SCM Express
Call Service Manual

INTRODUCTION

Purpose

This document describes the call service guide of the SCM.

Document Content and Organization

This manual consists of the following parts and an abbreviation as follows.

PART I. System Features

This part describes the system features.

PART II. User Features

This part describes the user features.

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.

CHECKPOINT

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

EDITION	DATE OF ISSUE	REMARKS
00	05. 2010	First Edition
01	12. 2010	Re-write based on version 3.1.x.x.
02	11. 2011	Updated for SCM version 3.2.2.x.
03	04.2012	Updated for SCM version 3.2.2.4
04	10.2012	Updated for SCM version 3.3.1

SAFETY CONCERNS

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

Symbols

	Caution Indication of a general caution.
	Restriction Indication for prohibiting an action for a product.
	Instruction Indication for commanding a specifically required action.

Warning

CAUTION

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

CAUTION

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

CAUTION

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

CAUTION

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

CAUTION

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

CAUTION

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

CAUTION

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

Contents

PART I. System Features

This part 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.

Chapter 1. System Features

System services are performed according to the data configured in the system, regardless of the user actions.

Anonymous Call Reject

The anonymous call reject service rejects anonymous incoming trunk calls without caller IDs.

An anonymous call's SIP message has anonymous@anonymous.invalid in the From header.

To use the anonymous call reject service, set [Anonymous Call Reject] to Anonymous Reject in the [Trunk Routing > Route] menu.

To use the nonumber call reject service, set [Anonymous Call Reject] to No Number Reject.

Also, To use both type rejection, set [Anonymous Call Reject] to Both Reject.

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.

You can view the maximum number of calls simultaneously supported by SCM under [Maximum Call] in the [CONFIGURATION > Miscellaneous > System Capacity] menu.

You can set the maximum usage ratio for CAC by call counts under [Local CAC Threshold] in the [MANAGEMENT > Call Admission Control > Local CAC] menu.

The number of calls allowed simultaneously by CAC by call counts is calculated by the following formula: [Maximum Call] * [Local CAC Threshold] / 100.

CAC by Location Bandwidth

The CAC by location bandwidth service restricts the calls made in excess of the bandwidth set for each location.

When calls are made between users or endpoints in different locations, the system calculates the bandwidth based on the codec used for the calls. Any calls exceeding the maximum bandwidth set for each location are restricted.

You can set the maximum bandwidth for locations under [Bandwidth] in the [CONFIGURATION > Location > Location] menu.

You can view the bandwidth required for each codec under [Codec Bandwidth] in the [CONFIGURATION > Miscellaneous > System Capacity] menu. This data cannot be changed.

You can set the maximum usage ratio for CAC by location bandwidth under [Bandwidth] in the [MANAGEMENT > Call Admission Control > Location Based CAC] menu.

CAC by Trunk Call Counts

The CAC by Trunk Call Counts service restricts calls made in excess of the maximum trunk call count for each route.

You can restrict calls based on inbound call, outbound call, and total trunk call.

You can also [Maximum call], [Maximum Inbound Call], and [Maximum Outbound Call] in the [COMFIGURATION > Trunk Routing > Route] menu. If you don't set the maximum count, calls are not 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.

You can restrict calls based on CPU and memory usage.

You can also check the CPU and memory usage level in the following three ways:

Use System Viewer in the lower-left corner of SCM Administrator.

View the real-time usage charts in the [PERFORMANCE > Main Monitor] menu.

Use the [PERFORMANCE > Server Resources > System] menu.

You can set the maximum usage ratio for CAC by system resources under [CPU Threshold for Resource Based CAC] and [Memory Threshold for Resource Based CAC] in the [MANAGEMENT > Call Admission Control > Resource Based CAC] menu.

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 based LCR, and Load Balance LCR) for each location of the calling party.

You can create route partitions in the [CONFIGURATION > Trunk Routing > Route Partition] menu.

The following items are mandatory. This menu is used for entering the detailed routes in the route partition.

Item	Description
User Group	Select a user group to which the route partition belongs.
Route Partition	Select a route partition in which to enter the data. If you want to create new partition, select < new > and enter the route partition name.
Location Select	Specify whether to select location Disable: do not select location Enable: specify a location mandatory.
Location	Select a location to set the routes.
Routing Type	Select a type of LCR for the location. - Special-Route: Select to use a special route sequence. - Normal-Route: Select to use a route sequence. - Route-Set: Select to use a route set.

When entering data into a route partition,, one of the following items is mandatory, depending on the type of the LCR selected.

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

Priority Routing

The Priority Routing feature allows automatic selection of alternative routes when the endpoint set as the default LCR is not available for call connections. Routes are assigned with priorities so that the route with the highest priority among those available is selected.

You can create route sequences in the [CONFIGURATION > Trunk Routing > Priority Routing] menu.

The following items are mandatory. This menu is used for entering the detailed routes in the route sequence.

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

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

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

Time Based Routing

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

You can create special route sequences in the [CONFIGURATION > Trunk Routing > Time Based Routing] menu.

The following items are mandatory. This menu is used for entering the detailed route sequences into the special route sequence.

Item	Description
User Group	Select a user group to which the time based routing belongs.
Time Based Routing Name	Select a time based routing where the data is entered. If you want to create new time-based routing, select < new > and enter the time based routing name.
Service Group	Select a service group to which the time based routing belongs.
Default Route Sequence	Specify the default route sequence used when selecting routes based on the settings of the special route sequence. You must select one of the route sequences created in the [CONFIGURATION > Routing > Route Sequence] menu.

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

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

Load Balance Routing

The feature allows the use of the set routes in an equally distributed manner according to the set ratio. Calls are distributed only between the routes identified as available for calls, and therefore there is no need for configuring alternative routes, as in other LCR methods.

You can create load balance routing in the [CONFIGURATION > Trunk Routing > Load Balance Routing] menu.

The following items are mandatory. This menu is used for entering the detailed routes in the route set.

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

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.

To restrict incoming calls or outgoing calls for users, set [Extension Lock] in the [CONFIGURATION > User > Single Phone User] menu.

NONE: No restriction is applied (all incoming/outgoing calls allowed).

Answering-only: All incoming calls are restricted (no incoming calls).

Dialing-only: All outgoing calls are restricted (no outgoing calls).

Both: All incoming calls and outgoing calls are restricted (no incoming/outgoing calls).

Route Lock

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

To restrict the use of a route, set [Route Lock] in the [CONFIGURATION > Trunk Routing > Route] menu.

None: All incoming and outgoing calls through the route are allowed.

Outbound Locked: Outgoing calls are restricted and only incoming calls through the route are allowed.

Inbound Locked: Incoming calls are restricted and only outgoing calls through the route are allowed.

All Locked: All incoming and outgoing calls through the route are restricted.

Restriction Policy

Call restriction policies can be applied by analyzing the calling number or called number when external calls are made to the users or external calls are made by the users through the trunk.

The call restriction tables created and configured in the menu described below can be applied to specific users, service groups, or user groups for call restriction.

If multiple call restriction policies are applied to a user, the policies are applied in the priority of user, service group, and user group.

In Tandem call case, Trunk is restricted by incoming trunk's restriction policy.

You can create call restriction tables in the [CONFIGURATION > Trunk Routing > Toll Restriction List] menu.

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

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

Item	Description
User Group	Select a user group to which the time-based restriction policy table belongs.
Name	Enter a name for the time-based restriction policy table.
Default Restriction Name	Select a call restriction table to use outside the set time periods.
Restriction List 1~3 Name	Specify up to three call restriction tables to apply according to time period.

Item	Description
Day 1~3	When time-based restriction policy tables are selected according to time, you can specify up to three types of days in which to reference the time-based restriction policy tables.
Start Time 1~3 End Time 1~3	When time-based restriction policy tables are selected according to time, you can specify the start time and end time for up to three time periods in which to reference the time-based restriction policy tables.

Number Translation

SCM can translate numbers for incoming calls,outgoing calls and local calls

Calling/Called Number Translation for Inbound Call

SCM provides two kinds of number translation service for the calling number and the called number of the inbound call. The configuration is served at the [Configuration > Trunk Routing > Route] menu.

First, you can make multiple rules to convert number to number in the [Configuration > Trunk Routing > Inbound MCN] menu. After that, select some rules for a trunk as you want to the [Inbound MCN] in the [Configuration > Trunk Routing > Route] menu. Only one matching rule can be applied call by call.

Item	Description
User Group	Select User Group which the number translation for inbound call is defined.
Name	Enter the name of Inbound MCN. When you make Inbound MCN list, you can see these Inbound MCN Names in the [Configuration > Trunk Routing > Route]. This name cannot be changed.
Number Type	Select a number type to convert. - Calling Number: Number Translation is applied to the calling number of the inbound call. - Called Number: Number Translation is applied to the called number of the inbound call.
Find Digits	Specifies the number replaced by another number from the first digit of the number of inbound call. 0-9, '*', '#', and '?' is possible for Find

Item	Description
	Digits. '#' is treated the same as the number. '*' means any number. If you want to use the Find Digits as a prefix not a full number, '?' is followed the desired pattern. - '1234': must be '1234'. - '1**4': all four digits starting with 1 and ending with 4 - '1234?': all numbers starting with '1234'. - '1**4?': all numbers prefixed by four digits starting with 1 and ending with 4, Any digit between '*'s is not allowed but duplicated '*' are supported. '?' is placed at the last digit of Find Digits. - '1*3*5': 3 is located between '*' and '*', so this pattern is not supported. Duplicated '*' without any interrupt by other digits are possible as like '**345', '1***5', and '123**' - '123**???': Multiple '?" are not supported. If the compared target number matched to several Find Digits, the most exactly matched rule is selected with number pattern and the pattern length. - The compared number is '12345' and there are two Find Digits, '12345' and '123**'. In this case, the Find Digits, '12345' is determined because it is matched with more digits. - The compared number is '12345' and there are two Find Digits, '123?' and '123**'. In this case, thd Find Digits, '123**' is determined because it is matched with the length including pattern. The compared number is '12345' and there are two Find Digits, '123?' and '12345?'. In this case, the Find Digits, '12345?' because its matched digit length is longer. -
Modified Digits	Specifies the converted number by the Find Digits. 0-9, '*', '#', and '?' can be entered to the Modified Digits. The count of '*' in the Find Digits and that of the Modified Digits must be same. If you use '?' in the Modified Digits, the digits corresponding to the '?' in the Find Digits are reused at the Modified Digits. - The compared number is '1234567'. The matched Find Digits is '123***7', and the Modified Digits is '00***111'. In this case, the number is converted to '00456111'.

Item	Description
	- The compared number is '1234567'. The matched Find Digits is '123?' and the Modified Digits is '00?'. In this case, the number is converted to '004567'. - The compared number is '1234567'. The matched Find Digits is '123?' and the Modified Digits is '00'. In this case, the number is converted to '00' -

After the number translation by Inbound MCN, simple delete and insert to a number is supported trunk by trunk for inbound call. If delete digits and insert digits are configured both, the delete procedure is precedence over insert procedure.

The number translation mentioned above is set in the [Configuration > Trunk Routing > Route].

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

Calling/Called Number Translation for Outbound Call

SCM provides two kinds of number translation service for the calling number and the called number of the outbound call. The configuration is served at the [Configuration > Trunk Routing > Priority Routing] menu or [Configuration > Trunk Routing > Load Balance Routing] menu. According to routing path, different number translation can be applied even if calls direct to a same trunk route.

First, you can make multiple rules to convert number to number for outbound call in the [Configuration > Trunk Routing > Outbound MCN] menu. After that, select some rules for a trunk as you want to the [Outbound MCN] in the [Configuration > Trunk Routing > Priority Routing] menu or [Configuration > Trunk Routing > Load Balance Routing] menu. Only one matching rule can be applied call by call.

Item	Description
User Group	Select User Group which the number translation for outbound call is defined.
Name	Enter the name of Outbound MCN. When you make Outbound MCN list, you can see these Outbound MCN Names in the [Configuration > Trunk Routing > Priority Routing & Load Balance Routing] This name cannot be changed.
Number Type	Select a number type to convert. - Calling Number: Number Translation is applied to the calling number of the outbound call. - Called Number: Number Translation is applied to the called number of the outbound call.
Find Digits	Specifies the number replaced by another number from the first digit of the number of outbound call. 0-9, '*', '#', and '?' is possible for Find Digits. '#' is treated the same as the number. '*' means any number. If you want to use the Find Digits as a prefix not a full number, '?' is followed the desired pattern. - '1234': must be '1234'. - '1**4': all four digits starting with 1 and ending with 4. - '1234?': all numbers starting with '1234'. - '1**4?': all numbers prefixed by four digits starting with 1 and ending with 4. Any digit between '*'s is not allowed but duplicated '*' are supported. '?' is placed at the last digit of Find Digits. - '1*3*5': 3 is located between '*' and '*', so this pattern is not supported. Duplicated '*' without any interrupt by other digits are possible as like '**345', '1***5', and '123**' - -'123**???': Multiple '?" are not supported. If the compared target number matched to several Find Digits, the most exactly matched rule is selected with number pattern and the pattern length. - The compared number is '12345' and there are two Find Digits, '12345' and '123**'. In this case, the Find Digits, '12345' is determined because it is matched with more

Item	Description
	digits. - The compared number is ‘12345’ and there are two Find Digits, ‘123?’ and ‘123**’. In this case, thd Find Digits, ‘123**’ is determined because it is matched with the length including pattern. - The compared number is ‘12345’ and there are two Find Digits, ‘123?’ and ‘12345?’. In this case, the Find Digits, ‘12345?’ because its matched digit length is longer. -
Modified Digits	Specifies the converted number by the Find Digits. 0-9, ‘*’, ‘#’, and ‘?’ can be entered to the Modified Digits. The count of ‘*’ in the Find Digits and that of the Modified Digits must be same. If you use ‘?’ in the Modified Digits, the digits corresponding to the ‘?’ in the Find Digits are reused at the Modified Digits. - The compared number is ‘1234567’. The matched Find Digits is ‘123***7’, and the Modified Digits is ‘00***111’. In this case, the number is converted to ‘00456111’. - The compared number is ‘1234567’. The matched Find Digits is ‘123?’ and the Modified Digits is ’00?’. In this case, the number is converted to ‘004567’. - The compared number is ‘1234567’. The matched Find Digits is ‘123?’ and the Modified Digits is ‘00’. In this case, the number is converted to ‘00’ -

After the number translation by Outbound MCN, simple delete and insert to a number is supported according to the routing path for outbound call. If delete digits and insert digits are configured both, the delete procedure is precedence over insert procedure.

The number translation mentioned above is set in the [Configuration > Trunk Routing > Priority Routing] or [Configuration > Trunk Routing > Load Balance Routing].

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

항목	설명
Length	calling number for outbound call.
Outbound CLI Insert Digits	Specifies the digits to insert from the first position of the calling number for outbound call.

Local Based Number Translation

SCM provides number translation service for the called number of the local call.

This feature is activated in same location and user group.

It can provide when using call-forward, blind-transfer and individual-speed-dial service.

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

Item	Description
Name	Enter the name of Local Based MCN. When you make Local Based MCN list, you can see these Local Based MCN Names in the [Configuration > Location > Local Based MCN]. This name cannot be changed.
Number Type	Select a number type to convert. Only Called Number is allowed. - Called Number: Number Translation is applied to the called number of the local call.
Find Digits	Specifies the number replaced by another number from the first digit of the number of inbound call. 0-9, '*', '#', and '?' is possible for Find Digits. '#' is treated the same as the number. '*' means any number. If you want to use the Find Digits as a prefix not a full number, '?' is followed the desired pattern. - '1234': must be '1234'. - '1**4': all four digits starting with 1 and ending with 4 - '1234?': all numbers starting with '1234'. - '1**4?': all numbers prefixed by four digits starting with 1 and ending with 4,

Item	Description
	Any digit between '*'s is not allowed but duplicated '*' are supported. '?' is placed at the last digit of Find Digits. - '1*3*5': 3 is located between '*' and '*', so this pattern is not supported. Duplicated '*' without any interrupt by other digits are possible as like '**345', '1***5', and '123**' - '123**???': Multiple '?" are not supported. If the compared target number matched to several Find Digits, the most exactly matched rule is selected with number pattern and the pattern length. - The compared number is '12345' and there are two Find Digits, '12345' and '123**'. In this case, the Find Digits, '12345' is determined because it is matched with more digits. - The compared number is '12345' and there are two Find Digits, '123?' and '123**'. In this case, thd Find Digits, '123**' is determined because it is matched with the length including pattern. The compared number is '12345' and there are two Find Digits, '123?' and '12345?'. In this case, the Find Digits, '12345?' because its matched digit length is longer. -
Modified Digits	Specifies the converted number by the Find Digits. 0-9, '*', '#', and '?' can be entered to the Modified Digits. The count of '*' in the Find Digits and that of the Modified Digits must be same. If you use '?' in the Modified Digits, the digits corresponding to the '?' in the Find Digits are reused at the Modified Digits. - The compared number is '1234567'. The matched Find Digits is '123***7', and the Modified Digits is '00***111'. In this case, the number is converted to '00456111'. - The compared number is '1234567'. The matched Find Digits is '123?' and the Modified Digits is '00?'. In this case, the number is converted to '004567'. - The compared number is '1234567'. The matched Find Digits is '123?' and the Modified Digits is '00'. In this case, the number is converted to '00'

Call Button

The call button feature allows directing multiple calls to the one phone number when the user's number is a single device and the call waiting service is in use.

If you are using a phone with programmable buttons, such as a Samsung phone, you can assign up to eight call buttons. If there are eight call buttons, the phone can control up to eight calls simultaneously. If there is no call button on the phone, it is treated as having two call buttons.

To create call buttons on the phone, set [Key] to CALL in the [CONFIGURATION > User > Phone Key Programming] menu.

Call Monitoring

SCM has the ability to route a media to the particular IP address to record the signaling and media easily.

When users activate the call monitoring service, it redirect the media from recording target to the specific IP address and port of media proxy inside the SCM Express.

It makes easy for eavesdropping equipment to collect the voice call associated with the particular IP address and ports under control of SCM.

SCM supports recording up to 100 subscribers simultaneously.

In case that the resource of media proxy is not enough, Fault event occurred to the Administrator, and the call will be failed.

The signaling packets are delivered from SCM Express to the Monitoring Equipment with specific interfaces.

In order to capture the media packets, you should place an L2 switch which has mirroring capability between the phones and the server. So the mirroring equipment can capture the packets from L2 switches and analyzing the media with the signaling packets which was taken from different interface.

You can configure call monitoring in the [CONFIGURATION > Miscellaneous > System Options] menu.

Item	Description
Call Monitoring Interworking Level	Specify the level of Interworking with Call monitoring Equipments. - Disable: SCME Administrator control Call Monitoring Option for User. - User Configure: Call Monitoring Server control Call Monitoring Option for User. - SIGNALING MESSAGE: Call Monitoring Server control Call Monitoring Option for User, and SIP Message is saved in SCM.

Item	Description
Call Monitoring Server IP Address	Enter Call Monitoring Server's IP Address.
Call Monitoring Server use TLS	Select Use TLS Connection Between SCM and Call Recording Server.

Administrator enables [Call Monitoring] option in the [Configuration > User > Single Phone User].

When call comes in to the user, SCME make a voice path to Media Proxy inside SCME, SCME send signaling messages to Call Recording Server, so Call Recording Server can mirroring the media packets.

CLI Number for Internal Call

Basically, a user makes a call by dialing Extension number within same User Group.

Extension Number should be assigned in unique within a User Group and it used as CLI when make a call and Destination number when finding a user.

But, exceptional cases like below, different number may be used instead of Extension Number.

- In case that a user makes a call to the users in a different User Group by dialing just Extension number, the call will be failed. Extension Number is a reachable number within same User Group.

  When a user makes a call to the users in a different User Group, the user should dial Extension Number followed by User Group Code. When a user make a call among different User Groups, both Calling and Called Number will be displayed as [User Group Code + Extension Number].

- Within same user group, extension number is used normally, but if there is [Service Group Code] in a User group, the users in same Service Group can make a call each other just using [Service Group Local Number] that is called "Station Number". In other words, Extension Number includes [Service Group Code] and [Service Group Local Number].

  To use [Service Group Local Number], create a service group and specify [Service Group Code] and then assign users to the service group. Only the Extension Numbers the prefix is starting with Service Group Code can be assigned to the service group.

  [Service Group Local Number] field in each User menu displays the digits which exclude the Service Group Code from Extension Number automatically. If there is Service Group Code, this field displays nothing. The Users within same service group can make a call using Extension number or Service Group Local Number.

- To display CLI with not Extension Number but [Service Group Local Number], operator should configure [Service Group Local CLI Number] to "Station Number" in [CONFIGURATION > User > Single Phone User] or [CONFIGURATION > User > Multi-Phone User].

If a user does not want to use Extension Number as CLI, use [Send Extension Number] instead.

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

CLI Number for Outbound Call

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

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

The default CLI number is the extension number without any configuration as describes above.

The CLI number for outbound call, by default, is applied in the order mentioned above. Depending on your needs, you can designate the CLI number type for a specific trunk. It is served by [Forced Send CLI Number] in the [Configuration > Trunk Routing > Route] menu. If you set the [Forced Send CLI Number] to none, outbound CLI number has priority according to the order mentioned above.

CLI Name for Outbound Call

It is a virtual calling name for outbound call at particular use. It is mainly used to insert a number to the 'Display Name' in the From header of the SIP message by the request of SIP ISP. There are two cases as described below.

Each user has [Send CLI Name] in the [Configuration > User > Single Phone User] or [Configuration > User > Multi-Phone User] menu, which can be used for 'Display Name' for a specific trunk. If [Send CLI Name for User] in the [Configuration > Trunk Routing > Route] menu set to Send CLI Name for the trunk, the [Send CLI Name] of each user is used for display name, or [Extension Name] in the [Configuration > User > Single Phone User] and [Configuration > User > Multi-Phone User] menu is used by default.

Tandem Call means the outbound call which is originating from a trunk. To configure the 'Display Name' of the tandem call, the [Send CLI Name for Inbound Call] is provided. If the [Send CLI Name for Inbound Call] in the [Configuration > Trunk Routing > Route] menu set to Receive CLI Number, the original received calling number is copied to the 'Display Name' regardless of the number translation.

Internal CLI Name

It is a virtual calling name for inbound call at particular use. It is mainly used to insert a name to verificate which GW's call.

This name display only in subscriber and service group. This name does not display in tandem call.

To use the Internal CLI Name, specify in the [CONFIGURATION > Trunk Routing > Route > Send CLI Name for Internal Call] menu.

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.

To use the CLIR feature, click [Act] and enable [Caller ID Block] in the [CONFIGURATION > Service > Feature Service > Service Activation] menu.

If a user set with CLIR calls another user set with CLIP, the calling user's information is restricted to the called user, as CLIR has precedence over CLIP.

CLI Routing

The CLI routing feature allows special processing of incoming trunk calls according to the caller number.

When entering a calling number, you can use wild cards (entered by *) to enter multiple numbers at a time.

Incoming trunk calls with caller numbers only and without caller names can be supported by caller name.

Incoming calls from specified callers can be rejected.

Incoming calls from specified callers can be assigned called numbers, regardless of the called DID numbers. Called numbers can also be assigned by time period. Time periods are defined by ring plans. For more information, see the section on "Ring Plans."

You can configure CLI routing using the [CONFIGURATION > Trunk Routing > CLI Routing] menu. The following items are specified.

Item	Description
User Group	Select a user group for which the CLI routing is defined.
CLI Number	Enter a caller number for incoming calls given special treatment. This is used as an identifier in the CLI routing table and cannot be changed. If necessary, you must delete it and enter it again.
Number to Name	You can enter the caller's name for the selected caller number and service the calls by the name.
Call Reject	Specify whether the incoming calls with the selected caller number is rejected.
ACD Queuing Level	Specify a level of priority for a call queues in ACD. You can select between Level 0 and Level 9. If higher the number, higher the priority and a waiting time to connect an agent is shorter.
Default Destination	You can specify a called number for incoming calls from the selected caller number, regardless of the DID number if the call is

Item	Description
	received outside the time periods defined by ring plans 1 through 15 or if the called number is not specified in the ring plan.
RP1~RP15 Destination	You can specify a called number for incoming calls from the selected caller number, regardless of the DID number if the call is received within the time periods defined by ring plans 1 through 15.
Logging	Specify whether to logging a SPAM Call. It logs with three cases like normal case, with announcement case, and without announcements case.

It is often necessary to register a lot of CLI routing information at once. Using SCM Administrator's batch feature, you can prepare the CLI routing information offline in an Excel spreadsheet and batch register the information with SCM.

You can batch register the CLI routing information prepared in an Excel spreadsheet using the [Tool > Customer Data Import/Export > CLI Routing] menu in the upper-left corner of SCM Administrator.

For more information on the Customer Data Import/Export, see the "Customer Data Import/Export" section of "2.3Batch Creating All Users"

Direct Inward Dialing (DID) Routing

The Direct Inward Dialing (DID) routing feature allows incoming trunk calls to be directed to different called numbers according to the DID number.

When entering a DID number, you can use wild cards (entered by *) to represents the multiple numbers at a time.

The system also performs translation of the DID number, the translated DID number can be used as the called number.

The called number according to the DID number can be assigned with the user number, the hunt group number, the ACD group number, and various feature codes, including the VMS access code and access code + external number. They can also be assigned by time period. Time periods are defined by ring plans. For more information, see the section on "Ring Plans."

You can configure DID routing using the [CONFIGURATION > Routing > DID Routing] menu. The following items are specified.

Item	Description
User Group	Select a user group to which the calls are directed.
DID Number	Enter a DID number for incoming trunk calls. This is used as an identifier in the DID routing table and cannot be changed. If necessary, you must delete it and enter it again.

Item	Description
DID Name	Specify a name for the DID number. The DID name makes it easy to identify the number dialed by the caller.
Default Destination	Specify a called number to which the incoming calls with the selected DID number is directed when calls are received outside the time periods defined in ring plans 1 through 15 or if the called number is not specified in the ring plan. If the called number is set to "B," a translated DID number is used as the called number. In the case of “E” for the called number, tandem call is not allowed. If the call is not termination at SCM, the call is rejected with invalid number response. However, the tandem calls by Call Forward or Transfer is not restricted.
Delete Length	DID number can be translated to be used as the called number. Here, you can specify the number of digits to delete from the first part of the DID number.
Insert Digit	DID number can be translated to be used as the called number. Here, you can specify the digits to insert into the first part of the DID number.
ACD Queuing Level	Specify a level of priority for a call queues in ACD. You can select between Level 0 and Level 9. If higher the number, higher the priority and a waiting time to connect an agent is shorter.
MOH ID	A specific on-hold tone can be played when incoming calls with the selected DID number are put on hold. Here, you can specify the ID of the MOH file to play for each DID number.
RP1~RP15 Destination	You can specify a called number to which the incoming calls with the selected DID number is directed when the calls are received within the time periods defined by ring plans 1 through 15. If the called number is set to "B," a translated DID number is used as the called number.

It is often necessary to register a lot of DID routing information at once. Using SCM Administrator’s batch feature, you can prepare the DID routing information offline in an Excel spreadsheet and batch register the information with SCM.

You can batch register the DID routing information prepared in an Excel spreadsheet using the [Tool > Customer Data Import/Export > CLI Routing] menu in the upper-left corner of SCM Administrator.

For more information on the Customer Data Import/Export, see the “Customer Data Import/Export” section.

Directory Service

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

You can search directory by name or number.

When you try to search a user by number, User’s extension number, Hunt Group number should be used.

When you try to search a user by name, User name, Hunt Group name and System Speed dial name should be used.

When you search by number, it required at least 3 digits, search by name, it requires at least two characters.

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

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

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

Item	**Description**
User Group	It displays the User Group of a user. It will be displayed by default.
Display Name	It displays the Extension name of a user. It will be displayed by default.

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

Direct Trunk Selection

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

To use DTS feature, the following configuration is required.

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

How to use the DTS feature is described below.

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

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

FMS (Fixed Mobile Substitution)

Fixed Mobile Substitution(FMS) is a Zone service for a specific trunk. FMS users which have a virtual number are mobile users in the specific zone. Although a FMS user is not registered, the trunk calls from or to FMS users are treated as if internal calls and a virtual extension number is used for CLI by default. Users can make a call with the virtual extension number or the mobile number to the FMS user.

How to work FMS call is as follows.

- FMS user receives a call: When SCM receives a call with an extension number of FMS user from a trunk or a normal user of SCM, SCM converts the called number to mobile phone number of FMS user and makes a call to FMS trunk which supports No.7.
- FMS user makes a call: When SCM receives a call from FMS trunk which supports No.7, SCM treats the call as if an internal call if the send CLI is an extension number of FMS user.

The configuration is required for FMS feature as follows.

- Make a Route which is connected to FMS zone in the [CONFIGURATION > Trunk Routing > Route] menu. And then, change [FMS Mode] to Enable.
- SCM is connected to FMS zone of mobile network by a FMS trunk which supports No.7. A mobile carrier issues FMS Zone ID for each FMS zone. To make a call to FMS zone, SCM has to use the FMS Zone ID.

  If Access Code to FMS trunk is 90 and assigned FMS Zone ID is 1234, SCM has to convert 90 to 901234. For this, Number Translation is applied to the outgoing calls to the FMS trunk. For more detail of Number Translation, refer Number Translation feature.
- Make a FMS zone in the [CONFIGURATION > Wireless Enterprise > FMS Zone] menu. Multiple FMS zone can be made and separate trunk has to be assigned to each FMS zone. Trunk Routing configuration is a default process for trunk calls.

- Make s FMS user in the [CONFIGURATION > User > Single Phone User] menu. FMS user needs [Mobile Phone Number] and [FMS zone name] which the FMS user is assigned to. [Phone Type] of a FMS user is FMS-Phone.
- For additional configuration of the FMS gateway, usually OS7500, refer to the manual for OS7500.

There is some limitation for a call from a FMS user.

- Some user services are restricted for a FMS user because FMS user is a virtual user which is not registered. Only [User Account Code] and [Send CLI Number] can be applied. Do not change other fields discretionally.
- When SCM receives a FMS call from a FMS trunk, [Inbound MCN] is not applied because a FMS call is treated as if an internal call. But [Outbound MCN] for the outgoing call to a FMS trunk can be supported.

At the [COFNIGURATION > Wireless Enterprise > FMS Zone], you can configure some options about FMS and FMS Smart Routing. Refer the table below.

Item	Description
User Group	Choose a user group which provides FMS zone service.
FMS Zone Name	Enter FMS Zone name.
FMS Trunk	Choose a trunk which is connected to FMS Zone.
FMS Default Access Code	Choose Default Access Code for FMS calls in the Access Code list. SCM converts FMS extension to 'FMS Default Access Code + Mobile number of the FMS user'.
Reroute FMS Call to Trunk	Set to Enable when you want to re-route the FMS call to a normal trunk if a designated response is received. For more details, refer to FMS Smart Routing.
Reroute Trunk call to FMS	Set to Enable when you want to use FMS call as a default. If a normal trunk call to a mobile number of a user re-route to FMS trunk because of account. For more details, refer to FMS Smart Routing.
Reroute Announcement	Set to Enable if you want to play an announcement during re-routing a FMS call to a normal trunk. For more details, refer to FMS Smart Routing.

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.

The following configurations are required for smart routing from FMS zone to normal trunk.

- First of all, [Alternative Route N] is required in the [CONFIGURATION > Trunk Routing > Priority Routing] for the FMS routing. The Alternative Route N is a normal trunk to reroute FMS call.
- [Reroute FMS call to Trunk] in the [CONFIGURATION > Wireless Enterprise > FMS Zone] has to set to Enable.
- Enter 480 or a designated response in the [Allow Reroute ReasonCode] of [CONFIGURATION > Trunk Routing > Route].
- SCM supports an announcement for smart routing from FMS zone to normal trunk. [Call Forward Announcement Iteration] in the [CONFIGURATION > Trunk Routing > Route] change to a value large than 1. You can check or change the announcement which ID is 1218(Announcement for Free zone to Trunk forwarding) in the [CONFIGURATION > Announcement > Service Announcement].

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.

- For smart routing from normal trunk to FMS zone, [Reroute Trunk call to FMS] has to set to Enable in the [CONFIGURATION > Wireless Enterprise > FMS Zone].

Emergency Group

In case of emergency, if user makes a call with pre-configured emergency group number, system automatically make a call to the members with emergency type access code.

When a member of emergency group response the call. The system automatically calls to the manager of the emergency group. The manager can only hear.

To use Emergency Group feature, ‘Emergency Type Access Code’ is required in the [Configuration > Trunk Routing > Access Code] menu.

And the following Items are required in the [Configuration > Service > Group Service > Emergency Group] menu.

Item	Description
User Group	Select a user group for which the emergency group will be configured

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

History Log

SCM provides history logging capability for the events like a SPAM call, incoming call, paging on answer call, wakeup call, feature set, registration fail etc.

You can configure Logging Service, using the [Configuration > Miscellaneous > System Capacity] and [Configuration > User Group > Change User Group > Detailed Event Logging Option] Menu.

Administrator can review logs at [Performance > Detailed History] menu..

The following items are information about [Configuration > Miscellaneous > System Capacity].

Item	Description
SPAM Call Log Lifetime (day)	Specify how many days do you want to keep the SPAM calls logs..
SPAM Call Log Record	Specify the maximum counts of SPAM call logs.
SPAM Call Log Target	Specify how many users do you want to log for SPAM call.
Incoming Call Log Lifetime (day)	Specify how many days do you want to keep the Incoming call.
Incoming Call Log Record	Specify the maximum counts of the Incoming call logs.
Incoming Call Log Target	Specify how many users do you want to log for Incoming call.
Paging Call Log Lifetime (day)	Specify how many days do you want to keep the Paging calls.
Paging Call Log Record	Specify the maximum counts of the Paging call logs.
Wakeup Call Log Lifetime (day)	Specify how many days do you want to keep the Wakeup call logs.

Item	Description
Wakeup Call Log Record	Specify the maximum counts of the Wakeup call logs.
Feature Set Log Lifetime (day)	Specify how many days do you want to keep the Feature Set calls logs.
Feature Set Log Record	Specify the maximum counts of the Feature Set call logs.
Register Fail Log Lifetime (day)	Specify how many days do you want to keep the Register Fail logs.
Register Fail Log Record	Specify the maximum counts of the Register Fail logs.
ACL Block Log Lifetime (day)	Specify how many days do you want to keep the ACL Block logs.
ACL Block Log Record	Specify the maximum counts of the ACL Block logs.

Follows are detailed description of [Configuration > User Group > Change User Group > Detailed Event Logging Option] menus.

All options should be configured, and it was disabled by default.

Item	Description
User Group	It displays the user group.
SPAM Call Logging	Specify whether to logging the SPAM calls. It logs a call from external only.
Incoming Call Logging	- Specify whether to logging the incoming calls from extension or trunk.Disable: it does not logging. - External Call: It logging for a call from external - Internal Call: It logging for a call from internal. - All Calls: It logging for a call from internal and external.
Paging On Answer Call Logging	Specify whether to logging the Paging On Answer calls.
Wakeup Call Logging	Specify whether to logging the Wakeup calls.
Wakeup Set Logging	Specify whether to logging the Wakeup Setting calls.
Call Block Feature Logging	Specify whether to logging the calls blocked by Cal Block feature.
ACL Block Logging	Specify whether to logging the calls blocked by ACL feature. This option is not available.
Register Fail Logging	Specify whether to logging for Register Failure.

Item	Description
	- Disable: It does not logging. - Phone Only: It logging the Register failure for Phone only. - All Devices: It logging the Register failure for all devices. -

SPAM Call History

SPAM Call History feature provides a user to logging the call with specific SPAM number from external judged as SAPM call.

You should enable the [SPAM Call Logging] option of [Configuration > User Group > Change User Group > Detailed Event Logging Option] and [Configuration > Trunk Routing > CLI Routing] menu.

You can review the history record of SPAM call at [Performance > Detailed Event History > Incoming Call History] menu.

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

Incoming Call History

Incoming Call History feature provides a user to logging the call from external or calls from internal.

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

Item	Description
Incoming Call Logging	Specify whether to logging an incoming call. - Disable: it does not logging. - No Answer Only: it logging the calls not answered only - All Call: It logging the calls with not answered, answered and abandoned by caller.

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

Item	Description
Date	It displays the time when Incoming Call event occurred.
Call Type	It displays the type of a call.. - Abandon: A caller cancelled the call. - Answer: the call was connected. - No Answer: The call was not answered by user.
User Group	It displays the User Group name of a user when the event occurred.
Extension	It displays the Extension number of a user when the event occurred.
CLI Number	It displays the CLI Number of the call from trunk when the event occurred.
DID Number	It displays the DID Number of the call from trunk when the event occurred.
Duration	It displays talk duration time. If Not Answer case, Ring time is displayed.

Paging On Answer Call History

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

Item	Description
Date	It display the time when Paging On Answer Call event was occurred.
Call Type	It displays five call Types for Paging On Answer Call. - COMMAND: The caller of the Paging On Answer call. - NOANSWER: In case that member has no answered.

Item	Description
	- ANSWER: In case that member was answered. BUSY: In case that member was busy. UNREACHABLE: In case that member was unreachable.
User Group	It displays the User Group name of a user when the event occurred.
Extension	It displays the Extension number of a user when the event occurred.
Group Number	It displays the Group Number of the Paging On Answer Group.
Ring Number	It display the members of the group who is in ringing state.

Wake-up Call History

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

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

Feature Set History

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

Item	Description
Date	It displays the time when Feature Set event occurred.
Feature	It displays the Types of Features.
User Group	It displays the User Group name of a user when the event occurred.
Event	It displays the Types of events.

Item	Description
Method	It displays who generate this event.
Index	It displays the index of Wake-up call.
Destination	It displays the destination number for the services like Forward All, Follow Me, and Remote Office.
Login ID	It displays the Login ID of a user for this event.
Login IP	It displays the Login IP Address of a user for this event.

Register Fail History

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

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

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.

To use the MPS, set [System Under NAT] to Enable and enter the public IP address allocated to the NAT system in [System Public IP Address] in the [CONFIGURATION > System Configuration > Dynamic Configuration] menu.

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.

For phones connected on the private IP network inside SCM's NAT but on a different subnet with routing, you can add the free zone and subnet mask in the [CONFIGURATION > Location > MPS Freezone] menu so that the voice and video data can be exchanged directly without using the MPS.

Example) Entering "192.168.10.255" into [Freezone] and "255.255.255.0" in [Subnet Mask] puts the users using the "192.168.10.x" subnet in the MPS-free zone. Calls made between a phone on the same subnet as the system and a phone in the MPS-free zone do not go through the MPS.

Lastly, for external connection endpoints, you can set the [NAT Traversal] option to Enable in the [CONFIGURATION > Routing > Endpoint Advanced Options] menu to provide the services through the MPS.

Hotel Service

There are additional menus for hotel services. Refer to the “SCM Express Hotel Service Guide” for detailed information.

PMS Interface

SCM should interworks with PMS for hotel services. Refer to the the “SCM Express Hotel Service Guide” for detailed information

Hot Desking

The hot desking feature allows a user to log in from a phone shared by multiple users.

The user can use a phone in the logged out status to enter his/her ID and password to log in and use the phone as his/her own phone until logged out. The user can log out when using Samsung phones' menu. If the user leaves the phone without logging it out, it is automatically logged out after a set period of time, preventing unauthorized users from using the phone. The default login expiration time is 8 hours.

If the user is already logged in through a phone but requests for login again through another phone using the same user ID, the new login request is processed by logging the previous phone out.

To use the hot desking feature, the following items must be configured.

In the [CONFIGURATION > User Group > Change User Group > Information] menu, hot desking must be enabled in the [Service Permission] section.

In the [CONFIGURATION > Service > Feature Service > Service Activation] menu, hot desking service must be enabled for the extension number used as a hot desking phone. Here, you can enter the login expiration time in [Hot Desk Expire Time (hour)].

Hot Line and Warm Line

The hot line feature allows automatic connection to a specified number when the handset of the selected phone is lifted.

If the call is connected without delay when the handset is lifted, it is called a hot line. If the call is automatically connected when the handset is lifted but no number is dialed for a set period of time, it is called a warm line.

To use the hot line feature, the following items must be configured.

In the [CONFIGURATION > User Group > Change User Group > Information] menu, hot line must be enabled in the [Service Permission] section.

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

Hunt Group

The hunt group service directs calls received by the pilot number of a hunt group appropriately within the hunt group using various routing methods.

When calls are received for a hunt group, the available member list excludes members unable to receive calls because they are unavailable, are subject to incoming call restriction policies, have logged out of the system, have user information that is locked out, or do not have their phones connected.

Called parties for calls received for a hunt group are determined in the following four ways.

Sequential

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

Circular

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

Parallel or Broadcast

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

Random

The call is randomly directed to one member in the hunt group at random.

When a call is received for the hunt group, the call is directed to a selected member. But if the call is not answered for a specified period of time, the call is canceled and directed to the next member. To set the time required for directing the call to the next member, set [Service No Answer Time (sec)] in the [CONFIGURATION > User Group > Change User Group > Timers] menu.

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

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

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

Item	Description
All Busy/Unavailable Destination	You can specify an alternative called number to use when the call cannot be directed to any member in the hunt group.
No Answer Destination	You can specify an alternative called number to use when the call is not answered by any member in the hunt group for a specified period of time. This works in the same way as no answer call

Item	Description
	forwarding.
No Answer Time(sec)	Specify a period of time during which the call must be answered by any member in the hunt group. If the call is not answered during this period, it is directed to the specified alternative called number.
External Ringback Tone Use	-None: Do not use Ringback tone -Internal: Use Ringback tone only for the originating call from user -External: Use Ringback tone only for the inbound call from the trunk.. -Both: User Ringback tone for both, the originating call from user and the inbound call from the trunk
External Ringback Tone Server	Select Application Server for Ringback tone. To select Ringback tone server, make External Ringback tone Server at [CONFIGURATION > Application > Other Application Server] menu first.

Hunt Group Login/Logout

You can temporarily prevent a hunt group member from receiving incoming calls for the hunt group. If a member logs out of his or her hunt group, the member is excluded from the available member list, and incoming hunt group calls are not directed to the member. If the member logs in again, he or she can receive incoming hunt group calls normally.

To use the hunt group login/logout feature, you must set the "Station Group - In / Station Group - Out" feature code in the [CONFIGURATION > Service > Feature Service > Feature Code] menu.

If the user dials feature code + hunt group number to set Station Group - In / Station Group - Out, the member is logged in to or out of the selected hunt group only. If the user only dials the feature code without a hunt group number to set Station Group - In / Station Group - Out, the member is logged in to or out of all his or her hunt groups.

Location Codec Negotiation

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.

Since codec negotiation only takes place between Internet phones, codec negotiation priority settings cannot be forced on calling or called phones which do not support the selected codec.

You can set the preferred codecs for locations using the [CONFIGURATION > Location > Location] menu.

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

Location Codec

You can designate Calling Location, Called Location, Video codec, Forced Audio Codec and Audio Codec at Location codec. You can specify the default audio codec, the default video codec, and the announcement codec for each location in order to change the priority of the codec list between the calling phone and the called phone. When you select forced Audio Codec, SCM sends only audio codec in Forced Audio Codec field to the destination.

Since codec negotiation only takes place between Internet phones, codec negotiation priority settings cannot be forced on calling or called phones which do not support the selected codec.

You can set the preferred codecs for locations using the [CONFIGURATION > Location > Location Codec] menu.

Item	Description
Calling Location	Select calling location.

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

Audio Codec

When 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.

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.

Default System Codec

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

Multiple Appearance

There are the following two multiple appearance services: assigning one phone number to multiple phones, or assigning multiple phone numbers to one phone.

These two services can be set independently or collectively.

You can configure multi-device and multi-number using the [CONFIGURATION > User > Multi-Extension Phone] menu and the [CONFIGURATION > User > Multi-Phone User] menu.

Multi-Device

The multi-device service assigns one user (phone number) to multiple devices (phones).

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 8 phone numbers can be assigned per phone.

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.

To use music on hold (MOH), administrator should set [MOH Enable] to ‘Enable‘and specify the ID of the sound source file in [MOH ID] in the [CONFIGURATION > User Group > Change User Group > Information] menu.

It is necessary to enable or disable MOH for each user group because there is limited number of channels for 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.

To support a specific music source per each subscribers, set [MOH Announcement ID] in the [User > Single Phone User] or [User > Multi-Phone User] menu.

The MOH ID of subscriber has priority over that of user group.

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.

Missed Call Display by Multi-Device

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

Missed Call Display by Hunt Answer

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

Missed Call Display by Multiring Answer

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

Missed Call Display for Pickup

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

Operator Group

An operator group is a special hunt group made up of members who act as operators. Different hunt groups can be specified as operator groups by time periods. In general, a user selected as a member of an operator group uses the phone in parallel to a PC application or uses a PC-based soft phone.

You can configure operator groups using the [CONFIGURATION > Service > Group Service > Operator Group] menu. When specifying an operator group, you can select one of the hunt groups configured in the [CONFIGURATION > Service > Group Service > Hunt Group] menu.

Item	Description
User Group	Select a user group for which the operator group will be defined.
Access Number	Enter a pilot number used for calling the operator group.
Operator Name	Specify a name for the operator group. The operator group name is useful for identifying the purpose of the operator group.
Default Operator Group	Specify a hunt group to use as the operator group when ring plans 1 through 15 are not applied.
RP1~10 Operator Group	Specify a hunt group to use as the operator group when ring plans 1 through 15 are applied.

Operator Recall

When a call is transferred or parked and then directed back to the original called party but the connection was not established, the operator recall service directs the call back to the operator group.

Calls are redirected to the operator group in the following cases:

Reconnection failure after call transfer failure: When call transfer fails for an incoming call for an extension number, the call is redirected to the extension number. Here, if the call is not answered by the extension number, it is redirected to the operator group.

Reconnection failure after call park: When an incoming call for an extension number is put on hold (call park) and the call is not answered for a set period of time, the call is redirected to the extension number. Here, if the call is not answered by the extension number, it is redirected to the operator group.

To use operator recall, set [Operator Recall] to Enable in the [CONFIGURATION > User Group > Change User Group > Options] menu.

To use operator recall, the following time settings are required in the [CONFIGURATION > User Group > Change User Group > Timers] menu.

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

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

Operator Transfer Recall

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

Calls are redirected to the operator in the following cases:

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

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

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 16 ring plans, including the 15 ring plans (ring plans 1 through 15) which can be assigned their own dates, days of the week and time of the day, and the default ring plan which is used when none of the former 15 ring plans is applied.

Calendar Exceptions

When User Group is created, calendars of twenty years are created by default. Specify holidays for the year and special days for the site in the calendars.

You can make detailed calendar settings using the [CONFIGURATION > Time Schedule > Calendar Exceptions] menu.

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

Ring Plan Schedule

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

Holiday Ring Plan Schedule

You can create ring plan schedule for user-defined days such as Holiday1, Holiday 2, User1, and User2.

You can enter detailed data for holiday ring plan schedules using the [CONFIGURATION > Time Schedule > Holiday Ring Plan Schedule] menu.

Item	Description
User Group	Select a user group for which the ring plan schedule will be configured.
Ring Plan	Specify a ring plan. You can select one from Ring Plan 1 through Ring Plan 10.
Day Type	Specify a type of the day. - Holiday1, 2: Set a type 1 or type 2 holiday. - User1, 2: Set a special day for a type 1 or type 2 site. Holiday 1, 2 and User 1, 2 are the types set in the calendar

Item	Description
	configuration menu.
Start Time End Time	Set the start time and the end time for the ring plan.

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.

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

You can set ring plan override using the [CONFIGURATION > Time Schedule > Ring Plan Override] menu.

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

Group Call Forward

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

Item	Description
User Group	Select a user group for which the group call forward will be configured.
Name	Enter the name of group call forward.
RP1 Forward Number ~ RP15 Forward Number	Enter the forward Number by ring plan.
Forced Forward Number	Enter the forward Number if you want to override the Forward Number by ring plan.
User Member Toll Restriction	Set to 'Enable' if you want to apply Toll Restriction to the members of the Call Forward Group.
Group Member	Select the members for the call forward group.

Service Group Local Number

Normally, call setup is made with an extension number. Service Group Local Number Feature provides a shorter number to call within a service group. An extension number in a SCM consists of 2 parts: the service group code and the service group local number. Each service group local number in a service group is unique and can be used directly without the service group code. This service group local number can have meaning only when Service Group Code in the [CONFIGURATION > User Group > Service Group] menu is configured. SCM provides a phone display method for a service group local number. An extension number or a service local number can be used for phone display. It is configured at the service group local CLI number in the [CONFIGURATION > User > Single Phone User] menu. This information is provisioned to the Samsung phone and used to determine phone number display.

The followings are limitations about a service group local number.

A service group code in the [CONFIGURATION > User Group > Service Group] menu can be changed only when no users are assigned to the service group.

To use a service group local number, a service group code for the user must be configured. You can check the service group local number in the [CONFIGURATION > User > Single Phone User] menu. It is displayed automatically.

When a user is assigned to a service group which has a service group code, the extension number of the user has to start with the service group code. An extension consists of the service group code and the service group local number.

A service group local number is available in the service group. When you setup a call to other service group, use an extension number for the user.

System Call Forward

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

Preset Call Forward All

This service performs preset call forward all based on the settings made by the system administrator even if preset call forward all is not set by users.

When administrator sets preset call forward all for users, this feature can be used to forward the call to the voice mail system for voice announcement, etc.

If the user has set call forwarding, the user settings have precedence over the system call forward settings. In other words, if the user has set call forward all, the system preset call forward preset all setting is ignored.

You can use the system preset call forward all feature by enabling [Preset Call Forward All] for selected extension numbers in the [CONFIGURATION > Service > Feature Service > Service Activation] menu.

Preset Call Forward Busy

This service performs call forward preset busy based on the settings made by the system administrator even if call forward preset busy is not set by users.

When the called user is busy, this feature can be used to forward the call to the voice mail system for voice announcement, etc.

If the user has set call forwarding, the user settings have precedence over the system call forward settings. In other words, if the user has set call forward all or call forward preset busy, the system call forward preset busy setting is ignored.

You can use the system call forward preset busy feature by enabling [Call Forward Preset Busy] for selected extension numbers in the [CONFIGURATION > Service > Feature Service > Service Activation] menu.

Preset Call Forward No Answer

This service performs call forward preset no answer based on the settings made by the system administrator even if call forward preset no answer is not set by users.

When the called user does not answer a call, this feature can be used to forward the call to the voice mail system for voice announcement, etc.

If the user has set call forwarding, the user settings have precedence over the system call forward settings. In other words, if the user has set call forward preset no answer, the system call forward preset no answer setting is ignored.

You can use the system call forward preset no answer feature by enabling [Call Forward Preset No Answer] for selected extension numbers in the [CONFIGURATION > Service > Feature Service > Service Activation] menu.

Preset Call Forward Unreachable

This service performs preset call Forward Unreachable based on the settings made by the system administrator even if preset call Forward Unreachable is not set by users.

When administrator sets preset call Forward Unreachable for users, this feature can be used to forward the call to the voice mail system for voice announcement, etc.

If the user has set call forwarding, the user settings have precedence over the system call forward settings. In other words, if the user has set call Forward Unreachable, the system preset call forward preset unavailable setting is ignored.

You can use the system preset call Forward Unreachable feature by enabling [Preset Call Forward Unreachable] for selected extension numbers in the [CONFIGURATION > Service > Feature Service > Service Activation] menu.

Preset Call Forward DND

This service performs preset call forward DND based on the settings made by the system administrator even if preset call forward DND is set by user.

When administrator sets preset call forward DND for users, this feature can be used to forward the call to the voice mail system for voice announcement, etc.

If the user has set DND service and administrator has set preset call forward DND service, incoming call is forwarded to the destination by setting of administrator.

You can use the system preset call forward DND feature by enabling [Preset Call Forward DND] for selected extension numbers in the [CONFIGURATION > Service > Feature Service > Service Activation] menu

VoIP Security

Signaling Encryption

The signaling encryption feature encrypts signaling information required for calls such as the SIP protocol. TLS is used in signaling encryption for VoIP connections between 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.

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

You can enable TLS of a Single Phone User by setting Protocol to TLS in the [CONFIGURATION > User > Single Phone User] menu.

You can enable TLS of a Multi-Extension Phone by setting Protocol to TLS in the [CONFIGURATION > User > Multi-Extension Phone] menu.

You can enable TLS for endpoints by setting Protocol to TLS in the [CONFIGURATION > Trunk Routing > Route] menu.

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

Media Encryption

The media encryption feature provides encryption for the voice data exchanged between the caller and the called party for calls established with signaling encryption. Media encryption can be enabled to calls between SIP phones or between a phone and an endpoint by applying secure RTP (sRTP), in which case, 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, endpoints, and 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.

You can enable media encryption for a Single Phone User by setting [Media] in the [CONFIGURATION > User > Single Phone User] menu.

You can enable media encryption for a Multi-Extension Phone by setting [Media] in the [CONFIGURATION > User > Multi-Extension Phone] menu.

RTP: No media encryption.

sRTP (AES/ARIA128): Encrypts media into the ARIA128 or AES protocol, and uses AES first.

sRTP (ARIA128/AES): Encrypts media into the ARIA128 or AES protocol, and uses ARIA128 first.

sRTP (AES/ARIA192): Encrypts media into the ARIA192 or AES protocol, and uses AES first.

sRTP (ARIA192/AES): Encrypts media into the ARIA192 or AES protocol, and uses ARIA192 first.

sRTP (AES): Encrypts media into the AES protocol.

sRTP (ARIA128): Encrypts media into the ARIA128 protocol.

sRTP (ARIA192): Encrypts media into the ARIA192 protocol.

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

Feature Services

Class of Service

SCM allows the administrator to set privileges for each user. But setting privileges for all the users individually is very complicated, so, it allows creating Class of Services which includes a set of service privileges and users can be assigned to their appropriate Class of Service.

When a user group is created, a default service group is automatically created for the user group. If you wish to apply a different service class than the default service class, you can create a new service class and use it.

Newly created Class of Service can be applied to user groups, service groups and users. Service classes are applied in the priority order of users > service groups > user groups.

Each service class can be set with different service privileges. It also supports override levels and privacy levels, whereby a particular service is provided only if the overriding user's override level is higher than the privacy level of the user being serviced.

The services involving override and privacy levels include the DND override feature and the barge-in with/without tone feature.

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

Item	Description
User Group	Select a user group for which the service class will be configured.
Name	Specify a name for the service class. This data is used as an identifier when selecting the service class in other menus.
Override Level Privacy Level	You can enter a level used for the barge-in with/without tone feature, etc. The override level is applied to the user using the service and the privacy level is applied to the user provided with the service. Available in levels 0 through 5. Higher levels have precedence over lower levels. A service is allowed only when the override level is higher than the privacy level.
Call Limitation Level	In an emergency situation, the call can be restricted by setting Call Limitation Level (Level 0 ~ Level 5). The priority of Call Limitation Level are as follows : Level 5 > Level 4 > Level 3 > Level 2 > Level 1 > Level 0 If the call level is lower than Call Limitation Level, this call is limited. For example, if the value of Call Limitation Level is level 3, all call of

Item	Description
	level 2 or level 1 or level 2 will be rejected including incoming and outgoing. The changed policy is applied to new calls only.
Service Permission	Specify allowed/inhibited settings of individual services for the service class. To allow a service, select the checkbox of a corresponding service. Only the services allowed in the [CONFIGURATION > User Group > User Group] menu can be set for permission.
Restriction Class	Even if user group is same, the call between service groups can be restricted.

Feature Lists

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

Service	Description
Absence	If enabled, when there is an incoming call, the absent announcement is played for the caller and the call is terminated.
Add-On Conference	It is including 2 types of conferences First. Conference member is added one by one. There are Ad-hoc, Conference On Answer, Barge-In and Multi-Device Conference. Second, conference services features related with UMS. There is Call Recording/AME.
Advanced Conference	Advanced conference includes predefined conference, progressive conference, and meet-me conference.
AME	If enabled, when there is an incoming call, the caller's voice is heard over the speaker as the caller leaves a voice message, just like an answering machine. You can also press the button to start the answering machine while ringing.
Auto Answer	If enabled, when there is an incoming call, the call is automatically answered in speaker mode.
Auto Record	If enabled, when a call is connected, voice mail is automatically connected for recording the call.

Service	Description
Auto Retry	If used when the number you dialed is busy, the number is automatically redialed repeatedly.
Barge-In with Tone	If used when the other person is engaged in a call, a three-way call is established.
Barge-In without Tone	If used when the other person is engaged in a call, a three-way call is established but your voice is muted.
BLF Key Create	If enabled, BLF key can be created.
Callback	If used when the number you dialed is busy, your phone will ring when the other person finishes the previous call. Answering the phone automatically redials the number.
Call Forward All	If enabled, all incoming calls are forwarded to another number.
Call Forward Busy	If enabled, incoming calls when busy are forwarded to another number.
Call Forward No Answer	If enabled, unanswered incoming calls are forwarded to another number.
Call Forward Unreachable	If enabled, incoming calls are forwarded to another number if the phone being called is not registered or otherwise unavailable.
Call Bridge	If enabled, a FXS user can join the bridged call.
Call Park Extension	If used during a call, the call is parked for an extension number.
Call Park Orbit	If used during a call, the call is parked for an orbit park ID.
Call Recording	If used during a call, the call conversation is recorded in voice mail.
Call Restriction by User	This service inhibits outgoing trunk calls. If enabled, only incoming trunk calls and extension calls are serviced.
Call Transfer	This feature allows you to transfer a call.
Call Transfer without Restriction Policy	If enabled, Call transfer to external number is allowed by restriction policy of held party phone.
Call Waiting	If used when the number you dialed is busy, this feature allows you to wait for the called party. If there is a call waiting, the phone will alert the user through the LED, LCD or tone.
Caller ID Block	If enabled, when there is an incoming call, the caller's number is

Service	Description
	not displayed.
Caller ID Display	If enabled, when there is an incoming call, the caller's number is displayed.
Direct Trunk Selection	This feature allows you to make a outgoing call by using DTS key.
DND	If enabled, when there is an incoming call, the DND announcement is played for the caller and the call is terminated.
DND Override	If used when the number you dialed is in DND mode, the DND setting is ignored and the call is made.
Follow Me	This feature allows you to use another phone to answer incoming calls for your phone.
Group Call Forward	This service is used for forwarding all incoming calls for all the phones in one hunt group to another number.
Hot Desking	This feature allows you to use any phone in any location by logging in with your user ID.
Hot Line	If enabled, when you lift your handset, the phone automatically dials a specified number.
Hotel Inter-Room Call Lock	If enabled, it blocks room to room calls in Hotel environments.
Individual Speed Dial	A user can register pairs of 3-digit Individual Speed Dial ID and the destination number. By dialing the IDs, the user can make a phone call to the destination.
Incoming Call Logging	If enabled, SCM keeps the incoming call logs. Incoming Call Logging Service should be activated in Service Activation Menu.
Malicious Call Trace	If used when there is a malicious call or an otherwise unwanted call, signaling will remain connected even if the caller has hung up the phone, so that the caller can be traced.
Meet-Me Conference	If the time and channel of a conference are reserved, the attendees can call into the conference room at the conference time to participate in the conference.
Move To Mobile	If enabled, the call can be moved to mobile. The call conversation is continued..
Mobile Auto Answer	If enabled, the mobile can answer automatically in case of move to mobile call.

Service	Description
MOBEX	The service also allows the subscriber to answer the call with his/her mobile phone and then when the subscriber returns to the office, the call can be transferred to the landline in the office and be picked up for continued conversation.
Music on Hold	This service plays the MOH when the call is put on hold.
Multi-ring	If enabled, when there is an incoming call, multiple phones ring so that one of the phones can be used to answer the call.
Multi-Device Conference	Even if one multi-device already joined a conference, the other multi-device using same number can also join the same conference.
No Ring	If enabled, when there is an incoming call, the phone does not ring but the call can be answered.
No Ring Override	If enabled, incoming call is allowed, although no ring service is activated.
One-Step Conference	Conference master calls Multiple members at once. It is including Predefined Conference, Progressive Conference, Intercom Conference, Dispatch Conference, Click 2 Conference and Emergency Conference.
Outbound Call Lock	This service inhibits outgoing trunk calls. If enabled, only incoming trunk calls and extension calls are serviced.
Operator Call	If enabled, operator call is allowed
Premium CID Service	If enabled , the detailed Information of caller will be shown on the callee display.
Preset Call Forward All	This service performs preset call forward all based on the settings made by the system administrator even if preset call forward all is not set by users.
Preset Call Forward Busy	This feature allows the administrator to forward incoming calls for a user without the call forward busy setting to another number when the user is busy.
Preset Call Forward No Answer	This feature allows the administrator to forward incoming calls for a user without the call forward no answer setting to another number when the user does not answer the phone.
Preset Call Forward Unreachable	This service performs preset call Forward Unreachable based on the settings made by the system administrator even if preset call

Service	Description
	Forward Unreachable is not set by users.
Preset Call Forward DND	This service performs preset call forward DND based on the settings made by the system administrator even if preset call forward DND is set by user.
Remote Office	This feature allows you to answer incoming calls for your phone from another location.
Retmote Extension Set	This feature allows you to set the DB of remote extension.
Restricted Call Forward	This feature allows you to exclude particular numbers from call forwarding.
Ring Plan Override	This feature allows you to set ring plan manually.
Station Paging	This feature allows paging extension numbers.
Paging On Answer	If enabled, user can make a paging on answer call, The member of paging on answer group should answer to listen paging announcement.
Temporary CID Restriction	If used when making a phone call, your number is not displayed in the called party's phone.
Wake-Up Call	If set with a time, an alarm will ring at the set time.

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.

You can configure feature codes using the [CONFIGURATION > Service > Feature Service > Feature Code] menu.

You must create new feature codes after the initial installation.

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

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

Feature Code + 1: Feature codes for enable, request, login, etc.

Feature Code + 0: Feature codes for disable, cancel, logout, etc.

Service	On/Off		Description
Absence	0	Cancel	This feature code requests for registration or cancelation of the absence service. If the service is enabled, when there is an incoming call, the absent announcement is played for the caller and the call is terminated.
	1	Set	
Account Code Voluntary			This feature code is used for entering an account code during a call.
ACD Agent Break	0	Cancel	This feature code requests for setting or unsetting of an ACD agent's break status.
	1	Set	
ACD Agent Login	0	Logout	This feature code requests for an ACD agent's login or logout of an ACD group.
	1	Login	
ACD Agent Wrap-up	0	Cancel	This feature code requests for setting or unsetting of an ACD agent's wrap-up status after an agent call.
	1	Set	
All Feature Clear			This feature code requests resetting of all features assigned to your number. Call Forward, DND, Absence, Extension Lock, etc. will be

Service	On/Off		Description
			cleared.
AME Enable	0	Cancel	This feature code enables or disables answering machine emulation. If the service is enabled, when there is an incoming call, the caller's voice is heard over the speaker as the caller leaves a voice message.
	1	Set	
AME Mode	0	Stop	This feature code, when there is an incoming call, directs the call to answering machine while ringing, or stops answering machine recording of a call and directs the call to the user.
	1	Start	
Attendant Continuous Call			This feature code is used for requesting a service which allows a call transferred by an IP attendant to be redirected to the IP attendant after called party hangs up the phone.
Auto Answer Mode	0	Cancel	This feature code requests for registration or cancelation of a service which automatically connects incoming calls in speaker mode.
	1	Set	
Auto Retry	0	Cancel	This feature code is used for requesting or canceling auto redials. If the service is used when the number you dialed is busy, the number is automatically redialed repeatedly.
	1	Set	
Barge-In with Tone			This feature code is used for requesting the barge-in service. If the service is used when the other person is engaged in a call, a three-way call is established.
Barge-In without Tone			This feature code is used for requesting the barge-in without tone service. If the service is used when the other person is engaged in a call, a three-way call is established but your voice is muted.
Callback	0	Cancel	This feature code is used for requesting registration or cancelation of the callback service. If the service is used when the number you dialed is busy, your phone will ring when the other person finishes the previous call. Answering the phone automatically redials the
	1	Set	

Service	On/Off		Description
			number.
Call Forward All	0	Cancel	This feature code is used for requesting registration or cancelation of a service which forwards all incoming calls to another number.
	1	Set	
Call Forward Busy	0	Cancel	This feature code is used for requesting registration or cancelation of a service which forwards incoming calls to another number when busy.
	1	Set	
Call Forward No Answer	0	Cancel	This feature code is used for requesting registration or cancelation of a service which forwards incoming calls to another number when not answered.
	1	Set	
Call Forward Busy/NoAnswer	0	Cancel	This feature code is used for requesting registration or cancelation of Call Forward Busy and Call Forward No Answer simultaneously.
	1	Set	
Call Forward Unreachable	0	Cancel	This feature code is used for requesting registration or cancelation of a service which forwards incoming calls to another number if the phone being called is not registered or otherwise unavailable.
	1	Set	
Call Forward Clear All			This feature code is used for requesting cancelation of all call forward services configured for your number.
Call Intercept			This feature code is used for an user intercept a call after barged-in.
Call Park Extension			This feature code is used for requesting parking of the current call for an extension number. If you dial the feature code without an extension number, the call will be parked for your number.
Call Park Orbit			This feature code is used for requesting parking of the current call for an orbit park ID.
Call Recording			This feature code is used for requesting recording of the current call conversation in voice mail.
Outbound Call Lock	0	Cancel	This feature code is used for requesting

Service	On/Off		Description
	1	Set	registration or cancelation of a service which inhibits outgoing trunk calls.
Call Waiting	0 1	Cancel Set	This feature code is used for requesting registration or cancelation of a service which allows you to wait for the called party when the number you dialed is busy.
Change Language			This feature code is used for requesting change of the language used for your number.
Change Password			This feature code is used for requesting change of the password used for your number.
Conference			This feature code is used for requesting a conference call.
Conference On Answer			This feature code is used for requesting a service which immediately starts a three-way conference call when answered by the called party during a call.
Direct Call Pickup			This feature code is used for requesting a service which allows you to dial into another number and answer its incoming call when the phone is ringing.
DND	0 1	Cancel Set	This feature code is used for requesting for registration or cancelation of the DND service. If the service is enabled, when there is an incoming call, the DND announcement is played for the caller and the call is terminated.
DND Override			This feature code is used for requesting the DND override service. If the service used when the number you dialed is in DND mode, the DND setting is ignored and the call is made.
Dispatch Conference			This feature code is used for making conference by using CSTA application. It calls multiple members at once.
Extended Alarm Reminder	0 1	Cancel Set	One subscriber can have multiple wakeup calls by using this feature code. This feature is aimed to support hotel services.

Service	On/Off		Description
Follow Me	0	Cancel	This feature code is used for requesting registration or cancelation of a service which allows you to use another phone to answer incoming calls for your phone.
	1	Set	
Forced Call Release			This feature code is used for an user release the call by force after barged-in.
FXS Make Call with CW Disable			This feature code is used for disabling call waiting during a call which was dialed from a gateway FXS phone.
FXS Call Waiting Disable			This feature code is used for disabling call waiting during a call on a gateway FXS phone.
FXS make Conference Call			This feature code is used for requesting a conference call during a call on a gateway FXS phone.
FXS Last Member Drop			This feature code is used for requesting deletion of the last joined attendee of a conference call on a gateway FXS phone.
Group Call Forward	0	Cancel	This feature code is used to activate or deactivate the forwarding feature of a Call Forward Group.
	1	Set	
Group Call Pickup			This feature code is used for requesting a service which allows you to dial into the Pickup group number of another number and answer its incoming call when the phone is ringing. If you dial the feature code without a Pickup group number, the call currently ringing for the Pickup group will be answered.
Hotel COS Change			This feature code is used for changing COS of a room phone. It is aimed to support hotel services.
Hotel Service			This feature code is used for changing status of room. It is aimed to support hotel services.
Hotel Staff Locate			This feature code is used for informing the location of staff. It is aimed to support hotel services.

Service	On/Off		Description
Hunt Group Login	0	Out	This feature code requests for login or logout for a hunt group. When there is an incoming call for the hunt group, the logged out members are excluded when determining the called party.
	1	In	
Intercom			This feature code requests for the intercom service which allows one-touch dialing and automatic answering of calls between the numbers registered with the intercom feature, such as executives and secretaries.
Intercom Conference			If user dial this feature code and hunt group number, caller and hunt group members can join conference. And the hunt group members answer automatically.
Last Incoming Redial			This feature code requests for redialing of the last incoming call's CLI number.
Last Outgoing Redial			This feature code requests for redialing of the last dialed number.
Individual Speed Dial	0	Insert	This feature code is used to add an Individual Speed Dial number.
	1	Delete	This feature code is used to delete an Individual Speed Dial number.
Individual Speed Dial -Call			This feature code is used for support speed dial per subscribers.
Malicious Call Trace			This feature code is used for requesting the malicious call trace service. If the service is used when there is a malicious call or an otherwise unwanted call, signaling will remain connected even if the caller has hung up the phone, so that the caller can be traced.
Meet Me Conference Join			This feature code is used for participating in a meet me conference, which is set up by reserving the time and channel of the conference and joined by calling into the conference room at the conference time.
MOBEX Call Pickup			This feature code is used for requesting the MOBEX on desk pick up service. If the service is

Service	On/Off		Description
			enabled, when the office phone and the mobile phone ring simultaneously by the multi-ring feature, you can answer the call on the mobile phone and then continue the call on the office phone.
Move to Mobile			This feature code is used for moving the call on the desk phone to mobile phone.
Multi-Device Conference			Even if one multi-device already joined a conference, the other multi-device using same number can also join the same conference. This feature code is used for supporting it.
Multi-ring Enable	0	Cancel	This feature code registers or cancels the multi-ring service. If the service is enabled, when there is an incoming call, multiple phones ring so that one of the phones can be used to answer the call.
	1	Set	
Multi-ring Member	0	Delete	This feature code requests adding/removing of members to/from the multi-ring service. If the service is enabled, when there is an incoming call, multiple phones ring so that one of the phones can be used to answer the call.
	1	Insert	
No Ring	0	Cancel	This feature code requests for registration or cancelation of the no ring service. If the service is enabled, when there is an incoming call, the phone does not ring but the call can be answered.
	1	Set	
Outbound Call Lock	0	Cancel	If this feature code is set, system does not let user make a outgoing call.
	1	Set	
Parked Call Retrieve			This feature code is used for picking up a call parked for the extension number or the orbit park ID.
Predefined Conference			This feature code is used for paging all the members registered in the conference system simultaneously for a conference.
Predefined Text Message			If users want to send text messages set by the administrator, user should dial this feature code

Service	On/Off		Description
			with destination number.
Progressive Conference			This feature code is used for entering numbers of multiple members in the conference system and then paging them simultaneously for a conference.
Remote Extension Lock			This feature code is used for an IP attendant to set incoming/outgoing call lock for another extension number.
Remote Extension Set			This feature code is used for an IP attendant to set features for another extension number.
Remote Office	0	Cancel	This feature code is used to activate or deactivate the Remote Office feature.
	1	Set	
Ring Plan Override	0	Cancel	This feature code is used to override time-based auto ring plan manually.
	1	Set	
Shared Call Retrieve			This feature code is used for picking up a call parked for another phone when using multi-device.
Station Paging			This feature code is used for requesting an extension announcement service.
Temporary CID Restriction			If used when making a phone call, your number is not displayed in the called party's phone.
VM Access			This feature code is used for dialing to access the voice mail system.
VM Administration			This feature code is used for accessing the voice mail system and changing it settings.
VM Memo			This feature code is used for accessing the voice mail system and leaving a message for another number or in your own mailbox.
VM Message			This feature code is used for accessing the voice mail system and listening to messages in your mailbox.
VM Transfer			This feature code is used for directing the current call to the voice mail system and connecting it the mailbox for another number.

Service	On/Off		Description
Wake-Up Call	0	Cancel	This feature code requests for registration or cancelation of the wake-up call service, which, if set with a time, an alarm will ring at the set time.
	1	Set	

Feature Activation

Those services not performed temporarily by user actions but configured in the database can be configured by the administrator.

The administrator can change service settings for individual users using the [CONFIGURATION > Service > Feature Service > Service Activation] menu.

Enter a user group and an extension number and press the Search button to change the current settings for the user.

Select an inactive service and press the Activate button to activate the service. Select an active service and press the Deactivate button to deactivate the service.

When activating a service, you may be required to enter additional information depending on the service type.

Item	Description
User Group	Select a user group.
Extension Number	Specify a user number for which the service will be activated.
Service Type	Specify a service to activate.
Forward/Hot/Dest Number	Specify a destination number if registering call forwarding, etc.
Service Date	Specify a service date if registering wake-up call, etc.
Wake-up Type	Specify whether to repeat the wake-up call if registering wake-up call.
Use Notification	Specify whether the phone needs to be notified of the status if registering call forwarding, etc.
No Answer Forward Time(sec)	Specify the time period used for determining no answer when registering no answer call forward.
Allow Other Ring	Specify whether incoming calls when busy will be forwarded to the multi-ring members if registering multi-ring.
Call Forward Preset Type	Specify a type of calls to forward if registering call forward presets. - Internal: Incoming calls from extension numbers are forwarded. - External: Incoming calls from trunks are forwarded.

Item	Description
	- Both: Incoming calls from both extension numbers and trunks are forwarded.
Auto Record Mailbox Number	Specify a mailbox where the recorded files will be stored if registering auto record.
Auto Record Call Type	Specify a type of calls to auto record if registering auto record. - Internal: Incoming calls from extension numbers are automatically recorded. - External: Incoming calls from trunks are automatically recorded. - Both: Incoming calls from both extension numbers and trunks are automatically recorded.
Hot Desk Expire Time(hour)	Specify a time period for which to wait before a logged in user is automatically logged out if registering hot desk.
Hot Line Waiting Time(sec)	Specify a time period for which to wait before the preset number is automatically dialed after the handset is lifted if registering hot line.

User Authentication

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

Local Authentication

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.

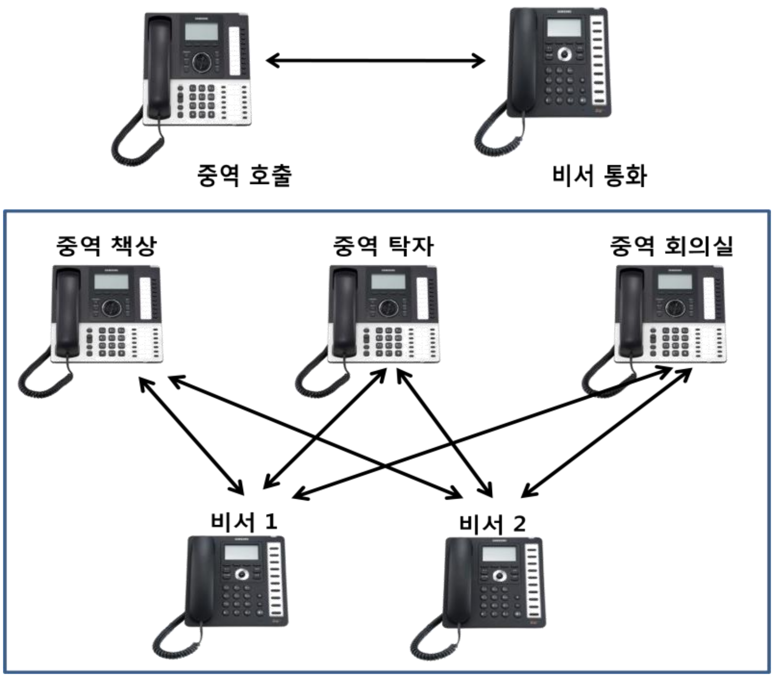
The user’s password stored in the LDAP server is received through a Search-Result message.

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.

Boss/Secretary

The boss/secretary feature allows a boss and a secretary to share one user number while using their own individual numbers and the intercom feature.

Bosses and secretaries can be connected 1:1 or M:N.



While configuration may vary by administrators, we recommend that you configure the feature in the following order for maximum efficiency.

1. Create a device in the [CONFIGURATION > User > Multi-Extension Phone] menu.
   - Phone Verification: Select IP Address or MAC Address for authentication method.
   - User Type: Select Manager or Secretary for device.
2. Create a user password for the manager/secretary in the [CONFIGURATION > User > Multi-Phone User] menu.
   - Call Appearance: Select SCA (SCA: Shared Call Appearance, MCA: Multiple Call Appearance).
3. Configure intercom between the individual numbers of the manager and the secretary in the [CONFIGURATION > Service > User Service > Intercom] menu.

The boss/secretary feature allows the phones to share the status of the shared line so that the users are informed of the current status of the shared line.

Example) Redirect an incoming call to the Boss

- Boss and Secretary have shared the same SCA number and each has its own private number also.
- A call is incoming to the SCA number, after the Secretary answered the call,
  - Secretary using its private number make an intercom call to the private number of Boss to explain who is calling to the Boss.

    If the Boss wants to talk, secretary make the Boss pickup the held SCA call by pressing SCA line button of the Boss's Phone.
  - Secretary redirect the SCA call by just pressing the BLF button pre-configured with the private number of the Boss.

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.

To use the BLF service, you must first configure the feature to the buttons of the phone. Buttons can be configured in the following ways:

Use the [CONFIGURATION > User > Phone Key Programming] menu to configure the built-in buttons on the phone.

Use the [CONFIGURATION > User > AOM Key Programming] menu to configure the buttons on the button extension box.

Fields	Description
User Group	Select a User Group
Phone Name	Select a phone number or Mac Address of AOM
#	Displays number and order of the buttons.
Display Name	Specify the Name of the button. It may be displayed in BLF button depends on Phone Model.
Key	Select the BLF type.
Value	Specify the Extension number want to monitor.
Extension Number	Multi-phone user case, you can specify one of the extension numbers which extension number wants to monitor.

DTMF Detection Service

During a call between SIP phones on an IP PBX, all data except SIP signaling for call connection is exchanged by the phones. Therefore, the numbers dialed for services—except the phone number included in the INVITE message for call connection—cannot be sent to the system using the standard protocol.

In order to receive the numbers dialed on the phone—except the INVITE message—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 when outbound call connected. The account code entered will be saved in the charging data record

(CDR), which can be used for calculating call charges for the user.

You can set the account code input method of a Single Phone User under [Account Code Use] in the [CONFIGURATION > User > Single Phone User] menu.

You can set the account code input method of a Multi-Phone User under [Account Code Use] in the [CONFIGURATION > User > Single Phone User] menu.

- None: No account code needs to be entered when making a trunk call.
- Force: An account code registered with SCM must be entered when making a trunk call.
- Voluntary: An account code can be entered by pressing the account code button when making a trunk call.

To use the voice announcement system for entering account codes, you should enable the item whose service type is [ACC] in the [CONFIGURATION > Service > DTMF Detection Service > Service Activation] menu.

The account codes entered for forced account code input can be registered in the [CONFIGURATION > Service > DTMF Detection Service > Account Code] menu.

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

1. The user dials an access code and an external number.
2. If the user's phone is set with forced account code input, SCM connected the call to its built-in voice announcement system.
3. The voice announcement system plays an announcement for the user to enter a registered account code.
4. The user enters an account code as instructed. The voice announcement system verifies that a valid account code has been entered.
5. If the account code entered is valid, SCM uses the access code received in step (1) to select a route and makes a call for the external number specified.
6. If the account code is invalid, 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.

To use the voice announcement system for authenticating call, you should enable the item whose service type is [AUTH] in the [CONFIGURATION > Service > DTMF Detection Service > Service Activation] menu.

The number for the voice announcement system which authenticates call authentication codes can be registered in the [CONFIGURATION > Service > DTMF Detection Service > Service Number] menu.

The call authentication codes used for call authentication can be registered in the [CONFIGURATION > Service > DTMF Detection Service > Authorize Code] menu.

When a user restricted from making external calls attempts to make a trunk call by using a call authentication code, actions are performed in the following order:

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.

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.

To use the voice announcement system for authenticating DISA users, you should enable the item whose service type is [DISA] in the [CONFIGURATION > Service > DTMF Detection Service > Service Activation] menu.

The number for the voice announcement system which authenticates DISA users can be registered in the [CONFIGURATION > Service > DTMF Detection Service > Service Number] menu.

To connect incoming DISA calls from trunks to the voice announcement system for authenticating DISA users, set the called number to the number of the DISA user authentication announcement system in the [CONFIGURATION > Trunk Routing > DID Routing] menu.

SCM can allow incoming DISA calls from registered callers to use DISA calls without user authentication. If you register caller numbers in the [CONFIGURATION > Service > DTMF Detection Service > DISA Approved CLI Number] menu in advance, the incoming DISA calls from the registered caller numbers are provided with the DISA service without having to enter user numbers or passwords.

System Speed Dial

This feature allows you to assign maximum 16-digit shortcut number to a phone number frequently dialed not by individual users but by all users of the system or to a lengthy phone number so that the number can be dialed just by using the shortcut number when necessary.

You can register system speed dial numbers using the [CONFIGURATION > Service > Speed Dial >System Speed Dial] menu.

Item	Description
User Group	Select a user group for which the system speed dial will be registered.
System Speed ID	Specify an ID for the system speed dial. The ID must be a number of two or longer digits. The ID must not be overlapped with other extension numbers and conference channel numbers.
System Speed Name	Specify a name for the system speed dial.
System Speed Number	Specify an actual phone number to be dialed by the system speed dial.

Call Bridge

This feature allows the gateway FXS user to join the conversion of bridged user by hook-off. After that, if bridged user is hook-off, FXS user has a continued conversation.

If bridged user is not busy status, the gateway FXS user listens dial-tone and makes a call.

In the [Configuration>Service>Feature Service>Service Activation>Hot Line], the bridge feature code and the extension number of bridged user should be entered for the gateway FXS user. And the Hot Line Delay Time should be greater than 0.

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.

License Key Registration

To use Wireless Enterprise Service, a wireless enterprise solution license is required.

The license key can be registered in the [CONFIGURATION > Miscellaneous > License] menu.

Creating the user for Wireless Enterprise Service

Creating the user for Wireless Enterprise Service is similar to how to create single phone user.

The user configuration for Wireless Enterprise Service can be configured as making single phone user in the [CONFIGURATION > User > Single Phone User].

For details, Refer to [PART II. Configuring SCM Server > 2.4 Adding Individual User > Making Single Phone User] in the SCM Express Operation Manual.

Normal Configuration is same as creating single phone user. Phone Type should be set to Samsung-Mobile-Phone and Mobile Phone Number should be set. Select the "Use Mobile Phone Number" option

The following describe "Use Mobile Phone Number" option.

Use Mobile Phone Number Option	Description
None	Do not use the mobile phone number. Only ring to the extension..
Ring Only	Both extension and mobile phone are ringing simulaneously. Multi ring service with extension and mobile phone.

Mobile Service Options

Wireless AP(Accees Point) SSID(Service Set Identifier) can be configured in the [CONFIGURATION > Wireless Enterprise > Mobile Service Options] menu. This SSID configured must be same the SSID of WiFi Configuration in the mobile phone..

When FMC client receives a call, "Wait Call/Later Call" menu is displayed on the FMC client UI in case that the FMC client user cannot receives that call. If the FMC client user selects "Wait Call or Later Call" menu on ringing, caller can listen to the announcement and then the call is held in case of "Wait Call" and is disconnected in case of "Later Call".

Wait Call/Later Call can be configured in the [CONFIGURATION > Wireless Enterprise > Mobile Service Options > Wait Call, Later Call] menu.

Upgrade Mobile Software

The Mobile phone user for Wireless Enterprise Service must download and install below the files.

File Name	Description
XX_Version.xml	This file include the latest FMC client information related to each Android Version.

File Name	Description
	(example : ICS_Version.xml, GB_Version.xml)
SMV_New_Version.apk	FMC Client Application The apk file name depends on file version and some files can be provided.

XX_Version.xml include the latest FMC client information related to each Android Version and the apk file supporting each android version is FMC Client application.

The files are uploaded using FTP. If there is no folder for updating FMC Client, upload the xml file and the apk file after creating the new folder below /DI/WEBCLI/. (example : /DI/WEBCLI/SMV)

Download Server IP, File folder is configured in the [CONFIGURATION > Wireless Enterprise > Upgrade Mobile Software] menu. The number of available is up to 5.

Mobile Phone Profile

The phones, which phone type is Samsung-Mobile-Phone are displayed in the [CONFIGURATION > Wireless Enterprise > Mobile Phone Profile] menu. The parameters related to FMC is configured in the [CONFIGURATION > Wireless Enterprise > Mobile Phone Profile] menu.

The following items by setting default values are used except for the special site.

Item	Description
Roaming Trigger	Roaming Trigger Parameter for busy FMC client (Default : -70) Comparing with the RSSI, this value is used to scanning WiFi
Roaming Delta	Roaming Delta Paramter for busy FMC client (Default : 10) When this value is more than the minimum roaming delta, FMC clients scan WiFi and Roam
Roaming Scan Period	Roaming Scan Period Parameter for busy FMC client (Default : 3)
Noise Supression TX	Select whether to use FMC Client Noise Suppression TX (Default : Disable)
Noise Supression RX	Select whether to use FMC Client Noise Suppression RX (Default : Disable)
AECM	Select whether to use FMC Client AECM (Default : Speaker Phone)
Echo Suppression	Echo Suppression (Default : Enable)

Item	Description
Enable Swing Free TX	Select whether to use Swing Free TX (Default : Enable)
Enable Swing Free RX	Select whether to use Swing Free RX (Default : Enable)
Enable CNG	Comfort Noise Generation (Default : Enable)
Select Download Server	Select the download server
Version	Latest Version
Media Start Port	FMC Client Media Start Port
Media End Port	FMC Client Media End Port
Multiframe Enable	Select whether to use Voice Engine Multiframe with Samsung AP (Default : Disable)
Multicast Enable	Select whether to use Voice Engine Multicast with Samsung AP (Default : Disable)
TOS	TOS for RTP in IP Header (Default: 224 (Decimal)
JBC Threshold	Jitter Buffer Size of Voice Engine (Default:4)

Mobile Configuration

Roaming parameter of each phone model can be configured in the [CONFIGURATION > Wireless Enterprise > Mobile Configuration > Roaming Parameter] menu. The priority of these setting values is higher than the setting values of each extension in Mobile Phone Profile menu.

Item	Description
Roaming Trigger	Roaming Trigger Parameter for busy FMC client Comparing with the RSSI, this value is used to scanning WiFi
Roaming Delta	Roaming Delta Paramter for busy FMC client When this value is more than the minimum roaming delta, FMC clients scan WiFi and Roam
Roaming Scan Period	Roaming Scan Period Parameter for busy FMC client

Codec Priority and Payload Type for FMC Client can be configured in the [CONFIGURATION > Wireless Enterprise > Mobile Configuration > Codec Priority] menu. Each codec does not have same codec priority. When the FMC client supporting SILK or AMR-WB negotiates a codec with phone (example : Desk Phone) that does not support SILK or AMR-WB, the codec negotiated is PCMA or PCMU.

Parameters for SILK or ARM-WB are configured in the [CONFIGURATION > Wireless Enterprise > Mobile Configuration > Codec Config] menu. The following values by setting the default values is used except for special testing purposes.

Item	Description
SILK : Sampling Freq.	SILK Sampling Frequency (Default : 24000)
SILK : Max Packet Time	SILK Maximum Packet Time (Default : 100)
SILK : DTX Use	Select whether to use SILK DTX (Default : Off)
SILK : FEC Use	Select whether to use SILK FEC (Default : On)
AMR-WB : Bit Rate	Select whether to use AMR-WB Bit Rate (Default : 23850)
AMR-WB : DTX Use	Select whether to use AMR-WB DTX (Default : Off)

Chapter 2. User Features

User features are the features executed or set by users.

A user service is only available to the users with privilege for the service which is given by the administrator. Since setting service privileges for all the users individually could be very complicated, Class of Service are be created with their own set of service privileges and users are then assigned to their appropriate Class of Service.

Service classes can be applied to user groups, service groups and users. Service classes are applied in the priority order of users > service groups > user groups.

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

To use a user feature defined by the user in advance, you can use the feature on SCM Personal Assistant. The user can also use a feature—including those services used temporarily—by pressing the feature button on the phone or dialing the feature code. To use a feature by pressing the feature button or dialing the feature code, the feature code must be defined in advance.

You can define feature codes using the [CONFIGURATION > Service > Feature Service > Feature Code] menu.

For more information on feature services, see the "Feature Service" section of "4.1. System Features".

Absence

The absence feature is used for notifying that the user is absent. If the absence feature is enabled, when there is an incoming call, an announcement is played to notify the caller of the absence status and the call is terminated.

To use the absence feature, the following items must be configured.

- The "Absence" service must be enabled in Class of Service.
- The "Absence -Set" and "Absence -Cancel" feature codes must be defined.

The user can register or cancel the absence feature in the following ways:

- The user can dial the "Absence -Set" feature code on the phone to register the absence status.
- The user can dial the "Absence -Cancel" feature code on the phone to cancel the absence status.
- The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the absence feature. You can also specify the time for enabling the absence feature.

Auto Answer

The auto answer feature is used when the user wishes to have his/her incoming calls answered automatically. If the auto answer feature is enabled, when there is an incoming call, the speaker will be turned on and the call will be answered automatically.

To use the auto answer feature, the following items must be configured.

- The “Auto Answer” service must be enabled in Class of Service.
- The “Auto Answer -Set” and “Auto Answer -Disable” feature codes must be defined.

The user can register or cancel the auto answer feature in the following ways:

- The user can dial the “Auto Answer -Enable” feature code on the phone to enable auto answer.
- The user can dial the “Auto Answer -Disable” feature code on the phone to disable auto answer.
- The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the auto answer feature.

You can use an attention tone which is used for indicating that an incoming call has been answered automatically. The attention tone setting is applied to all users of the selected user group.

Auto answer attention tone can be configured in the [CONFIGURATION > User Group > Change User Group > Options] menu.

- Auto Answer Attention Tone: Specify whether to play the attention tone when a call is answered automatically.
- Auto Answer Attention Tone Count: Specify how many times the 100ms tone should be repeated when the auto answer attention tone is played.

Automatic Retry

When the user makes an outgoing trunk call but the called party is busy or does not answer the call, the automatic retry service can be used to automatically redial the number after a set period of time. If the automatic retry is enabled, the phone’s speaker is automatically turned on at a set interval and the last dialed number is dialed again.

To use the automatic redial feature, the following items must be configured.

- The [Auto Retry] service must be enabled in [CONFIGURATION > Service > Class of Service].
- The [Auto Retry–Set] and [Auto Retry–Cancel] feature codes must be defined in the [CONFIGURATION > Service > Feature Service > Feature Code].

The user can register or cancel the automatic retry feature in the following ways:

- When the user makes an outgoing trunk call but the called party is busy or does not answer the call, the user can press the automatic redial button to use the automatic redial feature. When the automatic redial button is pressed on the phone, the phone dials the [Auto Retry –Set"] feature code to SCM.
- When the phone dials the [Auto Retry –Cancel] feature code, the automatic redial is canceled.

When the automatic redial feature is serviced, the following three timers are activated. The timers required for the automatic redial feature can be configured using the [CONFIGURATION > User Group > Change User Group> Timers] menu.

- Auto Retry No Answer Time (sec): When a trunk number is redialed by automatic retry and the called party does not answer the call, the phone waits for this period of time before it terminates the call as an unanswered call.
- Auto Retry Interval (sec): When a number is automatically redialed but is still busy or does not answer, the phone waits for this period of time before it redials the number.
- Auto Retry Service Duration (min): When automatic retry continues to fail, the phone tries for this period of time before it terminates the service.

Barge-In

This feature allows you to intrude into a user's current call for a three-way conference call. The call intrusion feature is also known as Barge In or Call Override.

The call intrusion feature is providing with warning or without warning depending on whether the user is notified that the call has been intruded into. 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.

The administrator needs to set [Application Type] to "Internal Conference" in the [CONFIGURATION > Application > Conference Server] menu to create a connection to the built-in conference system. This conference server should be included "Add-On Conference" in service list.

To use the call intrusion feature, both "Override Level" and "Privacy Level" must be defined in Class of Service. Call intrusion is allowed only when the override level is higher than the privacy level. The override level is applied to the user intruding and the privacy level is applied to the user being intruded into.

Barge-In with Tone

When you intrude into a call and establish a three-way conference, this service periodically plays a tone to notify the user being intruded into that the call has been intruded into.

To use the call intrusion feature, the following three items must be configured.

The "Call Intrusion" service must be enabled in Class of Service.

The “Call Intrusion” feature code must be defined in Class of Service.

The user can use the call intrusion feature in the following ways:

The user calls an extension number and if the called party is busy, the user can press the call intrusion button to intrude into the call.

The user can dial the call intrusion feature code + number of the user currently in a call to intrude into the call.

Barge-In without Tone

When a three-way conference call is established by intruding into a call, the user being intruded into is not given any notification and the intruding user’s voice is muted so that the call can be monitored in secret.

To use the call intrusion without tone feature, the following three items must be configured.

The “Call Intrusion without Tone” service must be enabled in Class of Service.

The “Call Intrusion without Tone” feature code must be defined in Class of Service.

The user can use the call intrusion without tone feature in the following ways:

The user calls an extension number and if the called party is busy, the user can press the call intrusion without tone button to intrude into the call without tone.

The user can dial the call intrusion without tone feature code + number of the user currently in a call to intrude into the call without tone.

Change Password

This service provides a user can change the PIN Number which is used for services by himself.

The Change Password Feature code must be configured.

User can dial the Change Password feature Code with previous Password once and news password twice like below:

Ex) *33 + (old Password) + (New Password) + (New Password)

Callback

When a user calls another user but if the called party is busy or does not answer, the caller can enable the callback feature so that when the called party becomes available, the caller’s phone will ring, and if the caller answers the phone, the called party’s number is redialed.

To use the callback feature, the following items must be configured.

The “Callback” service must be enabled in Class of Service.

The “Callback -Set” and “Callback -Cancel” feature codes must be defined.

The user can register or cancel the callback feature in the following ways:

When the user calls another user but the called party is busy or does not answer the call, the user can press the callback button to use the callback feature. When the callback button is pressed on the phone, the phone dials the “Callback –Reg” feature code to SCM.

The user can dial the “Callback -Cancel” feature code on the phone to cancel the callback feature.

When the callback feature is serviced, the following two timers are activated. The timers required for the callback feature can be configured using the [CONFIGURATION > User Group > Change User Group > Timers] menu.

Callback Ring No Answer Time (sec): When the caller is called back by the callback feature but the caller does not answer, this call which is made to notify the caller that the called party has now become available will be processed as a failed call after ringing for this period of time. If the callback notification call fails, the system waits until the called party uses the phone and become available again.

Callback Service Duration (min): When callback is enabled, if the callback service is not executed successfully during this period of time, the callback service is automatically terminated.

Call Forward

When there is an incoming call, this feature is used for forwarding the call to another number specified by the user. The call forward feature can have limitation by Restricted Call Forward in the [CONFIGURATION > Service > Feature Service > Class of Service] menu. If Restricted Call Forward is enabled, call cannot be forwarded to trunk.

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”.

To use the call forward all feature, the following items must be configured.

- The “Call Forward All” service must be enabled in Class of Service.
- The “Call Forward All -Set” and “Call Forward All -Cancel” feature codes must be defined.

The user can register or cancel the call forward all feature in the following ways:

- The user can dial the “Call Forward All -Set” feature code + destination phone number on the phone to enable call forward all.

- The user can dial the “Call Forward All -Cancel” feature code on the phone to cancel call forward all.
- The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the call forward all feature. If call forward is enabled on SCM Personal Assistant, the user can select two options, time and notification. Without time configuration, call forwarding is enabled permanently until the user disables the feature. However, If the user configures the time, call forward is worked during the time period. The user can know how many calls are forwarded with the notification option.

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 — Call Forward Preset Busy” section of “4.1. System Features”.

To use the call forward busy feature, the following items must be configured.

- The “Call Forward Busy” service must be enabled in Class of Service.
- The “Call Forward Busy -Set” and “Call Forward Busy -Cancel” feature codes must be defined.

The user can register or cancel the call forward busy feature in the following ways:

- The user can dial the “Call Forward Busy -Set” feature code + destination phone number on the phone to enable call forward busy.
- The user can dial the “Call Forward Busy -Cancel” feature code on the phone to cancel call forward busy.
- The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the call forward busy feature.

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 — Call Forward Preset No Answer” section of “4.1. System Features”.

You can use SCM Personal Assistant to set the time period for each user which is used by the call forward no answer service to determine the user’s an incoming call as an unanswered call and forward it to a specified number. If call forward no answer is enabled by pressing the feature code on the phone, the default time of 15 seconds is used.

To use the call forward no answer feature, the following items must be configured.

- The “Call Forward No Answer” service must be enabled in Class of Service.
- The “Call Forward No Answer -Set” and “Call Forward No Answer -Cancel” feature codes must be defined.

The user can register or cancel the call forward no answer feature in the following ways:

- The user can dial the “Call Forward No Answer -Set” feature code + destination phone number on the phone to enable call forward no answer.
- The user can dial the “Call Forward No Answer -Cancel” feature code on the phone to cancel call forward no answer.
- The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the call forward no answer feature. Here, you can also specify the time period used for determining the no answer status.

Call Forward Unavailable

If the call Forward Unreachable service enabled for a user, all incoming calls for the user are automatically forwarded to a specified number when the user’s phone is not registered, does not respond to signaling, or otherwise unavailable.

To use the call Forward Unreachable feature, the following items must be configured.

- The “Call Forward Unavailable” service must be enabled in Class of Service.
- The “Call Forward Unreachable -Set” and “Call Forward Unreachable -Cancel” feature codes must be defined.

The user can register or cancel the call Forward Unreachable feature in the following ways:

- The user can dial the “Call Forward Unreachable -Set” feature code + destination phone number on the phone to enable call Forward Unreachable.
- The user can dial the “Call Forward Unreachable -Cancel” feature code on the phone to cancel call Forward Unreachable.
- The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the call Forward Unreachable feature.

Selective Call Forward

This service only forwards a user’s incoming calls from specified numbers. Selective call forward can be serviced in the following two ways. If two features are enabled both, Call Forward Restriction has precedence over Call Forward Allowance.

Call Forward Allowance

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

- The user can use the [Selective Call Forward Allowance] menu on SCM Personal Assistant to register a list of caller numbers to allow selective call forward.

- The administrator can use the [CONFIGURATION > Service > Selective Service > Call Forward Allowance] menu to register a list of caller numbers to allow selective call forward for each user.

Call Forward Restriction

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

- The user can use the [Call Forward Restriction] menu on SCM Personal Assistant to register a list of caller numbers to restrict selective call forward.
- The administrator can use the [CONFIGURATION > Service > Selective Service > Call Forward Restriction] menu to register a list of caller numbers to restrict selective call forward for each user.

Call Forward No Response

No response from a User

It is one of cases of Call Forward Unavailable. It is different with Call Forward No Answer which is working after ring back tone.

If there is no response from a User within the specific time right after sending call setup message (initial INVITE), Call Forward Unavailable service is functioning.

If Call Forwarding Unavailable service was not set, the call will be dropped with announcements.

Administrator can configure the specific time for User at [CONFIGURATION > User Group > Change User Group > Timers > Internal Call No Response Time (sec)] and it is 3 seconds by default.

No Response from a Trunk

If there is no response from a Route within the specific time right after sending call setup message (initial INVITE), Call Forward Unavailable service is functioning.

If there is no Alternative Route for the Route, the call will be dropped with announcements.

Also, administrator can configure the specific time for Trunk at [CONFIGURATION > User Group > Change User Group > Timers > External Call No Response Time (sec)] and it is 10 seconds by default.

Call Hold

The call hold feature allows the user to park the current call and make another call for transfer or conference and then retrieve the hold call.

If the other person on the line is handling multiple calls (including call park, transfer, and conference), the call cannot put on hold.

Call Park

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

A park ID must be entered when parking a call, so that the call can be identified when picked up. There are the following two types of call park service depending on the park ID input method.

Call Park Extension

The call park service can be used by using an extension number as the park ID. Since an extension number is used as the park ID, only one call can be parked per extension number.

To use the call park feature, the following items must be configured.

The [Call Park Extension] service must be enabled in [CONFIGURATION > Service > Class of Service] menu.

The [Call Park Extension] feature code must be defined in [CONFIGURATION > Service > Feature Code] menu.

While on a call, the user can press the park button and then dial the call park feature code + an extension number to park the call for the extension number entered. Or, the user can just dial the call park feature code without an extension number to park the call for his/her own number.

Call Park Orbit

The call park orbit service can be used by using an independent orbit park number as the park ID. Since independent numbers are used as the park ID, multiple calls can be parked for each extension number. For Call Par Orbit, the park ID has a range from 01~99. Only one call can be parked with one park ID. If you want to use empty orbit park ID, use press 00 instead of the park ID. In that case, SCM finds an empty park ID and the call is parked with the park ID. You can check the parked ID on the display of the phone.

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

To use the orbit park feature, the following items must be configured.

The [Call Park Orbit] service must be enabled in [CONFIGURATION > Service > Class of Service] menu.

The [Call Park Orbit] feature code must be defined in [CONFIGURATION > Service > Feature Code] menu.

While on a call, the user can press the park button and then dial the orbit park feature code + an orbit park number to park the call for the orbit park number entered.

Parked Call Retrieve

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

To use the Parked Call Retrieve feature, the [Parked Call Retrieve] feature code must be defined in the [CONFIGURATION > Service > Feature Code] menu.

The user can dial the parked call retrieve feature code + park ID to pick up a parked call.

Park Recall

This feature allows a parked call to be redirected to the user who parked the call if the call is not picked up after a specified period of time.

The administrator can use [Park Recall Time (sec)] in the [CONFIGURATION > User Group > Change User Group > Timers] menu to specify the time for which parked calls will remained parked before being redirected.

If the redirected call is not answered, the call is redirected to the operator. For more information, see the "Operator Group — Operator Recall" section of "4.1. System Features".

Call Pickup

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

Direct Call Pickup

This feature allows you to pick up another user's incoming call by specifying the user's number.

To use the direct call Pickup feature, the following items must be configured.

The [Direct Call Pickup] feature code must be defined in the [CONFIGURATION > Service > Feature Code].

The user can press the direct call Pickup feature code + the number of the user whose phone is ringing to pick up the other user's incoming call which is currently ringing.

Group Call Pickup

If this feature is enabled, you can specify a call Pickup group number (instead of a user number) for which an incoming call is ringing to pick up the current incoming call for the selected group. You can also Pickup an incoming call ringing for your own call Pickup group.

To use the group call Pickup feature, the following three items must be configured.

The [Group Call Pickup] feature code must be defined in the [CONFIGURATION > Service > Feature Code].

Call Pickup groups must be configured in the [CONFIGURATION > Service > Group

Service > Pickup Group] menu. The following data must be configured.

Item	Description
User Group	Select a user group for which the call Pickup group will be created.
Group Number	Enter a number for the call Pickup group.
Group Name	Enter a name for the call Pickup group.
Ping Ring	Select a notice about no-answer for an incoming call during some period. Disable: There is no notice about no-answer call. Enable: There is a notice about no-answer call.
Ping Ring Time(sec)	Enter time to wait to make a notice to the members about no-answer call.
Group Member	Select members for the call Pickup group. A user can belong to one call Pickup group only.

The user can press the group call Pickup feature code + the number of the call Pickup group whose phone is ringing to pick up the group's incoming call which is currently ringing. Or, the user can dial just the group call Pickup feature code without a call Pickup group number to pick up the current incoming call for his/her own call Pickup group.

Ping Ring

SCM provides Ping Ring service for Group Call Pickup. When nobody picked up an incoming call ringing for own call Pickup Group during Ping Ring Time, members of the Pickup Group receive a notify to let know there is a incoming call to pickup. The Ping Ring Service set 'Enable' in the [CONFIGURATION > Service > Group Service > Pickup Group] menu.

Ping Ring Time can be configured in the [Pickup Group] as described below. If there is no configuration, SCM uses Ping Ring Time in the [CONFIGURATION > User Group > Change User Group > Timer].

Each member of Pickup Group can select a notify type for Ping Ring. It can be configured in the [Ping Ring Type] of [CONFIGURATION > User > Single Phone User].

Outbound Call Lock

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

To use Outbound Call Lock feature, the following items must be configured.

The [Outbound Call Lock] service must be enabled in [CONFIGURATION > Service

> Feature Service > Class of Service].

The [Outbound Call Lock-Set] and [Outbound Call Lock-Cancel] feature codes must be defined in the [CONFIGURATION > Service > Feature Service > Feature Code].

The user can register or cancel the Outbound Call Lock feature in the following ways:

The user can dial the "Outbound Call Lock -Set" feature code + password to enable call restriction.

The user can dial the "Outbound Call Lock -Cancel" feature code + password to cancel call restriction.

The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the call restrict by user feature.

The user can use the [User Information] menu on SCM Personal Assistant to change the password entered on the phone when enabling or canceling call restrict by user.

Call Transfer

The call transfer feature allows the user to put on hold the current call and transfer it to another number. If call transfer fails, the call is reconnected to the user who transferred the call.

To use the call transfer feature, the [Call Transfer] service must be enabled in [CONFIGURATION > Service > Feature Service > Class of Service].

Call Transfer Methods

The user can transfer calls in the following three ways.

Blind Transfer

This feature allows the user to transfer the call directly to another number without hold it. Although 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 transfer button, call another number, and then transfer the call by pressing the transfer button again while the phone is ringing.

Consultative Transfer

This feature allows the user to put on hold the current call by pressing the transfer button, call another number, and then transfer the call by pressing the transfer button again after the call is established.

Transfer Recall

This feature allows the transferred call to be redirected to the user who transferred the call when call transfer fails or when the transferred call is not answered.

The administrator can use [Transfer Recall Time (sec)] in the [CONFIGURATION > User Group > Change User Group > Timers] menu to specify the time after which the transferred call is redirected. If the user to whom the call is transferred does not answer the call during this period of time, the call is redirected to the user who transferred the call.

If the redirected call is not answered, the call is redirected to the operator. For more information, see the “Operator Group — Operator Recall” section of “4.1. System Features”.

Transfer Target Display for Transfer Recall

Transfer Ring Back Tone

SCM can play MOH itself or transparently deliver the media played from remote side to held party when blind or semi-blind transfer services.

The administrator needs to set [Transfer Ringback Tone] in the [CONFIGURATION > User Group > Change User Group > Options] menu.

Item	Description			
	When transfer target is subscriber		When transfer target is trunk	
Options	Use external ringback tone server	Not using external ringback tone server	Trunk side provides ringback tone	Trunk side doesn’t provides ringback tone
Internal Ringback Tone	External server plays	SCM plays ringback tone	SCM plays ringback tone	SCM plays ringback tone
MOH	Music On Hold	Music On Hold	Music On Hold	Music On Hold
External Ringback tone	External server plays	SCM plays ringback tone	Trunk plays ringback tone	SCM plays ringback tone

Call Waiting

If the call waiting feature is enabled, when there is an incoming call while the user is already engaged, the call is not terminated as a call when busy, but instead the user is notified that a call is waiting so that the user can park or end the previous call and Pickup the new call.

If there is an incoming call while the user is already engaged, a brief call waiting tone will be played for the user. If the user presses the call button to answer the new call, the previous call is automatically parked.

If the call waiting feature is enabled for a phone, the phone can receive all the calls it can accommodate. But if the call waiting feature is not enabled, all incoming calls while the phone is engaged are terminated as calls when busy.

If the call waiting feature is enabled for a phone, the phone can accommodate as many calls as the call buttons configured. If no call button is configured, all incoming calls while the phone is engaged are treated as calls when busy.

To use the call waiting feature, the following items must be configured.

- The “Call Waiting” service must be enabled in Class of Service.
- The “Call Waiting -Set” and “Call Waiting -Cancel” feature codes must be defined.

The user can register or cancel the call waiting feature in the following ways:

- The user can dial the “Call Waiting -Set” feature code on the phone to register the call waiting feature.
- The user can dial the “Call Waiting -Cancel” feature code on the phone to cancel the call waiting feature.
- The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the call waiting feature.

Call Intercept

Call Intercept can be provided under the 3-way conference by the Barge-In service. A User can make 2-way call with the party which is barged in by pressing Call Intercept soft key.

The menu “Call Intercept” and “Forced Call Release” will be displayed at the user who barged-in. When the user presses “Intercept” button, the opponent of barged-in user will be dropped and the barged-in target and the user who barged-in is directly connected.

Forced Call Release

Forced Call Release can be provided under the 3-way conference by the Barge-In service. By pressing Force Call Release soft key, a user can make 2-way call by releasing the party barged in.

The menu “Call Intercept” and “Forced Call Release” will be displayed at the user who barged-in. When the user presses “Forced Call Release” button, the barged-in target will be dropped and the opponent of barged-in target and the user who barged-in is directly connected.

CLI Control

Temporary CID Restriction

The temporary CID restriction feature allows the user to request that his/her number is not shown to the called party for a particular call.

To use the temporary CID restriction feature, the following items must be configured.

The "Temporary CID Restriction" service must be enabled in Class of Service.

The "Temporary CID Restriction" feature code must be defined.

When making a call, the user can dial the temporary CID restriction feature code + called party's phone number to request temporary CID restriction.

Distinctive Ring by CLI

The distinctive ring by CLI feature allows incoming calls to be distinguished by ringing different rings depending on the caller numbers.

The user can use the [Distinctive Ring] menu on SCM Personal Assistant to register a list of caller numbers for which distinctive ring by CLI will be used.

The administrator can use the [CONFIGURATION > Service > User Service > Distinctive Ring] menu to register a list of caller numbers for which distinctive rings will be serviced.

Do Not Disturb (DND)

When the 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.

To use the DND feature, the following items must be configured.

- The "DND (Do Not Disturb)" service must be enabled in Class of Service.
- The "DND (Do Not Disturb) -Set" and "DND (Do Not Disturb) -Cancel" feature codes must be defined.

The user can register or cancel the DND feature in the following ways:

- The user can dial the "DND (Do Not Disturb) -Set" feature code on the phone to register the DND feature.
- The user can dial the "DND (Do Not Disturb) -Cancel" feature code on the phone to cancel the DND feature.
- The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the DND feature.

DND White List

When there is an incoming call for a user with DND, this service prevents the call from getting rejected if the call originates from one of the caller numbers specified in advance.

The user can use the [DND White List] menu on SCM Personal Assistant to register a list of caller numbers to exclude from the DND service.

The administrator can use the [CONFIGURATION > Service > User Service > DND White List] menu to register a list of caller numbers which will be excluded from the DND service.

Item	Description
User Group	Select a user group for which the DND white list will be created.
Extension Number	Select an extension number for which the DND white list will be created.
White List	Enter the caller numbers for the DND white list.

DND Override

When there is an incoming call for a user with DND, this service allows the caller, while listening to the DND announcement, to ignore the DND status and have his/her call connected.

To use the DND override feature, both "Override Level" and "Privacy Level" must be defined in Class of Service. DND override is allowed only when the override level is higher than the privacy level. The override level is applied to the user overriding DND and the privacy level is applied to the user with DND.

To use the DND override feature, the following items must be configured.

- The "DND Override" service must be enabled in Class of Service.
- The "DND Override" feature code must be defined.

When there is an incoming call for a user with DND, the caller, while listening to the DND announcement, can press the DND override button to override DND and the called party's phone will ring. Even when the DND override feature is used, the call may not be connected if the called party is busy or otherwise unavailable.

Follow Me

When the caller has temporarily moved to another location, this service allows the caller to answer all incoming calls to his number by using another phone..

To use the follow me feature, the following items must be configured.

The “Follow Me” service must be enabled in Class of Service.

The “Follow Me -Set” and “Follow Me -Cancel” feature codes must be defined.

The user can register or cancel the follow me feature in the following ways:

The user can dial the “Follow Me -Set” feature code + his/her password + his/her phone number on another phone to enable follow me.

The user can dial the “Follow Me -Cancel” feature code + his/her password + his/her phone number on another phone to cancel follow me.

If follow me is enabled, it will be shown as enabled in the [Supplementary Service] menu on SCM Personal Assistant, where the feature can be canceled.

The user can use the [User Information] menu on SCM Personal Assistant to change the password entered when enabling or canceling follow me.

Individual Speed Dial

This feature allows you to assign a shortcut number to a phone number frequently dialed. The user should dial [Individual Speed Dial] feature code + [Speed Dial ID].

A speed dial ID is just a one digit. SCM supports 10 individual speed dial for each users.

To use this feature, it needs feature code configurations as follow :

- “Individual Speed Dial – Call” feature code
- “Individual Speed Dial Number – Insert” feature code
- “Individual Speed Dial – Delete” feature code

To set individual speed dial using phone, user should dial “Individual Speed Dial Number – Insert” feature code and speed dial ID(0~9) and dialing number on the phone.

To delete individual speed dial set, user should dial “Individual Speed Dial Number – Delete” feature code and speed dial ID(0~9) on the phone.

Intercom

When a call is made between the users for whom intercom is enabled, the call is automatically answered through the speaker. When using the manager/secretary feature, the intercom feature is used together.

To use the intercom feature, the following items must be configured:

- An intercom number must be specified for each user using the [CONFIGURATION > Service > User Service > Intercom] menu.
- The “Intercom” feature code must be defined.

A user with an intercom number can dial the intercom feature code + user number to page the selected user for a call.

The following items must be configured in the [CONFIGURATION > Service > User Service > Intercom] menu.

Item	Description
User Group	Select a user group for which the intercom will be created.
Extension Number	Select an extension number for which the intercom will be created.
Name	Specify a name for the intercom.
Intercom	Select an extension number which will be connected with the intercom.

Language Selection

This service allows the user to change the language displayed on their phone.

The user can use the [User Information] menu on SCM Personal Assistant to change his/her language.

The administrator can change the language of a Single Phone User by changing [Language] in the [CONFIGURATION > User > Single Phone User] menu.

The administrator can change the language of a Multi-Extension Phone by changing [Language] in the [CONFIGURATION > User > Multi-Phone User] menu.

Last Number Redial

The last number redial feature allows the user to redial the caller or the called party number of the most recent call.

For Call Forward All and Multi-Ring services, the number user dialed initially will be used as called number not the number finally reached.

The last number redial service allows redialing the called number of the last outgoing call or the caller number of the last incoming call. The following must be configured.

The “Last Outgoing Redial” feature code must be defined in order to be able to redial the last dialed number.

The “Last Incoming Redial” feature code must be defined in order to be able to redial the caller number of the last incoming call.

The user can use the last call redial feature in the following ways:

The user can dial the last outgoing feature code to redial the called number of the last outgoing call.

The user can dial the last incoming redial feature code to redial the caller number of the last incoming call.

Last Call Redial is independently operated by each node.

No Ring

The no ring feature prevents the phone from ringing when there is an incoming call for the user. This service is useful to prevent some phones from ringing when multiple phones are configured to ring at the same time by features such as multi-ring and multi-device.

To use the no ring feature, the following items must be configured.

- The “No Ring” service must be enabled in Class of Service.
- The “No Ring -Set” and “No Ring -Cancel” feature codes must be defined.

The user can register or cancel the no ring feature in the following ways:

- The user can dial the “No Ring -Set” feature code on the phone to register the no ring feature.
- The user can dial the “No Ring -Cancel” feature code on the phone to cancel the no ring feature.
- The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the no ring.

Multi-Ring

If the multi-ring feature is enabled, when there is an incoming call for the user, the call is directed to multiple phones at the same time, and when the call is answered by one of the phones, the call is connected to the phone and the ring on other phones are canceled. This service is useful for incoming calls to ring the landline and the mobile phone to ring at the same time.

To use the multi-ring feature, the following items must be configured:

The “Multi-Ring” service must be enabled in Class of Service.

The “Multi-Ring -Set” and “Multi-Ring -Cancel” feature codes must be defined.

Even if a user is set as a member on a multi-ring list, the multi-ring feature must be enabled for the user in order to use the multi-ring feature. The user can register or cancel the multi-ring feature in the following ways:

The user can dial the “Multi-Ring -Set” feature code on the phone to enable the multi-ring feature.

The user can dial the “Multi-Ring -Cancel” feature code on the phone to disable the multi-ring feature.

The administrator can add members to multi-ring lists by using the [CONFIGURATION > Service > Subscriber Service > Multi-ring List] menu.

The user can use the feature code to add members to multi-ring lists. For the user to add multi-ring members, the “Multi-Ring -Insert” and “Multi-Ring -Delete” feature codes must be defined.

The user can add or remove multi-ring members by using feature codes in the following ways:

The user can dial the add multi-ring member feature code + phone number to add a multi-ring member.

The user can dial the remove multi-ring member feature code + phone number to remove a multi-ring member.

When there is an incoming call, services enabled for the master user who enabled multi-ring will be provided, but the services enabled for the multi-ring members will not be provided. Note that the no ring service is provided to all users.

For example, when the user number 2000 is set as a multi-ring member for the user number 1000, if there is an incoming call for the user number 1000:

The call will be forwarded if call forwarding is enabled for the user number 1000, but the call will not be forwarded if call forwarding is enabled for the user number 2000.

The call will be rejected if DND is enabled for the user number 1000, but the call will not be rejected if DND is enabled for the user number 2000.

Only the user number 1000 will not ring if no ring is enabled for the user number 1000, and only the user number 2000 will not ring if no ring is enabled for the user number 2000.

When the master user who enabled multi-ring is busy, the incoming call is serviced according to the [Allow Other Ring] setting which is accessible through the pop-up window when the Enable button is selected after selecting the [CONFIGURATION > Service > Feature Service > Service Activation > Multi-ring] menu.

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.

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.

MOBEX Call Pickup

This service allows the call answered with an external mobile phone by the multi-ring feature to be transferred to the landline in the office and picked up for continued conversation.

To use the MOBEC Call Pickup feature, the “MOBEX Call Pickup ” feature code must be defined.

The user can dial the call Pickup on desk phone feature code on the master phone enabled with multi-ring to pick up the call from the mobile phone.

After answering an incoming call with a mobile phone enabled with multi-ring, the user can press the “MOBEX Call Pickup ” feature code on his/her master phone during the call to transfer the call to the master phone.

Transfer to Mobile Phone

This service allows the user to transfer a call to an external mobile phone specified as a multi-ring member without parking the call. It works in the same way as blind transfer.

The user can dial press the transfer button on the master phone enabled with multi-ring during a call to transfer the current call to the mobile phone.

To transfer a call, press the transfer button and a mobile phone number on the master phone during the call and end the call.

Remote Office

The remote office feature allows automatic forwarding of all incoming calls for a user to an internal number or an external number specified.

The remote office feature works in the same way as blind transfer but it is defined for remote use. It is also similar to the follow me to destination feature but it is different in that the calls can be forwarded to phone numbers outside the system.

To use the remote office feature, the following items must be configured.

The “Remote Office” service must be enabled in Class of Service.

The “Remote Office -Set” and “Remote Office -Cancel” feature codes must be defined.

The user can register or cancel the remote office feature in the following ways:

The user can dial the “Remote Office -Set” feature code + the destination phone number on the phone to register the remote office feature.

The user can dial the “Remote Office -Cancel” feature code on the phone to cancel the remote office feature.

The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the remote office feature.

Wake-Up Call

The wake-up call feature allows the user’s phone to ring at a wake-up time specified by the user. If the user answers the call, an announcement is played to notify that it is the wake-up time.

To use the wake-up call feature, the following items must be configured:

The “Wake-Up Call” service must be enabled in Class of Service.

The “Wake-Up Call -Set” and “Wake-Up Call -Cancel” feature codes must be defined.

The user can register or cancel the wake-up call feature in the following ways:

The user can dial the “Wake-Up Call -Set” feature code + the wake-up retry type (1: Once/2:Repeat) + the wake-up time on the phone to enable the wake-up call feature.

The user can dial the “Wake-Up Call -Cancel” feature code + the wake-up time (HHMM) on the phone to cancel the wake-up call feature.

The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the wake-up call feature. When enabling wake-up call, you can also set the wake-up time and the number of times the phone will ring.

운영자가 Wake-Up 서비스를 설정하는 경우는 [Configuration > Service > User Service > Wake-Up Call Lists] 메뉴에서 다음 항목을 설정해야 합니다.

The following items should be configured for wake-up service by operator in the [CONFIGURATION > Service > User Service > Wake-Up Call Lists] menu.

Item	Description
User Group	Select a user group for which the wake-up call will be created.
Extension Number	Select an extension number for which the wake-up call will be created.
Wake-Up Index	If subscriber needs more than 2 wakeup services, ‘Extended Alarm Reminder’ feature can be used. This feature is aimed to support hotel service. It supports 4 wakeup services per subscriber. So there is an index (1-4) to separate which one is set or cancel. ‘1’ is a default index for wakeup feature.
Wake-Up Retry Type	Select a service count for Once or Repeat
Wake-Up Time	Input wake-up ringing time. If service count is Once, should be input MMDDHHMM (Month, Date, Hour, Minute) or HHMM (Hour, Minute) type

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.

The administrator needs to set [Application Type] to "Internal UMS" in the [CONFIGURATION > Application > VM/AA Server] menu to create a connection to the built-in voice mail system.

There are three services for VM/AA server. The administrator can make several VM/AA servers. But only one service can only appoint one service.

- ✓ Voice Mail: All services except for Call Recording and Auto Attendant
- ✓ Call Recording
- ✓ Auto Attendant

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

The administrator needs to set [Application Type] to "Internal Conference" in the [CONFIGURATION > Application > Conference Server] menu to create a connection to the built-in conference system. This conference server should be included "Add-On Conference" in service list.

Answering Machine Emulation (AME)

If the AME feature is enabled, when there is an incoming call, the call is automatically answered by the voice mail system and the caller's message is recorded in the mailbox. The voice mail system announcement and the caller's voice message are heard over the phone's speaker.

AME Auto Start

This method allows the incoming calls to be connected to AME by configuring the AME feature in advance.

To use the AME auto start feature, the following items must be configured:

The "Answering Machine Emulation" service must be enabled in Class of Service.

The "AME-Enable" and "AME-Disable" feature codes must be defined.

The user can register or cancel the AME auto start feature in the following ways:

The user can dial the "AME-Enable" feature code on the phone to enable the AME auto start feature.

The user can dial the "AME-Disable" feature code on the phone to cancel the AME auto start feature.

The user can use the [Supplementary Service] menu on SCM Personal Assistant to register (enable) or cancel the AME auto start feature.

If the AME auto start feature is enabled, when there is an incoming call and the call is not answered, the call forward no answer feature is used for forwarding the call to the voice mail system to automatically start the AME.

AME Manual Start

This method allows the incoming calls to be connected to AME without configuring the AME feature in advance.

To use the AME manual start feature, the following items must be configured:

The “Answering Machine Emulation” service must be enabled in Class of Service.

The “AME -Manual Start” and “AME -Manual Stop” feature codes must be defined.

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.

To use the auto call record feature, the following items must be configured:

The “Call Recording” and “Auto Record” services must be enabled in Class of Service.

When a user for whom the auto call record feature is enabled is on a call, a three-way conference call will automatically be connected to the voice mail system and the call will be recorded.

When enabling the auto call record feature, you can specify a type of calls to record selectively.

The call type can be changed by [Auto Record Type] in the [CONFIGURATION > Service > Feature Service > Service Activation > Auto Record] menu.

Incoming: Incoming calls are recorded.

Outgoing: Outgoing calls are recorded.

Both: Both the incoming and outgoing calls are recorded.

Manual Call Record

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

To use the manual call record feature, the following items must be configured:

The “Call Record” service must be enabled in Class of Service.

The “Call Record” feature code must be defined.

If the user presses the “Call Record” button + the mailbox number during a call, a three-way conference call will be established with the voice mail system and the call will be recorded in the selected mailbox. If a mailbox number is not entered, the call will be recorded in the user’s mailbox.

Deflect to Voicemail

This service forwards allows the currently ringing call to be forwarded to the voice mail system by using the call forward no answer feature.

The voice mail system answers the call immediately and plays the no answer announcement so that the caller can leave a voice mail.

To use the deflect to voice mail feature, the “Deflect to Voicemail” feature code must be defined.

If the user presses the deflect to voice mail button on the phone which is ringing, the call will be processed for call forward no answer and be connected to the voice mail system.

Transfer to Voicemail

This feature allows the current call to be connected to a specified mailbox in the voice mail system so that the caller can leave a message.

If the current call is transferred to the voice mail system by a normal method, the voice mail system asks for the service code, mailbox number, password, etc. But if the transfer to voice mail feature is used for transferring the call, this step is skipped so that the caller can leave a voice message without entering anything.

To use the transfer to voice mail feature, the “UMS Transfer” feature code must be defined.

When the user dials the transfer to voice mail feature code + a mailbox number during a call and ends the call, the call will be transferred to the voice mail system and the caller will be allowed to leave a voice mail in the selected mailbox.

Personal SPAM Number

This feature allow user to configure SPAM numbers. When a call comes in from internal or external to the user, if the number is matched with pre-configured SPAM list then reject the call.

To use Personal SPAM Number feature, following items are required in the [Configuration > Service > User Service > Personal SPAM Number] Menu.

Item	Description
User Group	Select a user group for which the Personal SPAM number will be configured
Extension Number	Specify a user number for which the service will be activated.
SPAM Number	Enter the number specified as SPAM number.
Activation	Select ‘Yes’ or ‘No’ to use the Number as SPAM.

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’)

In the [Configuration>User Group>Timers>Pause Delay Time], the delay time between pause digits can be changed. And the delay time between normal digits after pause digit can be changed in the [Configuration>User Group>Timers>DTMF Duration Time].

This feature is served with the following services.

- Hot Line
- Speed Dial
- Call Forward
- Multi-Ring
- Paging on Answer
- Predefined Conference

The destination number for services should include pause digits. For the detailed configuration for services, refer to the menus of each service.

ABBREVIATION

A

AA	Auto Attendant
AAR	Automatic Alternative Routing
ACD	Automatic Call Distribution
AR	Alternative Route

B

BHCA	Busy Hour Call Attempt
BLF	Busy Lamp Field

C

CAC	Call Admission Control
CDR	Call Detailed Record
CLI	Calling Line Identification
CLIR	Calling Line Identification Restriction
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

I

ITSP	Internet Telephony Service Provider
IVR	Interactive Voice Response

L

LDAP	Lightweight Directory Access Protocol

M

MCS	Multimedia Conference System
MOH	Music On Hold
MWI	Message Waiting Indication

N

NMS	Network Management System

P

PBX	Private Branch eXchange
PSTN	Public Switched Telephone Network

R

RADIUS	Remote Authentication Dial In User Service
RFC	Request For Comments
RTP	Real Time Protocol

S

SBC	Single Board Computer
SCM	Samsung Communication Manager
SIP	Session Initiation Protocol
SNMP	Simple Network Management Protocol

T

TLS	Transport Layer Security

U

UMS	Unified Messaging System

V

VMS	Voice Mailing System
VoIP	Voice over Internet Protocol