU, memory, and hard disk. Also, CPU, memory and hard disk usage in the System Viewer screen in the bottom left corner.

If a network card stops working, it generates an alarm, which is cleared when the problem is rectified.

Viewing Resources by Processes

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.

Announcement Management

SCM's built-in sound source system can play voice announcements and system tones when necessary.

Release Announcements

This service plays voice announcements when calls are not processed normally due to errors, etc.

Select an announcement and click the Play button to have the selected announcement play through the PC's sound device.

Service Announcements

Voice announcements can be played when using call processing services.

Select an announcement and click the Play button to have the selected announcement play through the PC's sound device.

Music On Hold (MOH)

This Service manages 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.

Select a MOH and click the Play button to have the selected MOH play through the PC's sound device.

Language Settings

SCM 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.

Call Management

SCM 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. SCM also provides the trace feature which allows the administrator to trace calls or protocol messages.

Call Management

You can view the currently processed calls. 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.

Signaling Trace

SCM 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 protocol messages from the call initialization stage to the call termination stage are traced for a call for a user's extension number.
- Route Trace: All protocol messages from the call initialization stage to the call termination stage are traced for a call for a route.
- Protocol Trace: All protocol messages sent to and received from a specific IP address are traced.

Database Management

SCM provides a feature for backing up the database during operation.
When upgrading SCM to a newer version, you can back up the database, upgrade the version, and then restore the database for use.

Database Space

SCM provides a feature for displaying the current size of the database. The maximum allowed database size is shown in KB

Database Backup

You can configure periodical database backup to backup the database periodically or perform an immediate database backup.

User Management

SCM provides a feature for managing SCM users including phones, routes, gateways, and applications.

Extension Management

SCM manages extension users by adding, deleting or changing extension users in the database. Extension user information can be either phone information with physical properties or user information with logical properties.

Trunk Management

SCM 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.

Registration Management

SCM manages the registration status of the users-including phones, routes, gateways, and applications-which provide services by performing SIP registration with SCM and displays their current status.

Authentication Management

SCM can authenticate registration of the users-including phones, routes, gateways, and applications-which provide services by performing SIP registration with SCM. Also, when an extension user makes a call, SCM provides a service for allowing the call to be made after obtaining an external server's authentication.

Service Allowance Management

SCM can allow each individual extension user to use different sets of features by assigning them to service classes.

Maximum Calls Management

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.
SCM provides the Call Admission Control (CAC) feature which limits the maximum number of calls allowed. The CAC service provided by SCM includes CAC by call counts, CAC by location based bandwidth, and CAC by system resources.

3.4.5 Call Detail Records (CDR) Management

Account information includes Call Detail Records (CDR) and Station Message Detail Records (SMDR). Whenever a call starts or ends, SCM 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.

Local Store (Saving Account Information in SCM)

The CDR files generated are backed up and saved in the SCM hard disk without interoperating with any external account systems.

FTP Send (FTP Interoperation for Accounting System)

The CDR files generated are transferred to the external accounting system interoperating by FTP protocol.

RADIUS Send (RADIUS Interoperation for Accounting System)

The accounting information is send to the RADIUS server by interoperating with the external accounting system over RADIUS protocol.
SCM compiles RADIUS Accounting-Request messages in the following format and sends them to the RADIUS server. When a call starts, the RADIUS start record is sent to the RADIUS server. When a call ends, the RADIUS stop record is sent. The CDR data can be sent at the same time.

TCP Send (TCP Interoperation for Accounting System)

SCM 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 SCM.

TCP_SMDR Send (TCP Interoperation for Accounting System)

SCM interoperates with the external accounting system over a native TCP method. Whenever SMDR data is generated, the SMDR data is transferred to the TCP server. SMDR files are also backed up in SCM.

Billing Output by Call Types

This is a function that the CDR data is created by call types

Billing Delete Length

This is a function that deletes access code of trunk in the ‘connect number’ of CDR

TCP ACK Send (TCP ACK Interoperation for Accounting System)

SCM interoperates with the external accounting system over a native TCP method. Whenever CDR data is generated, the CDR data is transferred to the TCP ACK server. CDR files are also backed up in SCM. While it is disconnected from the TCP_ACK server, it will store the data inside the SCM. When it is connected with the TCP_ACK server, It transmits the CDR data stored in order of occurrence and then transfers real time CDR data. Further, after receiving the Acknowledge message from TCP_ACK server every time data is transmitted, it sends the next message.

3.4.6 Statistics Management

SCM provides statistical information for calls, resources and alarms generated in the system by hours, dates and months.
The statistical information is kept in the database for the duration specified by [Statistic DB Keep Up Lifetime].
The duration for which to query the statistical information must be entered with following conditions. 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.
SCM Express provides statistics on incoming calls and outgoing calls for individual users. This feature is not supported by SCM Enterprise.
You can view statistics on incoming calls and outgoing calls for individual users.
Only hourly statistics is available for incoming calls and outgoing calls for individual users.

3.4.7 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 SCM are categorized in the following way:

- 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.

In the bottom left corner of the SCM Administrator screen is the System Viewer window which displays the number of alarms generated in the SCM system. When an alarm is generated in SCM, the System Viewer window displays the number of alarms generated for each level.

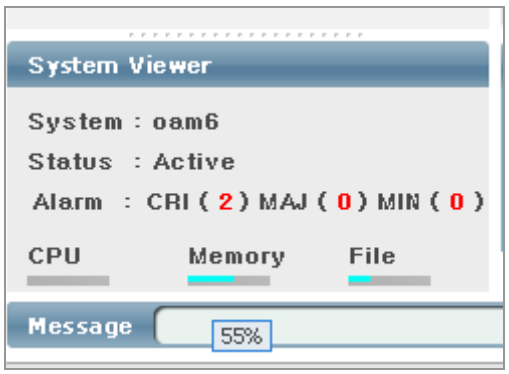


Figure 79. System Viewer

Setting

A setting contains detailed information for alarms, faults and status generated in SCM. The administrator can use this information to configure how SCM Administrator displays the alarm, fault and status information.

Viewers

Event Viewer provides real-time monitoring of alarms, faults, status, and alarm clear information in the SCM system.
Viewer-Event displays alarm, fault, status, and alarm clear information simultaneously in the order of alarms > faults > status > alarm clear from the top.
Viewer-Alarm provides real-time monitoring of alarms generated and then cleared in the SCM system.
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.

History

This feature allows you to manage the history of alarm, faults and status information generated in SCM.

Email Notification

The alarm, fault, and status information generated in SCM 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.

SMS Notification

The alarm, fault, and status information generated in SCM can be notified to the administrator by SMS. To allow this, the SMS field in the setting-alarm, setting fault or setting status must be set to Enable and the required settings for the SMS server must be configured in email settings.

3.4.8 License Management

SCM supports management of user license for call handling, license for UMS/MCS users, license of the ACD to be installed in other server, and license of IM/PS users.

User License

The user license management screen is provided where the manager can input the license key that is based on the server equipment information where SCM is installed and also check the result.

Embedded Application License

The embedded application License management screen is provided where the manager can input the license key required for starting Voice Mail of Meet-me Conference that is based on the equipment information of the server where SCM is installed and also check the result.

External Application License

The external application license management screen is provided for external servers where the manager can input the license key for starting external application that is based on the external server's equipment information and also check the result.

3.4.8.1 Activation Key

SCM supports management of activation key for call handling, UMS/MCS users and the ACD to be installed in other server.

SCM Package

The activation key management screen is provided where the manager can input the activation key that is based on the server equipment information where SCM is installed and also check the result.

3.4.9 Registration Management

The purpose of the registration management is to check the aliveness of the SIP phones, SIP gateways, and SIP endpoints. The SCM checks the aliveness by periodically checking the timestamp of heartbeat messages. The SCM uses REGISTER and OPTIONS methods as heartbeat messages. If the heartbeat messages are not exchanged within the expiration time, the SCM regards that the peer is not on the line and changes the status to Unregistered.

SIP Phone Registration by SIP REGISTER

SIP Phone Registration

An SIP phone periodically sends REGISTER messages to SCM, and the messages are authenticated by SCM.

Registration Clear upon Request by SIP Phone

An SIP phone sends a REGISTER message to SCM with the Expires value set to 0. When the message is received, SCM clears registration for the SIP phone.

Registration Clear upon Expire

SCM periodically checks the registration expiration time of SIP phones.

Gateway FXS Registration by SIP REGISTER Gateway FXS Registration

A gateway FXS periodically sends REGISTER messages to SCM, and the messages are authenticated by SCM.

Registration Clear upon Request by Gateway FXS

A gateway FXS sends a REGISTER message to SCM with the Expires value set to 0. When the message is received, SCM clears registration for the gateway FXS.

Registration Clear upon Expire

SCM periodically checks the registration update status of gateway FXS.

SIP Gateway Registration by SIP REGISTER

SIP Gateway Registration

An SIP gateway periodically sends REGISTER messages to SCM, and the messages are authenticated by SCM.

Registration Clear upon Request by SIP Gateway

An SIP gateway sends a REGISTER message to SCM with the Expires value set to 0. When the message is received, SCM clears registration for the SIP gateway.

Registration Clear upon Expire

SCM periodically checks the registration expiration time of SIP gateways.

Endpoint Registration by SIP REGISTER Endpoint Registration

SCM sends a REGISTER message to an endpoint. Once the message is successfully authenticated, the endpoint is registered.

Endpoint Registration Clear

SCM periodically sends REGISTER messages. If there is no response for a message, SCM attempts to resend the message for the maximum number of times specified.

Endpoint Registration by SIP OPTIONS

Endpoint Registration

If [Keep Alive] is set to Enable, SCM periodically sends OPTIONS messages to external connection endpoints. If a 200 OK message is received as a response to an OPTIONS message, the endpoint is registered.

Endpoint Registration Clear

If there is no response for an OPTIONS message or if response fails, SCM attempts to resend the message for the number of times. If all resending attempts fail, registration is cleared.
After clearing registration, SCM waits for the time before it resends the OPTIONS message.

Application Registration by SIP OPTIONS

Application Server Registration

If [Keep Alive] is set to Enable, SCM periodically sends OPTIONS messages to the selected application server. If a 200 OK message is received as a response to an OPTIONS message, the application server is registered.

Application Server Registration Clear

If there is no response for an OPTIONS message or if response fails, SCM attempts to resend the message for the number of times. If all resending attempts fail, the application sever registration is cleared.
After clearing registration for the application server, SCM waits for the time before it resends the OPTIONS message.

3.4.10 Photo File Management

The purpose of the photo file management is to check photo files which are used in SCME and to provide tools such as Compress, Delete, Adjust and AS Sync. Also, Upload and download of batch file can be provide through this menu.
The functions are as follow.

- Search: show the count and the size of photo files which is stored in SCM
- Command: provide Compress/Delete/Adjust/AS Sync processing.
- Upload: can change several photo files with a batch file formatted in TAR.
- Download: can get the stored photo files in SCM.

CHAPTER 4. Application Features

This part describes the applications provided by SCM and how to configure them. SCM includes a basic conference system and a basic ACD server. SCM Express also includes an advanced conference system and a voice mail system.

4.1 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.

4.1.1 Queuing Control

To put the call received by ACD group on hold, the Queuing service is performed internally within the system.

Message Greeting

First Greeting Message (Available)
This is the message that is played when some lines of the agents, who are members, are available when a call is received by an ACD group.

First Greeting Message (All Busy)
This is the message that is played when all lines of the agents, who are members, are busy when a call is received by an ACD group.

Second Greeting
This is the message that is repeated until an agent is available for the service, while all lines of the agents who are ACD group members are busy.

Maximum Queuing Count

This refers to the maximum number of calls that can be on hold in Queue after received by ACD group, and when this number is exceeded, the calls are forwarded to the Next Destination.

Next Agent Hunting

When the agent to whom the call is distributed is not answering, the function stops the ringing then calls the next member.

Next Destination

When agents are all busy
When all agents' lines are busy, the function waits for the Overflow Time then forwards the call to the All Busy Destination.

When all agents are all logout
When all Agents are logged out, the function forwards the call to the All Logout Destination instead of waiting for the Overflow Time.

Waiting Timeout to alternate destination
When there is no Agent for the service even after waiting for the Overflow Time, the function forwards the call to the Overflow Destination such as extension line, Hunt group, or ACD group.

When alternate destination is none
When there is no alternate destination for the service even after waiting for the Overflow Time, the call is forwarded to default Operator group.

4.1.2 Routing Control

It describes the distribution methods of the received calls distributed to ACD group.

Longest Idle Agent

It is the type of distributing the call to the agent who is at the idle state for the longest time when the call is received by the ACD group.

Least Occupied Agent

It is the type of distributing the call to the agent who had the shortest calling time during a certain period before the call being received by the ACD group.

Sequence Mode

It is the type of distributing the call to the agent in sequence.

4.1.3 Agent Status

It describes the status information of Agent. ACD Group calls are distributed according to the Agent status.

Login and Logout

Register whether Agent would receive the ACD group call distribution service or not.

Break and Work

Register an Agent in Break status when the agent cannot answer calls from ACD group call distribution service due to a meeting, break, etc. When the Break status is cleared, the status returns to Work status.

Wrap-Up

Refers to the state of time when an agent is writing a memo or performing other tasks that are related to the call just handled instead of taking the next call immediately. Calls are not distributed to the Agent with this status.

4.1.4 Real-Time Monitoring

Provides the real time status information of the ACD Group where the Agent belongs.

Number of wait calls

Number of the calls currently waiting for the call distribution

Longest ACD wait time (current)

Waiting time of the call that has waited for the longest time among the calls currently waiting

Number of Agents

The following Agent statistics information can be output.

- Number of Logged-in Agent
- Number of Agents that the call distribution service has available
- Number of Agents in use
- Number of Agents who are in status of getting ready after completing calls
- Number of Agents who are on a Break.

4.1.5 ACD Group Statistics

The following ACD Group statistics information can be output.

Abandoned Calls

This is the number of the calls abandoned before being answered by the agent, as received by the ACD group. When the number is high, it means that either more Agents are needed or the waiting time is too long.

Average Ring Time

This is the time from the moment of ringing after a call has been distributed to an Agent up to the moment when it is answered by the Agent, excluding unanswered calls.

Number of Times All Agents Busy

This shows the number of occurrences when all agents are busy or logout when a call has been received by ACD group.

Average Time in Queue

This shows the average waiting time of all calls that are waiting in the ACD queue.

Total Calls Received

This shows the total number of calls received by the ACD group. This number is the total sum of answered calls, calls transferred to other groups due to all agents being busy or logged out, and overflowed calls.
The statistics include the items below.

- Calls answered by Agent
- Overflowed Calls
- Calls Abandoned before being answered by Agent

Longest Queue Time Today

This shows the call that waited longest in the queue for today. The waiting time is calculated as follows.

- Queue time begins: start time of call waiting
- Queue time ends: end time of call completion
  - When Answered by Agent
  - When Abandoned by Caller
  - When Transferred due to Overflow

4.1.6 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 ‘4.4. External Applications’ section.

4.2 Conference

SCM provides Conference. Embedded Conference system provided in SCM refers to the system providing the voice conference function within SCM.

4.2.1 Conference Features

The conference functions provided by the Embedded Conference system are as following:

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.

Predefined Conference

The service allows a voice conference to be conducted with pre-assigned groups.
The master of conference prepares the list of the group and members in advance by using the management module. Also, the system calls each member using the list identifier for the conference then delivers the voices to each member after mixing the voice signals of each member into one voice signal.

Progressive Conference

The master dials the Progressive Conference feature code, and then enters the participant numbers according to the voice guidance. When the master is about to start the conference, the system calls each member, mixes the voice signals of the members then delivers the voice to each member in single voice signal.

Intercom Conference

Initiate a conference by dialing feature code + a station group number. The phones registered in the station group answer automatically to join the conference.

Dispatch Conference

This feature is provided for the CSTA applications to initiate a conference using its own conference group. It is not available from a phone.

Meet-Me Conference

The master prepares a conference room in advance, and then the members who participate in the conference input the conference ID to join. When each member tries to participate, the Embedded Conference system receives the conference ID and passcode through the voice guidance and DTMF before deciding whether to allow the entry to the conference.

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.

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.

4.2.2 Conference Control

The conference control function provides a monitoring function for conference status (participants and voice level) by the master of the conference call, voice level control (mute of a participant's voice), and removing a participant from the conference room functions.

Conference Monitoring

The master can view the current conference status.
He/she can view the participating members, mute, or eject members, lock the conference, terminate the conference

Conference Group Management

A user can view, create, change or delete the pre-define conference groups.

Conference Room Lock/Unlock

The master may lock/unlock his/her own conference rooms. If a conference is locked, no more member can attend the conference.

Sole Participant Audio

This is the function that plays music (or a tone) if a participant is in the conference room by him/herself.

Adaptive Codec Negotiation

It is the function that actively negotiates the codec according to the system load.
SCM supported following audio CODECs: G.711a, G.711u, G.729.

4.3 VoiceMail/Auto Attendant (VM/AA)

VM/AA provided by SCM refers to the integrated messaging system interfacing with IP interface within SCM. In other words, it provides voice mail, automatic repeater, e-mail server interface, and other functions to users.

4.3.1 Voice Mail

The Voice Mail function is the service provided when the number called is unavailable. In this case, the caller may be connected directly to the voice mail or connection to the voice mail after ringing may be selected. The call is connected to voice mail according to various status (busy, no answer, and busy/no answer).
The system answers the call with the user's pre-recorded message and then the caller may leave a message.
The user can access the voice mail then listens to the messages received from anywhere. Various options are available for handling the messages.

Answering Machine Emulation (AME)

This function allows monitoring of calls made to the voice mail by using a phone speaker, and is similar to a home answering machine.

Auto Conversation Recording

This function is used to record the phone conversation made between calling and receiving parties in the voice mail of the receiver. The recording starts automatically upon the beginning of the conversation and ends with the phone hanging up. Then the recording contents can be checked in the voice mail.

Auto Forward

This function forwards the voice message automatically to the mail box of another phone if the message left in the voice mail is not played within a certain amount of time. The Delay time can be set by the user, and the forwarded message can either be kept or deleted.

Auto Login

This function automatically logs into the voice mail without inputting the pin number and the authentication process.

Auto Message Play

This function automatically plays new messages that arrive in the voice mail when the user is logging in, and it can minimize the unnecessary operation of selecting digits to input.

Broadcasting

This function sends a voice message to all users of the system at once that can be performed by the user with the manager's authority.

Call Back

This function calls the sender who sent a voice message by pressing a button while listening to the message and can call both the extension and the office line. To call an office line, the Call ID should be available.

Call Forwarding to Voice Mail

This function forwards all incoming calls to the voice mail when the user's line is busy or cannot answer.

Call Record

This function records the contents of the phone conversation currently being made and saves it in the voice mail by pressing a button.

Call Back Requested Message

A caller may select the Call Back Request option when leaving a message, and once the number to receive the call back has input, the user is notified that the call back has been requested when playing the message. At this time, the user can call the caller directly by pressing a button.

Date and Time Stamp

This function notifies the accurate message arrival time when the user has logged in the voice mail then playing a new message.

Quick Memo

This function allows a caller to leave a message more conveniently with other users.

Distribution Lists

This function leaves a voice message to multiple users simultaneously. The list can include not only the users' phone numbers but also the lists containing the users' phone number. When a message is left on a list, the message is saved in all phones of the users who are on that list.

External Number Notification

This function gives notification of a voice message to a pre-registered home phone number, cell phone, or other phone when it is received by the user's voice mail.

Message Scheduled Delivery

When a user leaves a message with another user, the function allows the delivery time of the message to be set.

Message Grouping

This function is used to listen to the voice messages received by a user's voice mail per type (emergency, Callback, Fax, Reply Requested, Memo, etc.) in groups.

Individualized Voice Mail Greeting

This function is for a user to record a greeting message for his/her voice mail. When a caller is accessing the voice mail to leave a message, the recorded voice mail greeting can be played.

Individualized Voice Mail Name

It is the function of linking each voice mail to the name that is recorded by the user's own voice.

Message Address Verification

This function is used to play back the numbers pressed by the user. This is so that he can make sure they are correct when he leaves a voice message.

Message Delivery Options

When a voice message is left in a user's voice mail, this function notifies it to a pre-registered home phone number, cell phone, or other phone only at the registered time.

Message Alert Notification Schedule

This is the function to set options for sending a voice message, and the options are Emergency Message, Callback Request, Reply Request, Secret Message, Receive Notification, etc.

Message Listening Option

The function is used for setting options for listening to the voice messages in voice mail, and functions such as replay, save, delete, rewind, fast forward, and pause can be used.

Message Forward With Append

When forwarding a voice message received by a user, this function allows additional information to be recorded onto the original voice message and sends it together.

Message Length Control Option

It is the function of setting the length of a voice message received by a user's voice mail.

Message Delivery Cancellation

This is the function that allows the cancellation of the voice message left in a voice mail by the caller if the message has not been played by the called person.

Message Reply

This is the function of immediately sending a voice mail to the caller by pressing a button while the user is listening to the voice message received.

Message Scan

This is the function of partially listening to the beginning parts of all messages in the voice mail of a user and scanning for a message.

Message Skip

This is the function of skipping the message that is saved in the voice mail to the next one instead of listening to the end of the message.

Message Undelete

This is the function used to either listen or save a message in a voice mail after a user has listened and deleted the message.

Message Waiting Light Indication

When a voice message has arrived in the voice mail, the function notifies it to the user by blinking the phone lamp.

Multiple User Mailboxes Login

It is the function of several users accessing one mail box simultaneously then listening to a message. However, one message cannot be listened to by several users at the same time, and the message being played for a user becomes unavailable for other users.

Multiple User Mailboxes

Generally, the same number is used for both Mailbox and Extension, but different numbers can be used also for the each or using a Mailbox without an Extension is also possible.

One Touch Access

This is the function of logging in by using just one button or logging in with the Admin authority.

Retrieve Public Caller from Mail Box

This function notifies the user when he/she is logging in if a caller is currently leaving a voice message and connects to the caller after asking whether to connect or not.

Reminder

This function is for the user to leave a self memo such as a schedule or other items that need to be recorded.

4.3.2 Auto Attendant

Auto Attendant (AA) function is the function whereby VM/AA answers a caller automatically and plays a voice guidance message or messages according to the caller's dialing command. The caller may appoint a person or group by pressing the number.
AA answers the caller with a pre-recorded prompt and also plays several options that the caller may choose (selecting the number to transfer, connecting to the voice mail or Auto Attendant, accessing the pre-recorded information, etc.).

Multiple Alphabetic Directory

When a user does not know the phone number of the person to call, the function allows the system to search the phone number just with partial letters of the name then either connects to the person or to the voice mail.
Also, the system directory service is available to use by pressing '9' on AA after calling on the office line. When the name of the called person is input through the key pad, the call is connected to the person when the search result is one person or plays the names that are searched if two or more names are resulted where the caller can choose one name.

Auto Attendant Route

Automatic repeater is the function to either route or transfer a call according to the number the caller is pressing. By using the function, a call can be either transferred to a specific user or connected to a voice mail in VM/AA and also can interworking with functions such as Fax sending/receiving or E-mail receiving.

Automatic After Hours Answering

The function allows the greeting message of the Auto Attendant to be changed automatically according to the time and various modes.

Announce Hold Position

It is the function to put a call on hold in a waiting queue if the caller on the line desires then notifying the callers' location in the queue.

Announce Hold Time

This is the function to notify a caller the expected waiting time in current queue if the caller on the line wants to remain on hold. (same to the setting of Hold Location Notification Function)

Camp-On Support

This is the function to set when to pick up another call came while the caller is on the phone with other person.

Direct to Mailbox

This is the function of providing only the mailbox without a linked phone number (Station). This function is ideal in situations where there are few workers working in the office while many are working outside of the office.

Holiday and Special Events

This is the function for playing a special prompt to callers who made calls on holidays or certain company holidays.

Interruptible Voice Prompts

This is the function to select a desired service while a prompt is being played instead of listening to the end of the voice guidance or prompt.

Multi Call Handling

This function handles multiple calls simultaneously.

Multi Language Support

Supports multiple languages.

Operator Access

A caller may connect to an operator who is in ready state by pressing ‘0’ at any time.

Single Digit Call Routing

The menu processor can be set to perform a routing in a specific routine by recognizing one number. For instance, it can be set to connect to the number 2001 of Sales Department when the number ‘1’ is pressed on menu.)

4.3.3 Access Manager

Access Manager Function can individually manage the callers connecting the users.
The mail box owner can set his/her extension number to not ringing, connect a call to other extension number within the phone system, or search the call before answering.
These conditions can be set for specific times. VM/AA can be set to 'Find Me' to allow connection to the users in different places.

Call Blocking

While Call Blocking is set by a user, VM/AA does not connect callers to the user's extension. Instead, the call blocking greeting prompt is played immediately to the caller. If the call blocking prompt has not been recorded, VM/AA plays the user's existing no answer prompt. If the no answer prompt has not been recorded either, VM/AA plays the guidance prompt to the caller saying that the called number is currently not available and suggest other options.
The user sets call blocking using the access manager service. After setting call blocking, the user can set the blocking period. This function can be set to be activated for certain time periods such as until the end of today's business hours, until the start of the next business hours, until one day of the next week, from Monday to Sunday, until a certain date, until a specified time, or for certain hours; from 1 hour to 9 hours.

Call Forwarding

Callers are connected to an internal extension number. (cannot connect to an office line.) When the call is being connected to an extension, the prompt of 'You are being connected to A of B department. Calls are connected to C.' is played to the caller. Then when the called person answers the call, he will hear an automated announcement giving the information of the caller.
This function can be activated for between 1 and 9 hours, until the end of current business hours, until the beginning of the next business hours, until one day in the next week, from Monday to Sunday, or until a certain date.

Day/Night Personal Greeting

The Night Personal Greeting works in conjunction with the user's Weekly Availability Schedule. During the defined availability period, the VM/AA will automatically play the Primary No-Answer Greeting to callers, unless the user is busy on another call and has recorded a Busy Greeting or Call Blocking is set active, and a Call Blocking Greeting is recorded. During the time periods a user is not scheduled available to take calls, and does not have Call Blocking activated, the VM/AA plays the user's Night Greeting if recorded.

Find me

When this function is activated, the VM/AA attempts to deliver a call to a place designated by the user. The VM/AA first finds the user in the location designated by the user then makes the call to each of the user's saved phone numbers if necessary until all numbers are called. This function can be activated for between 1 and 9 hours, until the end of current business hours, until the beginning of the next business hours, until one day next week, from Monday to Sunday, or until a certain date.

Follow Me

A user can pick up a call that is automatically forwarded to a designated location. This is called the user's location designation. The location designation can be set for both internal extension and external phone number. This function can be activated for several hours from 1 to 9 hours, until the end of current business hours, until the beginning of the next business hour, until the day of the next week, from Monday to Sunday, or until a certain date.

Hold for Busy Station

VM/AA allows the caller to be placed on hold during a call. When a caller makes this choice, VM/AA puts the caller on hold. If another caller reaches the busy extension, the caller is put on hold with an on-hold notification and the expected waiting time until connection. Meanwhile, the called person is informed of a new incoming call.

Multiple Personal Greeting

When a user cannot receive a call, the VM/AA answers the call. VM/AA categorizes the user's reasons (5 categories) for being unable to pick up the call, and different greeting messages can be used for each reason. VM/AA provides several (1 to 9) greeting recording functions to users, and a user can use different greetings according to the situations.

Park and Overhead Paging

The VM/AA provides park and paging function for users who are frequently out of office. When a user is not answering a call and caller selects 'Park & Page,' the VM/AA notifies the user by turning on the speaker phones that are registered by the user.

Telephone Number Storing

This function is used to save the phone numbers that are frequently used by a user.

Weekly Schedule of Call Availability

This function registers a time of a day when the user can generally receive calls, and VM/AA plays a proper guidance message if the user cannot receive calls during that time.

4.3.4 E-mail Server Interface

An integrated messaging system can be configured by interfacing voice message through the E-mail server interface function.
Especially through the MS Outlook interface function, considerations have been given to existing E-mail users to use VM/AA more conveniently by allowing existing Outlook users for the service. Also, the integrated messaging system can be configured by interfacing with external servers (example: Exchange Server).

Outlook Interface

Users of VM/AA can send/receive all voice messages to/from the voice mail by using the Outlook.

Interface with E-mail Server

By interfacing (IMAP) with E-mail server, VM/AA can integrate and manage voice messages.

4.4 External Applications

4.4.1 Samsung Operator

The Samsung Operator Console is a PC-based console, which is a telephony application that inter-works with the SCM system based on CSTA Interface. Once the operator receives a request for call operation from the SCM system, the operator connects or transfers the call through the Samsung Operator Console.
For instance, when a caller dials a business phone, the operator answers the call, and transfers it to the requested number through the SCM system.
The SCM system, Samsung Operator Console is used for Intranet as shown in the figure below, and transfers/receives data in the TCP/IP protocol.
The Samsung Operator Console performs the functions described below:

Call Processing by Using the Keyboard/Mouse

The user can use the keyboard or mouse to operate calls. If the operator answers a call by lifting the handset, the operator can use the Samsung Operator Console and phone by turns when required.

Drag-and-Drop Call Processing

The operator can drag and drop a call on the station icon to transfer the call. For detailed information about the drag and drop, refer to the sections about drag-and-drop call processing in this manual.

Call Processing by Call Status

Calls are displayed in the <Incoming Calls> window, <Active Calls> window, and <Held Calls> window depending on call status. Caller information such as a caller ID is displayed in each window. Particularly in the <Active Calls> window, the caller information stored in the database is also displayed.

Real-time Display of Station Status

The station status registered with Busy Lamp Field (BLF) is displayed in the format of icons in real-time so that the user can easily check station status such as busy, dialing, and Do Not Disturb (DND).

Call Log

The call logs of the Samsung Operator Console are maintained for all the incoming/ outgoing calls made while operating the application. A call log includes the Notes field where data on a specific call are recorded. Use the log view to view a call log and search the log by categories.

For more details, see the Samsung Operator Console User Guide.

4.4.2 Samsung 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 two 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. All of which empowering the user, when connected, to make/receive calls to/from anywhere in the world as authorized by the organization.
The Samsung Communicator is an integrated client communications application that brings together two devices; your Business PC and your Business Telephone. This enables users/users on the same phone system to communicate via Voice and Video as well as share Information, and view Presence Awareness, when the Communicator is licensed in combination with the Messenger Application. We call this Unified Communications.
The Samsung Communicator can be installed in one of two Operation Modes; Basic and Professional. Basic provides two Device Modes; SoftPhone, UCPhone. These device Modes determine which device the Communicator will be working with as your telephone device.
In Soft Phone Mode the PC/Laptop becomes the telephone device. In UC Phone Mode the Communicator connects with the SMT-i Series Phones, while in Desk Phone Mode it connects with all other Samsung model phones
The Professional installation Mode includes all the same functionality as with Basic Mode and adds to it the Messenger functionality. The functionality added by Samsung Messenger are Presence Awareness, Instant Messaging, and information sharing like Chat & Video Chat, White Boarding, Screen Sharing, and File Sharing.

Soft Phone Mode (PC Phone Integration Device)

The Soft Phone Mode is also known as a PC Phone device. In this mode, the Communicator Soft Phone is the device the user uses to make/receive all their calls.
The Communicator Soft Phone is a software device that emulates a Samsung business telephone. The Communicator Soft Phone is registered to the Samsung switch installed at the Corporate office empowering the user, when connected, to make/receive calls to/from anywhere in the world as authorized by the organization, all through their PC and PC's Multi-Media functionality (PC MUST be equipped with Internal or external Mic and Speakers and/or headset).

UC Phone Mode (Samsung SMT-i Series Device Integration Only)

In this mode, the Samsung Communicator interoperates directly with a telephone device such as the SMT-i5243 terminal. In this mode, telephone functionality can be performed from either the PC or telephone device. However unlike the Communicator Soft Phone Mode while in UC Phone Mode you MUST use the SMT-i Series device to talk with the caller. When the Communicator PROFESSIONAL UCPhone Mode is paired with the SMT-i5243 device, content registration and Buddy List viewing functions are provided by the Collaboration with the Samsung Messenger Server.

BASIC mode

With Communicator BASIC mode, the user can select for the Communicator to run in one of two different device Modes; Soft Phone, UC Phone. Each gives the user control over.

- Audio Conference
- Audio Recording
- Answering/Making Calls
- Call Control: (Transfer, Hold, Forwarding, Speed Dial, etc…)
- Call/Missed Call Logs
- Dial from Outlook Contact List(s)
- (Access to) Easyset
- Free Dial
- Missed Call Notification
- Phonebook
- Schedule
- Screen Pop Contact List (both from OSC Phonebook and Outlook)
- SMS/Text Messages
- Video Calls
- And more…

PROFESSIONAL mode

With Communicator PROFESSIONAL mode Samsung Communicator provides not only the BASIC functions listed above but adds collaboration to a Samsung Messenger Server. Some of the collaboration added functions are

- Buddy List (requires SMT-i5243 Phone)
- Chat
- File Sharing
- Messenger Contents
- Presence Awareness
- Screen Sharing
- Video Chat (up to 5 members)
- White Boarding
- And More

For more details, see the Samsung Communicator User Guide.

4.4.3 ACD Agent

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
- Transfer Calls
- Conference Calls

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.

4.4.4 Third Party Applications

SCM can interface with external SIP Conference Server, SIP VMS, SIP UMS, SIP IVR (AA Server), and Legacy (PSTN) VMS, and can be categorized as shown below according to the interface type.

Using Standard SIP

SCM provides the interface function by using standard SIP protocols with application servers such as SIP Conference Server, SIP VMS Server, SIP UMS Server, and SIP IVR.

Using Standard CSTA

SCM provides the interface function using standard CSTA Phase-I, II protocols with application servers such as ACD Server and Operator Server.

CHAPTER 5. System Reports

5.1 Traffic Reports

5.1.1 Call Traffic Reports

SCM provides statistical information for calls generated in the system by hours, dates and months.

System Statistics

- Internal Calls: Calls were attempted between internal users.
- Outbound Calls: Calls internal users attempted to call external users.
- Inbound Calls: Calls external users attempted to call internal users.
- Tandem Calls: Calls external users attempted to call other external users through SCM.
- Call Failures: All failed calls including unknown calls.
- All Calls: This shows the statistical information for all calls.
- Failure Reasons Calls: All failure reasons.

User Group Statistics

- Outgoing: Counts users of user group attempted to make outgoing calls.
- Incoming: Counts attempted for users of the user group

Service Group Statistics

- Outgoing: Counts users of service group attempted to make outgoing calls.
- Incoming: Counts incoming calls were attempted for users of the service Group

Route Statistics

- Inbound: Counts incoming calls were attempted for internal users through the route.
- Outbound: Counts internal users attempted to call external users through the route.

Hunt Group Statistics

Incoming: Counts incoming calls were attempted for users of the hunt group.

User Statistics

- Outgoing: Counts the user attempted to make outgoing calls.
- Incoming: Counts incoming calls were attempted for the user.

System Service Statistics

- Mobile Remote Dial: Counts the system attempted the make mobile remote dial.
- Smart Routing: Counts the system attempted the make all smart routing.
- Smart Routing (Internal): Counts the system attempted the make internal smart routing.
- Smart Routing (External): Counts the system attempted the make inter SCM smart routing.
- Smart Handover: Counts the system attempted the make smart handover.
- Mobile Call: Counts Samsung mobile phone attempted the make calls.

Phone Usage by Type Statistics

- Internal Outgoing Usage: Counts outgoing calls were attempted between internal users
- Internal Incoming Usage: Counts incoming calls were attempted between internal users
- Outbound Usage: Counts internal users attempted to call external users .
- Inbound Usage: Counts incoming calls were attempted for internal users.
- Total Usage: This shows the statistical information for all calls.

Phone Registration by Type Statistics

- To Reg: Counts state change from unregistration to registration
- To Unreg: Counts state change from registration to unregistration
- Registration Average: Averages state is registration.

5.1.2 ACD Reports

ACD Group Statistics

- System Summary: All system-wide ACD calls.
- Group Summary: All calls for the ACD group.
- Overflow: Overflow occurred for ACD group.

ACD Agent Statistics

- Summary: all agents' calls.
- Utilization: all agents' level of contribution.
- Activity: all agents' activities.

ACD 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.

5.2 Performance Reports

5.2.1 System Performance

SCM check the current SCM system performance status including CPU, memory, and disk utilization on the right side of the main monitor. So user can also view the system and process resource usage.

System Resource Report

SCM 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 SCM include CPU, memory, hard disk drives, and network cards.

Process Resource Report

SCM monitors the memory and CPU usage by processes. 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

5.2.2 Call Performance Report

SCM 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. SCM also provides the trace feature which allows the administrator to trace calls or protocol messages.

Call Management

You can view the currently processed calls. 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.

5.2.3 Fault Reports

Alarm Profile

An alarm profile contains detailed information for alarms, faults and status generated in SCM.

Setting-Alarm

For the alarms indicated as requiring default values in the table below, the alarm levels can be changed according to the default values. Therefore, such alarms do not have default level values such as critical, major or minor.

Alarm Name	Category	Level	Critical (%)	Major (%)	Minor (%)
Abnormal Block State	Processing Error	Critical	-	-	-
Abnormal 3rd Party Application State	Processing Error	Critical	-	-	-
CPU Over Load (%)	Resource	-	100	95	90
Hard-Disk Over Used (%)	Resource	-	100	95	90
Memory Over Used (%)	Resource	-	100	95	90
CPU Over Used by Process (%)	Resource	-	100	95	90
Network Interface Down	Equipment	Critical	-	-	-
Standby System Down	Equipment	Critical	-	-	-
Maximum Call (%)	QoS	-	100	95	90
Maximum Subscriber (%)	QoS	-	100	95	-
Gateway Connection Lost	Communication	Minor	-	-	-
Location Bandwidth Used (%)	System Management	-	95	90	85
Resource Based CAC	System Management	Major	-	-	-
MailBox Over Used (%)	Resource	-	100	95	90
A/A Link Down	Communication	Critical	-	-	-
A/A License Expired	System Management	Minor	-	-	-
Recording Disk Space Used (%)	Resource	-	95	85	75
ALARM CLEARED	[GW] OfficeServ	-	-	-	-
FAN Out of Order	[GW] OfficeServ	Critical	-	-	-
CPU Overload	[GW] OfficeServ	Critical	-	-	-
DUAL PWR Error	[GW] OfficeServ	Critical	-	-	-
D-PWR FAN Error	[GW] OfficeServ	Critical	-	-	-
PoE PWR Error	[GW] OfficeServ	Critical	-	-	-
PoE FAN Error	[GW] OfficeServ	Critical	-	-	-
PoE Battery Error	[GW] OfficeServ	Critical	-	-	-
MAIN PWR Error	[GW] OfficeServ	Critical	-	-	-

Alarm Name	Category	Level	Critical (%)	Major (%)	Minor (%)
SYS High Temp	[GW] OfficeServ	Critical	-	-	-
IPC MSGQ Over	[GW] OfficeServ	Major	-	-	-
AC Pwr Loss	[GW] OfficeServ	Major	-	-	-
Low Battery	[GW] OfficeServ	Major	-	-	-
D-BD Init Fault	[GW] OfficeServ	Major	-	-	-
Card Init Fault	[GW] OfficeServ	Major	-	-	-
Sync Failure	[GW] OfficeServ	Major	-	-	-
Red Alarm	[GW] OfficeServ	Major	-	-	-
Yellow Alarm	[GW] OfficeServ	Major	-	-	-
Blue Alarm	[GW] OfficeServ	Major	-	-	-
SPID Init Error	[GW] OfficeServ	Minor	-	-	-
LPBK Error	[GW] OfficeServ	Minor	-	-	-
BRI DL Unavail	[GW] OfficeServ	Minor	-	-	-
PCM Loss	[GW] OfficeServ	Major	-	-	-
L2 Disconnect	[GW] OfficeServ	Major	-	-	-
MGI NTWK Error	[GW] OfficeServ	Major	-	-	-
MGI DSP Error	[GW] OfficeServ	Major	-	-	-
Trunk Fault	[GW] OfficeServ	Major	-	-	-
Trunk Disconnect	[GW] OfficeServ	Major	-	-	-
T1 Out Of Srv	[GW] OfficeServ	Minor	-	-	-
SLI Fault	[GW] OfficeServ	Minor	-	-	-
SYS FAN Stop	[GW] OfficeServ	Major	-	-	-
SIP Server Disc	[GW] OfficeServ	Critical	-	-	-
Gatekeeper Disc	[GW] OfficeServ	Major	-	-	-
ALARM CLEARED	[GW] iBG Series	-	-	-	-
FanTray Fail	[GW] iBG Series	Critical	-	-	-
FanUnit Fail	[GW] iBG Series	Critical	-	-	-
PoePower Fail	[GW] iBG Series	Major	-	-	-
Temperature Critical	[GW] iBG Series	Critical	-	-	-
Temperature Warning	[GW] iBG Series	Major	-	-	-
All Network-Clock Fail	[GW] iBG Series	Critical	-	-	-
CPU Over Load	[GW] iBG Series	Minor	-	-	-
Memory Over Used	[GW] iBG Series	Minor	-	-	-
T1E1 RLOS	[GW] iBG Series	Critical	-	-	-
T1E1 RLOF	[GW] iBG Series	Critical	-	-	-
T1E1 RAIS	[GW] iBG Series	Critical	-	-	-

Alarm Name	Category	Level	Critical (%)	Major (%)	Minor (%)
T1E1 RRAI	[GW] iBG Series	Critical	-	-	-
T1E1 TAIS	[GW] iBG Series	Major	-	-	-
T1E1 TRAI	[GW] iBG Series	Major	-	-	-
BRI LOF	[GW] iBG Series	Critical	-	-	-
Serial DSR	[GW] iBG Series	Critical	-	-	-
Ethernet LOS	[GW] iBG Series	Critical	-	-	-
Adsl Communication Fail	[GW] iBG Series	Critical	-	-	-
Adsl LOS	[GW] iBG Series	Critical	-	-	-
Maximum Call (Maj)	[GW] iBG Series	Major	-	-	-
Maximum Call (Min)	[GW] iBG Series	Minor	-	-	-
Maximum Channel (Maj)	[GW] iBG Series	Major	-	-	-
Maximum Channel (Min)	[GW] iBG Series	Minor	-	-	-
SIP Connection Fail	[GW] iBG Series	Major	-	-	-
H.323 Connection Fail	[GW] iBG Series	Major	-	-	-
FXO Connection Fail	[GW] iBG Series	Minor	-	-	-
DSP Fail	[GW] iBG Series	Critical	-	-	-
IVM Communication Fail	[GW] iBG Series	Critical	-	-	-
ALARM CLEARED	[GW] OS7500/OS7600	-	-	-	-
Rectifier Power Fail	[GW] OS7500/OS7600	Major	-	-	-
Battery Fail	[GW] OS7500/OS7600	Major	-	-	-
Fire	[GW] OS7500/OS7600	Major	-	-	-
Temperature Out of Range	[GW] OS7500/OS7600	Major	-	-	-
Humidity Out of Range	[GW] OS7500/OS7600	Major	-	-	-
Door Open	[GW] OS7500/OS7600	Major	-	-	-
FAN Out of Order	[GW] OS7500/OS7600	Major	-	-	-
System Overheat	[GW] OS7500/OS7600	Major	-	-	-
CPU Overload	[GW] OS7500/OS7600	-	90	80	70
Clock Fail	[GW] OS7500/OS7600	Major	-	-	-
Port OOS Over Limit	[GW] OS7500/OS7600	Minor	-	-	-
Trunk Group OOS Over Limit	[GW] OS7500/OS7600	Minor	-	-	-
MGM (GCM) Down	[GW] OS7500/OS7600	Minor	-	-	-
MGM (GCM) Dup. Link Down	[GW] OS7500/OS7600	Minor	-	-	-
Tone Source Fail	[GW] OS7500/OS7600	Major	-	-	-
LPM Simplex Down	[GW] OS7500/OS7600	Minor	-	-	-
LPM Duplex Down	[GW] OS7500/OS7600	Minor	-	-	-
SS7M Simplex Down	[GW] OS7500/OS7600	Minor	-	-	-

Alarm Name	Category	Level	Critical (%)	Major (%)	Minor (%)
SS7M Duplex Down	[GW] OS7500/OS7600	Minor	-	-	-
No7 Sig Link Down	[GW] OS7500/OS7600	Minor	-	-	-
No7 Sig Link Set Inactive	[GW] OS7500/OS7600	Minor	-	-	-
No7 Dest Inaccessible	[GW] OS7500/OS7600	Minor	-	-	-
Comm. Serv Device Fail	[GW] OS7500/OS7600	Minor	-	-	-
SMDR Total Buffer Full	[GW] OS7500/OS7600	-	-	100	90
SMDR User Buffer Full	[GW] OS7500/OS7600	-	-	100	90
CTI Serv. Link Down	[GW] OS7500/OS7600	Major	-	-	-
Comm.Serv. Link Down	[GW] OS7500/OS7600	Critical	-	-	-
GateKeeper Link Down	[GW] OS7500/OS7600	Major	-	-	-
Admin Serv. Link Down	[GW] OS7500/OS7600	Major	-	-	-
Infomobile Link Down	[GW] OS7500/OS7600	Critical	-	-	-
Ringer Fail	[GW] OS7500/OS7600	Minor	-	-	-
Message Waiting Power Fail	[GW] OS7500/OS7600	Minor	-	-	-
PDM out of Service	[GW] OS7500/OS7600	Major	-	-	-
Ringer out of Service	[GW] OS7500/OS7600	Major	-	-	-
Frame Warning	[GW] OS7500/OS7600	Minor	-	-	-
AIS Warning	[GW] OS7500/OS7600	Minor	-	-	-
TS16AIS Warning	[GW] OS7500/OS7600	Minor	-	-	-
Remote Warning	[GW] OS7500/OS7600	Minor	-	-	-
Multi-Frame Sync. Fail	[GW] OS7500/OS7600	Minor	-	-	-
Tree Bad Frame Alarm	[GW] OS7500/OS7600	Minor	-	-	-
Excessive Slip	[GW] OS7500/OS7600	Minor	-	-	-
Excessive Bit Error	[GW] OS7500/OS7600	Minor	-	-	-
Card out	[GW] OS7500/OS7600	Minor	-	-	-
QoS Monitor Link Down	[GW] OS7500/OS7600	Minor	-	-	-
SIP Server Link Down	[GW] OS7500/OS7600	Major	-	-	-
SPNet Link Down	[GW] OS7500/OS7600	Major	-	-	-
VCM Link Down	[GW] OS7500/OS7600	Major	-	-	-
VCM - OS7200 Link Down	[GW] OS7500/OS7600	Major	-	-	-
SIP CONNECTOR Link Down	[GW] OS7500/OS7600	Major	-	-	-
SCM Express Link Down	[GW] OS7500/OS7600	Major	-	-	-
Reference Clock Fail	[GW] OS7500/OS7600	Minor	-	-	-
Sync Fail	[GW] OS7500/OS7600	Minor	-	-	-
LPM-CARD Link Fail	[GW] OS7500/OS7600	Minor	-	-	-
Port Test Fail	[GW] OS7500/OS7600	Minor	-	-	-

Alarm Name	Category	Level	Critical (%)	Major (%)	Minor (%)
Trunk Seizure Test Fail	[GW] OS7500/OS7600	Minor	-	-	-
Digiphone Down	[GW] OS7500/OS7600	Minor	-	-	-
IP Termal Link Down	[GW] OS7500/OS7600	Minor	-	-	-
iNet Network Fault	[GW] OS7500/OS7600	Minor	-	-	-

Setting-Fault

Fault ID	Category
Authentication Fail	Communication
CDR FTP Send Fail	Processing Error
Illegal Call Try	Communication
DB Backup Fail	Processing Error
Unknown Registration	Communication
Evaluation License Expired	System Management
Sending Event Email Fail	Processing Error
Phone Upgrade Fail	Processing Error
MPS Channel Get Fail	Communication
A/A Data Send Fail	Communication
StandBy/Slave Data Save Fail	Communication
Active-Standby configuration mismatch	System Management
Unsupported Codec Request	Communication
MCP Reset	[GW] OfficeServ
LCP Restart	[GW] OfficeServ
PCM Switching	[GW] OfficeServ
Flash FORMAT Err	[GW] OfficeServ
Invalid MMC Halt	[GW] OfficeServ
MEDIA NOT UMOUNT	[GW] OfficeServ
HDLC Com Error	[GW] OfficeServ
Memory Alarm	[GW] OfficeServ
DTMF Fault	[GW] OfficeServ
Tone Fault	[GW] OfficeServ
CID DSP Fault	[GW] OfficeServ
Bit Error Alarm	[GW] OfficeServ
T1 Restart	[GW] OfficeServ
PRI Restart	[GW] OfficeServ
BRI Restart	[GW] OfficeServ

Fault ID	Category
MGI Restart	[GW] OfficeServ
MGI IP Duplicate	[GW] OfficeServ
MGI Self Restart	[GW] OfficeServ
TODC Error	[GW] OfficeServ
MGI Packet Loss	[GW] OfficeServ
MGI Packet Delay	[GW] OfficeServ
system reset	[GW] iBG Series
Card Out	[GW] iBG Series
Abnormal task	[GW] iBG Series
CF card full	[GW] iBG Series
FXS restart	[GW] iBG Series
FXO restart	[GW] iBG Series

Setting-Status

Status ID	Category
Block State Change	Equipment
HA Mode Change	Equipment
Wakeup Service	Communication
DB Backup	Communication
Data Sync	Communication
Long Duration Call	Communication
Malicious Call Claim	Communication
Database Over Used (%)	Resource
CDR FTP Send Complete	System Management
Phone Update Notified	System Management
Switch Over by CPU Overload	Equipment
Evaluation License Expired	System Management
Evaluation License Activated	System Management
Phone Upgrade Status	System Management
Switch Over by Memory Over Used	Equipment
TCP Send-Q Over Used	Resource
IP Conflict Detected	Environment
Switch Over Canceled	Equipment
Route Register Status	Communication
TLS Session Disconnected	Communication
A/A Node Configuration	System Management

Status ID	Category
Profile Creation	System Management
Subscriber Register Status	Communication
Evaluation License Notified	System Management
A/A License Notified	System Management
Sample License Activated	System Management
System Time Changed	Equipment
Switch Over by Each Block	Equipment
System Restart Reason	System Management
Sync Timeout Process	System Management
UMS Maintenance State Change	Equipment
Stopping Port	Communication
Power On Restart	[GW] OfficeServ
Button Restart	[GW] OfficeServ
MMC Restart	[GW] OfficeServ
Card Out	[GW] OfficeServ
Card In	[GW] OfficeServ
Manual Reset Req	[GW] OfficeServ
Card Active	[GW] OfficeServ
MEDIA CARD IN	[GW] OfficeServ
MEDIA CARD OUT	[GW] OfficeServ
Alarm Buff Clear	[GW] OfficeServ
PCMMC Connect	[GW] OfficeServ
PCMMC Disconnect	[GW] OfficeServ
Power On Restart	[GW] iBG Series
Manual Reset Req	[GW] iBG Series
CF card insertion	[GW] iBG Series
CF card removal	[GW] iBG Series
USB insertion	[GW] iBG Series
USB removal	[GW] iBG Series
MGM (GCM) Standby Side Loading Completed	[GW] OS7500/OS7600
MGM (GCM) Active Side Alive	[GW] OS7500/OS7600
MGM (GCM) Takeover	[GW] OS7500/OS7600
MGM (GCM) Standby Side Alive	[GW] OS7500/OS7600
MGM (GCM) Standby Side Out of Service	[GW] OS7500/OS7600
LPM Takeover	[GW] OS7500/OS7600
LPM Standby Side Alive	[GW] OS7500/OS7600
LPM Out Of Service	[GW] OS7500/OS7600

Status ID	Category
SS7M Takeover	[GW] OS7500/OS7600
SS7M Standby Side Alive	[GW] OS7500/OS7600
SS7M Standby Side Out Of Service	[GW] OS7500/OS7600
LPM Active Side Alive	[GW] OS7500/OS7600
SS7M Active Side Alive	[GW] OS7500/OS7600
Reference Clock Change	[GW] OS7500/OS7600
Sync Disconnection Status	[GW] OS7500/OS7600
MAT Log In	[GW] OS7500/OS7600
MAT Log Out	[GW] OS7500/OS7600
MAT Password Changed	[GW] OS7500/OS7600
Comm.Serv-MAT Link On	[GW] OS7500/OS7600
Comm.Serv-MAT Link Off	[GW] OS7500/OS7600
No.7 Active Signaling Link	[GW] OS7500/OS7600
No.7 Signaling Link Inhibit Denied	[GW] OS7500/OS7600
No.7 Signaling Link Inhibited	[GW] OS7500/OS7600
No.7 Signaling Link Uninhibit Denied	[GW] OS7500/OS7600
No.7 Signaling Link Uninhibited	[GW] OS7500/OS7600
No.7 Remote Signaling Link Inhibited	[GW] OS7500/OS7600
No.7 Remote Signaling Link Uninhibited	[GW] OS7500/OS7600
No.7 Signaling Link Remotely Blocked	[GW] OS7500/OS7600
No.7 Signaling Link Remotely Unblocked	[GW] OS7500/OS7600
No.7 Signaling Link Invalid	[GW] OS7500/OS7600
No.7 Local Change Over	[GW] OS7500/OS7600
No.7 Local Change Back	[GW] OS7500/OS7600
Alert Maintenance Personnel	[GW] OS7500/OS7600
Port MMC Block	[GW] OS7500/OS7600
Port MMC Unblock	[GW] OS7500/OS7600
Slot MMC Block	[GW] OS7500/OS7600
Slot MMC Unblock	[GW] OS7500/OS7600
E1 Slip Error	[GW] OS7500/OS7600
SMS Service Fail	[GW] OS7500/OS7600
SRBT Service Fail	[GW] OS7500/OS7600
Wake Up Fail-No Answer	[GW] OS7500/OS7600
Wake Up Fail-Blocked	[GW] OS7500/OS7600
Wake Up Fail-Busy	[GW] OS7500/OS7600
Wake Up Success	[GW] OS7500/OS7600
Wake Up Register	[GW] OS7500/OS7600

Status ID	Category
Wake Up Cancel	[GW] OS7500/OS7600
Wake Up All Cancel	[GW] OS7500/OS7600
Dispatch Line No Answer	[GW] OS7500/OS7600
Dispatch Line Busy	[GW] OS7500/OS7600
Dispatch Line Status	[GW] OS7500/OS7600
Conference Group Release	[GW] OS7500/OS7600
Malicious Call Trace	[GW] OS7500/OS7600
Conditional Interception All	[GW] OS7500/OS7600
Conditional Interception Busy	[GW] OS7500/OS7600
Conditional Interception No Answer	[GW] OS7500/OS7600
Bath Alarm	[GW] OS7500/OS7600
Bath Alarm Answer	[GW] OS7500/OS7600
Wake-Up Alarm Answer	[GW] OS7500/OS7600
Room Pilot Register	[GW] OS7500/OS7600
Room Pilot Cancel	[GW] OS7500/OS7600
Hotel Bill No. Register	[GW] OS7500/OS7600
Hotel Bill No. Cancel	[GW] OS7500/OS7600
Lan Port Change	[GW] OS7500/OS7600
Daily Takeover Fail	[GW] OS7500/OS7600
AFS cleared Status	[GW] OS7500/OS7600

5.3 Call Log Format

No.	Parameter	Description
Header		
1	Sequence Number	00000000-99999999 The sequence number when recording in Call log File. This increases by 1 when CDR is generated. Reset when new log file is generated.
2	DP Type	CDR creating point (Sending/Receiving)
3	Call ID	Call Identifier
Number		
4	Calling Number	Calling Only
5	Current Calling Number	The last number used to be connected to sending
6	Calling Name	The last name used to be connected to sending
7	Dialed Number	The number input by user or received from incoming (number before the number conversion)
8	Connect Number	The last number used to be connected to receiving
9	Connect Name	The last name used to be connected to sending
10	Billed Number	Bill Number (User's accounting code)
Usage (Time)		
11	Attempt Time	Time when the call is attempted
12	Call Duration	Call Duration Time (Disconnect Time-Answer Time)
13	Answer Time	Time when the call is connected
14	Disconnect Time	Call End Time
Call Forwarding Information		
15	Call Forwarding Flag	Call Forwarding status - 0: Call Forwarding not used - 1: Call Forwarding used
16	Call Forwarding Count	Number of Call Forwarding Attempts
17	Original Called Number	Initial Called Number
Calling Information in the SCM		
18	Calling Type	Caller Type - 0: Unknown - 1: User - 2: Service - 3: Trunk (End Point Type) - 4: Application Server Type
19	Calling IP	Calling IP Address
20	Calling SIP URI	Calling SIP URI
21	Calling User Group	User Group to which the caller belongs
22	Incoming Route	Incoming Route Number

No.	Parameter	Description
Called Information in the SCM		
23	Called Type	Called Type - 0: Unknown - 1: User - 2: Service - 3: Trunk (End Point Type) - 4: Application Server Type
24	Called IP	Called IP Address
25	Called SIP URI	Called SIP URI
26	Called User Group	User Group the called belongs
27	Outgoing Route	Outgoing Route Number
Disconnect Reason		
28	SIP Status Code	SIP Status Code
29	Q850 Release Cause	Q850 Release Cause
30	Internal Fail Code	Internal Fail Code

5.4 CDR Format

No.	Parameter	Description
Header		
1	Sequence Number	00000000-99999999 The sequence number when recording in Call log File. This increases by 1 when CDR is generated. Reset when new log file is generated.
2	DP Type	CDR creating point (Sending/Receiving)
Number		
3	Calling Number	Calling Only
4	Current Calling Number	The last number used to be connected to sending
5	Calling Name	The last name used to be connected to sending
6	Dialed Number	The number input by user or received from incoming (number before the number conversion)
7	Connect Number	The last number used to be connected to receiving
8	Connect Name	The last name used to be connected to sending
9	Billed Number	Bill Number (User's accounting code)
Usage (Time)		
10	Attempt Time	Time when the call is attempted
11	Call Duration	Call Duration Time (Disconnect Time-Answer Time)
12	Answer Time	Time when the call is connected
13	Disconnect Time	Call End Time
Calling Information in the SCM		
14	Calling Type	Caller Type - 0: Unknown - 1: User - 2: Service - 3: Trunk (End Point Type) - 4: Application Server Type
15	Calling IP	Calling IP Address
16	Calling User Group	User Group to which the caller belongs
17	Incoming Route	Incoming Route Number
Called Information in the SCM		
18	Called Type	Called Type - 0: Unknown - 1: User - 2: Service - 3: Trunk (End Point Type) - 4: Application Server Type
19	Called IP	Called IP Address

No.	Parameter	Description
20	Called User Group	User Group the called belongs
21	Outgoing Route	Outgoing Route Number
Disconnect Reason		
22	SIP Status Code	SIP Status Code
23	Q850 Release Cause	Q850 Release Cause
24	Internal Fail Code	SCM Internal Fail Code
System Node Information (optional-only used in Active-Active Mode)		
25	System Node ID	System Node ID - 0: system node 0 - 1: system node 1
26	Inter-Node Data	- 0: My Node's Data - 1: Peer Node's Data Peer Node's Data can be ignored
27	GMT Offset	GMT offset Time
Additional Information(optional) **Refer to [MANAGEMENT > CDR Storage Options > CDR Option > CDR Format].**		
28	Route Type	Code of route type - VOIP: FF00 - PSTN: 0000
29	Pilot Billed Number	Pilot Billed Number
30	Forward Type	Forward Call - 0: Non Forward Call - 1: Forward Call
31	Route Access Code	Route(Trunk) Access Code
32	Calling Dev Type	Calling device Type 0 = SAMSUNG_SIP 1 = SAMSUNG_SOFT 2 = SAMSUNG_MOBILE 3 = SAMSUNG_PC_ATTCON 4 = 3RD_SIP 5 = FXS 6 = FMS null = Call From route
33	Called Dev Type	Called device Type 0 = SAMSUNG_SIP 1 = SAMSUNG_SOFT 2 = SAMSUNG_MOBILE 3 = SAMSUNG_PC_ATTCON 4 = 3RD_SIP 5 = FXS 6 = FMS null = call to route

No.	Parameter	Description
34	User Digit	Digit pressed by user
35	Pick up	Pick up - 0: Normal Call - 1: Pickup Call
36	Smart Routing	- 1: internal SCM Smart Routing Call - 2: inter-SCM Smart Routing Call
37	Transferer	User Number who transfers the call
38	Service Type	Service Type which is applied to the call Refer to 'SCM Express_CDR Interoperability Guide'.
39	Auth Code	Authorization code used for outgoing call
40	Calling Reg Location	Calling Subscriber Register Location - 1: Internal Wi-Fi - 2: External Wi-Fi - 3: mVoIP
41	Called Reg Location	Called Subscriber Register Location - 1: Internal Wi-Fi - 2: External Wi-Fi - 3: mVoIP
42	Reserved field	Reserved field for CDR expansion
43	Reserved field	Reserved field for CDR expansion
44	Reserved field	Reserved field for CDR expansion
45	Reserved field	Reserved field for CDR expansion
46	Reserved field	Reserved field for CDR expansion
47	Reserved field	Reserved field for CDR expansion
48	Reserved field	Reserved field for CDR expansion
49	Reserved field	Reserved field for CDR expansion
50	Reserved field	Reserved field for CDR expansion

ANNEX A. Supported Specifications and RFC

A.1 SIP

RFC on SIP protocol and related RFCs being supported in SCM is as follows.

RFC	Description
RFC2246	The TLS Protocol Version 1.0
RFC2327	Session Description Protocol
RFC2833	RTP Payload for DTMF Digits, Telephony Tones and Telephony
RFC2976	The SIP INFO Method
RFC3261	Session Initiation Protocol
RFC3262	Reliability of Provisional Responses in SIP
RFC3263	SIP: Locating SIP Servers
RFC3264	An Offer/Answer Model with SDP
RFC3265	SIP-Specific Event Notification
RFC3311	The SIP UPDATE Method
RFC3323	A Privacy Mechanism for the SIP
RFC3325	Private Extensions to the SIP for Asserted Identity within Trusted Networks
RFC3326	The Reason Header Field for the SIP
RFC3428	SIP Extension for Instant Messaging
RFC3515	The SIP Refer Method
RFC3581	An Extension to the SIP for Symmetric Response Routing
RFC3665	SIP Basic Call Flow Examples
RFC3711	The Secure Real-time Transport Protocol (SRTP)
RFC3725	Best Current Practices for 3PCC in the SIP
RFC3824	Using E.164 numbers with the SIP
RFC3842	A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
RFC3891	The SIP Replaces Header

RFC	Description
RFC3960	Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)
RFC4028	Session Timers in SIP
RFC4492	Elliptic Curve Cryptography (ECC) Cipher Suites for Transport Layer Security (TLS)
RFC5246	The Transport Layer Security (TLS) Protocol Version 1.2
RFC5359	Session Initiation Protocol Service Examples

A.2 SNMP

Standards related to the SNMP protocol being supported in SCM are as follows.

- SNMP v1, v2

A.3 RADIUS

RFC on RADIUS protocol being supported in SCM is as follows.

- RFC 2865-Remote Authentication Dial In User Service (RADIUS)
- RFC 2866-RADIUS Accounting
- draft-ietf-radext-digest-auth-01: RADIUS Extension for Digest Authentication

A.4 LDAP

RFC that is related to the LDAP protocol supported by SCM are as follow.

- RFC2251: Lightweight Directory Access Protocol
- RFC2829: Authentication Methods for LDAP
- RFC2831: Using Digest Authentication as a SASL Mechanism

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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

H

HTTP	Hyper Text Transport Protocol
HTTP	Hyper Text Transport Protocol over Secure Sockets Layer

I

iBG	integrated Business Gateway
ICMP	Internet control message protocol
iES	integrated Ethernet Switch
IGMP	Internet Group Management Protocol
ITSP	Internet Telephony Service Provider
IVR	Interactive Voice Response

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
MPS	Media Proxy Service
MWI	Message Waiting Indication

N

NAT	Network Address Translation
NFC	Near Field Communication
NMS	Network Management System
NTP	Network Time Protocol

P

PBX	Private Branch eXchange
PMS	Property Management System
PNP	Plug and Play
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
SNTP	Simple Network Time Protocol
SRTP	Secure Real-time Transport Protocol

T

TCP	Transmission Control Protocol
TFTP	Trivial File Transfer Protocol
TLS	Transport Layer Security

U

UC	Unified Communication
UMS	Unified Messaging System
UPnP	Universal Plug and Play

V

VM	Voice Mail
VMS	Voice Mailing System
VoIP	Voice over Internet Protocol

W

WE	Wireless Enterprise
WiFi	Wireless Fidelity

SCM Express

System Description