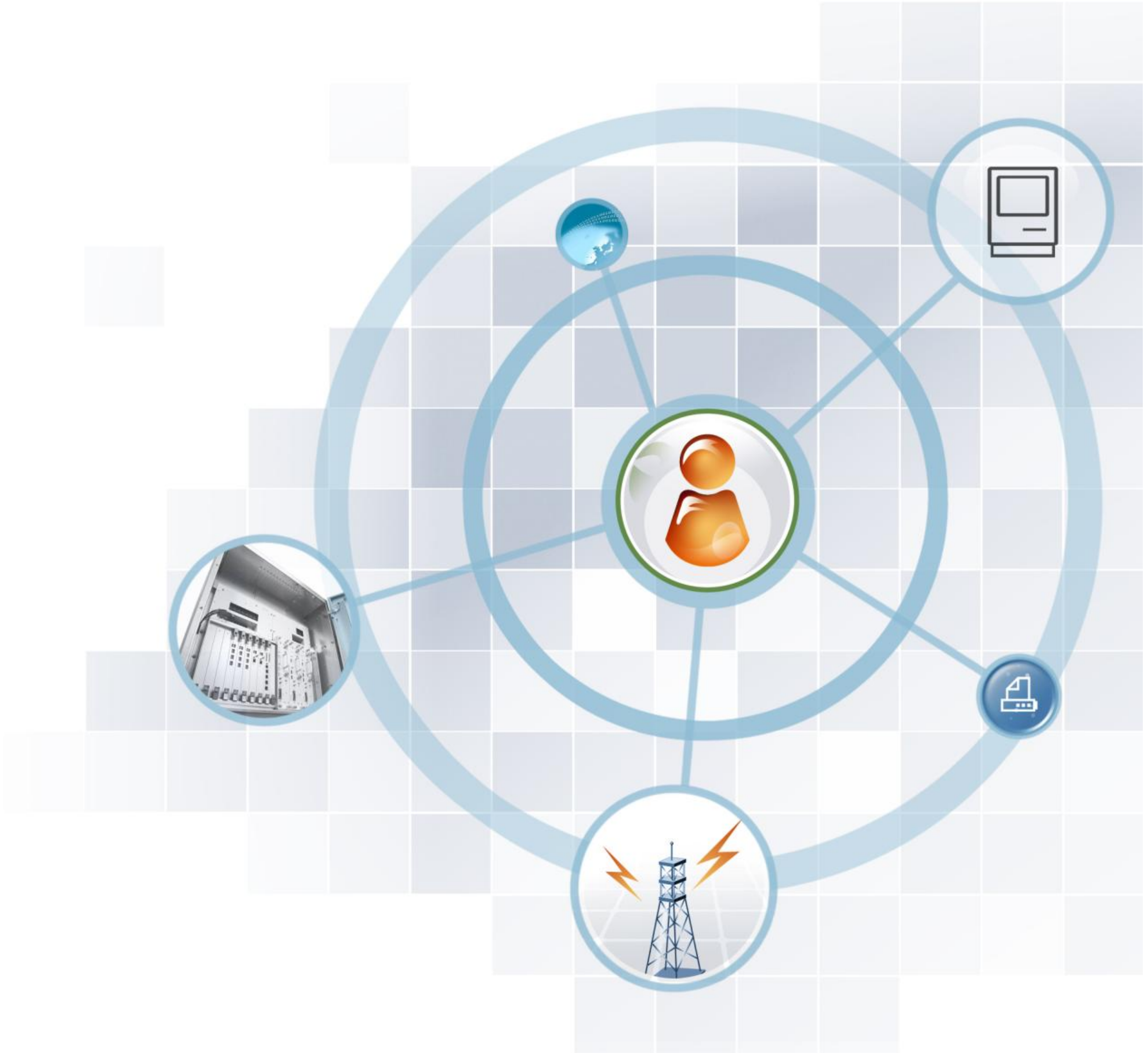
Auto Doc.code Ver. 1.0

**SCM Express**

# SIP Interoperability Guide



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## COPYRIGHT



## TRADEMARKS



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# INTRODUCTION

## Purpose

This manual describes SIP Interoperability Specification.

## Document Content and Organization

This document consists of three the following chapters.

### CHAPTER 1. SIP Trunk Interoperability

The part describes SIP Trunk Interoperability.

### CHAPTER 2. SIP Phone Interoperability

The part describes SIP Phone Interoperability.

### CHAPTER 3. SIP Application Interoperability

The part describes SIP Application Interoperability.

### ABBREVIATION

Provides the definitions of the abbreviations used in this manual.

## Conventions

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

WARNING

**WARNING**

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

CAUTION

**CAUTION**

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

CHECK

**CHECKPOINT**

Provides the operator with checkpoints for stable system operation.

NOTE

**NOTE**

Indicates additional information as a reference.

**OPERATION PROCEDURES**

Indicates the operational procedures that should be executed in sequence.

## Console Screen Output

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

## Revision History

| VERSION | DATE OF ISSUE | REMARKS |
|---|---|---|
| 1.0 | 12. 2012. | First edition for 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**<br>Indication of a general caution |
| | **Restriction**<br>Indication for prohibiting an action for a product |
| | **Instruction**<br>Indication for commanding a specifically required action |

# WARNING

| | |
|---|---|
| | When the JRE and Web Start versions are different, the SCM system operation can become slow or malfunction. The programs must be installed according to the SCM GUI installation procedures. |

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# CHAPTER 1. SIP Trunk Interoperability

## 1.1 Introduction

### 1.1.1 Network Topology

**Private Network with SBC**

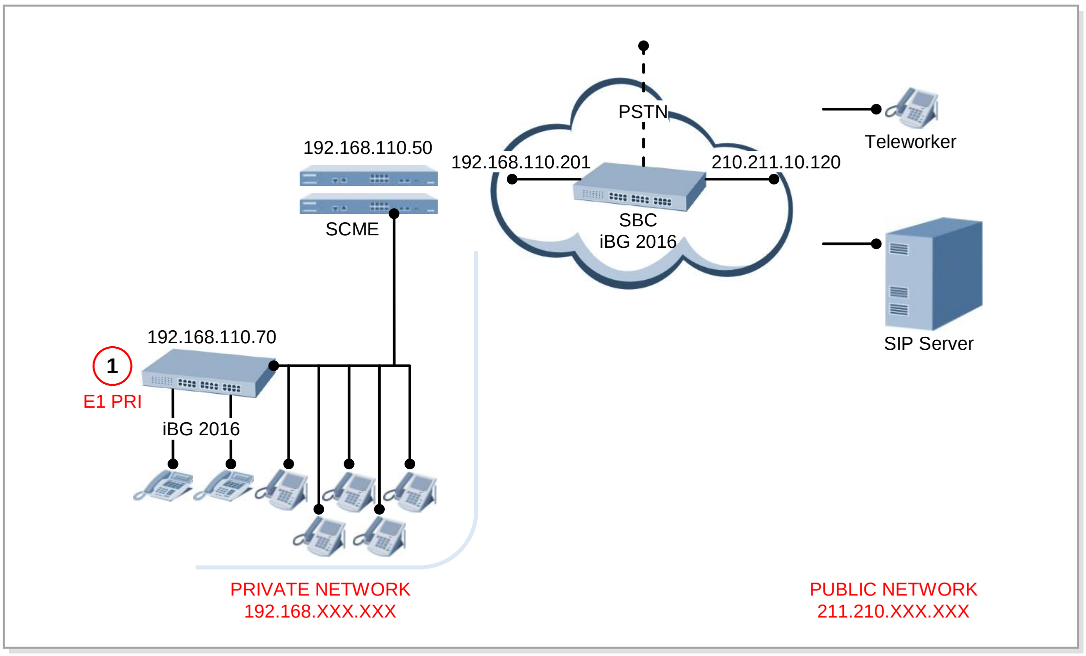


**Figure 1.1 Private Network with SBC**

This topology is the best recommended. SCM Express is located in Private network and interwork with SIP Trunk in public network through SBC. Therefore, this network configuration improves security.

SBC control all SIP signal between private and public network.

### Private Network with MPS and Router

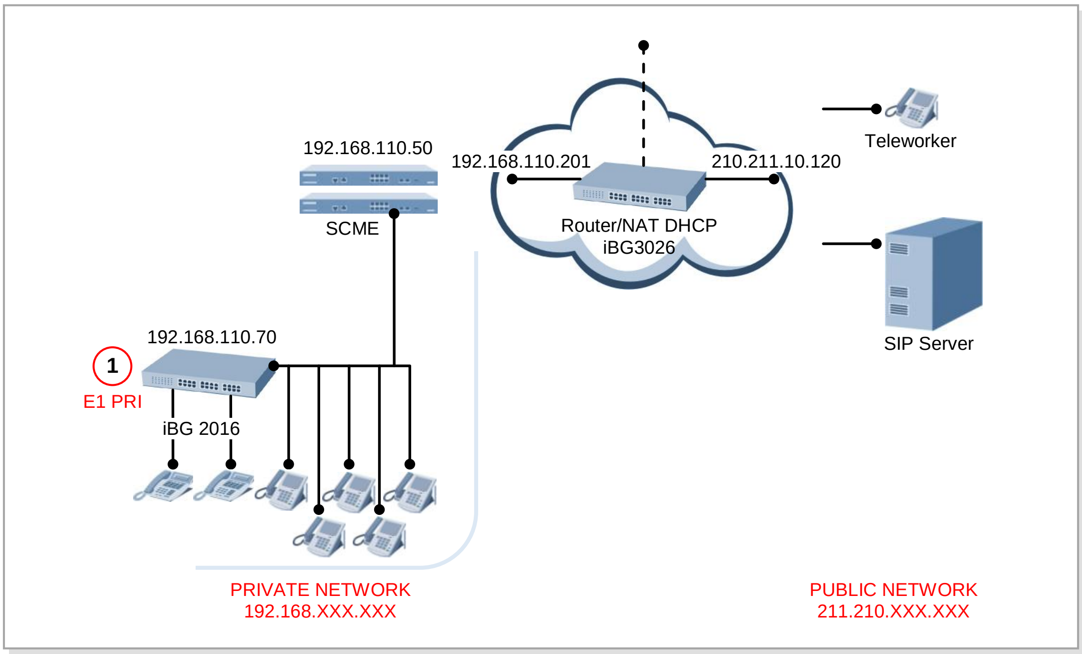


**Figure 1.2 Private Network with MPs and Router**

SCM Express is located in Private network and interwork with SIP Trunk in public network through the Router. In this configuration, SCM Express provides SIP NAT Traversal function for SIP signaling and RTP media. Router should have control rule for SIP port and MPS port range for this function.
By the way MPS have limitation for channel number. So when MPS channel can cover all users, it is useful.

### Public Network

SCM Express is located in Public network and interwork SIP Trunk. But it has security risk. So this topology is not recommended.

### 1.1.2 Supporting Protocol

- RFC 768 User Datagram Protocol (UDP)
- RFC 793 Transmission Control Protocol (TCP)
- RFC 3261 SIP: Session Initiation Protocol
- RFC 3262 Reliability of Provisional Responses in SIP
- RFC 3263 Locating SIP Servers
- RFC 3264 An Offer/Answer Model with the Session Description Protocol
- RFC 3515 The Session Initiation Protocol (SIP) Refer Method
- RFC 2327 SDP: Session Description Protocol
- RFC 2833 RTP Payload for DTMF Digits
- RFC 2246 The TLS Protocol Version 1.0
- RFC 2976 The SIP INFO Method
- RFC 4028 Session Timers in the Session Initiation Protocol (SIP)

### 1.1.3 SIP Trunk Functionality

- Registration
- Basic Call
- Calling Lind ID and Privacy
- E.164 Numbering
- Call Hold/Resume
- Call Transfer
- Call Redirect (Forwarding)
- Call Conference
- TLS, SRTP
- Session Timer
- DTMF

# 1.2 Configuring SIP Trunk

## 1.2.1 Overview

This chapter describes about basic configuration and function for SIP Trunk interworking.
The Basic Configurations are two, SIP Server configuration and Route configuration.
In SCM Express GUI, Priority Routing, Location Based Routing and Access Code menus are for Route configuration. Route is for SIP Server configuration.

1) Route
   Set SIP server various information. It includes account, server address and protocol.
2) Priority Routing
   A priority routing allows automatic selection of alternative routes when the default outgoing path becomes unavailable. Priority is assigned to the direct route and alternative routes. When calls going out through the high-priority route fail, they can be retried through the low-priority routes.
3) Location Based Routing
   A load-balanced route allows use of the selected routes in a specified ratio. Calls are distributed between the routes identified as available for calls, and therefore there is no need for configuring alternative routes.
4) Access Code
   Access codes are used for directing outgoing calls to trunks instead of extension numbers. They are also used for analyzing the destination numbers to determine which location-based routes to use for outgoing trunk calls.
5) DID Routing
   By default, incoming trunk calls are processed according to the routing settings by DID number.
   When entering a DID number, you can use wild cards (entered by *) to enter multiple numbers at a time. If the called number is set to ‘B’, a translated DID number is used as the called number.
   Also the called number is set to ‘E’, a translated DID number is used as the called number. Digit difference is allow tandem call or not.

## 1.2.2 Basic Configuration

Following shows GUI menu that creating a new Route. The mandatory items are Route Type, User Group, Route Name, Location, Register Type, Proxy Server, Port and User Name. If input Access Number with these, Priority Routing, Location Based Routing, Access Code are created automatically.

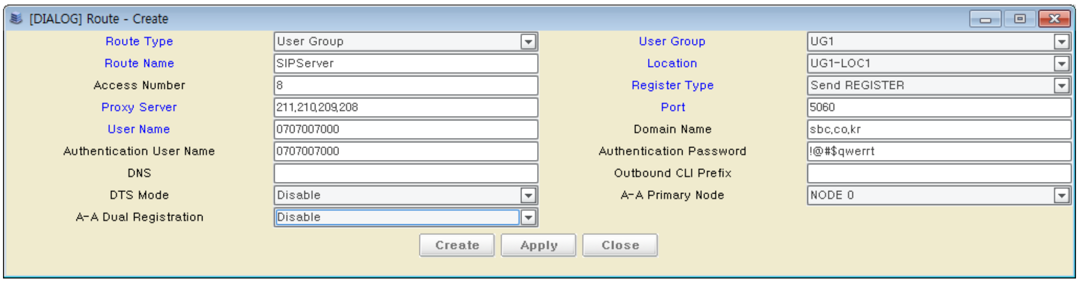


**Figure 1.3 Basic Configuration-Route**

Priority Routing

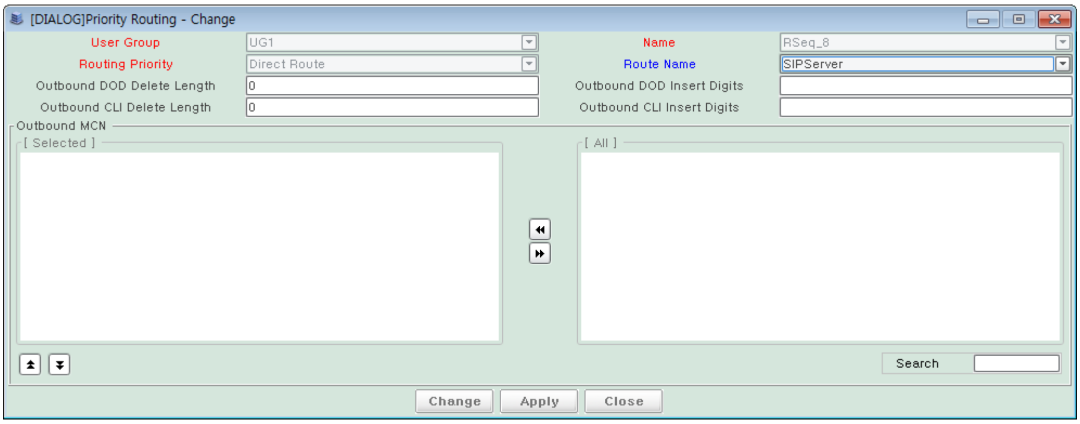


**Figure 1.4 Basic Configuration-Priority Routing**

Location Based Routing

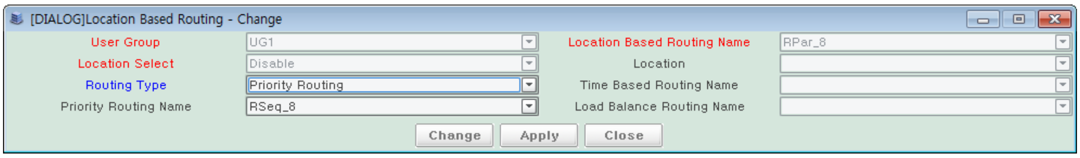


**Figure 1.5 Basic Configuration-Location Based Routing**

Access Code

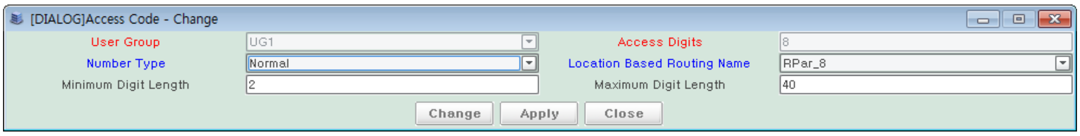


**Figure 1.6 Basic Configuration-Access Code**

The rule of input Proxy Server Address in Route Menu
Proxy Server and Domain name are the key that identity each Route. So the combination of two items has restriction.

1) Domain name must be unique.
2) Proxy Server is not duplicated originally but if Domain name is different, it can be allowed to have same Proxy Server value.

### 1.2.3 Registration

From SIP server's perspective, registration process has two meanings. One is to authenticate interacting SIP UAs, and the other is to locating SIP UAs. Though detailed registration processes may vary server to server, every server has to have those two mechanisms to provide VoIP services.

There are two types of Registration. One is Registration Mode and the other is Static Mode. It can be configured by Configuration → Trunk Routing → Route → Register Type option

1) Registration Mode
   (1) SCM Express send REGISTER message with account and SIP Server authenticate a client (SCM Express) through registration process.
   The registration process is repeated periodically
   (2) Configuration
   Route →Register Type: Send Register
   (3) Trunk Registration
   SCM Express supports only trunk registration. Trunk Registration means that the SCM Express does a single registration, whose credential data is shared by all the SIP connections between SCM Express and an outbound SIP server.
2) Static Mode
   (1) The SIP Server and SCM Express view each other as peer networks.
   The SIP Server can be configured with the domain name of static IP address.
   In static mode, OPTIONS may be used to check the status of each other.
   (2) Configuration
   ① Route → Register Type: None
   ② Route → Keep Alive: Enable (OPTIONAL)

### 1.2.4 DNS

SCM Express is able to determine the location of the outbound SIP Server (registrar or proxy) based on the resolution of SRV and A queries. SCM Express utilizes DNS servers specified in DNS & DNS 2 fields to resolve SIP server names.

Configuration → Trunk Routing → Route

1) DNS: Primary DNS Server address
2) DNS 2: Secondary DNS Server address
3) Blacklist Expires: The Blacklist means unavailable IP among resolved IP temporary. This option is to set the period of blacklist. If a IP address insert to blacklist, the IP address cannot be used for blacklist expires.

# 1.3 SIP Trunk Service

## 1.3.1 Registration

### 1.3.1.1 Send Register

#### Description

SCM Express sends REGISTER Message to SIP Server for registration.

#### Configuration

Configuration → Trunk Routing → Route

1) Register Type: Send Register
2) Register Expires: The interval of sending REGISTER periodically.
3) User Name, Authentication User Name, Authentication Password
4) Proxy Server, Port, Protocol Type
5) DNS: If Proxy Server has domain type, input DNS Server address to obtain IP Address.

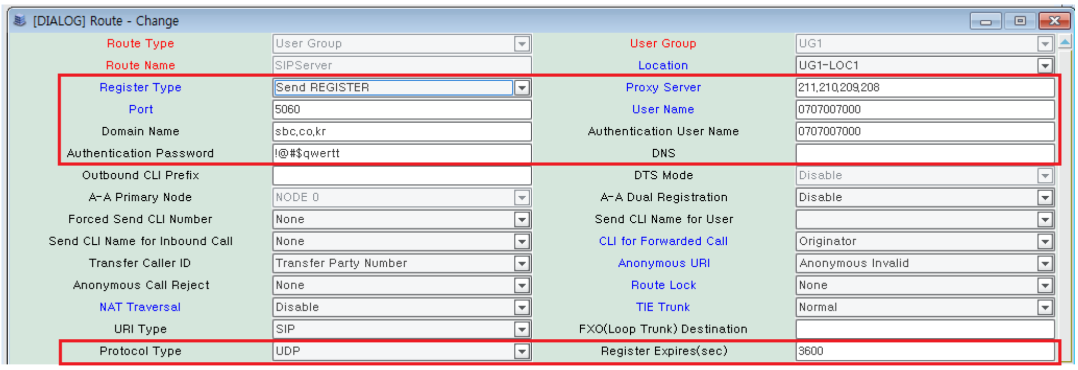


**Figure 1.7 Send Register**

#### Flow

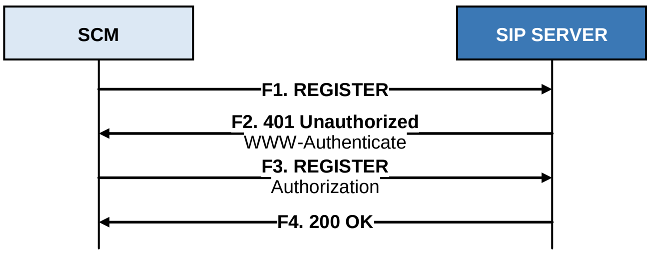


**Figure 1.8 REGISTER**

## Message

```
REG F1
REGISTER sip:sbc.co.kr SIP/2.0
From: <sip:0707007000@sbc.co.kr>;tag=c1a01c59-c25e-48d0-afcd-
9397b6313a2f
To: <sip:0707007000@sbc.co.kr>
Call-ID: CM59aACICIMEWf4EZFV80@scmex1
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-59106-
15be80ca-7600bcad
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Expires: 3600
Contact: <sip:0707007000@10.254.168.181:5060>
Content-Length: 0

REG F2
SIP/2.0 401 Unauthorized
From: <sip:0707007000@sbc.co.kr>;tag=c1a01c59-c25e-48d0-afcd-
9397b6313a2f
To: <sip:0707007000@sbc.co.kr>;tag=36b1800-0-13c4-50022-92d60-
5e392169-92d60
Call-ID: CM59aACICIMEWf4EZFV80@scmex1
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-59106-
15be80ca-7600bcad
Date: Wed, 12 Dec 2012 06:15:19 GMT
Expires: 3600
WWW-Authenticate: Digest
realm="sbc.co.kr",nonce="58292919cfb450d2fb3c84d9bd9afc87472fc021",sta
le=true,algorithm=MD5,qop="auth"
Contact: <sip:0707007000@10.254.168.181:5060>
User-Agent: SAMSUNG SCM 3.3.100
Content-Length: 0

REG F3
REGISTER sip:sbc.co.kr SIP/2.0
From: <sip:0707007000@sbc.co.kr>;tag=c1a01c59-c25e-48d0-afcd-
9397b6313a2f
To: <sip:0707007000@sbc.co.kr>;tag=36b1800-0-13c4-50022-92d60-
5e392169-92d60
Call-ID: CM59aACICIMEWf4EZFV80@scmex1
CSeq: 2 REGISTER
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-59106-
15be80ca-7600bcad
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Expires: 3600
Contact: <sip:0707007000@10.254.168.181:5060>
Authorization: Digest
username="0707007000",realm="sbc.co.kr",nonce="58292919cfb450d2fb3c84d
9bd9afc87472fc021",uri="sip:sbc.co.kr",response="dff607801bbc896ce0e01
fc7ba03dc24",algorithm=MD5,cnonce="c4f4fc814580f5ab",qop=auth,nc=00000
001
Content-Length: 0
```

```
REG F4
SIP/2.0 200 OK
From: <sip:0707007000@sbc.co.kr>;tag=c1a01c59-c25e-48d0-afcd-
9397b6313a2f
To: <sip:0707007000@sbc.co.kr>;tag=36b1800-0-13c4-50022-92d60-
5e392169-92d60
Call-ID: CM59aACICIMEWf4EZFV80@scmex1
CSeq: 2 REGISTER
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-59106-
15be80ca-7600bcad
Date: Wed, 12 Dec 2012 06:15:19 GMT
Expires: 3600
Contact: <sip:0707007000@10.254.168.181:5060>
User-Agent: SAMSUNG SCM 3.3.100
Content-Length: 0
```

## 1.3.1.2 None

### Description

SIP Server and SCM Express authenticate each other by IP address.

### Configuration

Configuration → Trunk Routing → Route

1) Register Type: None
2) User Name
3) Proxy Server, Port, Protocol Type
4) Keep Alive
   ① Keep Alive: It have EANBLE value, SCM Express send OPTIONS message to SIP Server.
   ② Keep Alive Interval
   ③ Keep Alive Retry Interval: When sends OPTIONS and receive failure response or no response, SCM Express retry to send OPTIONS after Keep Alive Retry Interval.
   ④ Keep Alive User Info: It have Enable value, From and To header have userinfo field that fill with User Name.

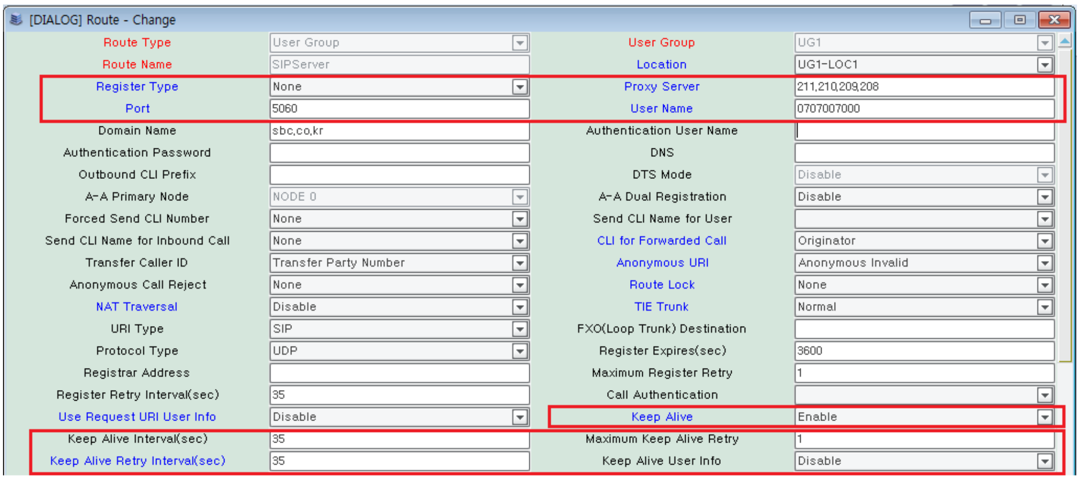


**Figure 1.9 None**

Flow

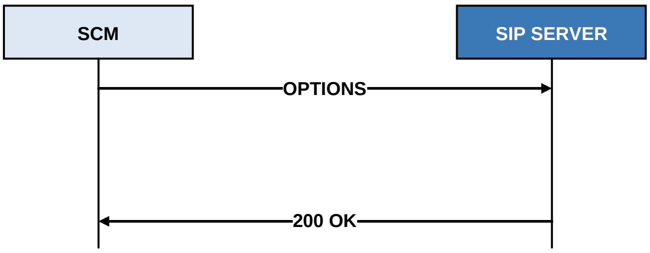


**Figure 1.10 OPTIONS**

## Message

1) Keep Alive User Info: Disable

```
OPTIONS sip:sbc.co.kr SIP/2.0
From: <sip:ug1.scm.com>;tag=c12386c8-a0b6-4bfa-92fb-14e265e54142
To: <sip:sbc.co.kr>
Call-ID: 8M55DIiios4heuL4IUIQQ@scmex1
CSeq: 1 OPTIONS
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-59f96-
15f76222-2578d6e4
Contact: <sip:10.254.168.181:5060>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Content-Length: 0
```

2) Keep Alive User Info: Enable

```
OPTIONS sip:0707007000@sbc.co.kr SIP/2.0
From: <sip:0707007000@ug1.scm.com>;tag=261fc79a-1d7f-415c-aef2-
fc61e5695575
To: <sip:0707007000@sbc.co.kr>
Call-ID: q11FOcm5OUI8EXAMFcsK5@scmex1
CSeq: 1 OPTIONS
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-5a0d3-
15fc386a-7918bda4
Contact: <sip:0707007000@10.254.168.181:5060>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Content-Length: 0
```

## 1.3.2 Authentication

### Description

When SIP Server requires authentication, SCM Express can support it. SCM support authorization for REGISTER, INVITE, re-INVITE, REFER method using MD5 algorithm.

### Flow

1) REGITSTER

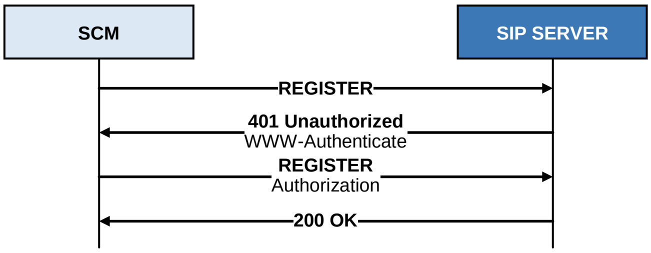


**Figure 1.11 Authentication-REGISTER**

2) INVITE

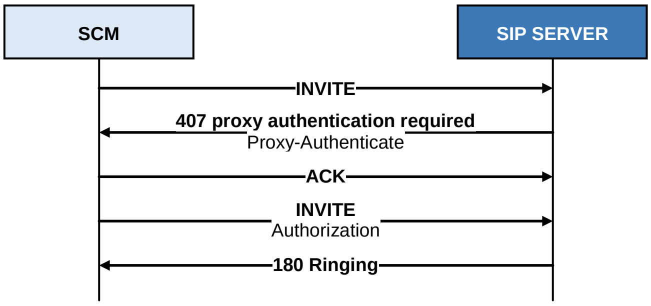


**Figure 1.12 Authentication-INVITE**

3) REFER

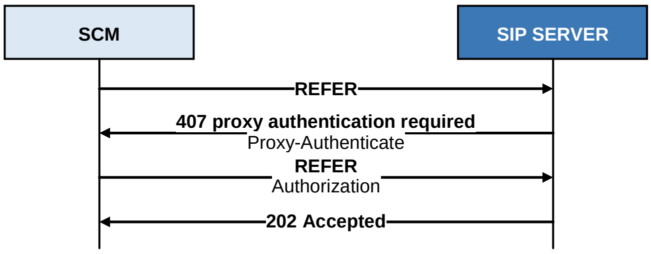


**Figure 1.13 Authentication-REFER**

### 1.3.3 Basic Call

#### Description

ITSP's SIP server will not accept any call request from SCM Express if the INVITE message does not show nor contain its valid public identity which the SIP server can recognize. In many cases, the valid identity is assigned by ITSP and also called as Primary number or Secondary number. It is shown as the caller ID in the From header in INVITE message. Some ITSP SIP server requires P-Asserted-ID header contains the Primary number and From header contains residential number. Mean while, some others require P-Asserted-ID header contains the residential number and From header contains Primary number.

#### Configuration

Configuration → Trunk Routing → Route

1) SIP-P-Asserted-ID-Type
   (1) Primary
      ① P-Asserted-Identity: User Name in Route
      ② From: residential CLI number
   (2) Secondary
      ① P-Asserted-Identity: residential CLI number
      ② From: User Name in Route
2) Use Request URI User Info: It field set to ENABLE, userinfo of Request-URI is used as called number. But generally To header is used as final recipient number.
   (1) DISABLE: Default Value. Get userinfo from To Header.
   (2) ENABLE: Get userinfo from Request-URI.

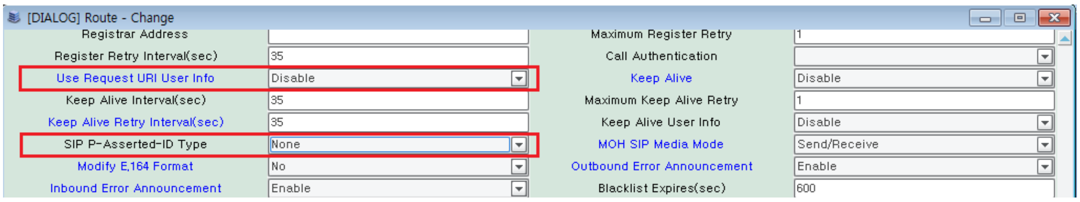


**Figure 1.14 Basic Call**

## Flow

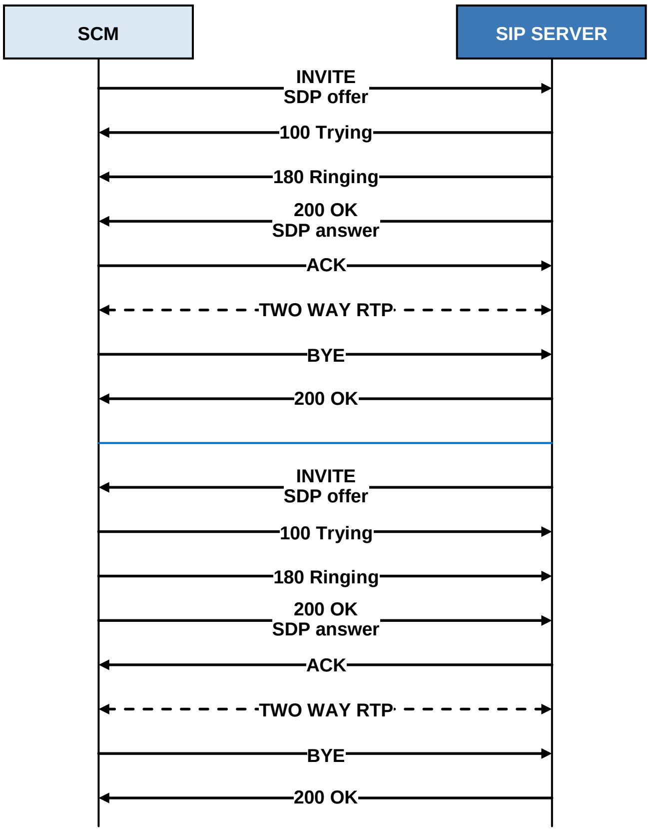


**Figure 1.15 Basic Call Flow**

## Message

1) SIP P-Asserted-ID Type: Primary

```
INVITE sip:0312793922@sbc.co.kr:5060 SIP/2.0
From: "John"<sip:0801000@sbc.co.kr>;tag=14bcd4ff-9345-4c84-99a6-
960f97deeead
To: <sip:0312793922@sbc.co.kr>
Call-ID: 9b16f42a-ce11-4b9c-a785-15b37de500b8@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-4aba-
123e8e0-29048041
P-Asserted-Identity: "0707007000" <sip:0707007000@sbc.co.kr>
Max-Forwards: 70
Allow:
INVITE,ACK,OPTIONS,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Contact: <sip:0801000@10.254.168.181:5060>
User-Agent: SAMSUNG SCM 3.3.100
Content-Type: application/sdp
Content-Length: 353
.........
```

2) SIP P-Asserted-ID Type: Secondary

```
INVITE sip:0312793922@sbc.co.kr:5060 SIP/2.0
From: "0707007000"<sip:0707007000@sbc.co.kr>;tag=53625e64-732b-4c2a-
ba77-c039ddb23c33
To: <sip:0312793922@sbc.co.kr>
Call-ID: 3b81d3da-d137-41f0-90fc-2fd5e3973cff@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-4b09-
1251ef3-21a726ee
P-Asserted-Identity: "John" <sip:0801000@sbc.co.kr>
Max-Forwards: 70
Allow:
INVITE,ACK,OPTIONS,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Contact: <sip:0707007000@10.254.168.181:5060>
User-Agent: SAMSUNG SCM 3.3.100
Content-Type: application/sdp
Content-Length: 356
………
```

3) SIP P-Asserted-ID Type: None

```
INVITE sip:0312793922@sbc.co.kr:5060 SIP/2.0
From: "John"<sip:0801000@sbc.co.kr>;tag=49a1c24e-94f2-4ba9-ab43-
1b323c2be2ba
To: <sip:0312793922@sbc.co.kr>
Call-ID: 48fa698e-5268-4093-aa2f-10c6099ab5d1@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-4b98-
1274d76-72a56343
Allow:
INVITE,ACK,OPTIONS,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Contact: <sip:0801000@10.254.168.181:5060>
Content-Type: application/sdp
Content-Length: 353
………
```

# 1.3.4 Calling Line ID and Privacy

## Description

SCM Express supports to configure CLIP for SIP Trunk Call. It can modify CLI Number, CLI Name.
For Anonymous call, USR type can be selected.

## Configuration

1) Calling Line ID
   (1) Configuration → User → Single User → Send CLI Number: Set userinfo field of From header
   (2) Configuration → User → Single User → Send CLI Name: Set display name field of From header

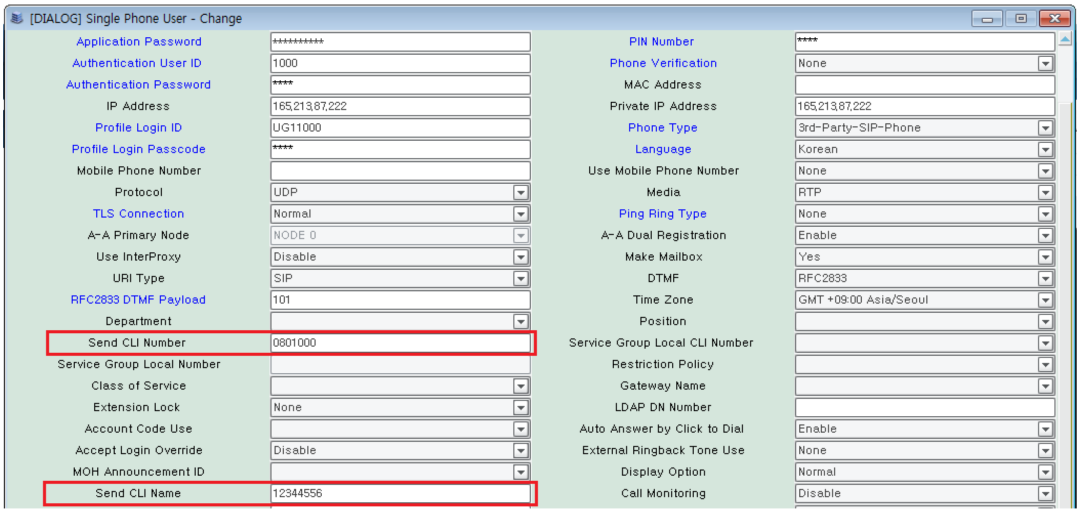


**Figure 1.16 Calling Line ID-Single Phone User**

① Configuration → Trunk Routing → Route → Send CLI Name for User: It can select setting value of From header.
② User Name: It will use Configuration → User → Single User → Extension Name
③ Send CLI Name: It will use Configuration → User → Single User → Send CLI Name

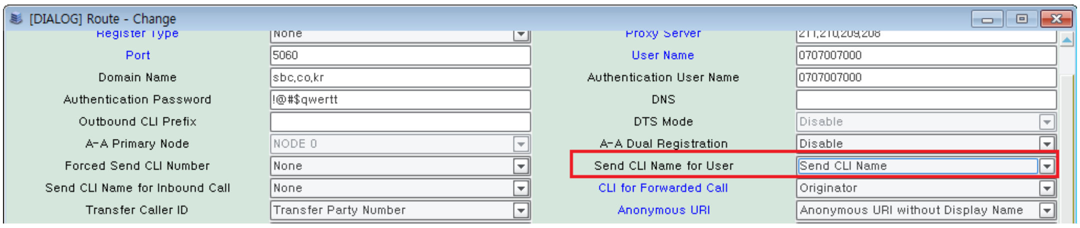


**Figure 1.17 Calling Line ID-Route**

2) Anonymous Call

(1) Configuration → Trunk Routing → Route → Anonymous URI

① Anonymous Invalid:
'Anonymous'<sip:anonymous@anonymous.invalid>

② User URI:
'Anonymous'<sip:anonymous@sbc.co.kr>

③ Anonymous Invalid without Display Name:
<sip:anonymous@anonymous.invalid>

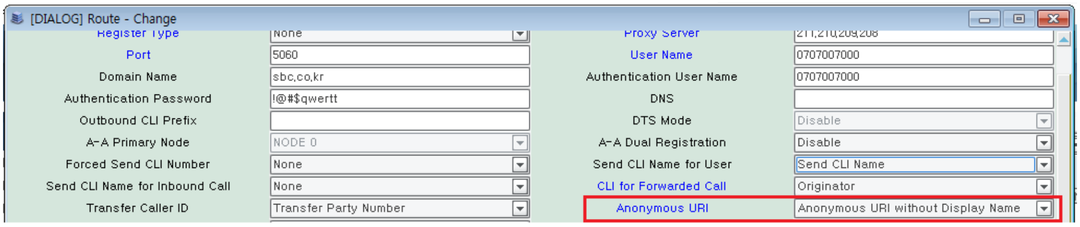


**Figure 1.18 Anonymous Call**

## Message

1) Calling Line ID

① Configuration → User → Single User → Send CLI Number: 0801000
Configuration → User → Single User → Send CLI Name: 12344556
Configuration → Trunk Routing → Route → Send CLI Name for User: User Name

```
INVITE sip:0312793922@sbc.co.kr:5060 SIP/2.0
From: "1000"<sip:0801000@sbc.co.kr>;tag=42ad5ea4-d7f9-498a-a0fa-
dd3ef1bfb7a7
To: <sip:0312793922@sbc.co.kr>
Call-ID: be275585-8bf0-4fc1-9942-7dd4fe4e6d28@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-3fb2-f8d0d8-
79ba4a7c
Allow:
INVITE,ACK,OPTIONS,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Contact: <sip:0801000@10.254.168.181:5060>
Content-Type: application/sdp
Content-Length: 353
………. .
```

② Configuration → User → Single User → Send CLI Number: 0801000
Configuration → User → Single User → Send CLI Name: 12344556
Configuration → Trunk Routing → Route → Send CLI Name for User: Send CLI Name

```
INVITE sip:0312793922@sbc.co.kr:5060 SIP/2.0
From: "12344556"<sip:0801000@sbc.co.kr>;tag=267433b6-4d7e-4b91-879a-
781eee683392
To: <sip:0312793922@sbc.co.kr>
Call-ID: 9bdfa1f1-ac55-4c4c-aeb1-592bfe3c27c7@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-4032-fac3d0-
3c6f94b3
Allow:
INVITE,ACK,OPTIONS,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Contact: <sip:0801000@10.254.168.181:5060>
Content-Type: application/sdp
Content-Length: 353
………..
```

2) Anonymous Call (SIP P-Asserted-ID Type: Primary)

① Anonymous Invalid

```
INVITE sip:0312793922@sbc.co.kr:5060 SIP/2.0
From: "Anonymous"<sip:anonymous@anonymous.invalid>;tag=348a9401-182e-
4dd7-a424
To: <sip:0312793922@sbc.co.kr>
Call-ID: effd87a5-9f84-490d-bcd6-386cd7beb22e@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-4953-
11e6f1d-32198c50
P-Asserted-Identity: "0707007000" <sip:0707007000@sbc.co.kr>
Privacy: id
Max-Forwards: 70
Allow:
INVITE,ACK,OPTIONS,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Contact: <sip:0801000@10.254.168.181:5060>
User-Agent: SAMSUNG SCM 3.3.100
Content-Type: application/sdp
Content-Length: 353
………..
```

② User URI

```
INVITE sip:0312793922@sbc.co.kr:5060 SIP/2.0
From: <sip:anonymous@anonymous.invalid>;tag=b5e4caae-d73b-4722-898c-
cf05fbd3a5da
To: <sip:0312793922@sbc.co.kr>
Call-ID: c46c57eb-0350-4828-8639-a3025a248da7@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-48b8-
11c10ca-1137efd
P-Asserted-Identity: "0707007000" <sip:0707007000@sbc.co.kr>
Privacy: id
Max-Forwards: 70
Allow:
INVITE,ACK,OPTIONS,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Contact: <sip:0801000@10.254.168.181:5060>
User-Agent: SAMSUNG SCM 3.3.100
Content-Type: application/sdp
Content-Length: 353
………..
```

③ Anonymous Invalid without Display Name

```
INVITE sip:0312793922@sbc.co.kr:5060 SIP/2.0
From: "Anonymous"<sip:anonymous@sbc.co.kr>;tag=07d0f17d-e660-4391-
bf5a-04a00c3bf4a0
To: <sip:0312793922@sbc.co.kr>
Call-ID: 80e6863b-5c5a-42c2-8a49-294d8f5e5ef8@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-4980-
11f1f5a-7c1ad888
P-Asserted-Identity: "0707007000" <sip:0707007000@sbc.co.kr>
Privacy: id
Max-Forwards: 70
Allow:
INVITE,ACK,OPTIONS,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Contact: <sip:0801000@10.254.168.181:5060>
User-Agent: SAMSUNG SCM 3.3.100
Content-Type: application/sdp
Content-Length: 353
………..
```

## 1.3.5 Call Hold/Resume

### Description

When use MOH, SCM Express can select media mode among ‘sendrecv’ or ‘sendlony’. If Configuration → User Group → Change User Group → Information → MOH Enable option has Disable, It will operate only hold without music.

### Configuration

Configuration → Trunk Routing → Route → MOH SIP Media Mode

1) Send/Receive: Default Value. The connection mode of SDP is set to ‘sendrecv’
2) Send Only: The connection mode of SDP is set to ‘sendrecv’.

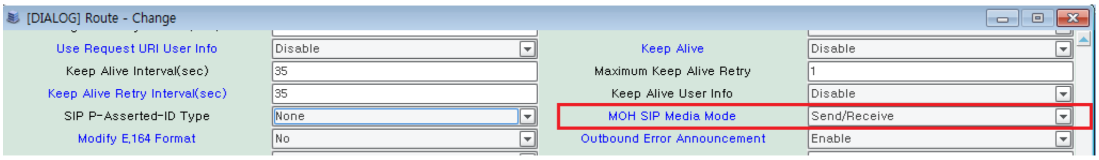


**Figure 1.19 Call Hold/Resume**

### Flow

1) MOH Disable

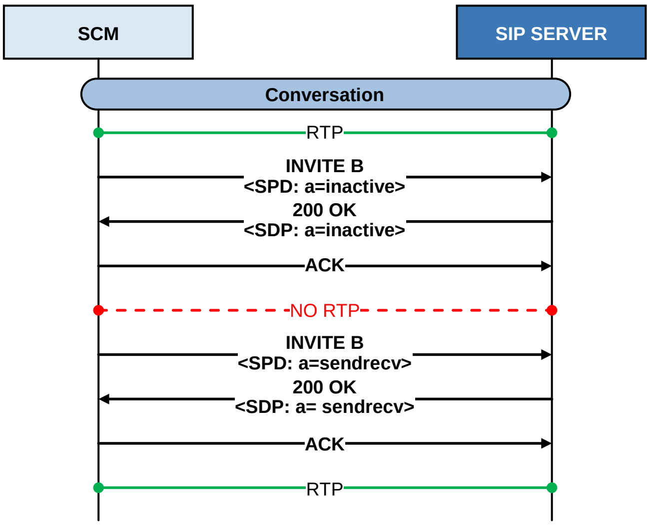


**Figure 1.20 Call Hold/Resume-MOH Disable**

2) Send/Receive

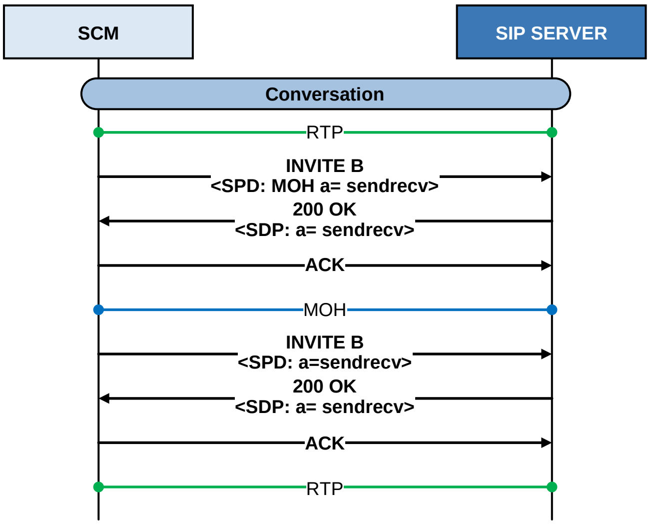


**Figure 1.21 Call Hold/Resume-Send/Receive**

3) SendOnly

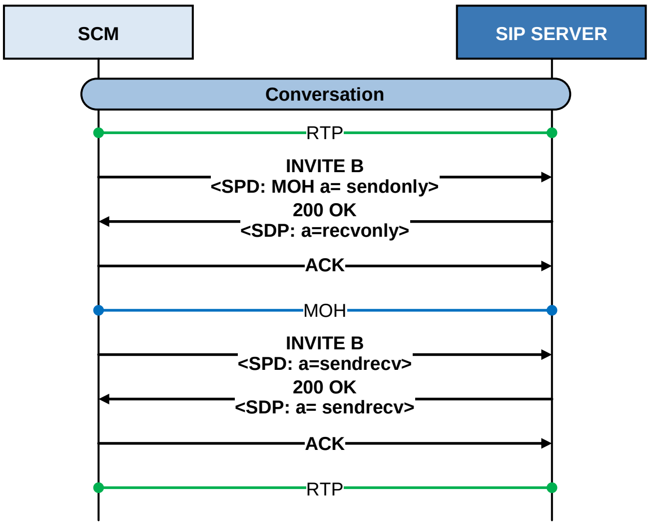


**Figure 1.22 Call Hold/Resume-Sendonly**

 

## Message

1) Send/Receive

```
INVITE sip:0312792002@10.254.168.82:5060;transport=udp SIP/2.0
From: "1005"<sips:1005@ug1.scm.com>;tag=2a3be3ec-5302-4558-a484-
60ae6fec1212
To: <sip:0312792002@10.254.168.82>;tag=41c0d220-52a8fe0a-13c4-50017-
4fafd064-596f2e7c-4fafd064
Call-ID: 1384a008-834d-4782-8359-9e56bf9335c4@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-1075f-
404cbbe-7aabeba4
Allow:
INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Contact: <sip:1005@10.254.168.181:5060>
Content-Type: application/sdp
Content-Length: 232

v=0
o=1005 100014 100014 IN IP4 10.254.168.181
s=Samsung IP PBX
c=IN IP4 10.254.168.181
t=0 0
a=sendrecv
m=audio 35000 RTP/AVP 0 8 18
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=sendrecv
```

2) SendOnly

```
INVITE sip:0312792002@10.254.168.82:5060;transport=udp SIP/2.0
From: "1005"<sips:1005@ug1.scm.com>;tag=ee38cea3-3032-49d1-9de2-
d5ad4ff3615c
To: <sip:0312792002@10.254.168.82>;tag=41c0e738-52a8fe0a-13c4-50017-
4fafd274-5e2279b0-4fafd274
Call-ID: c1cc07d0-d286-4cdc-8b1a-b4cd9e2398f8@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-1096d-
40cd51a-165d7b4e
Allow:
INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Contact: <sip:1005@10.254.168.181:5060>
Content-Type: application/sdp
Content-Length: 232

v=0
o=1005 100015 100015 IN IP4 10.254.168.181
s=Samsung IP PBX
c=IN IP4 10.254.168.181
t=0 0
a=sendonly
m=audio 35000 RTP/AVP 0 8 18
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=sendonly
```

## 1.3.6 Call Transfer

### Description

SCM Express provide Blind, Semi-Attended, Attended Transfer function for Sip Trunk call. And for all transfer type, there are two mode: re-INVITE relay, refer relay

- re-INVITE Relay: This is more general mechanism. Transfer function operate using only re-INVITE in SIP Trunk side.
- Refer Relay: SCM Express send REFER message to SIP Server, SIP Server make a transfer. By the way A small number of SIP Server can support REFER message. Therefore, this requires special care in handling.

### Configuration

Configuration → Trunk Routing → Route → Refer Relay

1) DISABLE: re-INVITE relay for Transfer
2) ENABLE: REFER relay for Transfer

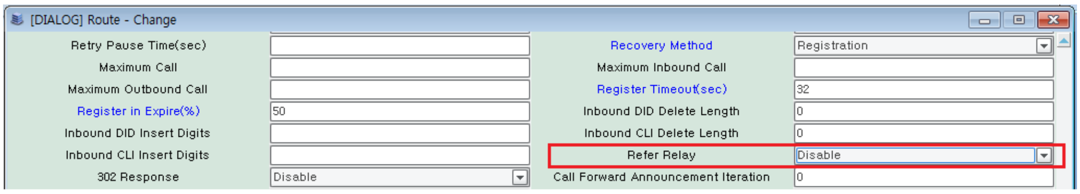


**Figure 1.23 Call Transfer**

## Flow

1) Blind Transfer

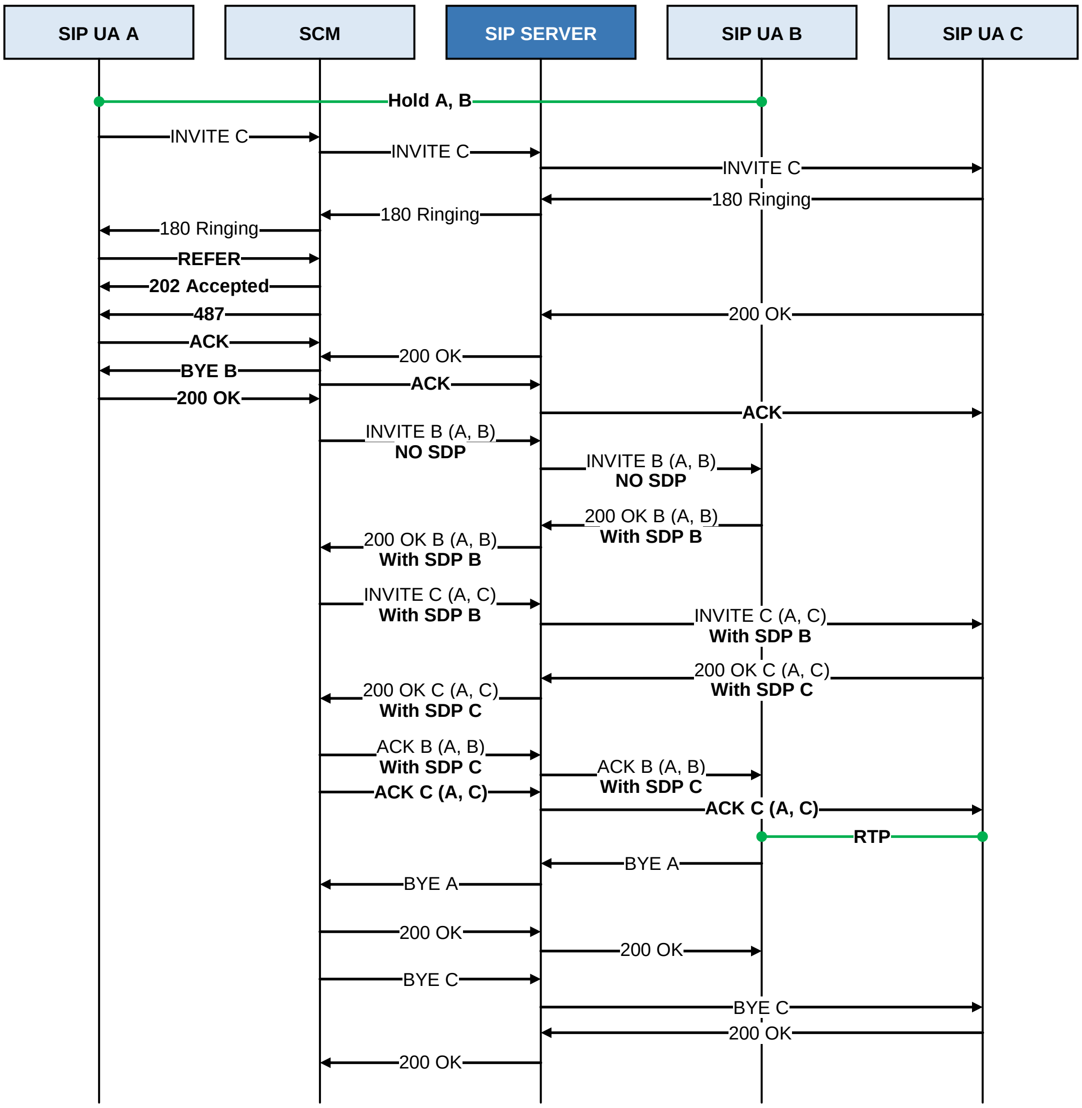


**Figure 1.24 Blind Transfer**

2) Attended Transfer

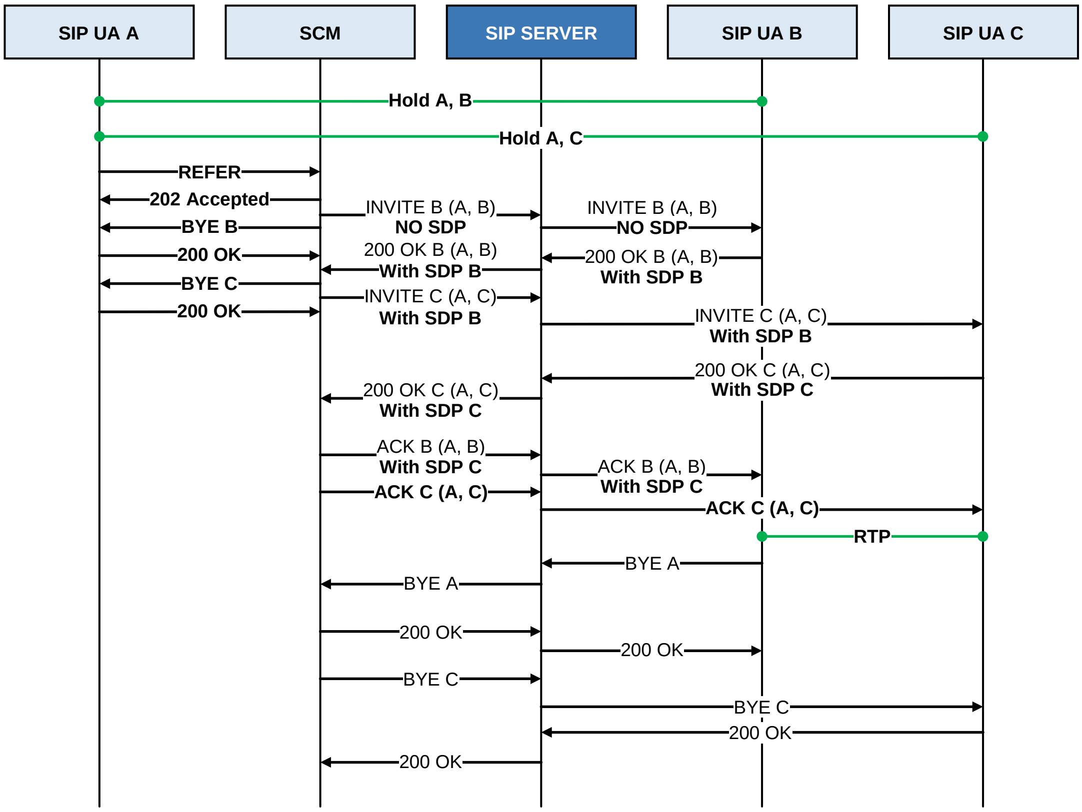


**Figure 1.25 Attended Transfer**

3) Refer Relay

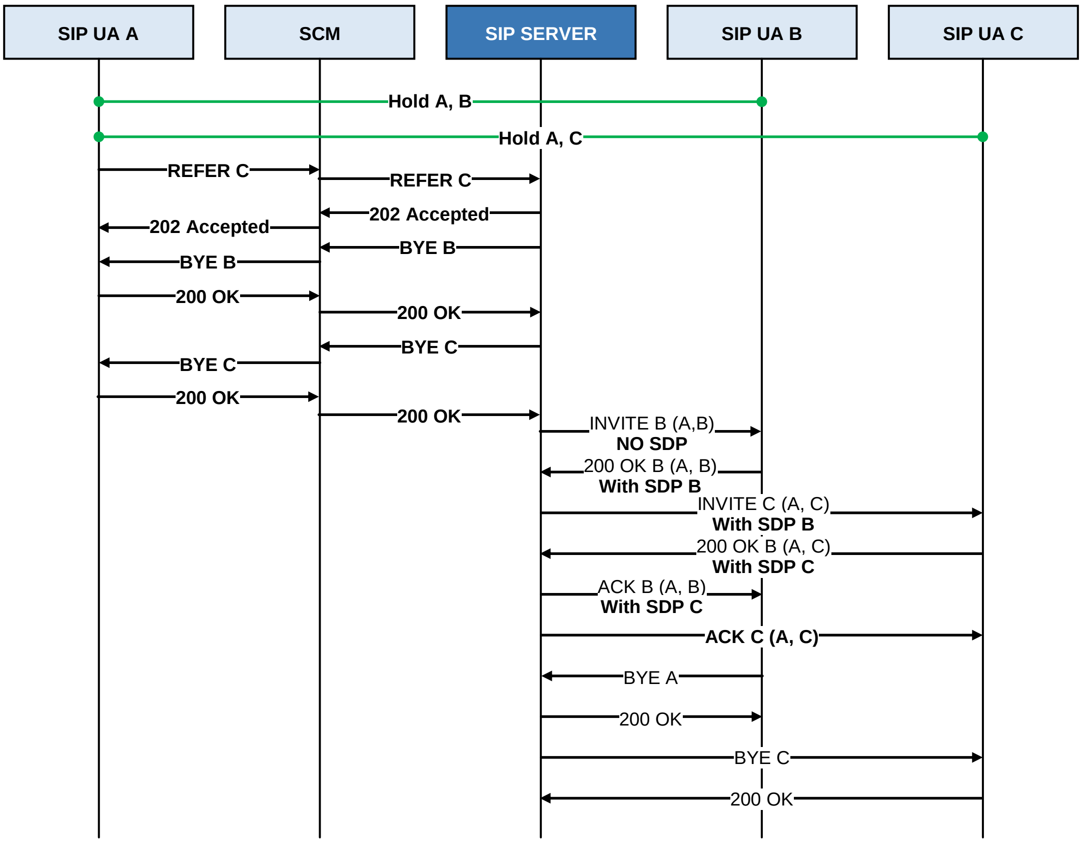


**Figure 1.26 Refer Relay**

## 1.3.7 Call Redirect (Forwarding)

### Description

If SCM Express is the one who sets the call forward, it can have two different options in doing it; either sending 302 Response back to the caller or forwarding the received INVITE to a designated destination. The former is that SCM Express asks the original caller to redirect the call to designated destination, and the latter is SCM Express itself redirects the call by sending an INVITE to the 3rd destination. The latter is more general Call Forwarding way.

### Configuration

Configuration → Trunk Routing → Route → 302 Response

1) DISABLE: Default value
2) ENABLE: SCM Express sends 302 response for incoming call that set call forward.

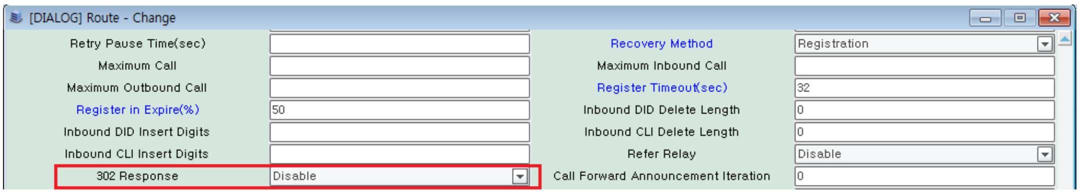


**Figure 1.27 Call Redirect (Forwarding)**

### Flow

1) Sending INVITE

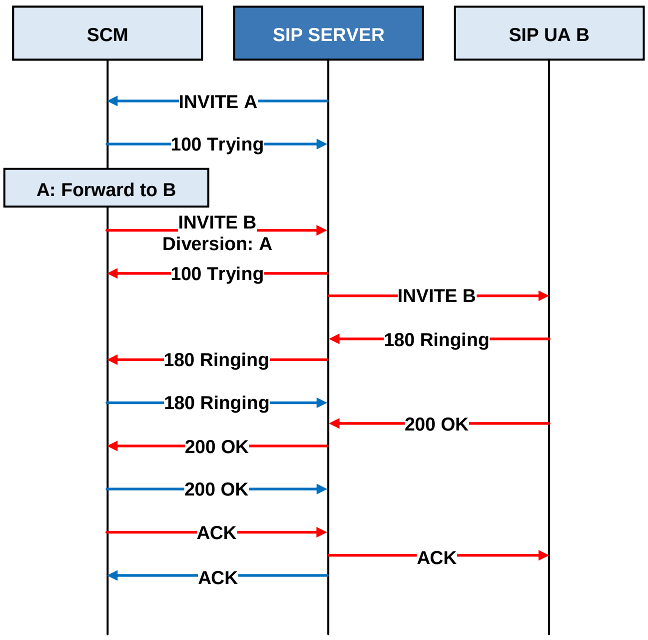


**Figure 1.28 Call Redirect - Sending INVITE**

2) 302 Response

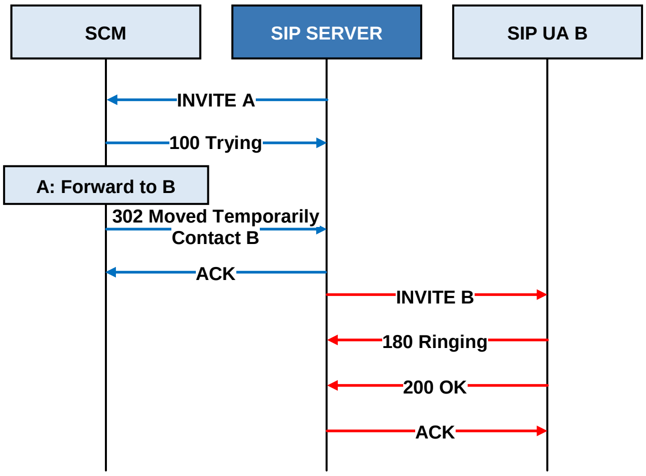


**Figure 1.29 Call Redirect – 302 Response**

### 1.3.8 Security

#### Description

SCM Express supports TLS, SRTP signaling for security.

- Signal Security: TLS v1.0 (RFC2246)
- Basic Encryption Algorithm: AES
- PKI (Public Key Infrastructure): RSA 2048 bit

When TLS connection is created between SCM Express and SIP Server, The connection must be lasted. SCM Express keep the connection. If the connection is terminated, SCM Express detects this event and make a connection and send REGISTER message to SIP Server.

#### Configuration

1) Configuration → Trunk Routing → Route
   (1) Protocol Type: TLS
   (2) TLS Connection: Reuse (It is recommended)
2) How to Load Certificate
   (1) Loading Certificate to SCM Express
   The certificate files is located to following directory.
      ① Directory path: /DI/BASE/data/COM
      ② Certificate type and file name format
         – root_caCert.pem: Root CA certificate
         – caCert.pem: CA certificate
         – myCert.pem: PBX (SCM Express) certificate
         – myPrvKey.pem: PBX (SCM Express) private key
      - 3 certificate files are included in SCM Express installation package.
        This certificate can use only between SAMSUNG Phone, Gateway.
        Basic Certificate is composed of caCert.pem, myCert.pem, myPrvKey.pem.
      - If being issued certificates from Certificate Authorities, modify filename like above format, replace all old certificate to new certificates.
      - To apply new certificate to SCM Express, it need to restart SCM.
      - Note: If encrypted private key is issued, that must be decrypted.
   (2) Loading Certificate to Samsung Phone
   If being issued certificates from Certificate Authorities, it need to reload certificate for Phone. Input Issued Certificates to the following directory, modify filename format, and Enable Use TLS Certification option. Then phone start to take new certificates.
      ① Directory Path: /tftpboot/sec_cert
      ② Configuration → Phone Setting → SIP Options → Use TLS Certification: YES

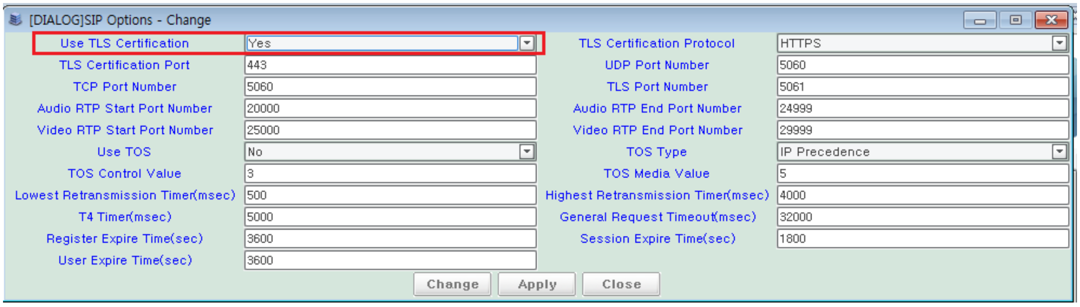


**Figure 1.30 SIP Options**

③ Certificate filename format

– {MAC}.der.pem: phone certificate

– {MAC}.key.pem: phone private key

*example: If the MAC of phone is 00214c9c8eef,

certificate: 00214c9c8eef.der.pem

private key: 00214c9c8eef.key.pem

- Note: If encrypted private key is issued, that must be decrypted.

## Flow

1) TLS Handshake Success

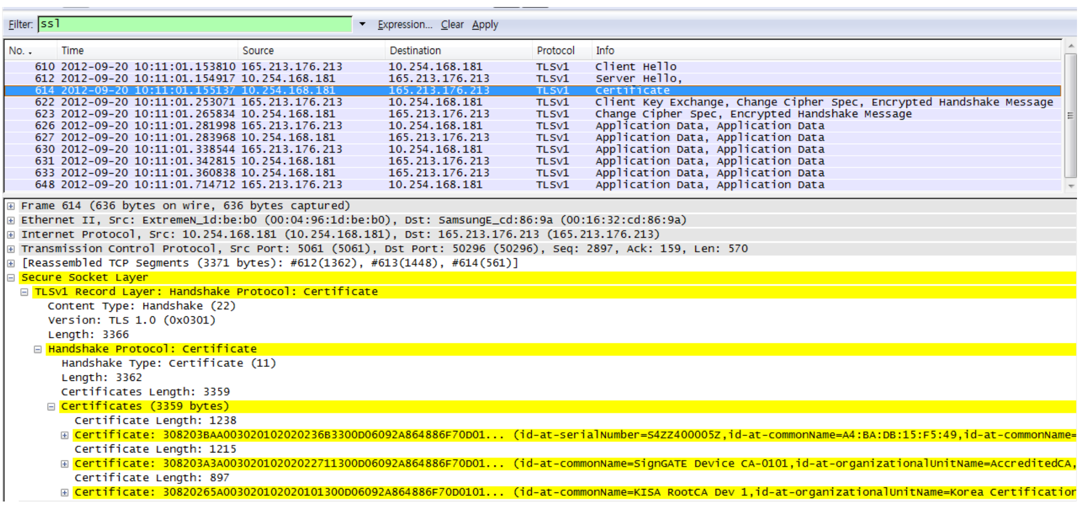


**Figure 1.31 TLS Handshake Success**

2) Mutual TLS Handshake Success

Filter: ssl Expression... Clear Apply

| No. | Time | Source | Destination | Protocol | Info |
|---|---|---|---|---|---|
| 915 | 2012-09-20 10:13:30.188542 | 165.213.176.213 | 10.254.168.181 | TLSv1 | Client Hello |
| 917 | 2012-09-20 10:13:30.189385 | 10.254.168.181 | 165.213.176.213 | TLSv1 | Server Hello, |
| 919 | 2012-09-20 10:13:30.189582 | 10.254.168.181 | 165.213.176.213 | TLSv1 | Certificate |
| 1025 | 2012-09-20 10:13:30.837164 | 165.213.176.213 | 10.254.168.181 | TCP | [TCP segment of a reassembled PDU] |
| 1027 | 2012-09-20 10:13:30.837483 | 165.213.176.213 | 10.254.168.181 | TLSv1 | Certificate |
| 1032 | 2012-09-20 10:13:30.852625 | 10.254.168.181 | 165.213.176.213 | TLSv1 | Change Cipher Spec, Encrypted Handshake Message |
| 1036 | 2012-09-20 10:13:30.861804 | 165.213.176.213 | 10.254.168.181 | TLSv1 | Application Data, Application Data |
| 1037 | 2012-09-20 10:13:30.863591 | 10.254.168.181 | 165.213.176.213 | TLSv1 | Application Data, Application Data |
| 1042 | 2012-09-20 10:13:30.900846 | 165.213.176.213 | 10.254.168.181 | TLSv1 | Application Data, Application Data |
| 1043 | 2012-09-20 10:13:30.904997 | 10.254.168.181 | 165.213.176.213 | TLSv1 | Application Data, Application Data |
| 1045 | 2012-09-20 10:13:30.924901 | 10.254.168.181 | 165.213.176.213 | TLSv1 | Application Data, Application Data |
| 1095 | 2012-09-20 10:13:31.206958 | 165.213.176.213 | 10.254.168.181 | TLSv1 | Application Data, Application Data |

```
⊞ Frame 919 (646 bytes on wire, 646 bytes captured)
⊞ Ethernet II, Src: ExtremeN_1d:be:b0 (00:04:96:1d:be:b0), Dst: SamsungE_cd:86:9a (00:16:32:cd:86:9a)
⊞ Internet Protocol, Src: 10.254.168.181 (10.254.168.181), Dst: 165.213.176.213 (165.213.176.213)
⊟ Transmission Control Protocol, Src Port: 5061 (5061), Dst Port: 33225 (33225), Seq: 2897, Ack: 159, Len: 580
    Source port: 5061 (5061)
    Destination port: 33225 (33225)
    Sequence number: 2897    (relative sequence number)
    [Next sequence number: 3477    (relative sequence number)]
    Acknowledgement number: 159    (relative ack number)
    Header length: 32 bytes
  ⊞ Flags: 0x18 (PSH, ACK)
    Window size: 6912 (scaled)
  ⊞ Checksum: 0x3ff2 [correct]
  ⊞ Options: (12 bytes)
    TCP segment data (580 bytes)
⊞ [Reassembled TCP Segments (3371 bytes): #917(1362), #918(1448), #919(561)]
⊟ Secure Socket Layer
  ⊟ TLSv1 Record Layer: Handshake Protocol: Certificate
      Content Type: Handshake (22)
      Version: TLS 1.0 (0x0301)
      Length: 3366
    ⊞ Handshake Protocol: Certificate
⊟ Secure Socket Layer
  ⊟ TLSv1 Record Layer: Handshake Protocol: Multiple Handshake Messages
      Content Type: Handshake (22)
      Version: TLS 1.0 (0x0301)
      Length: 14
    ⊟ Handshake Protocol: Certificate Request
        Handshake Type: Certificate Request (13)
        Length: 6
        Certificate types count: 3
      ⊞ Certificate types (3 types)
        Distinguished Names Length: 0
    ⊟ Handshake Protocol: Server Hello Done
        Handshake Type: Server Hello Done (14)
        Length: 0
```

**Figure 1.32 Mutual TLS Handshake Success**

3) TLS Handshake Failure-Unknown CA

: The reason for this problem is about certificates loading failure. You need to check that all certificate files are loaded to SCM Express properly.

Filter: ssl Expression... Clear Apply

| No. | Time | Source | Destination | Protocol | Info |
|---|---|---|---|---|---|
| 4 | 2012-09-20 10:19:43.898252 | 10.254.168.172 | 10.254.168.181 | TLSv1 | Client Hello |
| 6 | 2012-09-20 10:19:43.899176 | 10.254.168.181 | 10.254.168.172 | TLSv1 | Server Hello, |
| 9 | 2012-09-20 10:19:43.901215 | 10.254.168.181 | 10.254.168.172 | TLSv1 | Certificate |
| 11 | 2012-09-20 10:19:43.958841 | 10.254.168.172 | 10.254.168.181 | TLSv1 | Alert (Level: Fatal, Description: Unknown CA) |

**Figure 1.33 TLS Handshake Failure 1**

4) TLS Handshake Failure-Bad Certificate

: The reason for this problem is time zone. The time of SCM Express may be not present. You need to check SCM Express time setting.

| No. | Time | Source | Destination | Protocol | Info |
|---|---|---|---|---|---|
| 1 | 2010-01-12 18:04:25.670028 | 165.213.66.189 | 10.254.168.181 | SSL | Client Hello |
| 2 | 2010-01-12 18:04:25.670265 | 10.254.168.181 | 165.213.66.189 | TLSv1 | Server Hello, |
| 3 | 2010-01-12 18:04:25.670287 | 10.254.168.181 | 165.213.66.189 | TCP | [TCP Previous segment lost] [TCP segment of a reassembled PDU] |
| 4 | 2010-01-12 18:04:25.730163 | 165.213.66.189 | 10.254.168.181 | TLSv1 | Alert (Level: Fatal, Description: Bad Certificate) |

**Figure 1.34 TLS Handshake Failure 2**

## 1.3.9 ACL

### Description

ACL can prevent malicious access using SIP signal. When SCM Express receive SIP message from unauthorized sender, SCM Express block sender's IP address.
The Authorized entities are like the following:

- Route address
  (Trunk Routing → Route → Proxy Server, Secondary Proxy Server)
- Registered Phone
- Application Server

You can see blocked IP lists in Management → Access Control List (ACL) → Unauthorized SIP ACL

### Configuration

Management → Access Control List (ACL)

1) Unauthorized SIP ACL Status: DISABLE/ENABLE
2) SIP Common Message Block Timer: Set block period for all SIP message except REGISTER message.
3) SIP REGISTER Message Block Timer: Set block period for REGISTER message.
4) Unauthorized SIP ACL Degree: Select the range of block. If you select IP, Port, Protocol, only when received packet is matched with three conditions, it will be blocked.

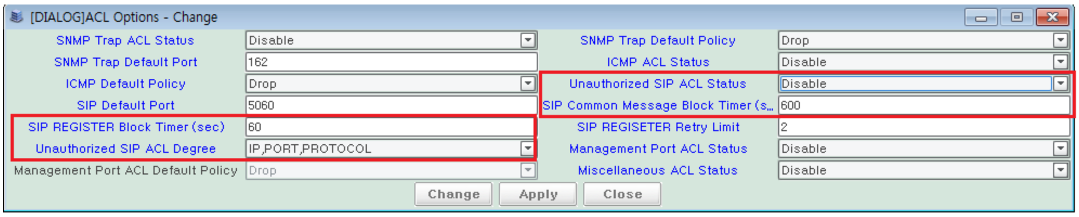


**Figure 1.35 ACL**

## 1.3.10 DTMF

### Description

SCM Express supports INFO method for sending DTMF.

### Configuration

Configuration → User → Single Phone User → DTMF: You can select DTMF type among RFC 2833, Invoice, Outband type.

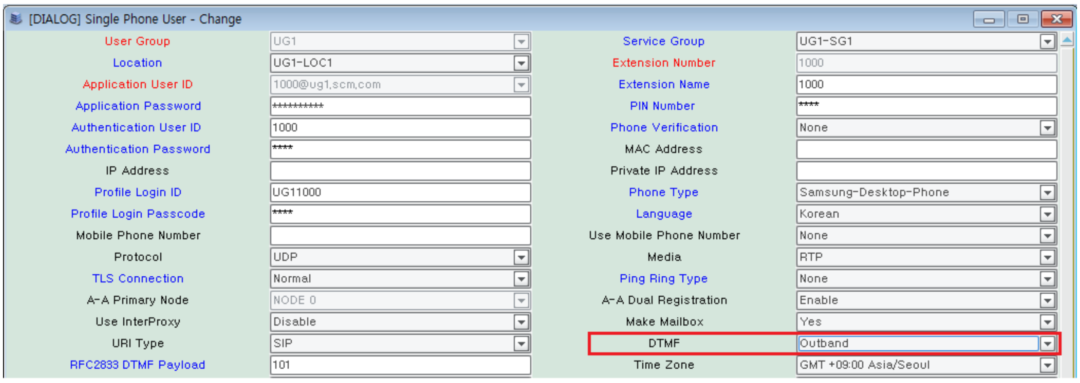


**Figure 1.36 DTMF**

## 1.3.11 Session Timer

### Description

This function checks the status of connecting call through sending a UPDATE request and receiving a response periodically. If this message exchange does not occur cyclically, the call is disconnected.
SCM Express supports to send UPDATE method as session refresh request. If receiver does not support a UPDATE method, SCM Express stops Session-Timer function.

### Configuration

Configuration → Miscellaneous → System Options

1) Call Session Used: Select to use session timer function.
2) Call Session Expires Timer: The internal of session refresh.
3) Call Minimum Session Expire Timer: The minimum internal of session refresh.
4) Call Session Max Refresh Retry: When expire session timer, SCM Express retry to send session refresh request (UPDPATE) up to this option's value and if SCM Express does not receive any response, disconnect the call finally.

## Flow

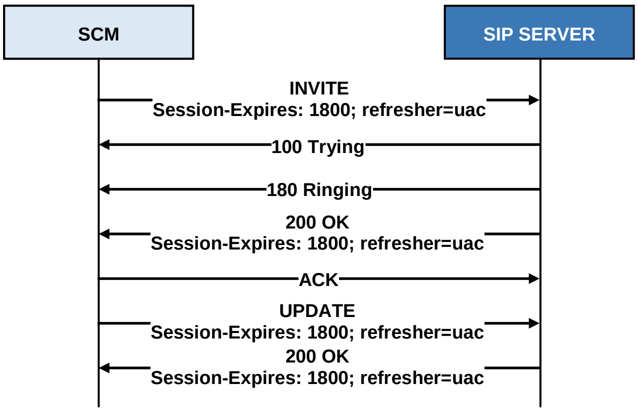


**Figure 1.37 Session Timer**

## Message

```
INVITE sip:202@domain.com:5060 SIP/2.0
From: "1003"<sip:1003@domain.com>;tag=ad40dc8b-7065-4da0-a6af-
6a98ffc3e501
To: <sip:202@domain.com>
Call-ID: 494c8e13-ddec-4acf-b1cf-9d16288f8e64@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-e506b-
37ea235c-3ef5b94e
Supported: replaces, 100rel, timer
Max-Forwards: 70
Allow:
INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Contact: <sip:1003@10.254.168.181:5060>
Session-Expires: 1800;refresher=uac
Min-SE: 90
User-Agent: SAMSUNG SCM 3.3.100
Content-Type: application/sdp
Content-Length: 329

v=0
o=1003 1349451896 0 IN IP4 165.213.176.159
s=Samsung IP PBX
c=IN IP4 165.213.176.159
t=0 0
a=sendrecv
m=audio 20006 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
SIP/2.0 100 Trying
From: "1003"<sip:1003@domain.com>;tag=ad40dc8b-7065-4da0-a6af-
6a98ffc3e501
To: <sip:202@domain.com>
Call-ID: 494c8e13-ddec-4acf-b1cf-9d16288f8e64@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport=5060;branch=z9hG4bK-e506b-
37ea235c-3ef5b94e
Contact: <sip:202@10.254.168.82:5060;transport=udp>
User-Agent: OfficeServ 7100
Content-Length: 0


SIP/2.0 183 Session Progress
From: "1003"<sip:1003@domain.com>;tag=ad40dc8b-7065-4da0-a6af-
6a98ffc3e501
To: <sip:202@domain.com>;tag=41bf50d0-52a8fe0a-13c4-50017-4fc3bc32-
364b9240-4fc3bc32
Call-ID: 494c8e13-ddec-4acf-b1cf-9d16288f8e64@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport=5060;branch=z9hG4bK-e506b-
37ea235c-3ef5b94e
Contact: <sip:202@10.254.168.82:5060;transport=udp>
User-Agent: OfficeServ 7100
Content-Type: application/sdp
Content-Length: 341

v=0
o=SAMSUNG_SIP_GATEWAY 2492933144 0 IN IP4 10.254.168.82
s=SIP_CALL
c=IN IP4 10.254.168.82
t=0 0
m=audio 30006 RTP/AVP 0 8 18 18 4 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv


SIP/2.0 200 OK
From: "1003"<sip:1003@domain.com>;tag=ad40dc8b-7065-4da0-a6af-
6a98ffc3e501
To: <sip:202@domain.com>;tag=41bf50d0-52a8fe0a-13c4-50017-4fc3bc32-
364b9240-4fc3bc32
Call-ID: 494c8e13-ddec-4acf-b1cf-9d16288f8e64@ug1.scm.com
CSeq: 1 INVITE
```

```
Via: SIP/2.0/UDP 10.254.168.181:5060;rport=5060;branch=z9hG4bK-e506b-
37ea235c-3ef5b94e
Contact: <sip:202@10.254.168.82:5060;transport=udp>
Require: timer
User-Agent: OfficeServ 7100
Supported: timer
Session-Expires: 1800;refresher=uac
Allow:
REGISTER,INVITE,ACK,BYE,REFER,NOTIFY,CANCEL,INFO,OPTIONS,PRACK,SUBSCRI
BE,UPDATE
Content-Type: application/sdp
Content-Length: 220

v=0
o=SAMSUNG_SIP_GATEWAY 2492933144 1 IN IP4 10.254.168.82
s=SIP_CALL
c=IN IP4 10.254.168.82
t=0 0
m=audio 30006 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv


ACK sip:202@10.254.168.82:5060;transport=udp SIP/2.0
From: "1003"<sip:1003@domain.com>;tag=ad40dc8b-7065-4da0-a6af-
6a98ffc3e501
To: <sip:202@domain.com>;tag=41bf50d0-52a8fe0a-13c4-50017-4fc3bc32-
364b9240-4fc3bc32
Call-ID: 494c8e13-ddec-4acf-b1cf-9d16288f8e64@ug1.scm.com
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-e506e-
37ea3196-3792ed08
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Contact: <sip:1003@10.254.168.181:5060>
Content-Length: 0


UPDATE sip:202@10.254.168.82:5060;transport=udp SIP/2.0
From: "1003"<sip:1003@domain.com>;tag=ad40dc8b-7065-4da0-a6af-
6a98ffc3e501
To: <sip:202@domain.com>;tag=41bf50d0-52a8fe0a-13c4-50017-4fc3bc32-
364b9240-4fc3bc32
Call-ID: 494c8e13-ddec-4acf-b1cf-9d16288f8e64@ug1.scm.com
CSeq: 2 UPDATE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-e53f2-
37f7eb07-14b0196e
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Contact: <sip:1003@10.254.168.181:5060>
Session-Expires: 1800;refresher=uac
Min-SE: 90
Content-Length: 0
```

```
SIP/2.0 200 OK
From: "1003"<sip:1003@domain.com>;tag=ad40dc8b-7065-4da0-a6af-
6a98ffc3e501
To: <sip:202@domain.com>;tag=41bf50d0-52a8fe0a-13c4-50017-4fc3bc32-
364b9240-4fc3bc32
Call-ID: 494c8e13-ddec-4acf-b1cf-9d16288f8e64@ug1.scm.com
CSeq: 2 UPDATE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport=5060;branch=z9hG4bK-e53f2-
37f7eb07-14b0196e
Contact: <sip:202@10.254.168.82:5060;transport=udp>
Require: timer
User-Agent: OfficeServ 7100
Session-Expires: 1800;refresher=uac
Content-Length: 0


UPDATE sip:202@10.254.168.82:5060;transport=udp SIP/2.0
From: "1003"<sip:1003@domain.com>;tag=ad40dc8b-7065-4da0-a6af-
6a98ffc3e501
To: <sip:202@domain.com>;tag=41bf50d0-52a8fe0a-13c4-50017-4fc3bc32-
364b9240-4fc3bc32
Call-ID: 494c8e13-ddec-4acf-b1cf-9d16288f8e64@ug1.scm.com
CSeq: 3 UPDATE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport;branch=z9hG4bK-e5776-
3805a6ed-3b630bc0
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.3.100
Contact: <sip:1003@10.254.168.181:5060>
Session-Expires: 1800;refresher=uac
Min-SE: 90
Content-Length: 0

SIP/2.0 200 OK
From: "1003"<sip:1003@domain.com>;tag=ad40dc8b-7065-4da0-a6af-
6a98ffc3e501
To: <sip:202@domain.com>;tag=41bf50d0-52a8fe0a-13c4-50017-4fc3bc32-
364b9240-4fc3bc32
Call-ID: 494c8e13-ddec-4acf-b1cf-9d16288f8e64@ug1.scm.com
CSeq: 3 UPDATE
Via: SIP/2.0/UDP 10.254.168.181:5060;rport=5060;branch=z9hG4bK-e5776-
3805a6ed-3b630bc0
Contact: <sip:202@10.254.168.82:5060;transport=udp>
Require: timer
User-Agent: OfficeServ 7100
Session-Expires: 1800;refresher=uac
Content-Length: 0
```

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# CHAPTER 2. SIP Phone Interoperability

## 2.1 Introduction

This part describe 3rd-Party-SIP-Phone specification.

# 2.2 Configuring SIP Phone

## 2.2.1 Creating 3rd Party User

Configuration → User → Single Phone User

1) Extension Number: Specify the value of user's extension number
2) Phone Type: 3rd-Party-SIP-Phone
3) Authentication User ID/Password: Specify a user ID/Password used for authentication during user registration.

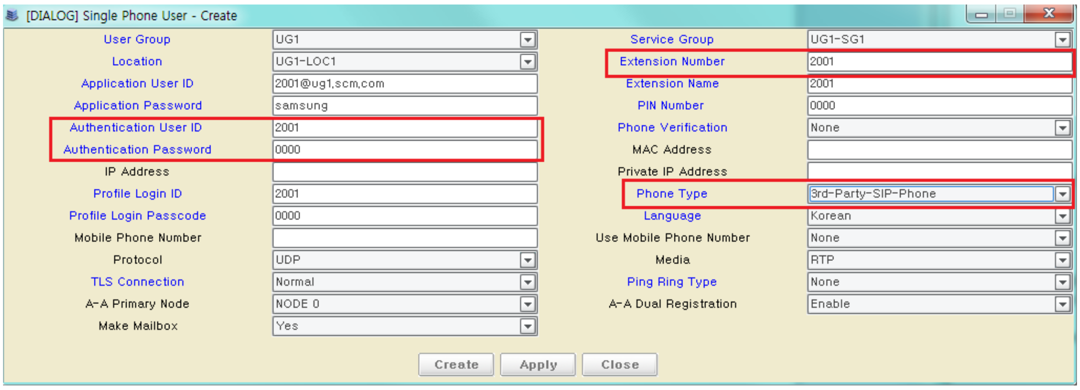



**Figure 2.1 Creating 3rd Party User**

# 2.3 SIP Phone Service

## 2.3.1 Registration with Authentication Challenge

3rd-Party-SIP-Phone must send REGISTER Message to SCM Express for registration.

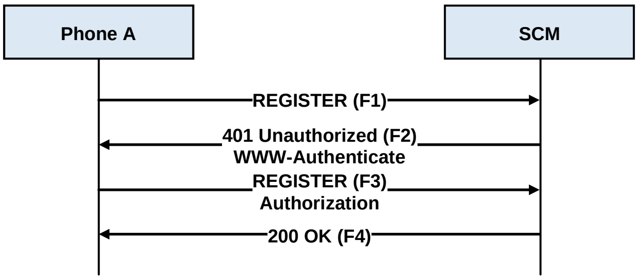



**Figure 2.2 Registration with Authentication Challenge**

**(F1)**

```
REGISTER sip:scm.com SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3fd5f8-ceb0d5a5-13c4-50022-48e4-
43f2dfa6-48e4
To: "SIP9004"<sip:9004@scm.com>
Call-ID: 405ae0-ceb0d5a5-13c4-50022-37-7219e739-37
CSeq: 13 REGISTER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-48e4-11cbb04-
2fd50d00
Max-Forwards: 70
Supported: timer,replaces
User-Agent: SAMSUNG-SMT-i5243/V1.02
Expires: 3600
Contact: <sip:9004@165.213.176.206>;mac="00:16:32:cd:86:c8"
Content-Length: 0
```

**(F2)**

```
SIP/2.0 401 Unauthorized
From: "SIP9004"<sip:9004@scm.com>;tag=3fd5f8-ceb0d5a5-13c4-50022-48e4-
43f2dfa6-48e4
To: "SIP9004"<sip:9004@scm.com>;tag=af70f68-0-13c4-50022-5178-
4828a259-5178
Call-ID: 405ae0-ceb0d5a5-13c4-50022-37-7219e739-37
CSeq: 13 REGISTER
Date: Sat, 15 May 2010 09:22:44 GMT
Min-Expires: 3600
WWW-Authenticate: Digest
realm="scm.com",nonce="58915364544c568404529fd421d068a153487852",stale
=false,algorithm=MD5,qop="auth"
Contact: <sip:9004@165.213.176.206>
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-48e4-11cbb04-
2fd50d00
Content-Length: 0
```

**(F3)**

```
REGISTER sip:scm.com SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3fd5f8-ceb0d5a5-13c4-50022-48e4-
43f2dfa6-48e4
To: "SIP9004"<sip:9004@scm.com>
Call-ID: 405ae0-ceb0d5a5-13c4-50022-37-7219e739-37
CSeq: 14 REGISTER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-48e4-11cbb18-
5d1ea26
Max-Forwards: 70
Supported: timer,replaces
User-Agent: SAMSUNG-SMT-i5243/V1.02
Expires: 3600
Authorization: Digest
username="9004",realm="scm.com",nonce="58915364544c568404529fd421d068a
153487852",uri="sip:scm.com",response="c78e093303bfd05fafe914adcc8e821
3",algorithm=MD5,cnonce="11cbb18",qop=auth,nc=00000001
Contact: <sip:9004@165.213.176.206>;mac="00:16:32:cd:86:c8"
Content-Length: 0
```

**(F4)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=3fd5f8-ceb0d5a5-13c4-50022-48e4-
43f2dfa6-48e4
To: "SIP9004"<sip:9004@scm.com>;tag=af71108-0-13c4-50022-5178-
6188bc87-5178
Call-ID: 405ae0-ceb0d5a5-13c4-50022-37-7219e739-37
CSeq: 14 REGISTER
Date: Sat, 15 May 2010 09:22:44 GMT
Expires: 3600
Contact: <sip:9004@165.213.176.206>
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-48e4-11cbb18-
5d1ea26
Content-Length: 0
```

### 2.3.2 Call Origination by Phone

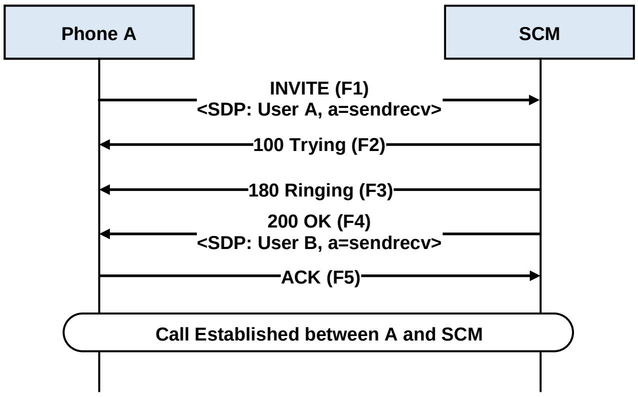



**Figure 2.3 Call Origination by Phone**

**(F1)**

```
INVITE sip:9005@scm.com SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=4893d8-ceb0d5a5-13c4-50022-
19325-ca32bf6-19325
To: <sip:9005@scm.com>
Call-ID: 48d5e8-ceb0d5a5-13c4-50022-19325-2606d1d0-19325
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-19325-626cae0-
50e91247
Max-Forwards: 70
Supported: timer,replaces
P-Preferred-Identity: "SIP9004"<sip:9004@scm.com>
Contact: <sip:9004@165.213.176.206>
Session-Expires: 1800;refresher=uac
Min-SE: 90
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Type: application/sdp
Content-Length: 367

v=0
o=SAMSUNG_IP_PHONE 1275911121 0 IN IP4 165.213.176.206
s=SAMSUNG_VOIP_CALL
c=IN IP4 165.213.176.206
t=0 0
m=audio 20000 RTP/AVP 8 0 18 9 102 101
```

```
b=TIAS:64000
b=AS:82
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F2)**

```
SIP/2.0 100 Trying
From: "SIP9004"<sip:9004@scm.com>;tag=4893d8-ceb0d5a5-13c4-50022-
19325-ca32bf6-19325
To: <sip:9005@scm.com>
Call-ID: 48d5e8-ceb0d5a5-13c4-50022-19325-2606d1d0-19325
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-19325-626cae0-
50e91247
Contact: <sip:9005@scm.com>
Content-Length: 0
```

**(F3)**

```
SIP/2.0 180 Ringing
From: "SIP9004"<sip:9004@scm.com>;tag=4893d8-ceb0d5a5-13c4-50022-
19325-ca32bf6-19325
To: <sip:9005@scm.com>;tag=039d133a-4525-45b1-89b5-7500bc488d46
Call-ID: 48d5e8-ceb0d5a5-13c4-50022-19325-2606d1d0-19325
CSeq: 1 INVITE
P-Asserted-Identity: "SIP9005" <sip:9005@scm.com>
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-19325-626cae0-
50e91247
Contact: <sip:9005@10.254.176.120:5060>
Content-Length: 0
```

**(F4)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=4893d8-ceb0d5a5-13c4-50022-
19325-ca32bf6-19325
To: <sip:9005@scm.com>;tag=039d133a-4525-45b1-89b5-7500bc488d46
Call-ID: 48d5e8-ceb0d5a5-13c4-50022-19325-2606d1d0-19325
CSeq: 1 INVITE
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
P-Asserted-Identity: "SIP9005" <sip:9005@scm.com>
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-19325-626cae0-
50e91247
Contact: <sip:9005@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 247

v=0
o=9005 1275911115 1 IN IP4 10.254.176.120
s=Samsung IP PBX
c=IN IP4 10.254.168.97
t=0 0
m=audio 20000 RTP/AVP 0 101
b=TIAS:64000
b=AS:82
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F5)**

```
ACK sip:9005@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=4893d8-ceb0d5a5-13c4-50022-
19325-ca32bf6-19325
To: <sip:9005@scm.com>;tag=039d133a-4525-45b1-89b5-7500bc488d46
Call-ID: 48d5e8-ceb0d5a5-13c4-50022-19325-2606d1d0-19325
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-19327-626d2ec-
6ceaa250
Max-Forwards: 70
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Contact: <sip:9004@165.213.176.206>
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Length: 0
```

## 2.3.3 Call Origination by SCM

Phone should display COLP with P-Asserted-Identity Header.
P-Asserted-Identity: ‘Display Name’ <sip:user@host>

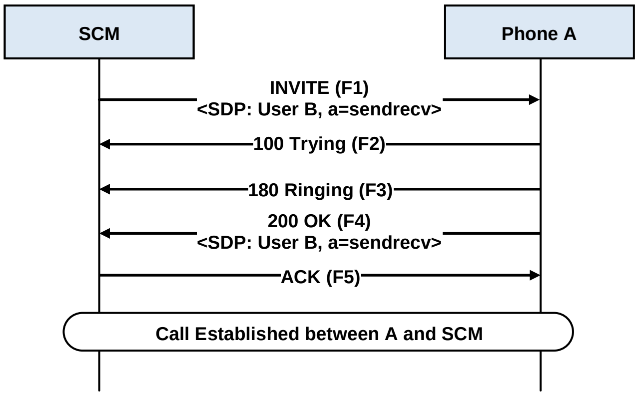



**Figure 2.4 Call Origination by SCM**

**(F1)**

```
INVITE sip:9005@10.254.168.97 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3f37247c-3953-47ac-b07e-
7e6d21dc2d6e
To: "SIP9005"<sip:9005@scm.com>
Call-ID: 13f15cdb-ce2b-4c28-9c8b-3234122aa4c7@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-13a41-
4cb91c9-228e28e1
P-Asserted-Identity: "SIP9004" <sip:9004@scm.com>
Max-Forwards: 70
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
Contact: <sip:9004@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 329

v=0
o=9004 1275911121 0 IN IP4 165.213.176.206
s=Samsung IP PBX
c=IN IP4 165.213.176.206
t=0 0
m=audio 20000 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

**(F2)**

```
SIP/2.0 100 Trying
From: "SIP9004"<sip:9004@scm.com>;tag=3f37247c-3953-47ac-b07e-
7e6d21dc2d6e
To: "SIP9005"<sip:9005@scm.com>
Call-ID: 13f15cdb-ce2b-4c28-9c8b-3234122aa4c7@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-13a41-
4cb91c9-228e28e1
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@10.254.168.97>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

**(F3)**

```
SIP/2.0 180 Ringing
From: "SIP9004"<sip:9004@scm.com>;tag=3f37247c-3953-47ac-b07e-
7e6d21dc2d6e
To: "SIP9005"<sip:9005@scm.com>;tag=440f78-61a8fe0a-13c4-50022-18e-
3ef4fdc5-18e
Call-ID: 13f15cdb-ce2b-4c28-9c8b-3234122aa4c7@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-13a41-
4cb91c9-228e28e1
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@10.254.168.97>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

**(F4)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=3f37247c-3953-47ac-b07e-
7e6d21dc2d6e
To: "SIP9005"<sip:9005@scm.com>;tag=440f78-61a8fe0a-13c4-50022-18e-
3ef4fdc5-18e
Call-ID: 13f15cdb-ce2b-4c28-9c8b-3234122aa4c7@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-13a41-
4cb91c9-228e28e1
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@10.254.168.97>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Type: application/sdp
Content-Length: 261

v=0
o=SAMSUNG_IP_PHONE 1275911115 1 IN IP4 10.254.168.97
s=SAMSUNG_VOIP_CALL
c=IN IP4 10.254.168.97
t=0 0
m=audio 20000 RTP/AVP 0 101
b=TIAS:64000
b=AS:82
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F5)**

```
ACK sip:9005@10.254.168.97 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3f37247c-3953-47ac-b07e-
7e6d21dc2d6e
To: "SIP9005"<sip:9005@scm.com>;tag=440f78-61a8fe0a-13c4-50022-18e-
3ef4fdc5-18e
Call-ID: 13f15cdb-ce2b-4c28-9c8b-3234122aa4c7@scm.com
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-13a43-
4cb996b-628e1a64
Max-Forwards: 70
Contact: <sip:9004@10.254.176.120:5060>
Content-Length: 0
```

### 2.3.4 Call Origination with Authentication Challenge

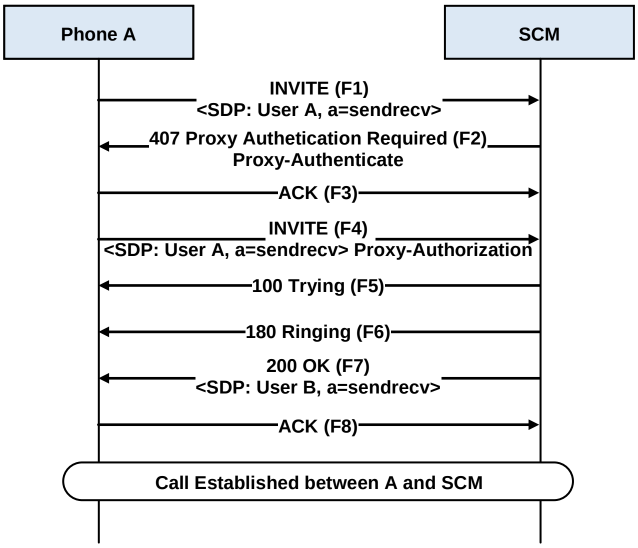



**Figure 2.5 Call Origination with Authentication Challenge**

**(F1)**

```
INVITE sip:9005@scm.com SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3ff440-ceb0d5a5-13c4-50022-
5ef41-5dceb83f-5ef41
To: <sip:9005@scm.com>
Call-ID: 401a78-ceb0d5a5-13c4-50022-5ef41-6826fb89-5ef41
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-5ef41-172e99a8-
2ea80b5c
Max-Forwards: 70
Supported: timer,replaces
P-Preferred-Identity: "SIP9004"<sip:9004@scm.com>
Contact: <sip:9004@165.213.176.206>
Session-Expires: 1800;refresher=uac
Min-SE: 90
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Type: application/sdp
Content-Length: 367
```

```
v=0
o=SAMSUNG_IP_PHONE 1276307701 0 IN IP4 165.213.176.206
s=SAMSUNG_VOIP_CALL
c=IN IP4 165.213.176.206
t=0 0
m=audio 20050 RTP/AVP 8 0 18 9 102 101
b=TIAS:64000
b=AS:82
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F2)**

```
SIP/2.0 407 Proxy Authentication Required
From: "SIP9004"<sip:9004@scm.com>;tag=3ff440-ceb0d5a5-13c4-50022-
5ef41-5dceb83f-5ef41
To: <sip:9005@scm.com>;tag=ac7e708-0-13c4-50022-74753-29434003-74753
Call-ID: 401a78-ceb0d5a5-13c4-50022-5ef41-6826fb89-5ef41
CSeq: 1 INVITE
Proxy-Authenticate: Digest
realm="scm.com",nonce="ac7e7087763f",stale=false,algorithm=MD5,qop="au
th"
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-5ef41-172e99a8-
2ea80b5c
Content-Length: 0
```

**(F3)**

```
ACK sip:9005@scm.com SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3ff440-ceb0d5a5-13c4-50022-
5ef41-5dceb83f-5ef41
To: <sip:9005@scm.com>;tag=ac7e708-0-13c4-50022-74753-29434003-74753
Call-ID: 401a78-ceb0d5a5-13c4-50022-5ef41-6826fb89-5ef41
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-5ef41-172e99a8-
2ea80b5c
Max-Forwards: 70
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Contact: <sip:9004@165.213.176.206>
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Length: 0
```

**(F4)**

```
INVITE sip:9005@scm.com SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3ff440-ceb0d5a5-13c4-50022-
5ef41-5dceb83f-5ef41
To: <sip:9005@scm.com>
Call-ID: 401a78-ceb0d5a5-13c4-50022-5ef41-6826fb89-5ef41
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-5ef41-172e99bc-
252932ba
Max-Forwards: 70
Supported: timer,replaces
P-Preferred-Identity: "SIP9004"<sip:9004@scm.com>
Contact: <sip:9004@165.213.176.206>
Proxy-Authorization: Digest
username="9004",realm="scm.com",nonce="ac7e7087763f",uri="sip:9005@scm
.com",response="14250d38c18692cf56e7cc13d918bc27",algorithm=MD5,cnonce
="172e99bc",qop=auth,nc=00000001
Session-Expires: 1800;refresher=uac
Min-SE: 90
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Type: application/sdp
Content-Length: 367

v=0
o=SAMSUNG_IP_PHONE 1276307701 0 IN IP4 165.213.176.206
s=SAMSUNG_VOIP_CALL
c=IN IP4 165.213.176.206
t=0 0
m=audio 20050 RTP/AVP 8 0 18 9 102 101
b=TIAS:64000
b=AS:82
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F5)**

```
SIP/2.0 100 Trying
From: "SIP9004"<sip:9004@scm.com>;tag=3ff440-ceb0d5a5-13c4-50022-
5ef41-5dceb83f-5ef41
To: <sip:9005@scm.com>
Call-ID: 401a78-ceb0d5a5-13c4-50022-5ef41-6826fb89-5ef41
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-5ef41-172e99bc-
252932ba
Contact: <sip:9005@scm.com>
Content-Length: 0
```

**(F6)**

```
SIP/2.0 180 Ringing
From: "SIP9004"<sip:9004@scm.com>;tag=3ff440-ceb0d5a5-13c4-50022-
5ef41-5dceb83f-5ef41
To: <sip:9005@scm.com>;tag=711c04ea-4230-4583-90f2-978872b3b3d7
Call-ID: 401a78-ceb0d5a5-13c4-50022-5ef41-6826fb89-5ef41
CSeq: 2 INVITE
P-Asserted-Identity: "SIP9005" <sip:9005@scm.com>
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-5ef41-172e99bc-
252932ba
Contact: <sip:9005@10.254.176.120:5060>
Content-Length: 0
```

**(F7)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=3ff440-ceb0d5a5-13c4-50022-
5ef41-5dceb83f-5ef41
To: <sip:9005@scm.com>;tag=711c04ea-4230-4583-90f2-978872b3b3d7
Call-ID: 401a78-ceb0d5a5-13c4-50022-5ef41-6826fb89-5ef41
CSeq: 2 INVITE
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
P-Asserted-Identity: "SIP9005" <sip:9005@scm.com>
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-5ef41-172e99bc-
252932ba
Contact: <sip:9005@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 248

v=0
o=9005 1276307693 1 IN IP4 10.254.176.120
s=Samsung IP PBX
c=IN IP4 165.213.176.39
t=0 0
m=audio 20020 RTP/AVP 0 101
```

```
b=TIAS:64000
b=AS:82
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F8)**

```
ACK sip:9005@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3ff440-ceb0d5a5-13c4-50022-
5ef41-5dceb83f-5ef41
To: <sip:9005@scm.com>;tag=711c04ea-4230-4583-90f2-978872b3b3d7
Call-ID: 401a78-ceb0d5a5-13c4-50022-5ef41-6826fb89-5ef41
CSeq: 2 ACK
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-5ef43-172ea01a-
5625472b
Max-Forwards: 70
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Contact: <sip:9004@165.213.176.206>
Proxy-Authorization: Digest
username="9004",realm="scm.com",nonce="ac7e7087763f",uri="sip:9005@scm
.com",response="14250d38c18692cf56e7cc13d918bc27",algorithm=MD5,cnonce
="172e99bc",qop=auth,nc=00000001
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Length: 0
```

## 2.3.5 Call Release by Phone

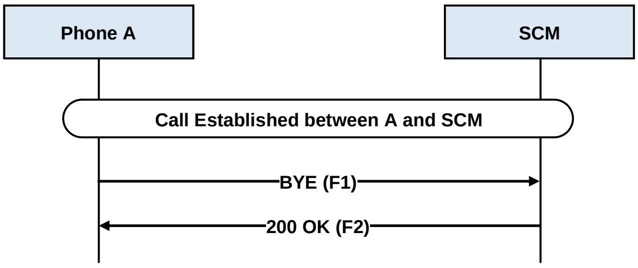



**Figure 2.6 Call Release by Phone**

**(F1)**

```
BYE sip:9005@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=4893d8-ceb0d5a5-13c4-50022-
19325-ca32bf6-19325
To: <sip:9005@scm.com>;tag=039d133a-4525-45b1-89b5-7500bc488d46
Call-ID: 48d5e8-ceb0d5a5-13c4-50022-19325-2606d1d0-19325
CSeq: 2 BYE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1932b-626e07a-
323044ee
Max-Forwards: 70
Supported: timer,replaces
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Length: 0
```

**(F2)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=4893d8-ceb0d5a5-13c4-50022-
19325-ca32bf6-19325
To: <sip:9005@scm.com>;tag=039d133a-4525-45b1-89b5-7500bc488d46
Call-ID: 48d5e8-ceb0d5a5-13c4-50022-19325-2606d1d0-19325
CSeq: 2 BYE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1932b-626e07a-
323044ee
Content-Length: 0
```

## 2.3.6 Call Release by SCM

3rd-Party-SIP-Phone should display reason using Reason Header including Control protocol when receive Request or Response Message.
Reason: Control; cause=[value]; text=[description]; charset=utf-8
3rd-Party-SIP-Phone do not display reason Reason Header protocol has SIP or Q.850.

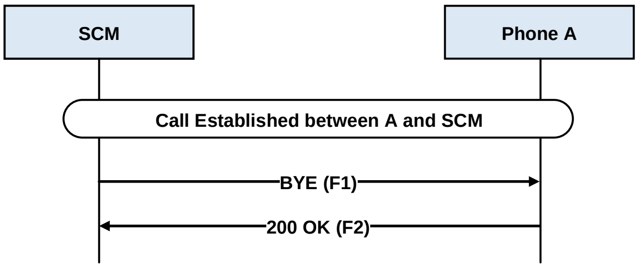



**Figure 2.7 Call Release by SCM**

**(F1)**

```
BYE sip:9005@10.254.168.97 SIP/2.0
From: “SIP9004”<sip:9004@scm.com>;tag=3f37247c-3953-47ac-b07e-
7e6d21dc2d6e
To: “SIP9005”<sip:9005@scm.com>;tag=440f78-61a8fe0a-13c4-50022-18e-
3ef4fdc5-18e
Call-ID: 13f15cdb-ce2b-4c28-9c8b-3234122aa4c7@scm.com
Cseq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-13a47-
4cba6ff-3c94f476
Reason: Q.850 ;cause=16 ;text=”Normal Release”
Max-Forwards: 70
Content-Length: 0
```

**(F2)**

```
SIP/2.0 200 OK
From: “SIP9004”<sip:9004@scm.com>;tag=3f37247c-3953-47ac-b07e-
7e6d21dc2d6e
To: “SIP9005”<sip:9005@scm.com>;tag=440f78-61a8fe0a-13c4-50022-18e-
3ef4fdc5-18e
Call-ID: 13f15cdb-ce2b-4c28-9c8b-3234122aa4c7@scm.com
Cseq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-13a47-
4cba6ff-3c94f476
Supported: timer,replaces
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

## 2.3.7 Call Cancel by SCM

3rd-Party-SIP-Phone should display Absence State when receive CANCEL Message.

3rd-Party-SIP-Phone do not display Absence State when receive CANCEL Message including prototol=Q.850, cause=31 or protocol=SIP, cause=200.
Reason: Q.850; cause=31; text='Normal Or Unspecified'; charset=euc-kr
Reason: SIP; cause=200; text='Call completed elsewhere'; charset=euc-kr

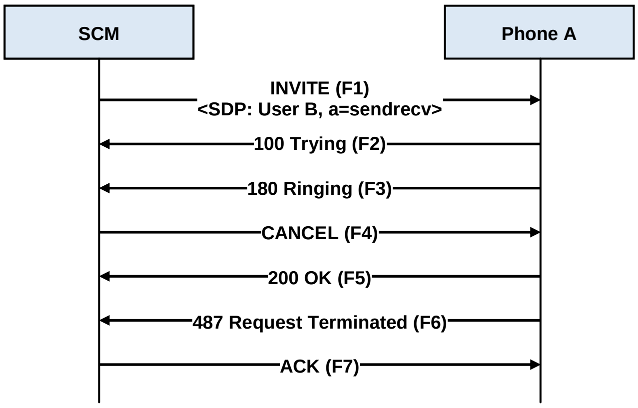



**Figure 2.8 Call Cancel by SCM**

**(F1)**

```
INVITE sip:9005@165.213.176.39 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3c7d4e2a-8eab-47ac-ab02-
56d37e9505d3
To: "SIP9005"<sip:9005@scm.com>
Call-ID: ccde5bc1-9d2e-4b91-8eb2-036d550a1dfd@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-93a9c-
240cf4bd-3bf10777
P-Asserted-Identity: "SIP9004" <sip:9004@scm.com>
Max-Forwards: 70
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
Contact: <sip:9004@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 329
v=0
o=9004 1276435526 0 IN IP4 165.213.176.206
s=Samsung IP PBX
c=IN IP4 165.213.176.206
t=0 0
m=audio 20056 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

**(F2)**

```
SIP/2.0 100 Trying
From: "SIP9004"<sip:9004@scm.com>;tag=3c7d4e2a-8eab-47ac-ab02-
56d37e9505d3
To: "SIP9005"<sip:9005@scm.com>
Call-ID: ccde5bc1-9d2e-4b91-8eb2-036d550a1dfd@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-93a9c-
240cf4bd-3bf10777
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@165.213.176.39>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

**(F3)**

```
SIP/2.0 180 Ringing
From: "SIP9004"<sip:9004@scm.com>;tag=3c7d4e2a-8eab-47ac-ab02-
56d37e9505d3
To: "SIP9005"<sip:9005@scm.com>;tag=3ffaa0-27b0d5a5-13c4-50022-5f77f-
768619d4-5f77f
Call-ID: ccde5bc1-9d2e-4b91-8eb2-036d550a1dfd@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-93a9c-
240cf4bd-3bf10777
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@165.213.176.39>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

**(F4)**

```
CANCEL sip:9005@165.213.176.39 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3c7d4e2a-8eab-47ac-ab02-
56d37e9505d3
To: "SIP9005"<sip:9005@scm.com>
Call-ID: ccde5bc1-9d2e-4b91-8eb2-036d550a1dfd@scm.com
CSeq: 1 CANCEL
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-93a9c-
240cf4bd-3bf10777
Max-Forwards: 70
Content-Length: 0
```

**(F5)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=3c7d4e2a-8eab-47ac-ab02-
56d37e9505d3
To: "SIP9005"<sip:9005@scm.com>;tag=3ffaa0-27b0d5a5-13c4-50022-5f77f-
768619d4-5f77f
Call-ID: ccde5bc1-9d2e-4b91-8eb2-036d550a1dfd@scm.com
CSeq: 1 CANCEL
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-93a9c-
240cf4bd-3bf10777
Supported: timer,replaces
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

**(F7)**

```
SIP/2.0 487 Request Terminated
From: "SIP9004"<sip:9004@scm.com>;tag=3c7d4e2a-8eab-47ac-ab02-
56d37e9505d3
To: "SIP9005"<sip:9005@scm.com>;tag=3ffaa0-27b0d5a5-13c4-50022-5f77f-
768619d4-5f77f
Call-ID: ccde5bc1-9d2e-4b91-8eb2-036d550a1dfd@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-93a9c-
240cf4bd-3bf10777
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

**(F8)**

```
ACK sip:9005@165.213.176.39 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3c7d4e2a-8eab-47ac-ab02-
56d37e9505d3
To: "SIP9005"<sip:9005@scm.com>;tag=3ffaa0-27b0d5a5-13c4-50022-5f77f-
768619d4-5f77f
Call-ID: ccde5bc1-9d2e-4b91-8eb2-036d550a1dfd@scm.com
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-93a9c-
240cf4bd-3bf10777
Max-Forwards: 70
Contact: <sip:9004@10.254.176.120:5060>
Content-Length: 0
```

### 2.3.8 Hold by Phone

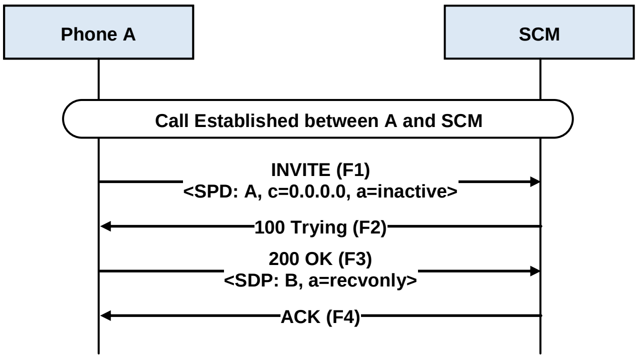



**Figure 2.9 Hold by Phone**

**(F1)**

```
INVITE sip:9005@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=4898a0-ceb0d5a5-13c4-50022-
1969c-69e5a7bf-1969c
To: <sip:9005@scm.com>;tag=0ef5b5a9-fb78-4d82-b15f-9bc78d63cd15
Call-ID: 48d7c8-ceb0d5a5-13c4-50022-1969c-651eca11-1969c
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-196a1-634654c-
1277cd80
Max-Forwards: 70
Supported: timer,replaces
P-Preferred-Identity: "SIP9004"<sip:9004@scm.com>
Contact: <sip:9004@165.213.176.206>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Type: application/sdp
Content-Length: 359

v=0
o=SAMSUNG_IP_PHONE 1275912008 1 IN IP4 165.213.176.206
s=SAMSUNG_VOIP_CALL
c=IN IP4 0.0.0.0
t=0 0
m=audio 20002 RTP/AVP 8 0 18 9 102 101
b=TIAS:64000
b=AS:82
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=inactive
```

**(F2)**

```
SIP/2.0 100 Trying
From: "SIP9004"<sip:9004@scm.com>;tag=4898a0-ceb0d5a5-13c4-50022-
1969c-69e5a7bf-1969c
To: <sip:9005@scm.com>;tag=0ef5b5a9-fb78-4d82-b15f-9bc78d63cd15
Call-ID: 48d7c8-ceb0d5a5-13c4-50022-1969c-651eca11-1969c
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-196a1-634654c-
1277cd80
Contact: <sip:9005@10.254.176.120:5060>
Content-Length: 0
```

**(F3)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=4898a0-ceb0d5a5-13c4-50022-
1969c-69e5a7bf-1969c
To: <sip:9005@scm.com>;tag=0ef5b5a9-fb78-4d82-b15f-9bc78d63cd15
Call-ID: 48d7c8-ceb0d5a5-13c4-50022-1969c-651eca11-1969c
CSeq: 2 INVITE
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
P-Asserted-Identity: "SIP9005" <sip:9005@scm.com>
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-196a1-634654c-
1277cd80
Contact: <sip:9005@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 191

v=0
o=9005 1275912002 2 IN IP4 10.254.176.120
s=Samsung IP PBX
c=IN IP4 10.254.168.97
t=0 0
m=audio 20002 RTP/AVP 0
b=TIAS:64000
b=AS:82
a=rtpmap:0 PCMU/8000
a=ptime:20
a=recvonly
```

**(F4)**

```
ACK sip:9005@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=4898a0-ceb0d5a5-13c4-50022-
1969c-69e5a7bf-1969c
To: <sip:9005@scm.com>;tag=0ef5b5a9-fb78-4d82-b15f-9bc78d63cd15
Call-ID: 48d7c8-ceb0d5a5-13c4-50022-1969c-651eca11-1969c
CSeq: 2 ACK
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-196a1-6346646-
351e1c24
Max-Forwards: 70
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Contact: <sip:9004@165.213.176.206>
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Length: 0
```

## 2.3.9 Hold by SCM (Music On Hold)

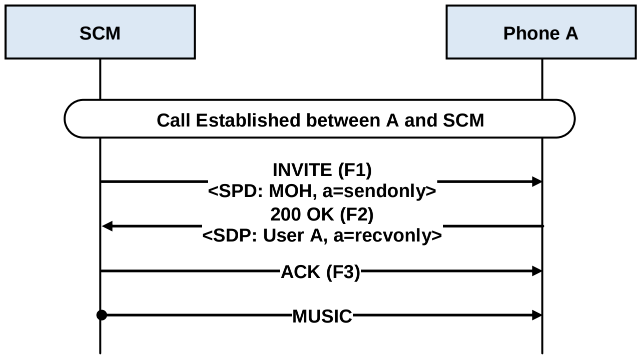



**Figure 2.10 Hold by SCM (Music On Hold)**

a=sendonly may be sendrecv for NAT Traveral

**(F1)**

```
INVITE sip:9005@10.254.168.97 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=fe9fcaf2-0c3f-4289-b1dd-
30b7d22fdb2b
To: "SIP9005"<sip:9005@scm.com>;tag=4412a8-61a8fe0a-13c4-50022-505-
1b211811-505
Call-ID: a2fa2930-1442-4010-8fad-03ffd31a78f9@scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-13dbd-
4d92bdc-4462e7f7
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
P-Asserted-Identity: "SIP9004" <sip:9004@scm.com>
Max-Forwards: 70
Contact: <sip:9004@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 268

v=0
o=9004 100009 100009 IN IP4 10.254.176.120
s=Samsung IP PBX
c=IN IP4 10.254.176.120
t=0 0
m=audio 35000 RTP/AVP 0 8 4 18
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=ptime:10
a=rtpmap:4 G723/8000
a=ptime:30
a=rtpmap:18 G729/8000
a=ptime:10
a=sendonly
```

**(F2)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=fe9fcaf2-0c3f-4289-b1dd-
30b7d22fdb2b
To: "SIP9005"<sip:9005@scm.com>;tag=4412a8-61a8fe0a-13c4-50022-505-
1b211811-505
Call-ID: a2fa2930-1442-4010-8fad-03ffd31a78f9@scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-13dbd-
4d92bdc-4462e7f7
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@10.254.168.97>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Type: application/sdp
Content-Length: 205

v=0
o=SAMSUNG_IP_PHONE 1275912002 2 IN IP4 10.254.168.97
s=SAMSUNG_VOIP_CALL
c=IN IP4 10.254.168.97
t=0 0
m=audio 20002 RTP/AVP 0
b=TIAS:64000
b=AS:82
a=rtpmap:0 PCMU/8000
a=ptime:20
a=recvonly
```

**(F3)**

```
ACK sip:9005@10.254.168.97 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=fe9fcaf2-0c3f-4289-b1dd-
30b7d22fdb2b
To: "SIP9005"<sip:9005@scm.com>;tag=4412a8-61a8fe0a-13c4-50022-505-
1b211811-505
Call-ID: a2fa2930-1442-4010-8fad-03ffd31a78f9@scm.com
CSeq: 2 ACK
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-13dbd-
4d92c9c-13e23647
Max-Forwards: 70
Contact: <sip:9004@10.254.176.120:5060>
Content-Length: 0
```

## 2.3.10 Hold by SCM (Simple Hold)

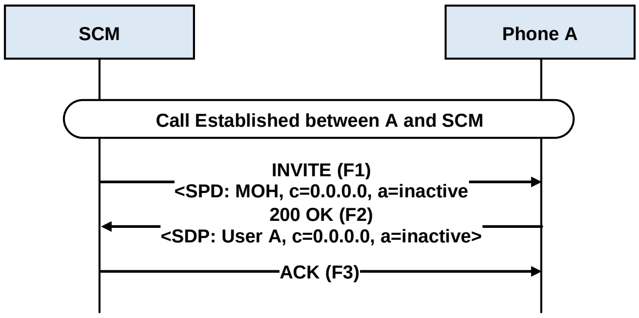



**Figure 2.11 Hold by SCM (Simple Hold)**

**(F1)**

```
INVITE sip:9005@10.254.168.97 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=130d2c2e-2e20-4f9a-b369-
9ac97d5e4609
To: "SIP9005"<sip:9005@scm.com>;tag=441908-61a8fe0a-13c4-50022-60b-
2efb0f32-60b
Call-ID: 5c145066-c2b9-4dfa-a12e-6c1b3da3d023@scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-13edd-
4dd90bf-549721c9
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
P-Asserted-Identity: "SIP9004" <sip:9004@scm.com>
Max-Forwards: 70
Contact: <sip:9004@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 313

v=0
o=9004 1275912270 1 IN IP4 0.0.0.0
s=Samsung IP PBX
c=IN IP4 0.0.0.0
t=0 0
m=audio 20004 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=inactive
```

**(F2)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=130d2c2e-2e20-4f9a-b369-
9ac97d5e4609
To: "SIP9005"<sip:9005@scm.com>;tag=441908-61a8fe0a-13c4-50022-60b-
2efb0f32-60b
Call-ID: 5c145066-c2b9-4dfa-a12e-6c1b3da3d023@scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-13edd-
4dd90bf-549721c9
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@10.254.168.97>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_IP_PHONE 1275912263 2 IN IP4 10.254.168.97
s=SAMSUNG_VOIP_CALL
c=IN IP4 0.0.0.0
t=0 0
m=audio 20004 RTP/AVP 0 101
b=TIAS:64000
b=AS:82
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=inactive
```

**(F3)**

```
ACK sip:9005@10.254.168.97 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=130d2c2e-2e20-4f9a-b369-
9ac97d5e4609
To: "SIP9005"<sip:9005@scm.com>;tag=441908-61a8fe0a-13c4-50022-60b-
2efb0f32-60b
Call-ID: 5c145066-c2b9-4dfa-a12e-6c1b3da3d023@scm.com
CSeq: 2 ACK
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-13edd-
4dd91a4-5e907166
Max-Forwards: 70
Contact: <sip:9004@10.254.176.120:5060>
Content-Length: 0
```

## 2.3.11 Media Change by Phone

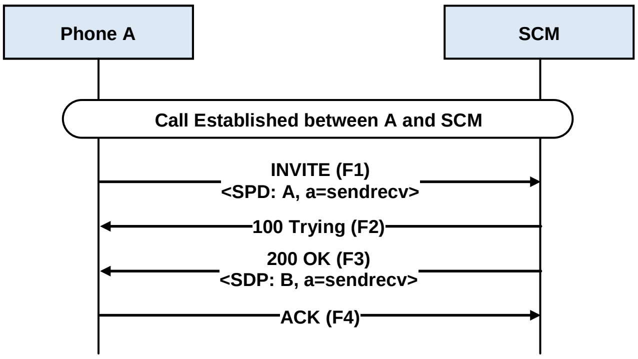



**Figure 2.12 Media Change by Phone**

**(F1)**

```
INVITE sip:9005@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=48aa28-ceb0d5a5-13c4-50022-
19921-66e4522d-19921
To: <sip:9005@scm.com>;tag=c1d007c8-f13c-42cf-b708-53d33063c2a6
Call-ID: 48db88-ceb0d5a5-13c4-50022-19921-6d4e099f-19921
CSeq: 5 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1996a-63f4728-
5a3bb61d
Max-Forwards: 70
Supported: timer,replaces
P-Preferred-Identity: "SIP9004"<sip:9004@scm.com>
Contact: <sip:9004@165.213.176.206>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Type: application/sdp
Content-Length: 367

v=0
o=SAMSUNG_IP_PHONE 1275912653 4 IN IP4 165.213.176.206
s=SAMSUNG_VOIP_CALL
c=IN IP4 165.213.176.206
t=0 0
m=audio 20006 RTP/AVP 8 0 18 9 102 101
b=TIAS:64000
b=AS:82
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F2)**

```
SIP/2.0 100 Trying
From: "SIP9004"<sip:9004@scm.com>;tag=48aa28-ceb0d5a5-13c4-50022-
19921-66e4522d-19921
To: <sip:9005@scm.com>;tag=c1d007c8-f13c-42cf-b708-53d33063c2a6
Call-ID: 48db88-ceb0d5a5-13c4-50022-19921-6d4e099f-19921
CSeq: 5 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1996a-63f4728-
5a3bb61d
Contact: <sip:9005@10.254.176.120:5060>
Content-Length: 0
```

**(F3)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=48aa28-ceb0d5a5-13c4-50022-
19921-66e4522d-19921
To: <sip:9005@scm.com>;tag=c1d007c8-f13c-42cf-b708-53d33063c2a6
Call-ID: 48db88-ceb0d5a5-13c4-50022-19921-6d4e099f-19921
CSeq: 5 INVITE
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
P-Asserted-Identity: "SIP9005" <sip:9005@scm.com>
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1996a-63f4728-
5a3bb61d
Contact: <sip:9005@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 247

v=0
o=9005 1275912647 5 IN IP4 10.254.176.120
s=Samsung IP PBX
c=IN IP4 10.254.168.97
t=0 0
m=audio 20006 RTP/AVP 0 101
b=TIAS:64000
b=AS:82
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F4)**

```
ACK sip:9005@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=48aa28-ceb0d5a5-13c4-50022-
19921-66e4522d-19921
To: <sip:9005@scm.com>;tag=c1d007c8-f13c-42cf-b708-53d33063c2a6
Call-ID: 48db88-ceb0d5a5-13c4-50022-19921-6d4e099f-19921
CSeq: 5 ACK
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1996a-63f47c8-
25df00ca
Max-Forwards: 70
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Contact: <sip:9004@165.213.176.206>
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Length: 0
```

## 2.3.12 Media Change by SCM (with SDP)

3rd-Party-SIP-Phone must answer with one of offered media info when receive Re-INVITE with SDP

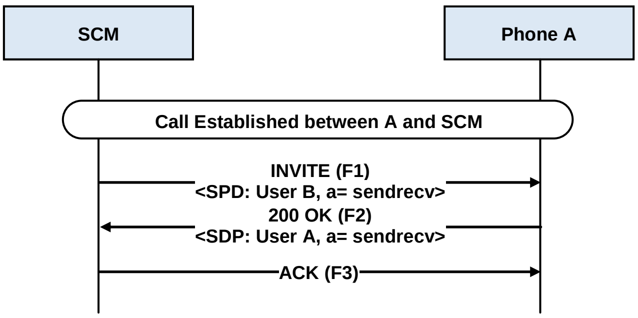



**Figure 2.13 Media Change by SCM**

**(F1)**

```
INVITE sip:9005@10.254.168.97 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=dae99f68-ff59-441f-bf45-
5f580f1bf816
To: "SIP9005"<sip:9005@scm.com>;tag=441f68-61a8fe0a-13c4-50022-78a-
23dc2f6b-78a
Call-ID: a9e0f85a-9df1-4b0b-b8d2-318ee67aebdb@scm.com
CSeq: 5 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-14086-
4e40d4b-7be0c25b
```

```
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
P-Asserted-Identity: "SIP9004" <sip:9004@scm.com>
Max-Forwards: 70
Contact: <sip:9004@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 329

v=0
o=9004 1275912653 4 IN IP4 165.213.176.206
s=Samsung IP PBX
c=IN IP4 165.213.176.206
t=0 0
m=audio 20006 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

**(F2)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=dae99f68-ff59-441f-bf45-
5f580f1bf816
To: "SIP9005"<sip:9005@scm.com>;tag=441f68-61a8fe0a-13c4-50022-78a-
23dc2f6b-78a
Call-ID: a9e0f85a-9df1-4b0b-b8d2-318ee67aebdb@scm.com
CSeq: 5 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-14086-
4e40d4b-7be0c25b
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@10.254.168.97>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Type: application/sdp
Content-Length: 261

v=0
o=SAMSUNG_IP_PHONE 1275912647 5 IN IP4 10.254.168.97
s=SAMSUNG_VOIP_CALL
c=IN IP4 10.254.168.97
t=0 0
m=audio 20006 RTP/AVP 0 101
```

```
b=TIAS:64000
b=AS:82
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F3)**

```
ACK sip:9005@10.254.168.97 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=dae99f68-ff59-441f-bf45-
5f580f1bf816
To: "SIP9005"<sip:9005@scm.com>;tag=441f68-61a8fe0a-13c4-50022-78a-
23dc2f6b-78a
Call-ID: a9e0f85a-9df1-4b0b-b8d2-318ee67aebdb@scm.com
CSeq: 5 ACK
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-14086-
4e40ddd-5f130efb
Max-Forwards: 70
Contact: <sip:9004@10.254.176.120:5060>
Content-Length: 0
```

## 2.3.13 Media Change by SCM (without SDP)

3rd-Party-SIP-Phone must answer with every media info in the phone capability when receive Re-INVITE without SDP.

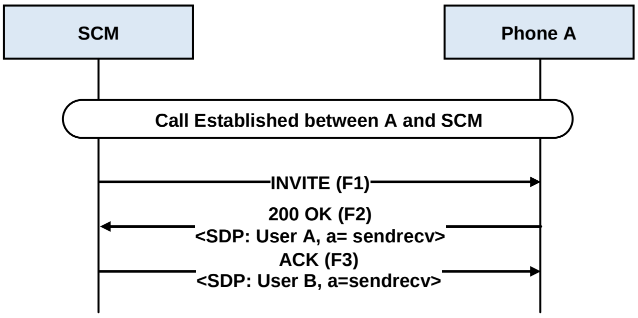



**Figure 2.14 Media Change by SCM (without SDP)**

**(F1)**

```
INVITE sip:9001@10.254.168.82 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=75c32024-28f0-4ff9-a480-
36f086fb1dd8
To: "SIP9001"<sip:9001@scm.com>;tag=959b0f78-afea852-13c4-1c0c3-
78195260-1c0c3
Call-ID: 7ce59293-9d3b-4243-a804-ecc6ddf4647d@scm.com
CSeq: 5 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-1416d-
4e792ae-7e7412b4
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
P-Asserted-Identity: "SIP9005" <sip:9005@scm.com>
Max-Forwards: 70
Contact: <sip:9005@10.254.176.120:5060>
Content-Length: 0
```

**(F2)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=75c32024-28f0-4ff9-a480-
36f086fb1dd8
To: "SIP9001"<sip:9001@scm.com>;tag=959b0f78-afea852-13c4-1c0c3-
78195260-1c0c3
Call-ID: 7ce59293-9d3b-4243-a804-ecc6ddf4647d@scm.com
CSeq: 5 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-1416d-
4e792ae-7e7412b4
Supported: replaces,timer
```

```
User-Agent: SAMSUNG,ITP-5112L,0113,1101,0150
Accept: application/sdp
Contact: <sip:9001@10.254.168.82>
Allow: INVITE,ACK,CANCEL,BYE,PRACK,REFER,SUBSCRIBE,NOTIFY,UPDATE
Allow-Events: talk,hold
Content-Type: application/sdp
Content-Length: 274

v=0
o=SAMSUNG_IP_PHONE 57455086 4 IN IP4 10.254.168.82
s=SAMSUNG_VOIP_CALL
c=IN IP4 10.254.168.82
t=0 0
m=audio 27058 RTP/AVP 8 18 0 101
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

**(F3)**

```
ACK sip:9001@10.254.168.82 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=75c32024-28f0-4ff9-a480-
36f086fb1dd8
To: "SIP9001"<sip:9001@scm.com>;tag=959b0f78-afea852-13c4-1c0c3-
78195260-1c0c3
Call-ID: 7ce59293-9d3b-4243-a804-ecc6ddf4647d@scm.com
CSeq: 5 ACK
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-1416d-
4e7937d-3024be06
Max-Forwards: 70
Contact: <sip:9005@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 247

v=0
o=9005 1275912945 2 IN IP4 10.254.176.120
s=Samsung IP PBX
c=IN IP4 10.254.168.97
t=0 0
m=audio 20008 RTP/AVP 0 101
b=TIAS:64000
b=AS:82
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

### 2.3.14 Blind Transfer by Phone

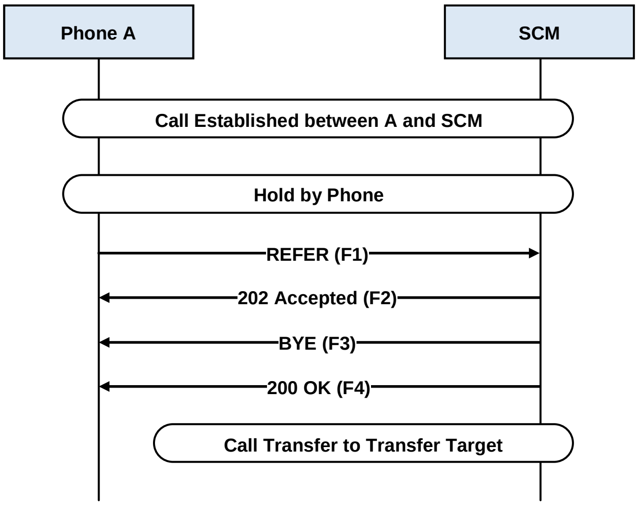



**Figure 2.15 Blind Transfer by Phone**

**(F1)**

```
REFER sip:9005@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3fd2c8-ceb0d5a5-13c4-50022-81-
406ba601-81
To: <sip:9005@scm.com>;tag=e4af7a50-9c58-41cb-acf7-3e3253d141ca
Call-ID: 4014d8-ceb0d5a5-13c4-50022-81-3339312b-81
CSeq: 3 REFER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-a1-277fe-2417e951
Refer-To: <sip:9003@scm.com>
Referred-By: <sip:9004@scm.com>
Max-Forwards: 70
Supported: timer,replaces
Contact: <sip:9004@165.213.176.206>
Content-Length: 0
```

**(F2)**

```
SIP/2.0 202 Accepted
From: "SIP9004"<sip:9004@scm.com>;tag=3fd2c8-ceb0d5a5-13c4-50022-81-
406ba601-81
To: <sip:9005@scm.com>;tag=e4af7a50-9c58-41cb-acf7-3e3253d141ca
Call-ID: 4014d8-ceb0d5a5-13c4-50022-81-3339312b-81
CSeq: 3 REFER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-a1-277fe-2417e951
Contact: <sip:9005@10.254.176.120:5060>
Content-Length: 0
```

**(F3)**

```
BYE sip:9004@165.213.176.206 SIP/2.0
From: <sip:9005@scm.com>;tag=e4af7a50-9c58-41cb-acf7-3e3253d141ca
To: "SIP9004"<sip:9004@scm.com>;tag=3fd2c8-ceb0d5a5-13c4-50022-81-
406ba601-81
Call-ID: 4014d8-ceb0d5a5-13c4-50022-81-3339312b-81
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-158cc-
542dd34-48ab1dcb
Max-Forwards: 70
Content-Length: 0
```

**(F4)**

```
SIP/2.0 200 OK
From: <sip:9005@scm.com>;tag=e4af7a50-9c58-41cb-acf7-3e3253d141ca
To: "SIP9004"<sip:9004@scm.com>;tag=3fd2c8-ceb0d5a5-13c4-50022-81-
406ba601-81
Call-ID: 4014d8-ceb0d5a5-13c4-50022-81-3339312b-81
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-158cc-
542dd34-48ab1dcb
Supported: timer,replaces
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

### 2.3.15 Consultative Transfer after Answer by Phone

Consultative Call Transfer After Answer
Screened Transfer

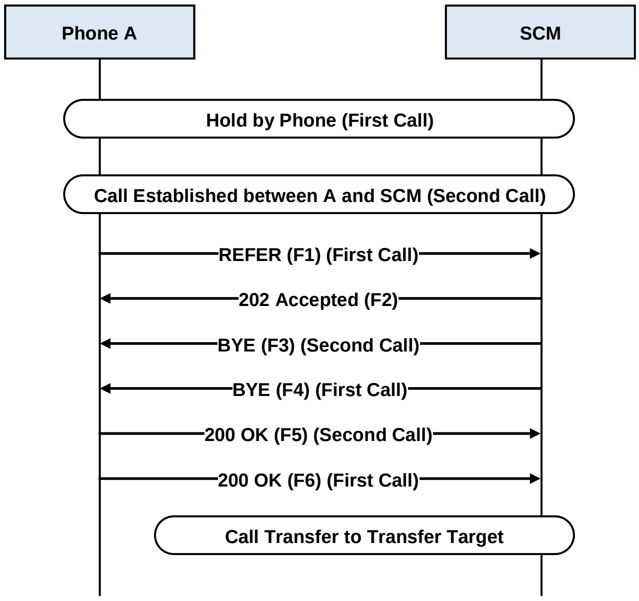



**Figure 2.16 Consultative Transfer after Answer by Phone**

**(F1)**

```
REFER sip:9005@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3fd928-ceb0d5a5-13c4-50022-54d-
427c6694-54d
To: <sip:9005@scm.com>;tag=b4266b70-338c-4a2c-a082-4740156dfe19
Call-ID: 4025b8-ceb0d5a5-13c4-50022-54d-22cfd596-54d
CSeq: 3 REFER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-558-14e29a-
7af80a04
Refer-To: <sip:9003@scm.com?Replaces=402798-ceb0d5a5-13c4-50022-552-
5701294c-552%3Bto-tag%3Deaf0ca4a-7647-45c0-8c0f-650b800c241a%3Bfrom-
tag%3D3fdac0-ceb0d5a5-13c4-50022-552-5c8e3faa-552>
Referred-By: <sip:9004@scm.com>
Max-Forwards: 70
Supported: timer,replaces
Contact: <sip:9004@165.213.176.206>
Content-Length: 0
```

**(F2)**

```
SIP/2.0 202 Accepted
From: "SIP9004"<sip:9004@scm.com>;tag=3fd928-ceb0d5a5-13c4-50022-54d-
427c6694-54d
To: <sip:9005@scm.com>;tag=b4266b70-338c-4a2c-a082-4740156dfe19
Call-ID: 4025b8-ceb0d5a5-13c4-50022-54d-22cfd596-54d
CSeq: 3 REFER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-558-14e29a-
7af80a04
Contact: <sip:9005@10.254.176.120:5060>
Content-Length: 0
```

**(F3)**

```
BYE sip:9004@165.213.176.206 SIP/2.0
From: <sip:9003@scm.com>;tag=eaf0ca4a-7647-45c0-8c0f-650b800c241a
To: "SIP9004"<sip:9004@scm.com>;tag=3fdac0-ceb0d5a5-13c4-50022-552-
5c8e3faa-552
Call-ID: 402798-ceb0d5a5-13c4-50022-552-5701294c-552
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-15d82-
5554774-36fa7b85
Max-Forwards: 70
Content-Length: 0
```

**(F4)**

```
BYE sip:9004@165.213.176.206 SIP/2.0
From: <sip:9005@scm.com>;tag=b4266b70-338c-4a2c-a082-4740156dfe19
To: "SIP9004"<sip:9004@scm.com>;tag=3fd928-ceb0d5a5-13c4-50022-54d-
427c6694-54d
Call-ID: 4025b8-ceb0d5a5-13c4-50022-54d-22cfd596-54d
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-15d83-
55547cd-13d43d9a
Max-Forwards: 70
Content-Length: 0
```

**(F5)**

```
SIP/2.0 200 OK
From: <sip:9003@scm.com>;tag=eaf0ca4a-7647-45c0-8c0f-650b800c241a
To: "SIP9004"<sip:9004@scm.com>;tag=3fdac0-ceb0d5a5-13c4-50022-552-
5c8e3faa-552
Call-ID: 402798-ceb0d5a5-13c4-50022-552-5701294c-552
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-15d82-
5554774-36fa7b85
Supported: timer,replaces
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

**(F6)**

```
SIP/2.0 200 OK
From: <sip:9005@scm.com>;tag=b4266b70-338c-4a2c-a082-4740156dfe19
To: "SIP9004"<sip:9004@scm.com>;tag=3fd928-ceb0d5a5-13c4-50022-54d-
427c6694-54d
Call-ID: 4025b8-ceb0d5a5-13c4-50022-54d-22cfd596-54d
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-15d83-
55547cd-13d43d9a
Supported: timer,replaces
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

## 2.3.16 Consultative Transfer before Answer by Phone

Consultative Call Transfer Before Answer
Unscreened Transfer

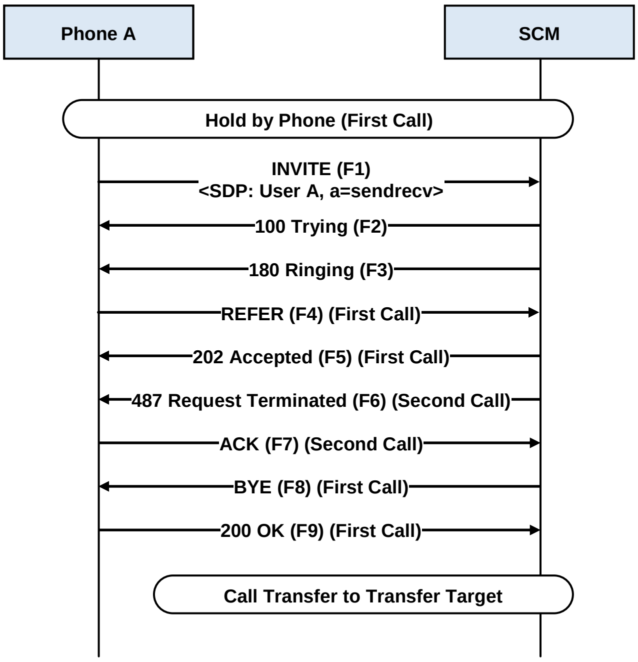



**Figure 2.17 Consultative Transfer before Answer by Phone**

**(F1)**

```
INVITE sip:9003@scm.com SIP/2.0
From: "SIP5004"<sip:9004@scm.com>;tag=681768-ceb0d5a5-13c4-50022-
17a8e-23901212-17a8e
To: <sip:9003@scm.com>
Call-ID: 685a38-ceb0d5a5-13c4-50022-17a8e-42fb2534-17a8e
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-17a8e-5c6bc0e-
36a86cfc
Max-Forwards: 70
Supported: timer,replaces
P-Preferred-Identity: "SIP5004"<sip:9004@scm.com>
User-Agent: SAMSUNG-SMT-i5243/V1.00
Contact: <sip:9004@165.213.176.206>
```

```
Session-Expires: 1800;refresher=uac
Min-SE: 90
Allow: INVITE,ACK,CANCEL,BYE,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Allow-Events: talk,hold
Content-Type: application/sdp
Content-Length: 367

v=0
o=SAMSUNG_IP_PHONE 1278767394 0 IN IP4 165.213.176.206
s=SAMSUNG_VOIP_CALL
c=IN IP4 165.213.176.206
t=0 0
m=audio 20114 RTP/AVP 8 0 18 9 102 101
b=TIAS:64000
b=AS:82
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F2)**

```
SIP/2.0 100 Trying
From: "SIP5004"<sip:9004@scm.com>;tag=681768-ceb0d5a5-13c4-50022-
17a8e-23901212-17a8e
To: <sip:9003@scm.com>
Call-ID: 685a38-ceb0d5a5-13c4-50022-17a8e-42fb2534-17a8e
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-17a8e-5c6bc0e-
36a86cfc
Contact: <sip:9003@scm.com>
Content-Length: 0
```

**(F3)**

```
SIP/2.0 180 Ringing
From: "SIP5004"<sip:9004@scm.com>;tag=681768-ceb0d5a5-13c4-50022-
17a8e-23901212-17a8e
To: <sip:9003@scm.com>;tag=8f53616f-2552-45fc-aa15-d625856568a2
Call-ID: 685a38-ceb0d5a5-13c4-50022-17a8e-42fb2534-17a8e
CSeq: 1 INVITE
P-Asserted-Identity: "SIP5003" <sip:9003@scm.com>
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-17a8e-5c6bc0e-
36a86cfc
Contact: <sip:9003@10.254.176.120:5060>
Content-Length: 0
```

**(F4)**

```
REFER sip:9005@10.254.176.120:5060 SIP/2.0
From: "SIP5004"<sip:9004@scm.com>;tag=681108-ceb0d5a5-13c4-50022-
17a84-2f92187c-17a84
To: <sip:9005@scm.com>;tag=e8b5f5f6-ce93-4b2c-8bd0-eafc882a680d
Call-ID: 685858-ceb0d5a5-13c4-50022-17a84-a36b8fe-17a84
CSeq: 3 REFER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-17a91-5c6c71c-
529e5728
Refer-To: <sip:9003@scm.com?Replaces=685a38-ceb0d5a5-13c4-50022-17a8e-
42fb2534-17a8e%3Bto-tag%3D8f53616f-2552-45fc-aa15-d625856568a2%3Bfrom-
tag%3D681768-ceb0d5a5-13c4-50022-17a8e-23901212-17a8e>
Referred-By: <sip:9004@scm.com>
Max-Forwards: 70
Supported: timer,replaces
Contact: <sip:9004@165.213.176.206>
Content-Length: 0
```

**(F5)**

```
SIP/2.0 202 Accepted
From: "SIP5004"<sip:9004@scm.com>;tag=681108-ceb0d5a5-13c4-50022-
17a84-2f92187c-17a84
To: <sip:9005@scm.com>;tag=e8b5f5f6-ce93-4b2c-8bd0-eafc882a680d
Call-ID: 685858-ceb0d5a5-13c4-50022-17a84-a36b8fe-17a84
CSeq: 3 REFER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-17a91-5c6c71c-
529e5728
Contact: <sip:9005@10.254.176.120:5060>
Content-Length: 0
```

**(F6)**

```
SIP/2.0 487 Request Terminated
From: "SIP5004"<sip:9004@scm.com>;tag=681768-ceb0d5a5-13c4-50022-
17a8e-23901212-17a8e
To: <sip:9003@scm.com>;tag=8f53616f-2552-45fc-aa15-d625856568a2
Call-ID: 685a38-ceb0d5a5-13c4-50022-17a8e-42fb2534-17a8e
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-17a8e-5c6bc0e-
36a86cfc
Content-Length: 0
```

**(F7)**

```
ACK sip:9003@scm.com SIP/2.0
From: "SIP5004"<sip:9004@scm.com>;tag=681768-ceb0d5a5-13c4-50022-
17a8e-23901212-17a8e
To: <sip:9003@scm.com>;tag=8f53616f-2552-45fc-aa15-d625856568a2
Call-ID: 685a38-ceb0d5a5-13c4-50022-17a8e-42fb2534-17a8e
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-17a8e-5c6bc0e-
36a86cfc
Max-Forwards: 70
User-Agent: SAMSUNG-SMT-i5243/V1.00
Allow: INVITE,ACK,CANCEL,BYE,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Contact: <sip:9004@165.213.176.206>
Allow-Events: talk,hold
Content-Length: 0
```

**(F8)**

```
BYE sip:9004@165.213.176.206 SIP/2.0
From: <sip:9005@scm.com>;tag=e8b5f5f6-ce93-4b2c-8bd0-eafc882a680d
To: "SIP5004"<sip:9004@scm.com>;tag=681108-ceb0d5a5-13c4-50022-17a84-
2f92187c-17a84
Call-ID: 685858-ceb0d5a5-13c4-50022-17a84-a36b8fe-17a84
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-19cf6-
64d24d2-11f0ff9
Max-Forwards: 70
Content-Length: 0
```

**(F9)**

```
SIP/2.0 200 OK
From: <sip:9005@scm.com>;tag=e8b5f5f6-ce93-4b2c-8bd0-eafc882a680d
To: "SIP5004"<sip:9004@scm.com>;tag=681108-ceb0d5a5-13c4-50022-17a84-
2f92187c-17a84
Call-ID: 685858-ceb0d5a5-13c4-50022-17a84-a36b8fe-17a84
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-19cf6-
64d24d2-11f0ff9
Supported: timer,replaces
Allow: INVITE,ACK,CANCEL,BYE,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Allow-Events: talk,hold
Content-Length: 0
```

## 2.3.17 Ad-hoc Conference

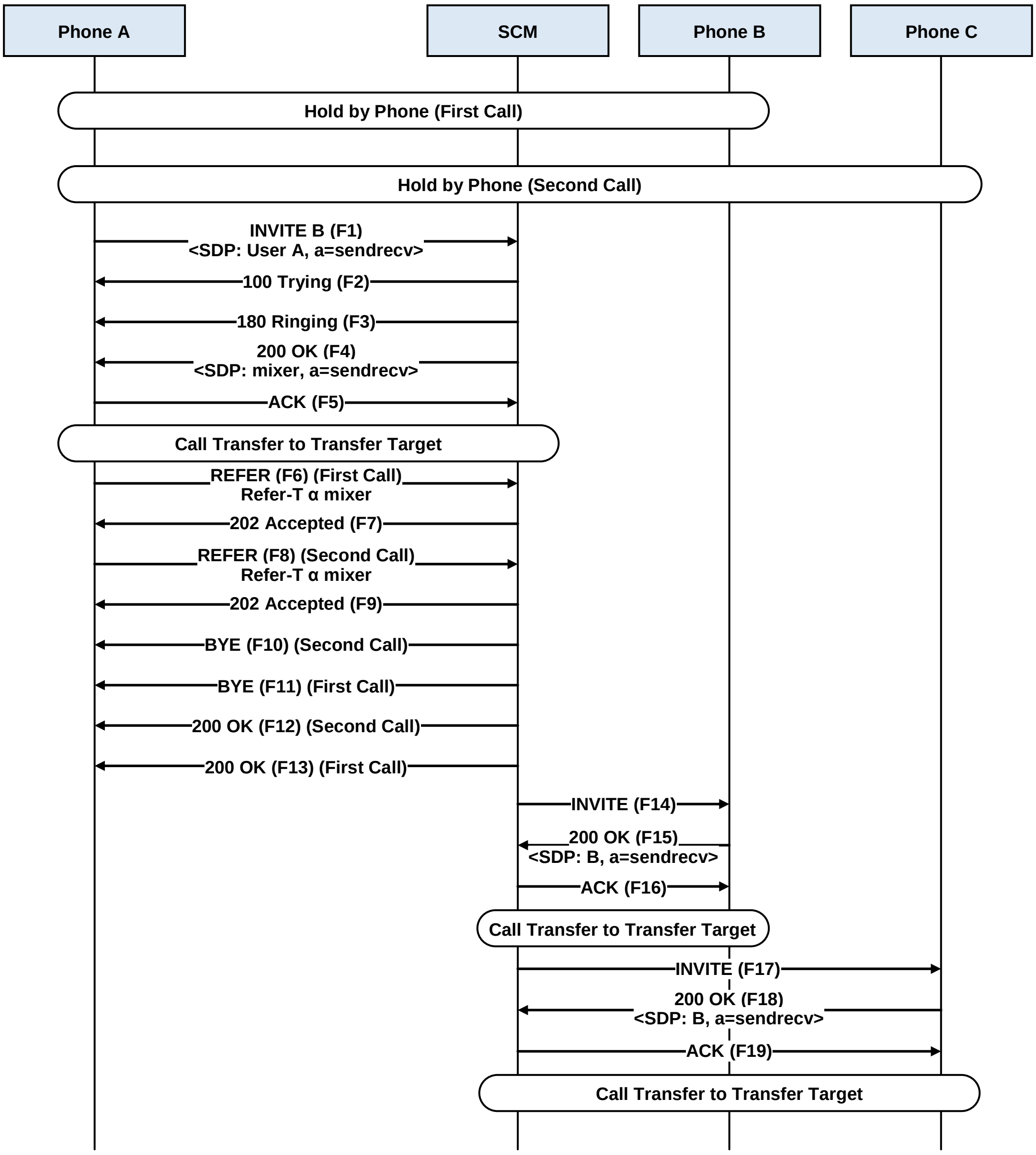



**Figure 2.18 ad-hoc Confernece**

**(F1)**

```
INVITE sip:*42@scm.com SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3fede0-ceb0d5a5-13c4-50022-
1ee72-722b1847-1ee72
To: <sip:*42@scm.com>
Call-ID: 403698-ceb0d5a5-13c4-50022-1ee72-2488c219-1ee72
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1ee72-78b70de-
2c8c378d
Max-Forwards: 70
Supported: timer,replaces
P-Preferred-Identity: "SIP9004"<sip:9004@scm.com>
Contact: <sip:9004@165.213.176.206>
Session-Expires: 1800;refresher=uac
Min-SE: 90
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Type: application/sdp
Content-Length: 367

v=0
o=SAMSUNG_IP_PHONE 1276045350 0 IN IP4 165.213.176.206
s=SAMSUNG_VOIP_CALL
c=IN IP4 165.213.176.206
t=0 0
m=audio 20036 RTP/AVP 8 0 18 9 102 101
b=TIAS:64000
b=AS:82
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F2)**

```
SIP/2.0 100 Trying
From: "SIP9004"<sip:9004@scm.com>;tag=3fede0-ceb0d5a5-13c4-50022-
1ee72-722b1847-1ee72
To: <sip:*42@scm.com>
Call-ID: 403698-ceb0d5a5-13c4-50022-1ee72-2488c219-1ee72
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1ee72-78b70de-
2c8c378d
Contact: <sip:*42@scm.com>
Content-Length: 0
```

**(F3)**

```
SIP/2.0 180 Ringing
From: "SIP9004"<sip:9004@scm.com>;tag=3fede0-ceb0d5a5-13c4-50022-
1ee72-722b1847-1ee72
To: <sip:*42@scm.com>;tag=fe771969-d00e-4074-bbf8-b7359a7bd000
Call-ID: 403698-ceb0d5a5-13c4-50022-1ee72-2488c219-1ee72
CSeq: 1 INVITE
P-Asserted-Identity: <sip:*42@scm.com>
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1ee72-78b70de-
2c8c378d
Contact: <sip:*42@10.254.176.120:5060>
Content-Length: 0
```

**(F4)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=3fede0-ceb0d5a5-13c4-50022-
1ee72-722b1847-1ee72
To: <sip:*42@scm.com>;tag=fe771969-d00e-4074-bbf8-b7359a7bd000
Call-ID: 403698-ceb0d5a5-13c4-50022-1ee72-2488c219-1ee72
CSeq: 1 INVITE
P-Asserted-Identity: <sip:*42@scm.com>
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1ee72-78b70de-
2c8c378d
Contact: <sip:8002@10.254.176.120:5060>;isfocus
Content-Type: application/sdp
Content-Length: 395

v=0
o=8002 0 0 IN IP4 10.254.176.120
s=Samsung IP PBX
u=sip:8002@scm.com
c=IN IP4 10.254.176.120
t=0 0
a=x-master:9004@scm.com
a=x-confInfo:<conf type="35" conf_id="8002" ug="1" rec="0"><aud_opt
mixmode="1" /><srtp use="0" /></conf>
a=sendrecv
m=audio 45002/2 RTP/AVP 8 101
c=IN IP4 10.254.176.120
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

**(F5)**

```
ACK sip:8002@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3fede0-ceb0d5a5-13c4-50022-
1ee72-722b1847-1ee72
To: <sip:*42@scm.com>;tag=fe771969-d00e-4074-bbf8-b7359a7bd000
Call-ID: 403698-ceb0d5a5-13c4-50022-1ee72-2488c219-1ee72
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1ee73-78b73ae-
6bc7a355
Max-Forwards: 70
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Contact: <sip:9004@165.213.176.206>
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/V1.02
Content-Length: 0
```

**(F6)**

```
REFER sip:9005@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3fe918-ceb0d5a5-13c4-50022-
1ee6c-3d5b8221-1ee6c
To: <sip:9005@scm.com>;tag=acc5b9a0-9806-461a-843a-4eb51cfc3c04
Call-ID: 4034b8-ceb0d5a5-13c4-50022-1ee6c-38066253-1ee6c
CSeq: 3 REFER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1ee73-78b73fe-
109cc2d9
Refer-To: <sip:8002@scm.com>
Referred-By: <sip:9004@scm.com>
Max-Forwards: 70
Supported: timer,replaces
Contact: <sip:9004@165.213.176.206>
Content-Length: 0
```

**(F7)**

```
SIP/2.0 202 Accepted
From: "SIP9004"<sip:9004@scm.com>;tag=3fe918-ceb0d5a5-13c4-50022-
1ee6c-3d5b8221-1ee6c
To: <sip:9005@scm.com>;tag=acc5b9a0-9806-461a-843a-4eb51cfc3c04
Call-ID: 4034b8-ceb0d5a5-13c4-50022-1ee6c-38066253-1ee6c
CSeq: 3 REFER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1ee73-78b73fe-
109cc2d9
Contact: <sip:9005@10.254.176.120:5060>
Content-Length: 0
```

**(F8)**

```
REFER sip:9003@10.254.176.120:5060 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=3fe120-ceb0d5a5-13c4-50022-
1ee62-3cc93fdb-1ee62
To: <sip:9003@scm.com>;tag=9624d9dc-75ec-4d77-8b6e-627ae93c08ea
Call-ID: 4032d8-ceb0d5a5-13c4-50022-1ee62-1f09d86d-1ee62
CSeq: 3 REFER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1ee73-78b749e-
3c17096c
Refer-To: <sip:8002@scm.com>
Referred-By: <sip:9004@scm.com>
Max-Forwards: 70
Supported: timer,replaces
Contact: <sip:9004@165.213.176.206>
Content-Length: 0
```

**(F9)**

```
SIP/2.0 202 Accepted
From: "SIP9004"<sip:9004@scm.com>;tag=3fe120-ceb0d5a5-13c4-50022-
1ee62-3cc93fdb-1ee62
To: <sip:9003@scm.com>;tag=9624d9dc-75ec-4d77-8b6e-627ae93c08ea
Call-ID: 4032d8-ceb0d5a5-13c4-50022-1ee62-1f09d86d-1ee62
CSeq: 3 REFER
Via: SIP/2.0/UDP 165.213.176.206:5060;branch=z9hG4bK-1ee73-78b749e-
3c17096c
Contact: <sip:9003@10.254.176.120:5060>
Content-Length: 0
```

**(F10)**

```
BYE sip:9004@165.213.176.206 SIP/2.0
From: <sip:9005@scm.com>;tag=acc5b9a0-9806-461a-843a-4eb51cfc3c04
To: "SIP9004"<sip:9004@scm.com>;tag=3fe918-ceb0d5a5-13c4-50022-1ee6c-
3d5b8221-1ee6c
Call-ID: 4034b8-ceb0d5a5-13c4-50022-1ee6c-38066253-1ee6c
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-34696-
ccbba61-4fc13240
Max-Forwards: 70
Content-Length: 0
```

**(F11)**

```
SIP/2.0 200 OK
From: <sip:9005@scm.com>;tag=acc5b9a0-9806-461a-843a-4eb51cfc3c04
To: "SIP9004"<sip:9004@scm.com>;tag=3fe918-ceb0d5a5-13c4-50022-1ee6c-
3d5b8221-1ee6c
Call-ID: 4034b8-ceb0d5a5-13c4-50022-1ee6c-38066253-1ee6c
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-34696-
ccbba61-4fc13240
Supported: timer,replaces
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

**(F12)**

```
BYE sip:9004@165.213.176.206 SIP/2.0
From: <sip:9003@scm.com>;tag=9624d9dc-75ec-4d77-8b6e-627ae93c08ea
To: "SIP9004"<sip:9004@scm.com>;tag=3fe120-ceb0d5a5-13c4-50022-1ee62-
3cc93fdb-1ee62
Call-ID: 4032d8-ceb0d5a5-13c4-50022-1ee62-1f09d86d-1ee62
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-34696-
ccbbb5f-6db77d2b
Max-Forwards: 70
Content-Length: 0
```

**(F13)**

```
SIP/2.0 200 OK
From: <sip:9003@scm.com>;tag=9624d9dc-75ec-4d77-8b6e-627ae93c08ea
To: "SIP9004"<sip:9004@scm.com>;tag=3fe120-ceb0d5a5-13c4-50022-1ee62-
3cc93fdb-1ee62
Call-ID: 4032d8-ceb0d5a5-13c4-50022-1ee62-1f09d86d-1ee62
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-34696-
ccbbb5f-6db77d2b
Supported: timer,replaces
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

**(F14)**

```
INVITE sip:9005@165.213.176.39 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=8890416c-aacd-4a0e-85ce-
cc5e0dd6f4b8
To: "SIP9005"<sip:9005@scm.com>;tag=3fe2b8-27b0d5a5-13c4-50022-358-
213c2c66-358
Call-ID: b12af8ce-cb3b-4dea-b6dd-e1c63d8fdb5a@scm.com
CSeq: 3 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-34696-
ccbbb87-1b7f7279
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
P-Asserted-Identity: "8002" <sip:8002@scm.com>
Max-Forwards: 70
Contact: <sip:8002@10.254.176.120:5060>;isfocus
Content-Length: 0
```

**(F15)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=8890416c-aacd-4a0e-85ce-
cc5e0dd6f4b8
To: "SIP9005"<sip:9005@scm.com>;tag=3fe2b8-27b0d5a5-13c4-50022-358-
213c2c66-358
Call-ID: b12af8ce-cb3b-4dea-b6dd-e1c63d8fdb5a@scm.com
CSeq: 3 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-34696-
ccbbb87-1b7f7279
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@165.213.176.39>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Type: application/sdp
Content-Length: 365

v=0
o=SAMSUNG_IP_PHONE 1276045336 2 IN IP4 165.213.176.39
s=SAMSUNG_VOIP_CALL
c=IN IP4 165.213.176.39
t=0 0
m=audio 20010 RTP/AVP 8 0 18 9 102 101
b=TIAS:64000
b=AS:82
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F16)**

```
ACK sip:9005@165.213.176.39 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=8890416c-aacd-4a0e-85ce-
cc5e0dd6f4b8
To: "SIP9005"<sip:9005@scm.com>;tag=3fe2b8-27b0d5a5-13c4-50022-358-
213c2c66-358
Call-ID: b12af8ce-cb3b-4dea-b6dd-e1c63d8fdb5a@scm.com
CSeq: 3 ACK
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-34696-
ccbbc43-64ca0888
Max-Forwards: 70
Contact: <sip:8002@10.254.176.120:5060>;isfocus
Content-Type: application/sdp
Content-Length: 395

v=0
o=8002 0 0 IN IP4 10.254.176.120
s=Samsung IP PBX
u=sip:8002@scm.com
c=IN IP4 10.254.176.120
t=0 0
a=x-master:9004@scm.com
a=x-confInfo:<conf type="35" conf_id="8002" ug="1" rec="0"><aud_opt
mixmode="1" /><srtp use="0" /></conf>
a=sendrecv
m=audio 45002/2 RTP/AVP 8 101
c=IN IP4 10.254.176.120
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

**(F17)**

```
INVITE sip:9003@165.213.176.174 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=29ecb85e-ab86-4b81-85ff-
821c49482312
To: "SIP9003"<sip:9003@scm.com>;tag=80ed33b8-a5d5b0ae-13c4-50022-
62b61-18eac78b-62b61
Call-ID: 410a960d-efd1-48a5-8ed7-744c92b96ee0@scm.com
CSeq: 3 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-34696-
ccbbc68-4f6a472a
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
P-Asserted-Identity: "8002" <sip:8002@scm.com>
Max-Forwards: 70
Contact: <sip:8002@10.254.176.120:5060>;isfocus
Content-Length: 0
```

**(F18)**

```
SIP/2.0 200 OK
From: "SIP9004"<sip:9004@scm.com>;tag=29ecb85e-ab86-4b81-85ff-
821c49482312
To: "SIP9003"<sip:9003@scm.com>;tag=80ed33b8-a5d5b0ae-13c4-50022-
62b61-18eac78b-62b61
Call-ID: 410a960d-efd1-48a5-8ed7-744c92b96ee0@scm.com
CSeq: 3 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-34696-
ccbbc68-4f6a472a
Supported: timer,replaces
P-Preferred-Identity: "SIP9003"<sip:9003@scm.com>
Contact: <sip:9003@165.213.176.174>
Session-Expires: 1800;refresher=uas
Allow: INVITE,ACK,CANCEL,BYE,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Allow-Events: talk,hold
Content-Type: application/sdp
Content-Length: 315

v=0
o=SAMSUNG_IP_PHONE 1276045350 2 IN IP4 165.213.176.174
s=SAMSUNG_VOIP_CALL
c=IN IP4 165.213.176.174
t=0 0
m=audio 20010 RTP/AVP 8 0 18 101
b=TIAS:64000
b=AS:82
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F19)**

```
ACK sip:9003@165.213.176.174 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=29ecb85e-ab86-4b81-85ff-
821c49482312
To: "SIP9003"<sip:9003@scm.com>;tag=80ed33b8-a5d5b0ae-13c4-50022-
62b61-18eac78b-62b61
Call-ID: 410a960d-efd1-48a5-8ed7-744c92b96ee0@scm.com
CSeq: 3 ACK
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-34696-
ccbbcd7-18c95cd1
Max-Forwards: 70
Contact: <sip:8002@10.254.176.120:5060>;isfocus
Content-Type: application/sdp
Content-Length: 395
```

```
v=0
o=8002 0 0 IN IP4 10.254.176.120
s=Samsung IP PBX
u=sip:8002@scm.com
c=IN IP4 10.254.176.120
t=0 0
a=x-master:9004@scm.com
a=x-confInfo:<conf type="35" conf_id="8002" ug="1" rec="0"><aud_opt
mixmode="1" /><srtp use="0" /></conf>
a=sendrecv
m=audio 45002/2 RTP/AVP 8 101
c=IN IP4 10.254.176.120
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

## 2.3.18 Calling Line Identification Blocking by SCM

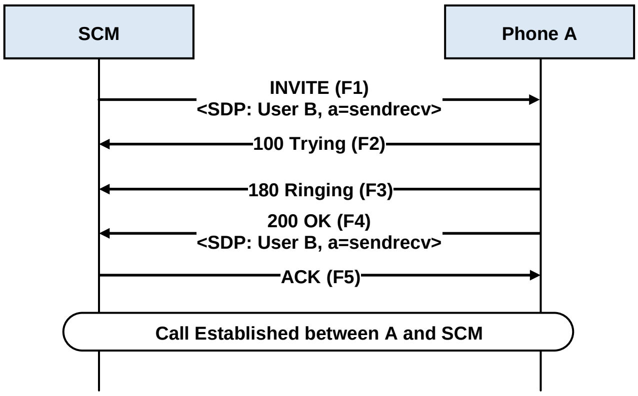



**Figure 2.19 Calling Line Identification Blocking by SCM**

**(F1)**

```
INVITE sip:9005@165.213.176.39 SIP/2.0
From: "Anonymous"<sip:anonymous@anonymous.invalid>;tag=4050e1bf-e492-
4b28-b917-31a41cbe5596
To: "SIP9005"<sip:9005@scm.com>
Call-ID: 418fa8c2-a1a4-4748-be54-07517bed090d@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-3abb2-
e56af7d-7ab2f10d
P-Asserted-Identity: "SIP9004" <sip:9004@scm.com>
Privacy: id
```

```
Max-Forwards: 70
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
Contact: <sip:9004@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 329

v=0
o=9004 1276071237 0 IN IP4 165.213.176.206
s=Samsung IP PBX
c=IN IP4 165.213.176.206
t=0 0
m=audio 20044 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

**(F2)**

```
SIP/2.0 100 Trying
From: "Anonymous"<sip:anonymous@anonymous.invalid>;tag=4050e1bf-e492-
4b28-b917-31a41cbe5596
To: "SIP9005"<sip:9005@scm.com>
Call-ID: 418fa8c2-a1a4-4748-be54-07517bed090d@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-3abb2-
e56af7d-7ab2f10d
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@165.213.176.39>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

**(F3)**

```
SIP/2.0 180 Ringing
From: "Anonymous"<sip:anonymous@anonymous.invalid>;tag=4050e1bf-e492-
4b28-b917-31a41cbe5596
To: "SIP9005"<sip:9005@scm.com>;tag=400430-27b0d5a5-13c4-50022-687d-
6cad8fd5-687d
Call-ID: 418fa8c2-a1a4-4748-be54-07517bed090d@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-3abb2-
e56af7d-7ab2f10d
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@165.213.176.39>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Length: 0
```

**(F4)**

```
SIP/2.0 200 OK
From: "Anonymous"<sip:anonymous@anonymous.invalid>;tag=4050e1bf-e492-
4b28-b917-31a41cbe5596
To: "SIP9005"<sip:9005@scm.com>;tag=400430-27b0d5a5-13c4-50022-687d-
6cad8fd5-687d
Call-ID: 418fa8c2-a1a4-4748-be54-07517bed090d@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-3abb2-
e56af7d-7ab2f10d
Supported: timer,replaces
P-Preferred-Identity: "SIP9005"<sip:9005@scm.com>
Contact: <sip:9005@165.213.176.39>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISHtalk,h
old
Allow-Events: talk,hold
Content-Type: application/sdp
Content-Length: 263

v=0
o=SAMSUNG_IP_PHONE 1276071229 1 IN IP4 165.213.176.39
s=SAMSUNG_VOIP_CALL
c=IN IP4 165.213.176.39
t=0 0
m=audio 20014 RTP/AVP 0 101
b=TIAS:64000
b=AS:82
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

**(F5)**

```
ACK sip:9005@165.213.176.39 SIP/2.0
From: "Anonymous"<sip:anonymous@anonymous.invalid>;tag=4050e1bf-e492-
4b28-b917-31a41cbe5596
To: "SIP9005"<sip:9005@scm.com>;tag=400430-27b0d5a5-13c4-50022-687d-
6cad8fd5-687d
Call-ID: 418fa8c2-a1a4-4748-be54-07517bed090d@scm.com
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-3abb3-
e56b6e7-1d484740
Max-Forwards: 70
Contact: <sip:9004@10.254.176.120:5060>
Content-Length: 0
```

## 2.3.19 Message Waiting Indication (Unsolicited)

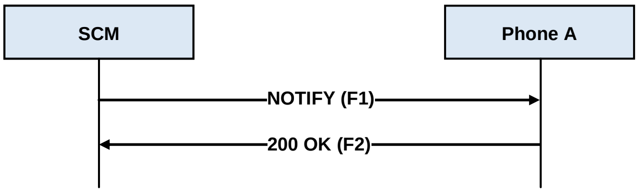



**Figure 2.20 Message Waiting Notification (Unsolicited)**

**(F1)**

```
NOTIFY sip:9004@165.213.176.206 SIP/2.0
From: <sip:7000@scm.com>;tag=b55cc98-0-13d8-50022-15faf-2f5e94d0-15faf
To: <sip:9004@scm.com>
Call-ID: b55cc98-0-13d8-50022-15faf-5e0e96c1-15faf
CSeq: 1513186487 NOTIFY
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-15faf-
55dc6ba-357827b1
Event: message-summary
Contact: <sip:7000@10.254.176.120:5060>
Max-Forwards: 70
Content-Type: application/simple-message-summary
Content-Length: 87

Messages-Waiting:yes
Message-Account:sip:9004@10.254.176.120
Voice-Message:1/0(0/0)
```

**(F2)**

```
SIP/2.0 200 OK
From: <sip:7000@scm.com>;tag=b55cc98-0-13d8-50022-15faf-2f5e94d0-15faf
To: <sip:9004@scm.com>;tag=3fe2b8-ceb0d5a5-13c4-50022-785-7c2c6ae1-785
Call-ID: b55cc98-0-13d8-50022-15faf-5e0e96c1-15faf
CSeq: 1513186487 NOTIFY
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-15faf-
55dc6ba-357827b1
Supported: timer,replaces
Content-Length: 0
```

## 2.3.20 Auto Answer

3rd-Party-SIP-Phone must answer with 200 OK when receive INVITE including following Call-Info Header.
Call-Info: <http://127.0.0.1>; answer-after=1
Answer-after value means repeating numbers for phone to ring bell.

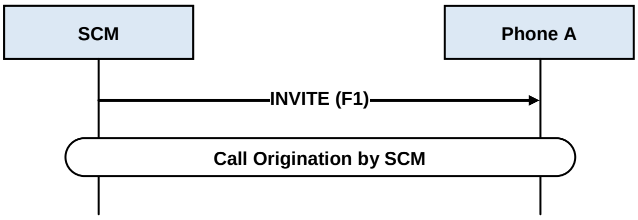



**Figure 2.21 Auto Answer**

**(F1)**

```
INVITE sip:9005@10.254.168.97 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=ec6e9659-88f7-44b5-97ba-
e2fce3ce2d06
To: "SIP9005"<sip:9005@scm.com>
Call-ID: 2a7de19b-8919-4f04-82e8-4f046710c4f1@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-143da-
4f10e63-6c643eaa
P-Asserted-Identity: "SIP9004" <sip:9004@scm.com>
Call-Info: <http://127.0.0.1>; answer-after=1
Max-Forwards: 70
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
Contact: <sip:9004@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 329
```

```
v=0
o=9004 1275913578 0 IN IP4 165.213.176.206
s=Samsung IP PBX
c=IN IP4 165.213.176.206
t=0 0
m=audio 20016 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

## 2.3.21 Distinctive Ring

3rd-Party-SIP-Phone must ring a bell when receive INVITE including following Alert-Info Header.
Alert-Info: <http://127.0.0.1/Bellcore-dr1>
Bellcore-dr1 specify the bell type to ring.

If 3rd-Party-SIP-Phone already have a embeded distinctive ring functionality, the phone may ignore Alert-Info Header.

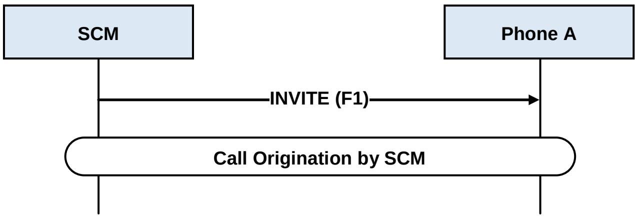



**Figure 2.22 Distinctive Ring**

**(F1)**

```
INVITE sip:9005@10.254.168.97 SIP/2.0
From: "SIP9004"<sip:9004@scm.com>;tag=ec6e9659-88f7-44b5-97ba-
e2fce3ce2d06
To: "SIP9005"<sip:9005@scm.com>
Call-ID: 2a7de19b-8919-4f04-82e8-4f046710c4f1@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-143da-
4f10e63-6c643eaa
P-Asserted-Identity: "SIP9004" <sip:9004@scm.com>
```

```
Alert-Info: <http://127.0.0.1/Bellcore-dr1>
Max-Forwards: 70
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE
Contact: <sip:9004@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 329

v=0
o=9004 1275913578 0 IN IP4 165.213.176.206
s=Samsung IP PBX
c=IN IP4 165.213.176.206
t=0 0
m=audio 20016 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

## 2.3.22 Display Control using INFO

SCM Express use application/vnd.sec-display at Content-Type of INFO.
Also, Body is including follow text.

```
Text="display this"
Charset=euc-kr
```

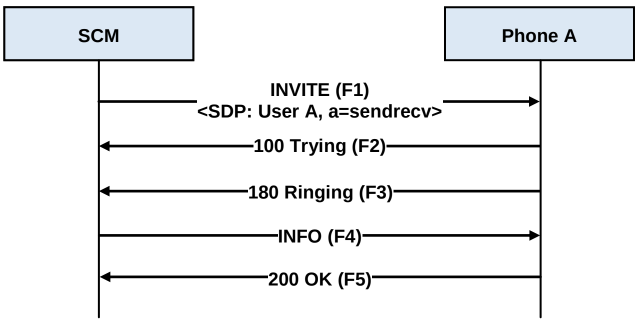



**Figure 2.23 Display Control using INFO**

**(F1)**

```
INVITE sip:9005@165.213.176.39 SIP/2.0
From: "SIP5004"<sip:9004@scm.com>;tag=32129aae-0ffa-4393-9b9f-
2543bd75a0da
To: "SIP5005"<sip:9005@scm.com>
Call-ID: b6851e66-0c4b-4dfc-a58c-9b8d76865706@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-46bbb-
1144d4dc-4c280462
P-Asserted-Identity: "SIP5004" <sip:9004@scm.com>
Max-Forwards: 70
Allow:
INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Contact: <sip:9004@10.254.176.120:5060>
Content-Type: application/sdp
Content-Length: 325

v=0
o=9004 121188 0 IN IP4 165.213.176.206
s=Samsung IP PBX
c=IN IP4 165.213.176.206
t=0 0
m=audio 20030 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

**(F2)**

```
SIP/2.0 100 Trying
From: "SIP5004"<sip:9004@scm.com>;tag=32129aae-0ffa-4393-9b9f-
2543bd75a0da
To: "SIP5005"<sip:9005@scm.com>
Call-ID: b6851e66-0c4b-4dfc-a58c-9b8d76865706@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-46bbb-
1144d4dc-4c280462
Supported: timer,replaces
P-Preferred-Identity: "SIP5005"<sip:9005@scm.com>
Contact: <sip:9005@165.213.176.39>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISH
Allow-Events: talk,hold
Content-Length: 0
```

**(F3)**

```
SIP/2.0 180 Ringing
From: "SIP5004"<sip:9004@scm.com>;tag=32129aae-0ffa-4393-9b9f-
2543bd75a0da
To: "SIP5005"<sip:9005@scm.com>;tag=43bae0-27b0d5a5-13c4-50022-54-
5d3ab1bf-54
Call-ID: b6851e66-0c4b-4dfc-a58c-9b8d76865706@scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-46bbb-
1144d4dc-4c280462
Supported: timer,replaces
P-Preferred-Identity: "SIP5005"<sip:9005@scm.com>
Contact: <sip:9005@165.213.176.39>
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISH
Allow-Events: talk,hold
Content-Length: 0
```

**(F4)**

```
INFO sip:9005@165.213.176.39 SIP/2.0
From: "SIP5004"<sip:9004@scm.com>;tag=32129aae-0ffa-4393-9b9f-
2543bd75a0da
To: "SIP5005"<sip:9005@scm.com>;tag=43bae0-27b0d5a5-13c4-50022-54-
5d3ab1bf-54
Call-ID: b6851e66-0c4b-4dfc-a58c-9b8d76865706@scm.com
CSeq: 2 INFO
Via: SIP/2.0/UDP 10.254.176.120:5060;rport;branch=z9hG4bK-46bbe-
1144de8e-230544f1
Max-Forwards: 70
Contact: <sip:9004@10.254.176.120:5060>
Content-Type: application/vnd.sec-display
Content-Length: 44

Text="9003, Trasfered from 9004"
Charset=euc-kr
```

**(F5)**

```
SIP/2.0 200 OK
From: "SIP5004"<sip:9004@scm.com>;tag=32129aae-0ffa-4393-9b9f-
2543bd75a0da
To: "SIP5005"<sip:9005@scm.com>;tag=43bae0-27b0d5a5-13c4-50022-54-
5d3ab1bf-54
Call-ID: b6851e66-0c4b-4dfc-a58c-9b8d76865706@scm.com
CSeq: 2 INFO
Via: SIP/2.0/UDP 10.254.176.120:5060;rport=5060;branch=z9hG4bK-46bbe-
1144de8e-230544f1
Supported: timer,replaces
Allow:
INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISH
Contact: <sip:9005@165.213.176.39>
Allow-Events: talk,hold
Content-Length: 0
```

## 2.4 Session Timer

3rd-Party-SIP-Phone must support session timer function defined at RFC4028.
Session timer function may be use UPDATE, Re-INVITE. In case of Re-INVITE, the last session-id, session-version must be applied.

# 2.5 DTMF

3rd-Party-SIP-Phone must support three different types of DTMF.

## 2.5.1 RFC2833

DTMF is negotiated based on following chart.

| Offer | Answer | Negotiated |
|---|---|---|
| RFC2833 | RFC2833 | RFC2833 |
| RFC2833 | In-band (In-voice) | In-band (In-voice) |
| In-band (In-voice) | In-band (In-voice) | In-band (In-voice) |

## 2.5.2 In-band (In-voice)

3rd-Party-SIP-Phone must answer In-band DTMF when In-band DTMF offered by peer

## 2.5.3 INFO

Content-Type with ‘application/dtmf-relay’ is INFO type of DTMF

# 2.6 Early media

## 2.6.1 18x without SDP

3rd-Party-SIP-Phone must play local ringback.

## 2.6.2 18x with SDP

3rd-Party-SIP-Phone must play media info in the 183 SDP.

## 2.6.3 18x with SDP after 18x without SDP

3rd-Party-SIP-Phone must play media info in the 183 SDP after local ringback.

## 2.6.4 18x without SDP after 18x with SDP

3rd-Party-SIP-Phone must play local ringback after playing media in the 183 SDP.

# CHAPTER 3. SIP Application Interoperability

## 3.1 Introduction

This chapter describes SIP Application Configuration and SIP usages for SIP Application interoperability.

# 3.2 Configuring Application

## 3.2.1 Application License

SCM License scheme defines SIP application channel capabilities. Operators can insert the external license key issued by a license server using [SCM Administrator/CONFIGURATION/ Miscellaneous/License] menu.

### 3.2.1.1 Voice Mail/Recording/Auto Attendant/Conference/External Ringback Tone License

In order to integrate with application servers such as Voice Mail, Recording, Auto Attendant, Conference and External Ringback Tone server, the [SIP Application Channels] of External License should be assigned. The [SIP Application Channels] defines SIP application channel capabilities. Each application needs to set SIP channels in each application menu. The number of each application license cannot exceed the total number of SIP application channel capacity. Total SIP application channels are be shown as below.

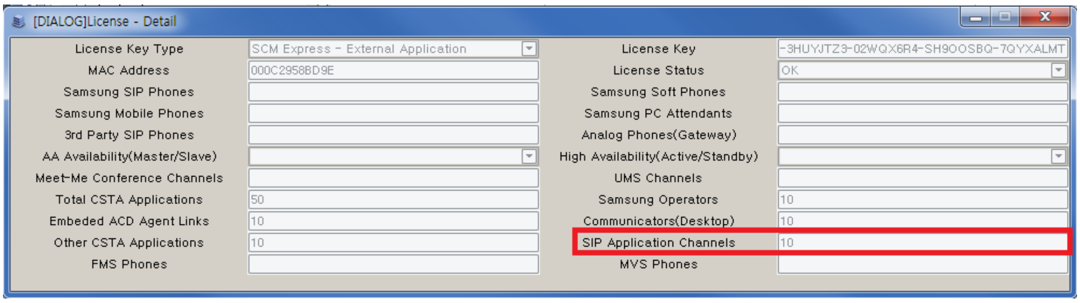



**Figure 3.1 SIP Application Licence**

## 3.2.2 Application Server Service Group

### 3.2.1.1 Configuring Application Server Service Group

First of all, operates should make an Application Server Service Group using [SCM Administrator/CONFIGURATION/Application/Application Server Service Group] menu.

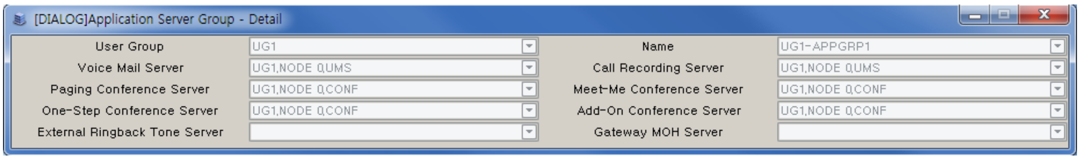



**Figure 3.2 Application Server Service Group**

### 3.2.1.2 Assigning Application Server Service Group

In order to provide a specific service using an application server, operators can assign an Application Server Service Group in user or service group or user group menu.
The application server specified in user menu has a higher priority than the application server of service group or user group. And the application server of service group has a higher priority than that of user group.
For example, if a user should be connected to application server for a specific service, SCM first tries to fine an application server in user menu. If there is no available application server, SCM checks an Application Server Service Group of service group. After that, if SCM fail to find an application server in service group, the Application Server Service Group of user group is checked. If it is also failed, the call will fail.

## In User Menu

In [SCM Administrator/CONFIGURATION/User/Single User] menu, operator can assign Application Server Service Group.

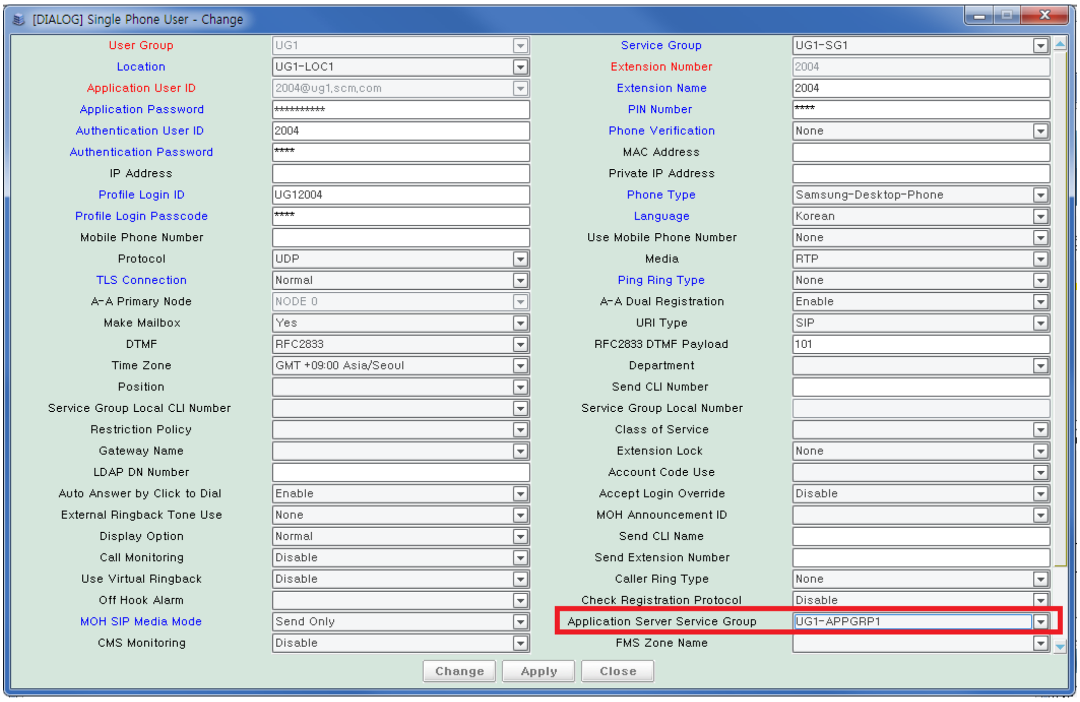


**Figure 3.3 Assigning Application Server Service Group-User**

## In Service Group Menu

In [SCM Administrator/CONFIGURATION/User Group/Service Group] menu, operator can assign Application Server Service Group.

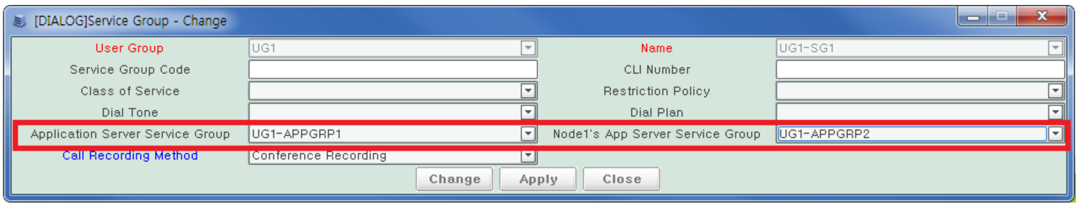


**Figure 3.4 Assigning Application Server Service Group-Service Group**

### In User Group Menu

In [SCM Administrator/CONFIGURATION/User Group/Change User Group/Options] menu, operator can assign Application Server Service Group.
Refer to above Service Group to know about differences between Application Server Service Group and Node1's App Server Service Group.

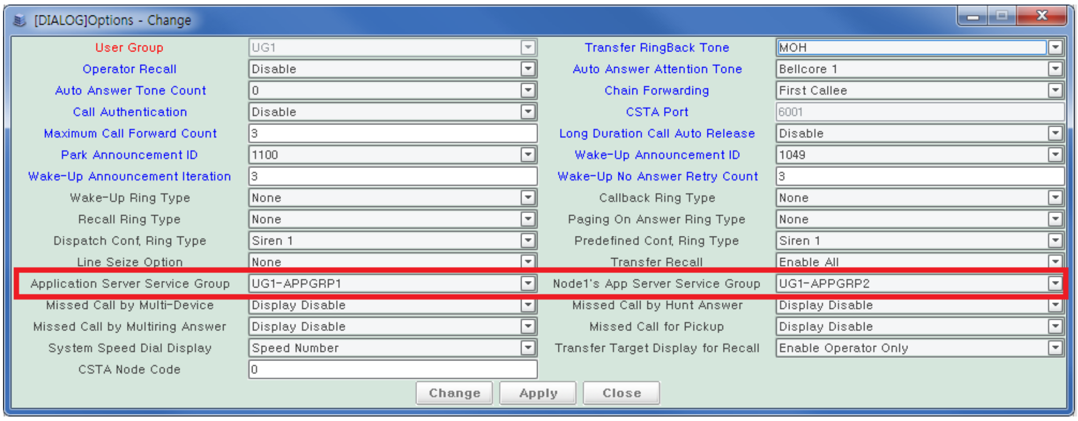



**Figure 3.5 Application Server Service Group-User Group**

## 3.2.3 Active-Active Node

If SCM version is 3.3 or higher and Active-Active mode is set, the below menu is available.

In [SCM Administrator/Configuration/Application] menu, [VM/AA Server], [Conference Server], [Other Application Server] needs select [A-A Primary Node].

If application server support dual-registration, select [A-A Dual Registration] option enable.
In this case, [A-A Primary Node] is required.
Specified Primary node provides application service.
When Primary node can't provide service, Secondary Node provides application service.

Otherwise, If application server can't support dual-registration, select [A-A Dual Registration] option disable. It also [A-A Primary Node] is required.
Difference point is that Secondary Node does not provide application service call.
When Primary node does not provide application service, call is failure.

When [A-A Dual Operation] option is enabled, Application Service does not check node.
[A-A Dual Operation] option should specify [A-A Dual Registration] enabled.

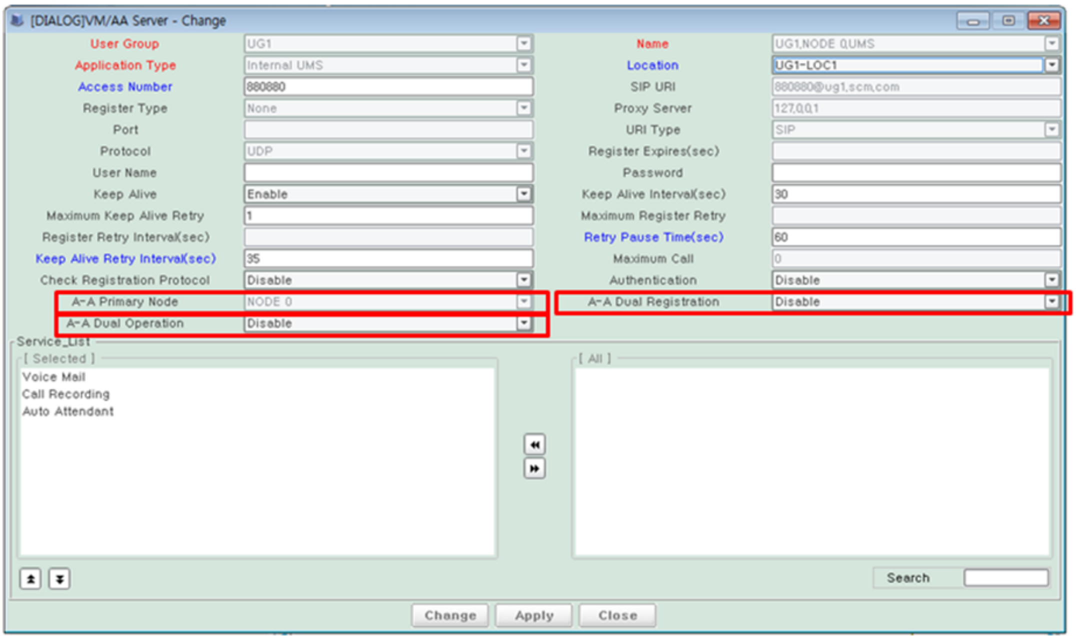



**Figure 3.6 Application Server A-A Configuration**

### 3.2.4 Voice Mail Server

For Subscriber or group, SCM can provide voice mail service using 3rd party application.

#### 3.2.4.1 Configuring Voice Mail Server

In [SCM Administrator/Configuration/Application] Menu, operator can create voice mail server.
When operator create voice mail server, Following menu items are necessary.

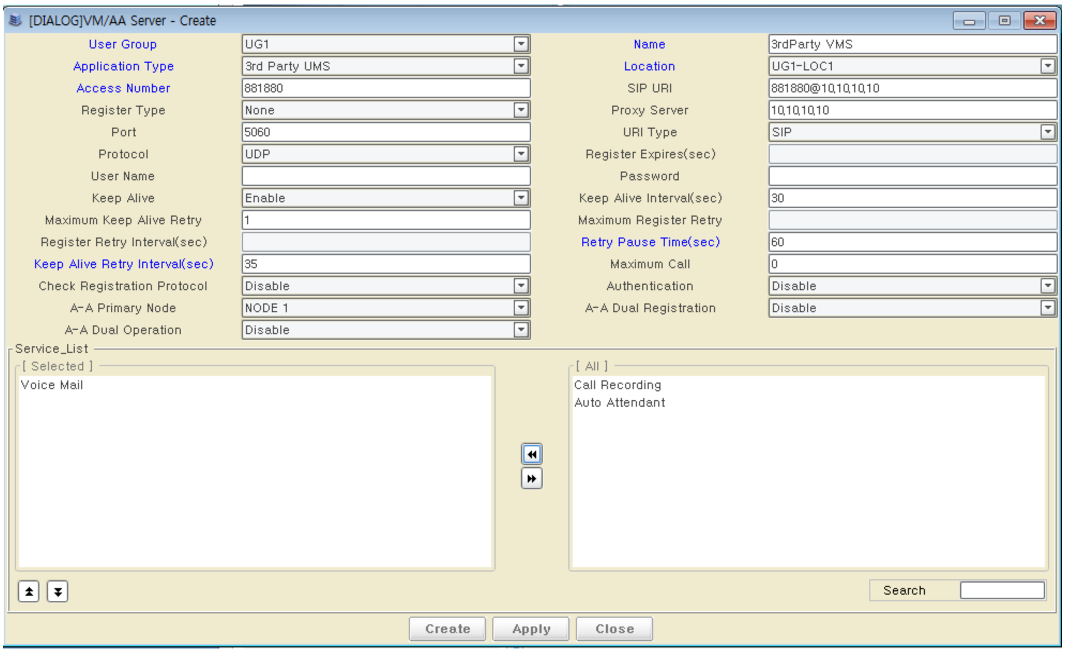



**Figure 3.7 Voice Mail Server-Create**

**Table 3.1 Voice Mail Server Create**

| Options | Explains |
|---|---|
| User Group | Operator selects User Group that using this application server. |
| Name | Specify Application Server Name |
| Application Type | Select 3rd Party UMS Type |
| Location | Select Location in SCM Express |
| Access Number | Specify Application Server Number.<br>Access number not duplicate number with extension number, conference channel number and system speed dial number. |
| SIP URI | Specify SIP URI, like E-mail Address. Ex) Access Number@IP Address |
| Register Type | Default Selection Option is None.<br>If [Keep Alive] Option is Enable, SCM Send Option Message recursively [Keep Alive Interval] time.<br>Otherwise [Keep Alive] Option is Disable, SCM does not check Application Server's Status.<br>If Application Server can receive register message or send register message, scm select reverse REGISTER. |
| Proxy Server | Specify application server's ip address. |
| Port | Default port number is 5060.<br>If application server use other port, input other port number. |
| Protocol | Default Value is UDP Protocol.<br>If application server allow other protocol, select TCP or TLS protocol |
| Keep Alive | When [Register Type] option is none type, this option also need enabled.<br>When [Keep Alive] option is enabled, SCM send optional message for connection checking. |
| Keep Alive Interva (Sec) | Default Interval value is 30 second.<br>[Keep Alive] option is enabled, SCM uses this interval value. |
| Maximum Call | Specify max channel counts.<br>If Maximum call value is not defined, 3rd party Application call is rejected.<br>Maximum call value is necessary to [SIP Application Channels] value within License key. |
| Service List | In Service List, move Voice Mail option in [selected].<br>If SCM version is before 3.2.4, Application Server is not duplicated service list. |

### 3.2.4.2 Voice Mail Operation

An operator should specify feature codes for Voice Mail Service.

In [SCM Administrator/Configuration/Service/Feature Service] Menu, operator can change [VM Access] feature code. default feature code is *88.
Subscriber can use voice mail service by dialed *88.

[VM Transfer] Service is provided following subscriber’s action.
When subscriber answered a call, phone is displaying [VM Transfer] menu. When phone is ringing, [Deflect to VM] Menu is displayed.
To use [VM Transfer] and [Deflect to VM] is required by specified [VM Transfer] feature code. Default feature code is *87.

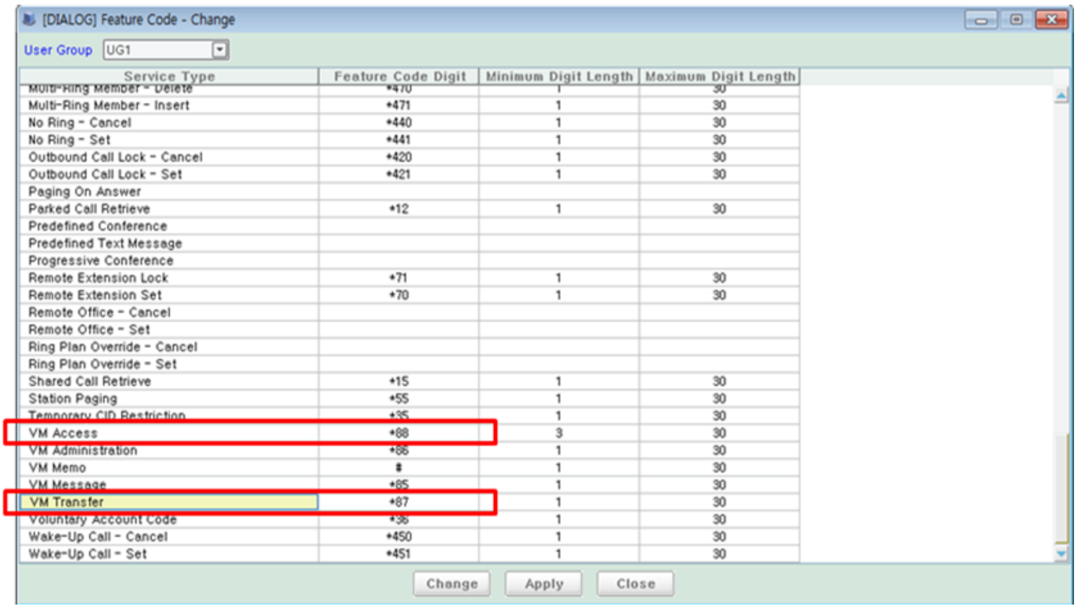



**Figure 3.8 Voice Mail Server Feature Code Configuration**

Subscriber can use voice mail server by call forward destination.
It is provided by specify the call forward service in [SCM Administrator/Configuration/ Service/Feature Service/Service Activation] menu.
If [CallForwardBusy] destination is set to *88, subscriber can use voice mail server to other call when answer.

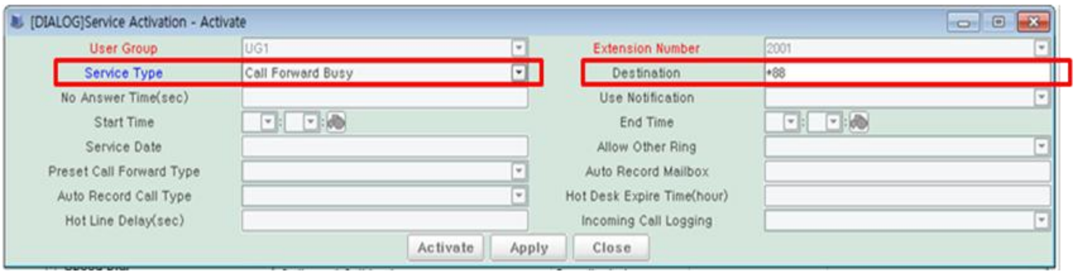



**Figure 3.9 Voice Mail Server Call Forward Setting**

## 3.2.5 Recording Server

SCM provides a recording service. There are two ways for recording, using conference server and phone bridge recording.
In order to provide a specific service using a recording server, operators can assign a Call Recording Method in user or service group menu. The recording method specified in user menu has a higher priority than the recording method of service group.
Case of using conference server, the conversation between users is recorded through the recording server. The recording server should receive the mixed RTP data by a conference server. For more detailed information about a conference server, refer to 3.2.10 chapter.
Case of phone bridge recording, the recording server should receive the mixed RTP data by a phone.

### 3.2.5.1 Configuring Recording Server

In [SCM Administrator/Configuration/Application/VM/AA Server] menu, create recording server.
Following items are necessary for creating recording server.

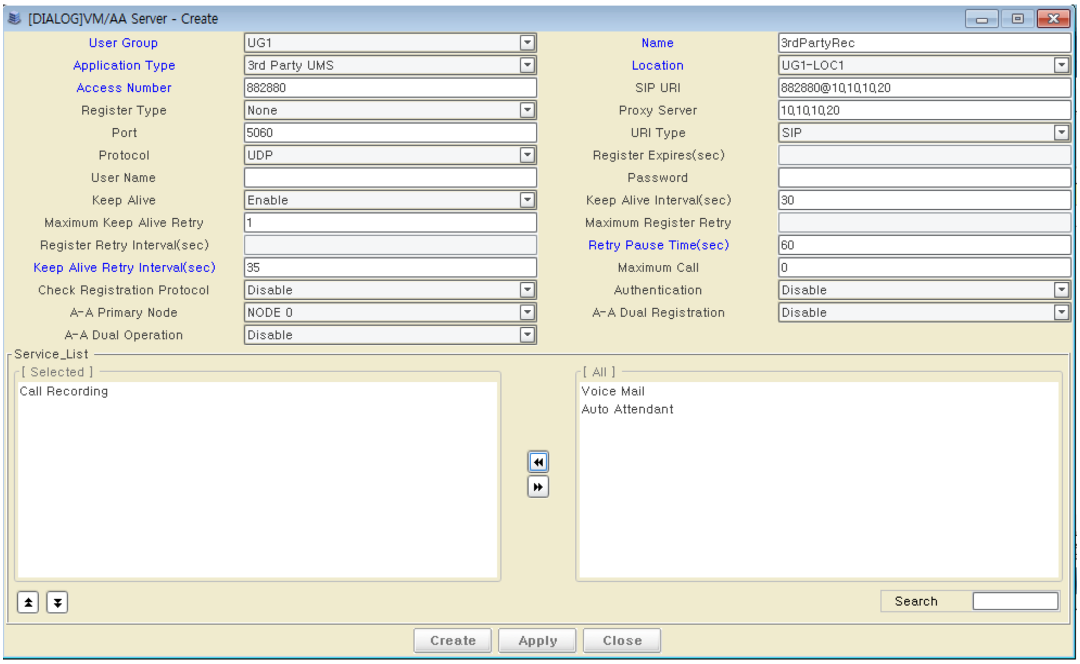



**Figure 3.10 Create Recording Server**

**Table 3.2 Create Recording Server**

| Options | Explains |
|---|---|
| User Group | Operator selects User Group that using this application server. |
| Name | Specify Application Server Name |
| Application Type | Select 3rd Party UMS Type |
| Location | Select Location in SCM Express |
| Access Number | Specify Application Server Number.<br>Access number not duplicate number with extension number, conference channel number and system speed dial number. |
| SIP URI | Specify SIP URI, like E-mail Address. Ex) Access Number@IP Address |
| Register Type | Default Selection Option is None.<br>If [Keep Alive] Option is Enable, SCM Send Option Message recursively [Keep Alive Interval] time.<br>Otherwise [Keep Alive] Option is Disable, SCM does not check Application Server's Status.<br>If Application Server can receive register message or send register message, scm select reverse REGISTER. |
| Proxy Server | Specify application server's ip address. |
| Port | Default port number is 5060.<br>If application server use other port, input other port number. |
| Protocol | Default Value is UDP Protocol.<br>If application server allow other protocol, select TCP or TLS protocol |
| Keep Alive | When [Register Type] option is none type, this option also need enabled.<br>When [Keep Alive] option is enabled, SCM send optional message for connection checking. |
| Keep Alive Interva (Sec) | Default Interval value is 30 second.<br>[Keep Alive] option is enabled, SCM uses this interval value. |
| Maximum Call | Specify max channel counts.<br>If Maximum call value is not defined, 3rd party Application call is rejected.<br>Maximum call value is necessary to [SIP Application Channels] value within License key. |
| Service List | In Service List, move [Call Recording] option in [selected].<br>If SCM version is before 3.2.4, Application Server is not duplicated service list. |

### 3.2.5.2 Conference Recording Operation

Operators should specify a feature code for call recording service.

In [SCM Administrator/Configuration/Service/Feature Service] Menu, operator can change [call recording] feature code. default feature code is *22.

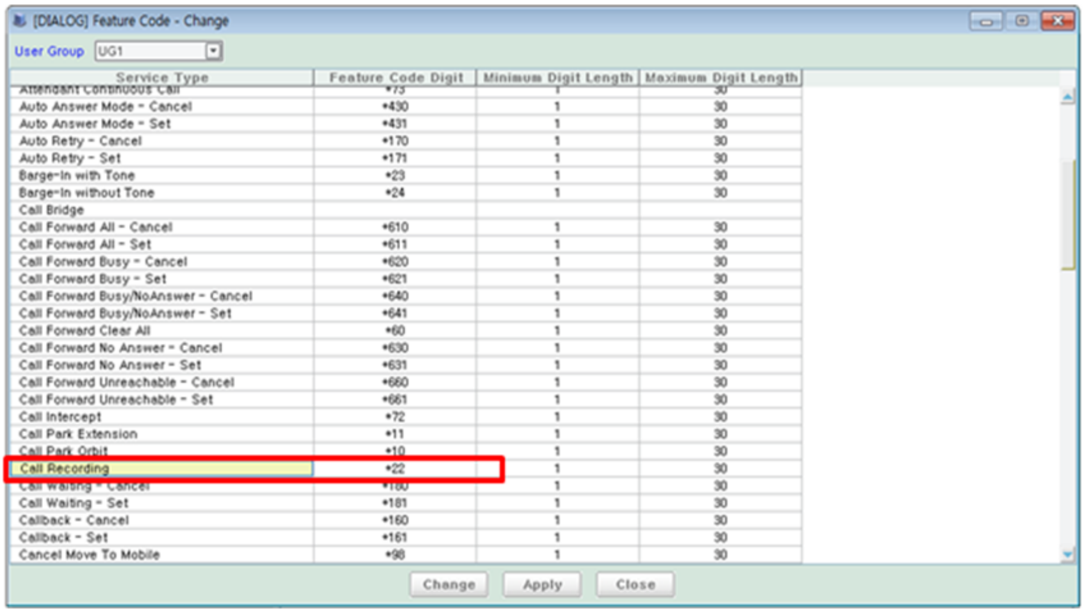



**Figure 3.11 Voice Mail Server Feature Code Configuration**

In [SCM Administrator/CONFIGURATION/User Group/Service Group] menu, operator can assign Application Server Service Group.
In same menu, operator should select 'Conference Recording' in Call Recording Method.

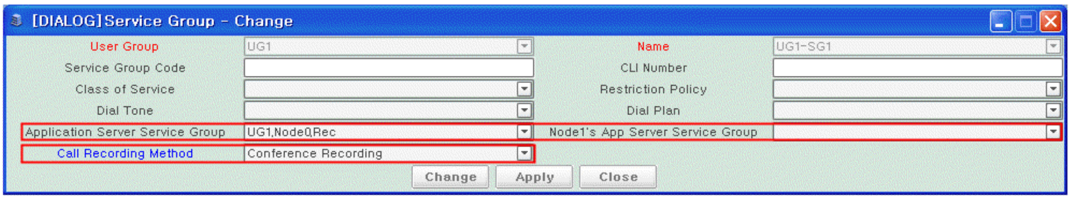



**Figure 3.12 Select Call Recording Method in Service Group Menu**

In [SCM Administrator/Configuration/User/Single Phone User] menu, operator can assign Application Server Service Group.
In same menu, operator should select 'Conference Recording' in Call Recording Method.

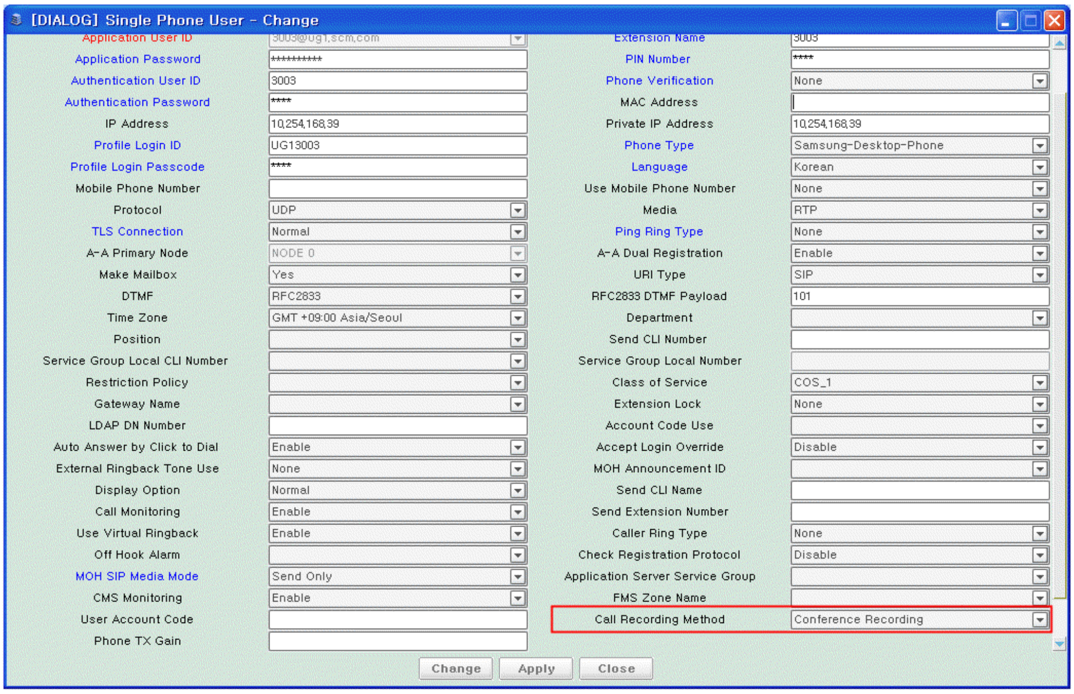



**Figure 3.13 Select Call Recording Method**

When subscriber answered a call, phone is displaying [call recording] menu.
If Subscriber select [call recording] menu, call recording service is started.
In [SCM Administrator/Configuration/Service/Feature Service/Service Activation] menu, Auto call recording service is provided.

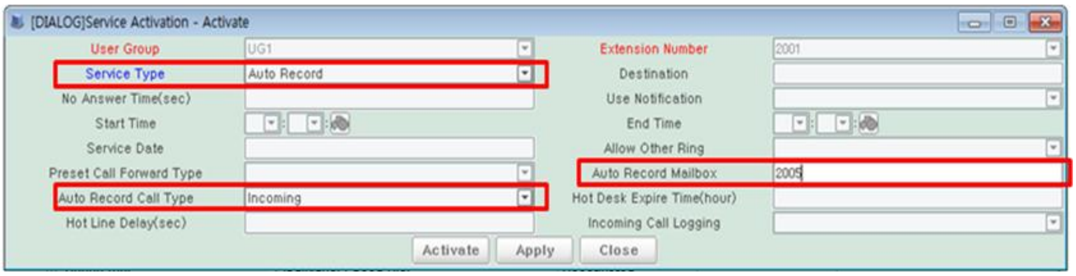



**Figure 3.14 Recording Server-Auto Record Configuration**

### 3.2.5.3. Phone Bridge Recording Operation

In [SCM Administrator/CONFIGURATION/User Group/Service Group] menu, operator can assign Application Server Service Group.
In same menu, operator should select 'Phone Recording' in Call Recording Method.

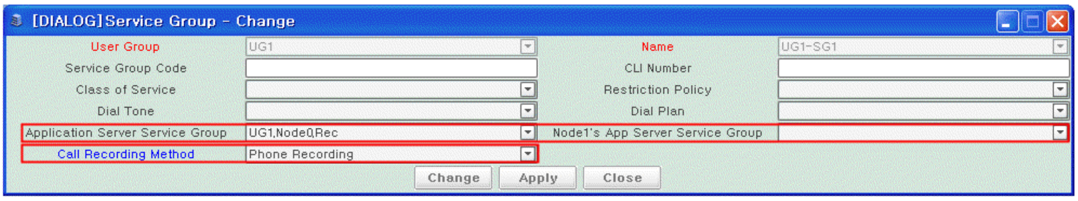



**Figure 3.15 Select Call Recording Method in Service Group Menu**

In [SCM Administrator/Configuration/User/Single Phone User] menu, operator can assign Application Server Service Group.
In same menu, operator should select 'Phone Recording' in Call Recording Method.

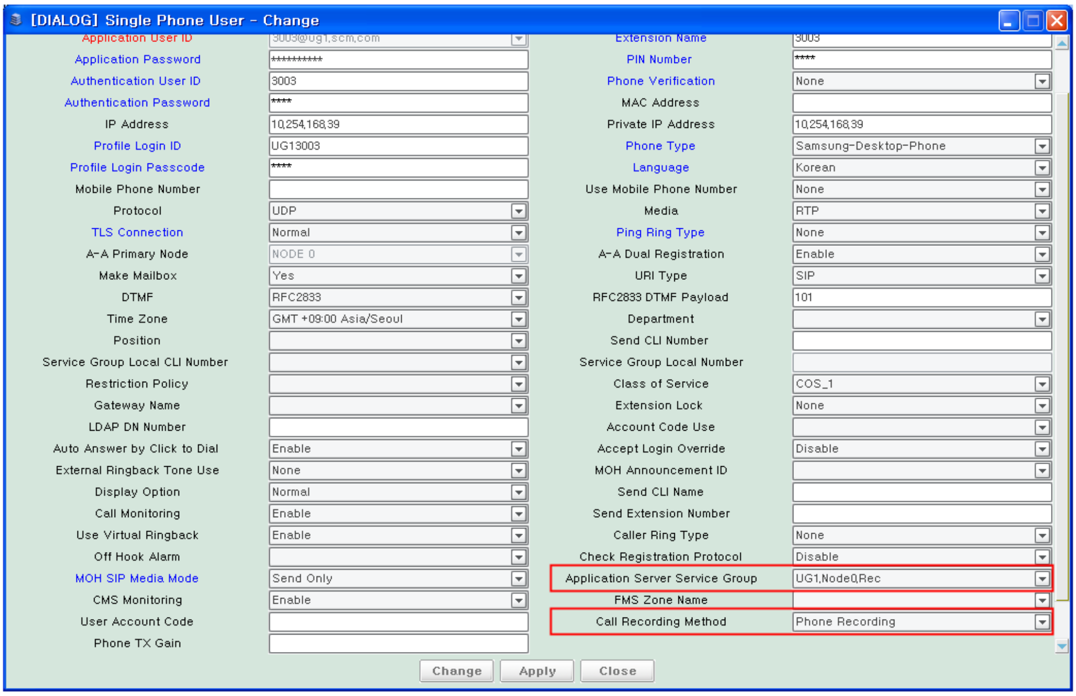



**Figure 3.16 Select Call Recording Method in User Menu**

When subscriber answered a call, phone is displaying [call recording] menu.
If Subscriber select [call recording] menu, call recording service is started.
In [SCM Administrator/Configuration/Service/Feature Service/Service Activation] menu, Auto call recording service is provided.

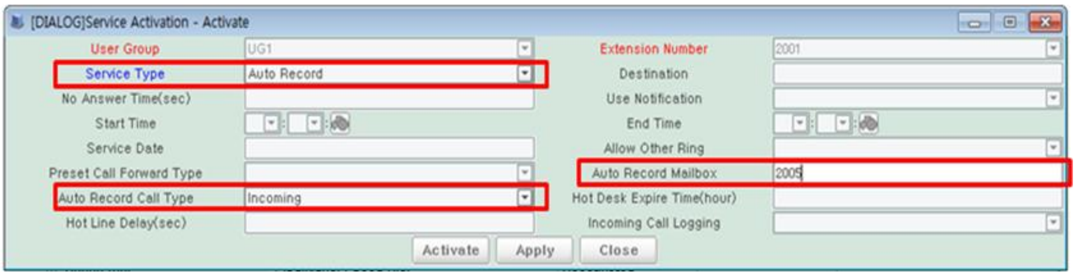



**Figure 3.17 Recording Server Auto-Record Configuration**

## 3.2.6 IVR/Auto Attendant Server

SCM provides call forward service for incoming trunk call, and call forward destination is Auto Attendant Server.

### 3.2.6.1 Configuring IVR/Auto Attendant Server

By using application or single phone user, create IVR/Auto Attendant Server.
If IVR/Auto Attendant Server needs other information, DID Number table, application server method is recommended.

#### Create Auto Attendant Server-using Application Server

In [SCM Administrator/Configuration/Application/VM/AA Server] Menu, create Auto Attendant Server.
Following items are necessary to create Auto Attendant Server.

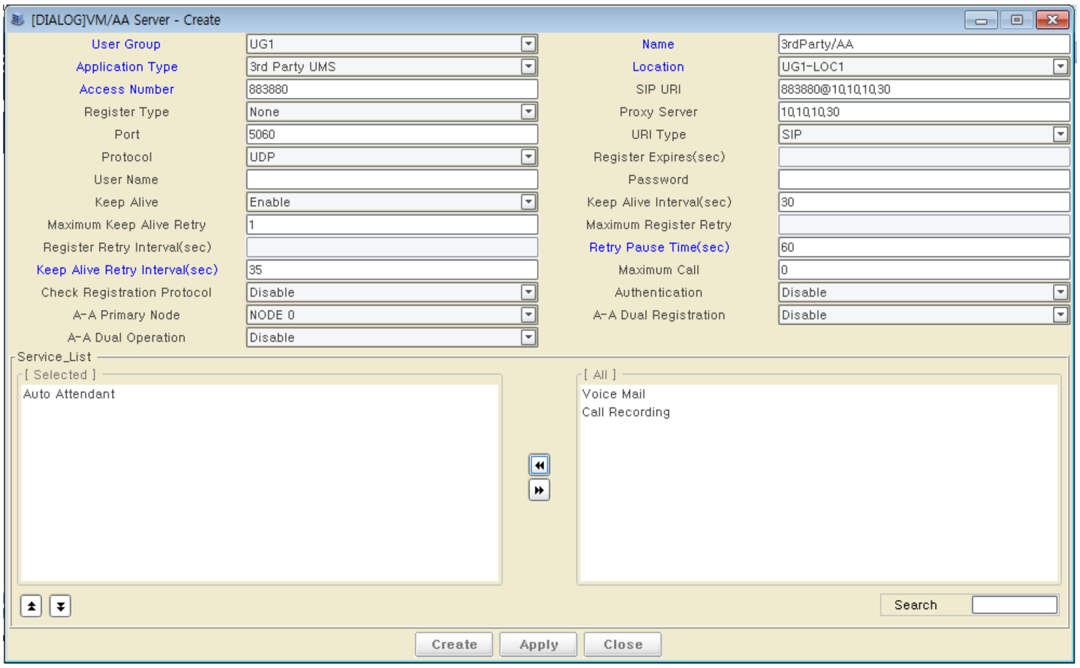



**Figure 3.18 Create Auto Attendant Server**

Table 3.3 Create Auto Attendant Server-Application Case

| Options | Explains |
|---|---|
| User Group | Operator selects User Group that using this application server. |
| Name | Specify Application Server Name |
| Application Type | Select 3rd Party UMS Type |
| Location | Select Location in SCM Express |
| Access Number | Specify Application Server Number.<br>Access number not duplicate number with extension number, conference channel number and system speed dial number. |
| SIP URI | Specify SIP URI, like E-mail Address. Ex) Access Number@IP Address |
| Register Type | Default Selection Option is None.<br>If [Keep Alive] Option is Enable, SCM Send Option Message recursively [Keep Alive Interval] time.<br>Otherwise [Keep Alive] Option is Disable, SCM does not check Application Server's Status.<br>If Application Server can receive register message or send register message, scm select reverse REGISTER. |
| Proxy Server | Specify application server's ip address. |
| Port | Default port number is 5060.<br>If application server use other port, input other port number. |
| Protocol | Default Value is UDP Protocol.<br>If application server allow other protocol, select TCP or TLS protocol |
| Keep Alive | When [Register Type] option is none type, this option also need enabled.<br>When [Keep Alive] option is enabled, SCM send optional message for connection checking. |
| Keep Alive Interva (Sec) | Default Interval value is 30 second.<br>[Keep Alive] option is enabled, SCM uses this interval value. |
| Maximum Call | Specify max channel counts.<br>If Maximum call value is not defined, 3rd party Application call is rejected.<br>Maximum call value is necessary to [SIP Application Channels] value within License key. |
| Service List | In Service List, move [Auto Attendant] option in [selected].<br>If SCM version is before 3.2.4, Application Server is not duplicated service list. |

If SCM version is 3.2.4 or higher, the [3rd Party IVR] Type is available. The operation and configuration for an IVR service are the same as the service using the [3rd Party UMS]. But there is a difference in setting of the DID table. In case of [3rd Party UMS] Type, the UMS feature code should be inserted to the destination number of a DID table. In case of [3rd Party IVR] type, an access number should be inserted to that.

### Create Auto Attendant Server-using single phone user

In [SCM Administrator/Configuration/User/Single Phone User] menu, operator can create auto attendant server.

Operator set the Phone Type [3rd Party-SIP-Phone]. Other configuration is same as single phone user.

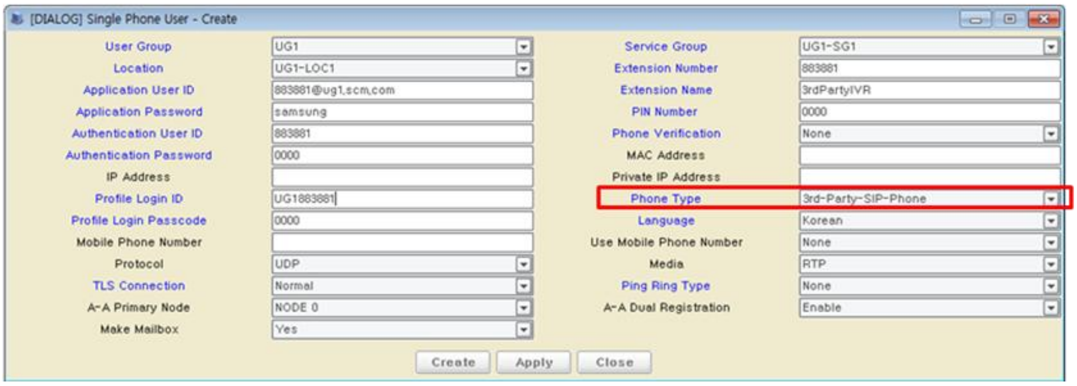



**Figure 3.19 Create Auto Attendant Server-using single phone user**

## 3.2.6.2 IVR/Auto Attendant Operation

In [SCM Administrator/Configuration/Trunk Routing/DID routing] menu, operator create DID Number for Auto Attendant Operation.

After creating an Application Server, the default destination number of a DID table should be '*88'.

If an application server type is 3rd party IVR, the default destination number of a DID table should be an application server's access number.

If an auto attendant server is created as a single phone user, the default destination number of a DID table should be single user's extension number.

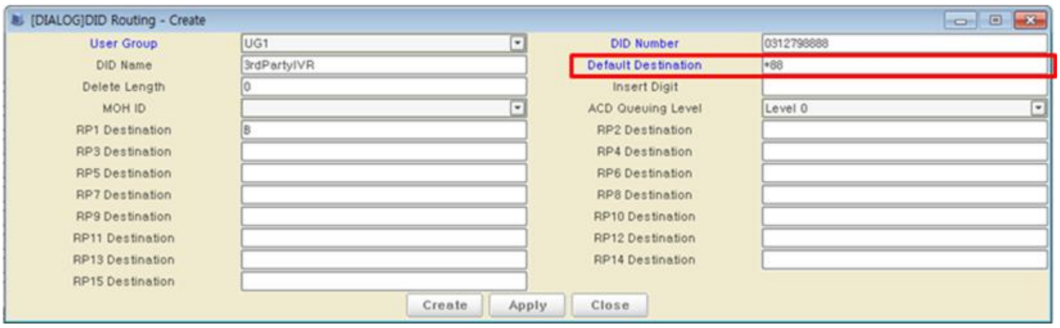



**Figure 3.20 Auto Attendant Operation-DID Configuration**

## 3.2.7 Paging Conference Server

SCM supports Paging feature and Paging on Answer features using Conference Server.

### 3.2.7.1 Configuring Paging Conference Server

Conference Server type of Paging can be created in the [CONFIGURATION > Application > Conference Server] menu.
The follow information can be configured for creating a Conference Server.

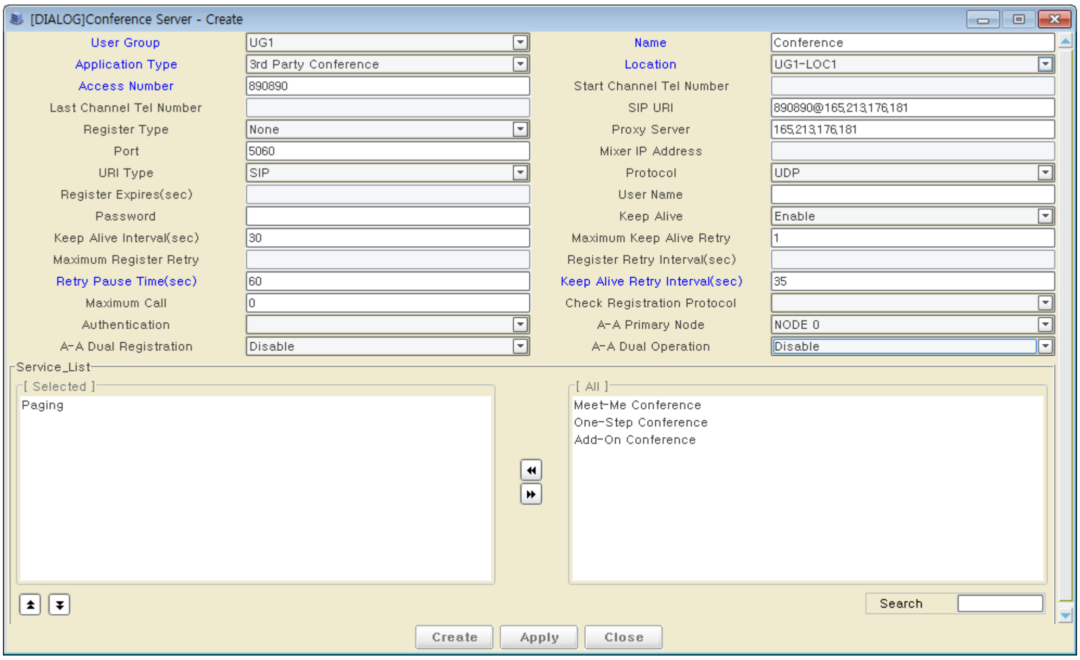



**Figure 3.21 Creating Conference Server for Paging**

**Table 3.4 Creating Conference Server for Paging**

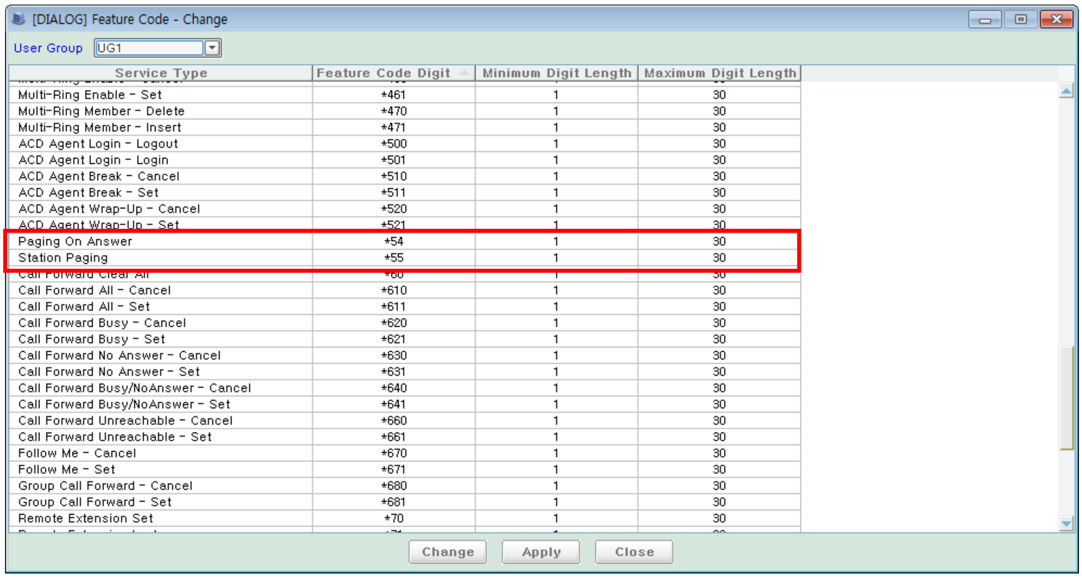
| Item | Description |
|---|---|
| User Group | Select the User Group. |
| Name | Enter the name of Application. |
| Application Type | Select the type of 3rd Party Conference. |
| Location | Select one of the configured Location Information. |
| Access Number | Enter the number used for calling the conference system.<br>This is the key data identifying each Conference Server and must not be entered in duplicate with any extension number, any access number, any system speed dial or any extension group number. |
| SIP URI | Enter the form of an e-mail address. Ex) Access Number@IP address |

Table 3.4 Creating Conference Server for Paging (Continued)

| Item | Description |
|---|---|
| Register Type | Select to None type.<br>If Keep Alive is set to Enable, it is used to verify the connection using SIP OPTION message.<br>If Keep Alive is set to Disable, it is not verified the connection with application server when calling to conference server.<br>If you want to verify the connection using REGISTER message, select to Receive REGISTER or Send REGISTER type. |
| Proxy Server | Enter the IP address of Application Server. |
| Port | Enter 5060 port.<br>If Application Server use the other port, enter that port number. |
| URI Type | Keep the default value to SIP. |
| Protocol | Select a UDP protocol.<br>If the other protocol is used to connecting Application server, enter that protocol. |
| Keep Alive | Select the Enable.<br>This item is activated only if Register Type is selected to None.<br>If this menu is set to Enable, SCM sends the OPTIONS message to the application to check the registration status. |
| Keep Alive Interval (sec) | Keep the default value to 30.<br>This item is activated only if Keep Alive is set to Enable.<br>SCM sends the OPTIONS message to the application sever every Keep Alive seconds to check the registration status. |
| Maximum Call | Enter the maximum available channel count.<br>If this item is blank or 0, all call using this conference server is rejected.<br>This item can be assigned within SIP Application Channels in external application license. |
| Service List | Select the type of Paging. |
| A-A Primary Node | Select the Node number of SCM to register.<br>Specify Primary node when Active-Active mode. |
| A-A Dual Registration | Select the Disable.<br>Specify dual registration enable or not when Active-Active mode.<br>If select dual registration enable, Conference Server must be registered to both node. |
| A-A Dual Operation | Select the Disable. |

### 3.2.7.2 Paging Conference Server Operation

Configure a feature code for Paging or Paging on Answer.
Feature code can be changed in [CONFIGURATION/Service/Feature Service/Feature Code] menu.
Default value of Paging feature code is *55 and default value of Paging on Answer feature code is none.


[DIALOG] Feature Code - Change
User Group UG1

| Service Type | Feature Code Digit | Minimum Digit Length | Maximum Digit Length |
|---|---|---|---|
| Multi-Ring Enable - Set | *461 | 1 | 30 |
| Multi-Ring Member - Delete | *470 | 1 | 30 |
| Multi-Ring Member - Insert | *471 | 1 | 30 |
| ACD Agent Login - Logout | *500 | 1 | 30 |
| ACD Agent Login - Login | *501 | 1 | 30 |
| ACD Agent Break - Cancel | *510 | 1 | 30 |
| ACD Agent Break - Set | *511 | 1 | 30 |
| ACD Agent Wrap-Up - Cancel | *520 | 1 | 30 |
| ACD Agent Wrap-Up - Set | *521 | 1 | 30 |
| Paging On Answer | *54 | 1 | 30 |
| Station Paging | *55 | 1 | 30 |
| Call Forward Clear All | *60 | 1 | 30 |
| Call Forward All - Cancel | *610 | 1 | 30 |
| Call Forward All - Set | *611 | 1 | 30 |
| Call Forward Busy - Cancel | *620 | 1 | 30 |
| Call Forward Busy - Set | *621 | 1 | 30 |
| Call Forward No Answer - Cancel | *630 | 1 | 30 |
| Call Forward No Answer - Set | *631 | 1 | 30 |
| Call Forward Busy/NoAnswer - Cancel | *640 | 1 | 30 |
| Call Forward Busy/NoAnswer - Set | *641 | 1 | 30 |
| Call Forward Unreachable - Cancel | *660 | 1 | 30 |
| Call Forward Unreachable - Set | *661 | 1 | 30 |
| Follow Me - Cancel | *670 | 1 | 30 |
| Follow Me - Set | *671 | 1 | 30 |
| Group Call Forward - Cancel | *680 | 1 | 30 |
| Group Call Forward - Set | *681 | 1 | 30 |
| Remote Extension Set | *70 | 1 | 30 |

Change Apply Close

**Figure 3.22 Conference Server for Paging or Paging on Answer-Feature Code**

## Paging Group

Paging Group can be created in [CONFIGURATION/Service/Group Service/Paging Group] menu.
Users must dial to Paging Group Number (or Station Paging Feature Code + Paging Group Number) to use paging feature.

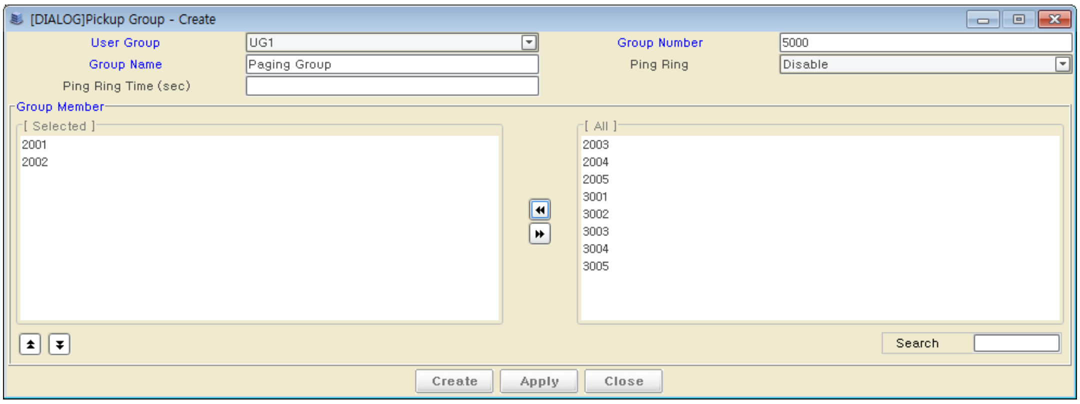



**Figure 3.23 Creating Paging Group**

## Paging On Answer Group

Paging On Answer Group can be created in [CONFIGURATION/Service/Group Service/Paging On Answer Group] menu.
Users must be dial to Paging On Answer Group Number (or Paging On Answer Feature Code + Paging On Answer Group Number) to use paging on answer feature.

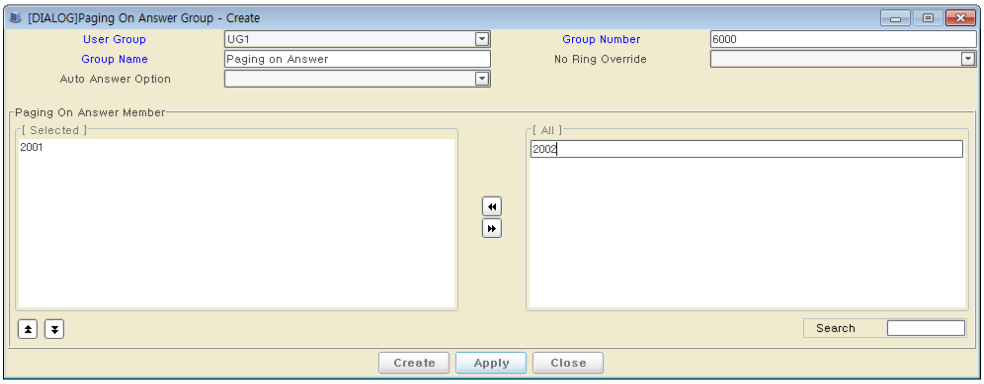



**Figure 3.24 Creating Paging On Answer Group**

- Auto Answer Option
  Activate a Auto Answer feature about subscriber or remote SCM's subscriber that connected by SIP trunk.

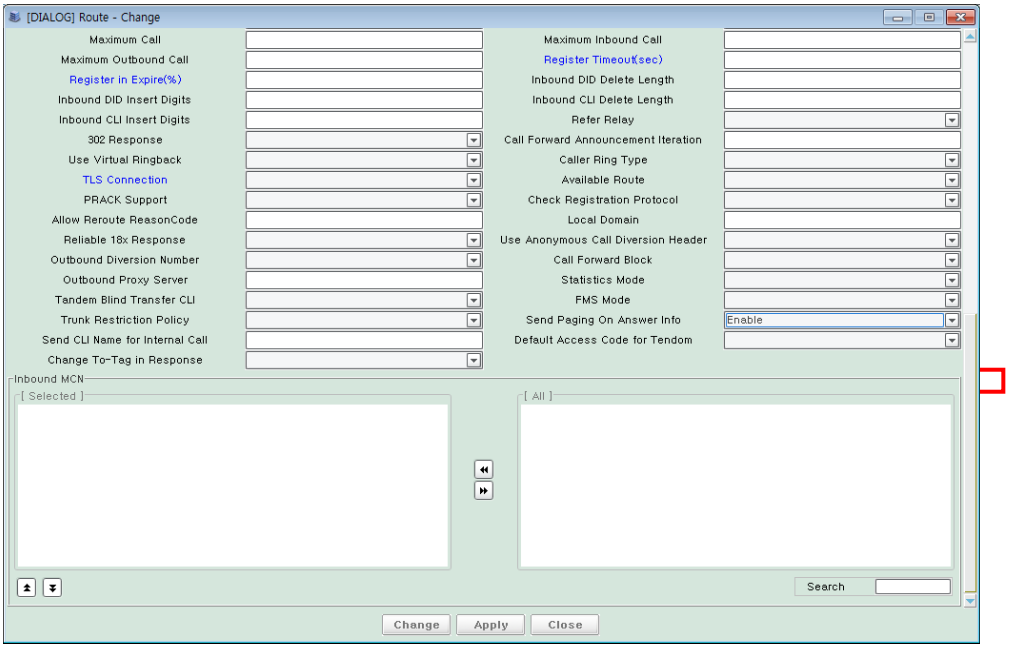



**Figure 3.25 Send Paging On Answer Info**

- Send Paging On Answer Info
  To use a Auto Answer feature about subscriber or remote SCM's subscriber that connected by SIP trunk, this option must be activated in [CONFIGURATION/Trunk Routing/Route] menu.

# 3.2.8 Meet-me Conference Server

## 3.2.8.1 Configuring Meet-me Conference Server

Conference Server type of Meet-Me Conference can be created in the [CONFIGURATION > Application > Conference Server] menu.
The follow information can be configured for creating a Conference Server.

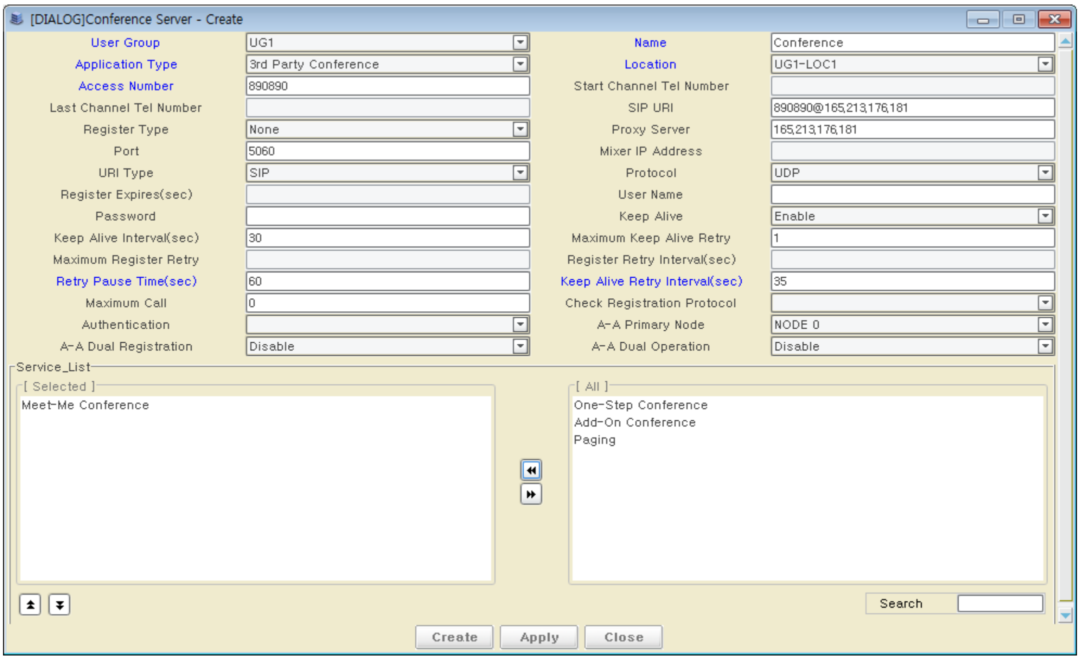



**Figure 3.26 Creating Conference Server for Meet-Me**

**Table 3.5 Creating Conference Server for Meet-Me**

| Item | Description |
|---|---|
| User Group | Select the User Group. |
| Name | Enter the name of Application. |
| Application Type | Select the type of 3rd Party Conference. |
| Location | Select one of the configured Location Information. |
| Access Number | Enter the number used for calling the conference system.<br>This is the key data identifying each Conference Server and must not be entered in duplicate with any extension number, any access number, any system speed dial or any extension group number. |
| SIP URI | Enter the form of an e-mail address. Ex) Access Number@IP address |
| Register Type | Select to None type.<br>If Keep Alive is set to Enable, it is used to verify the connection using SIP OPTION message.<br>If Keep Alive is set to Disable, it is not verified the connection with application server when calling to conference server.<br>If you want to verify the connection using REGISTER message, select to Receive REGISTER or Send REGISTER type. |

**Table 3.5 Creating Conference Server for Meet-Me (Contnued)**

| Item | Description |
| --- | --- |
| Proxy Server | Enter the IP address of Application Server. |
| Port | Enter 5060 port.<br>If Application Server use the other port, enter that port number. |
| URI Type | Keep the default value to SIP. |
| Protocol | Select a UDP protocol.<br>If the other protocol is used to connecting Application server, enter that protocol. |
| Keep Alive | Select the Enable.<br>This item is activated only if Register Type is selected to None.<br>If this menu is set to Enable, SCM sends the OPTIONS message to the application to check the registration status. |
| Keep Alive Interval (sec) | Keep the default value to 30.<br>This item is activated only if Keep Alive is set to Enable.<br>SCM sends the OPTIONS message to the application sever every Keep Alive seconds to check the registration status. |
| Maximum Call | Enter the maximum available channel count.<br>If this item is blank or 0, all call using this conference server is rejected.<br>This item can be assigned within SIP Application Channels in external application license. |
| Service List | Select the type of Paging. |
| A-A Primary Node | Select the Node number of SCM to register.<br>Specify Primary node when Active-Active mode. |
| A-A Dual Registration | Select the Disable.<br>Specify dual registration enable or not when Active-Active mode.<br>If select dual registration enable, Conference Server must be registered to both node. |
| A-A Dual Operation | Select the Disable. |

### 3.2.8.2 Meet-me Conference Server Operation

Configure a feature code for Meet-Me Conference Join.
Feature code can be changed in [CONFIGURATION/Service/Feature Service/Feature Code] menu.
Default value of Meet-Me Conference Join feature code is *82.

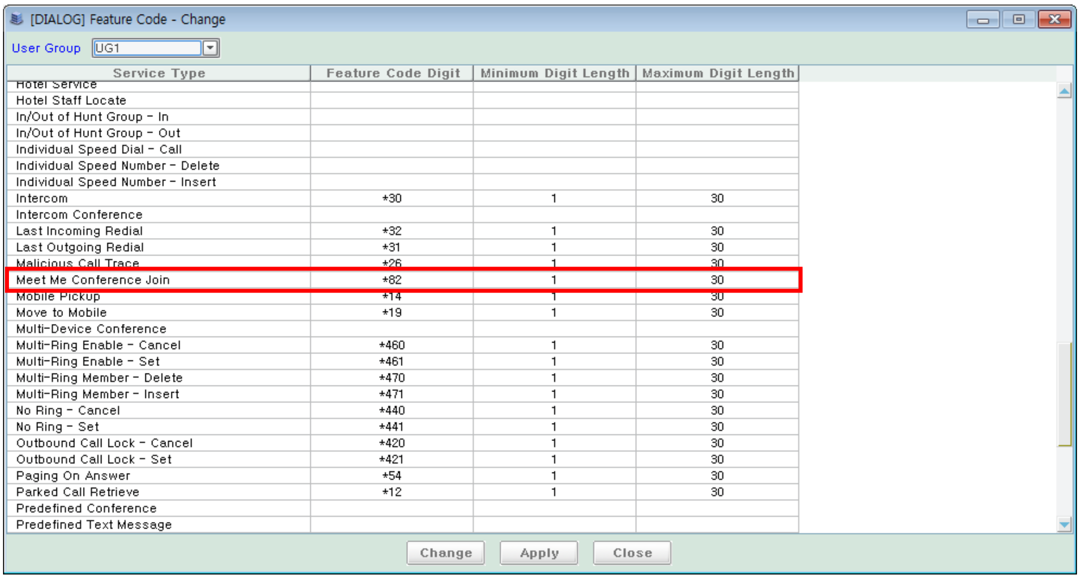



**Figure 3.27 Conference Server for Meet Me Conference Join-Feature Code**

## Meet-Me Reservation

Meet-Me Conference can be reserved in [CONFERENCE/Conference Management/Meet Me Reservation] menu.
Users must dial to Meet Me Conference Join feature code + Conference ID to join Meet-Me Conference.

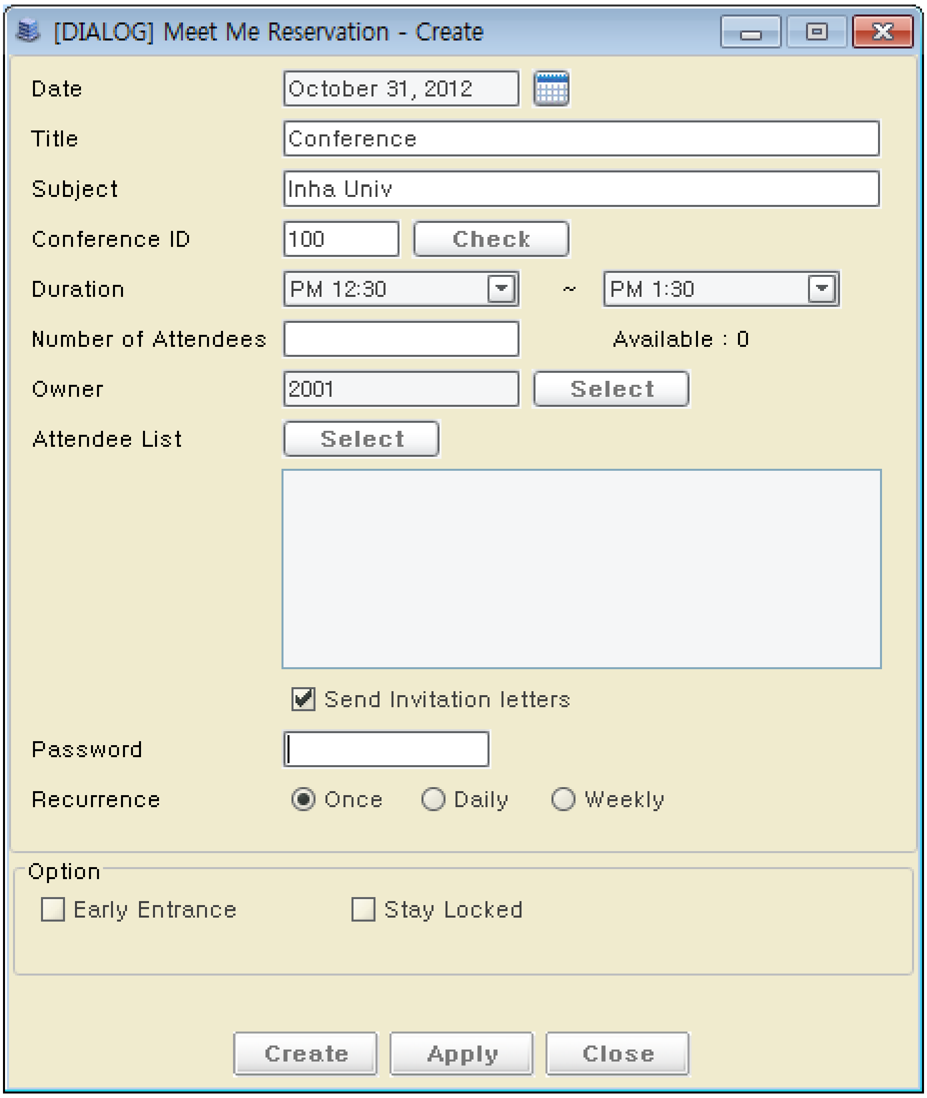



**Figure 3.28 Reserving Meet-Me Conference**

## 3.2.9 One-Step Conference Server

One-Step Conference means that Conference Server calls multiple members.

### 3.2.9.1 Configuring One-Step Conference Server

Conference Server type of One-Step Conference can be created in the [CONFIGURATION > Application > Conference Server] menu.

The follow information can be configured for creating a Conference Server.

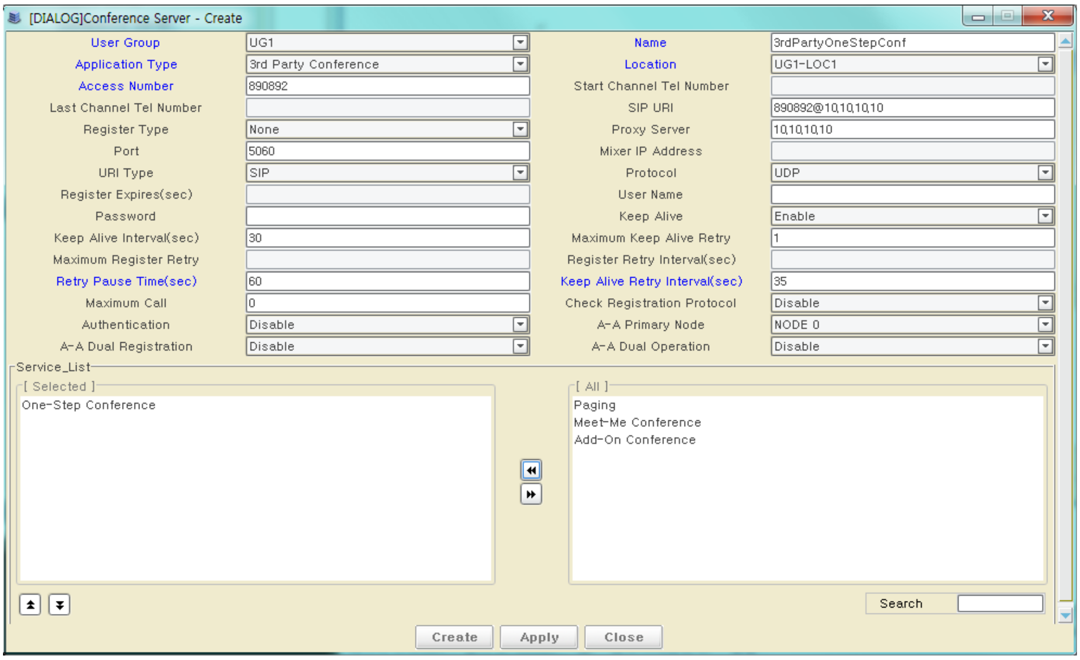



**Figure 3.29 Creating One-Step Conference Server**

**Table 3.6 Creating Conference Server for One-Step Conference**

| Item | Description |
|---|---|
| User Group | Select the User Group. |
| Name | Enter the name of Application. |
| Application Type | Select the type of $3^{rd}$ Party Conference. |
| Location | Select one of the configured Location Information. |
| Access Number | Enter the number used for calling the conference system.<br>This is the key data identifying each Conference Server and must not be entered in duplicate with any extension number, any access number, any system speed dial or any extension group number. |
| SIP URI | Enter the form of an e-mail address. Ex) Access Number@IP address |

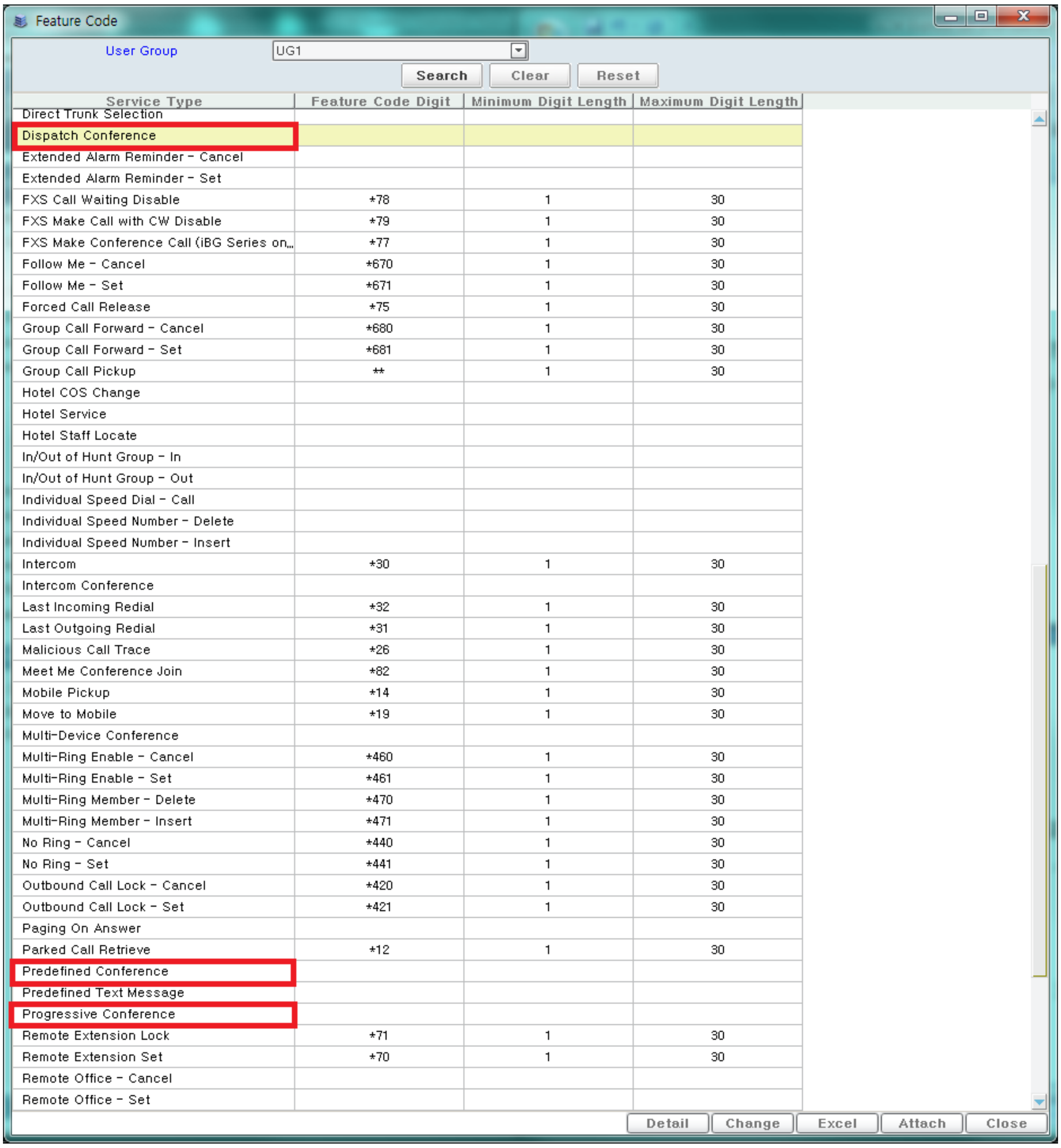
Table 3.6 Creating Conference Server for One-Step Conference (Continued)

| Item | Description |
| --- | --- |
| Register Type | Select to None type.<br>If Keep Alive is set to Enable, it is used to verify the connection using SIP OPTION message.<br>If Keep Alive is set to Disable, it is not verified the connection with application server when calling to conference server.<br>If you want to verify the connection using REGISTER message, select to Receive REGISTER or Send REGISTER type. |
| Proxy Server | Enter the IP address of Application Server. |
| Port | Enter 5060 port.<br>If Application Server use the other port, enter that port number. |
| URI Type | Keep the default value to SIP. |
| Protocol | Select a UDP protocol.<br>If the other protocol is used to connecting Application server, enter that protocol. |
| Keep Alive | Select the Enable.<br>This item is activated only if Register Type is selected to None.<br>If this menu is set to Enable, SCM sends the OPTIONS message to the application to check the registration status. |
| Keep Alive Interval (sec) | Keep the default value to 30.<br>This item is activated only if Keep Alive is set to Enable.<br>SCM sends the OPTIONS message to the application sever every Keep Alive seconds to check the registration status. |
| Maximum Call | Enter the maximum available channel count.<br>If this item is blank or 0, all call using this conference server is rejected.<br>This item can be assigned within SIP Application Channels in external application license. |
| Service List | Select the type of One-Step Conference. |
| A-A Primary Node | Select the Node number of SCM to register.<br>Specify Primary node when Active-Active mode. |
| A-A Dual Registration | Select the Disable.<br>Specify dual registration enable or not when Active-Active mode.<br>If select dual registration enable, Conference Server must be registered to both node. |
| A-A Dual Operation | Select the Disable. |

### 3.2.9.2 One-Step Conference Server Operation

Configure a feature code for One-Step Conference.
Feature code can be changed in [CONFIGURATION/Service/Feature Service/Feature Code] menu.




| Service Type | Feature Code Digit | Minimum Digit Length | Maximum Digit Length |
|---|---|---|---|
| Direct Trunk Selection | | | |
| Dispatch Conference | | | |
| Extended Alarm Reminder - Cancel | | | |
| Extended Alarm Reminder - Set | | | |
| FXS Call Waiting Disable | *78 | 1 | 30 |
| FXS Make Call with CW Disable | *79 | 1 | 30 |
| FXS Make Conference Call (iBG Series on.. | *77 | 1 | 30 |
| Follow Me - Cancel | *670 | 1 | 30 |
| Follow Me - Set | *671 | 1 | 30 |
| Forced Call Release | *75 | 1 | 30 |
| Group Call Forward - Cancel | *680 | 1 | 30 |
| Group Call Forward - Set | *681 | 1 | 30 |
| Group Call Pickup | ** | 1 | 30 |
| Hotel COS Change | | | |
| Hotel Service | | | |
| Hotel Staff Locate | | | |
| In/Out of Hunt Group - In | | | |
| In/Out of Hunt Group - Out | | | |
| Individual Speed Dial - Call | | | |
| Individual Speed Number - Delete | | | |
| Individual Speed Number - Insert | | | |
| Intercom | *30 | 1 | 30 |
| Intercom Conference | | | |
| Last Incoming Redial | *32 | 1 | 30 |
| Last Outgoing Redial | *31 | 1 | 30 |
| Malicious Call Trace | *26 | 1 | 30 |
| Meet Me Conference Join | *82 | 1 | 30 |
| Mobile Pickup | *14 | 1 | 30 |
| Move to Mobile | *19 | 1 | 30 |
| Multi-Device Conference | | | |
| Multi-Ring Enable - Cancel | *460 | 1 | 30 |
| Multi-Ring Enable - Set | *461 | 1 | 30 |
| Multi-Ring Member - Delete | *470 | 1 | 30 |
| Multi-Ring Member - Insert | *471 | 1 | 30 |
| No Ring - Cancel | *440 | 1 | 30 |
| No Ring - Set | *441 | 1 | 30 |
| Outbound Call Lock - Cancel | *420 | 1 | 30 |
| Outbound Call Lock - Set | *421 | 1 | 30 |
| Paging On Answer | | | |
| Parked Call Retrieve | *12 | 1 | 30 |
| Predefined Conference | | | |
| Predefined Text Message | | | |
| Progressive Conference | | | |
| Remote Extension Lock | *71 | 1 | 30 |
| Remote Extension Set | *70 | 1 | 30 |
| Remote Office - Cancel | | | |
| Remote Office - Set | | | |

#### Progressive Conference

Similar to the predefined method, the conference attendees are not registered in advance but the attendees' phone numbers are entered one by one according to the interactive voice announcement. When the call is made, those attendees answering the call are automatically included in the conference.

### Predefined Conference

A list of conference attendees are registered in advance and the attendees are paged using the conference group number. Those attendees answering the call are automatically included in the conference.

### Dispatch Conference

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

## 3.2.10 Basic Conference Server

### 3.2.10.1 Configuring Basic Conference Server

Conference Server type of One-Step Conference can be created in the [CONFIGURATION > Application > Conference Server] menu.
The follow information can be configured for creating a Conference Server.

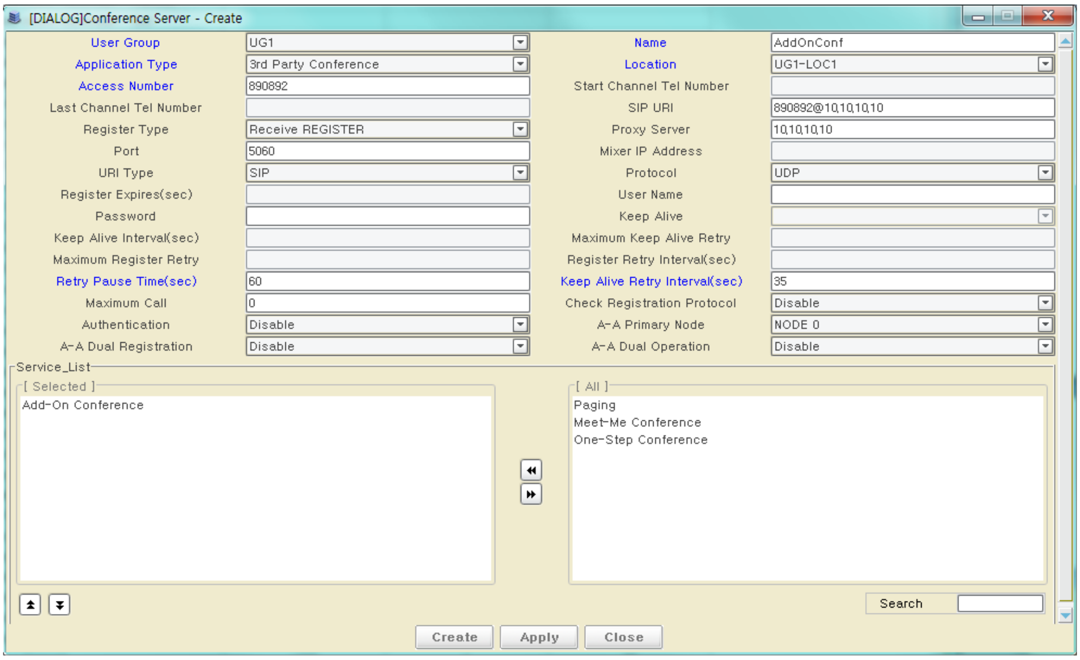



Figure 3.30 Creating Add-On Conference Server

Table 3.7 Creating Conference Server for Add-On Conference

| Item | Description |
|---|---|
| User Group | Select the User Group. |
| Name | Enter the name of Application. |
| Application Type | Select the type of 3rd Party Conference. |
| Location | Select one of the configured Location Information. |

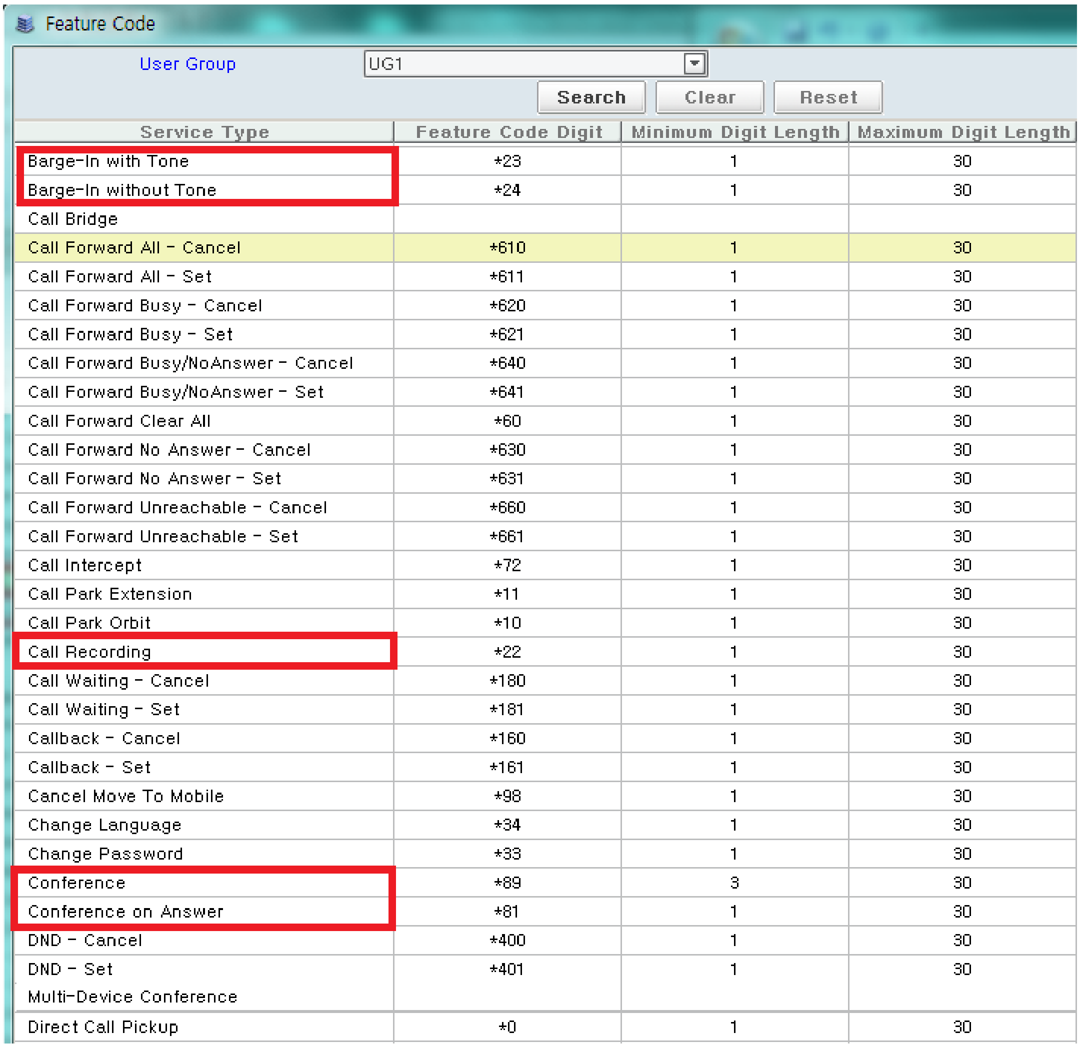
 

Table 3.7 Creating Conference Server for Add-On Conference (Continued)

| Item | Description |
|---|---|
| Access Number | Enter the number used for calling the conference system.<br>This is the key data identifying each Conference Server and must not be entered in duplicate with any extension number, any access number, any system speed dial or any extension group number. |
| SIP URI | Enter the form of an e-mail address. Ex) Access Number@IP address |
| Register Type | Select to None type.<br>If Keep Alive is set to Enable, it is used to verify the connection using SIP OPTION message.<br>If Keep Alive is set to Disable, it is not verified the connection with application server when calling to conference server.<br>If you want to verify the connection using REGISTER message, select to Receive REGISTER or Send REGISTER type. |
| Proxy Server | Enter the IP address of Application Server. |
| Port | Enter 5060 port.<br>If Application Server use the other port, enter that port number. |
| URI Type | Keep the default value to SIP. |
| Protocol | Select a UDP protocol.<br>If the other protocol is used to connecting Application server, enter that protocol. |
| Keep Alive | Select the Enable.<br>This item is activated only if Register Type is selected to None.<br>If this menu is set to Enable, SCM sends the OPTIONS message to the application to check the registration status. |
| Keep Alive Interval (sec) | Keep the default value to 30.<br>This item is activated only if Keep Alive is set to Enable.<br>SCM sends the OPTIONS message to the application sever every Keep Alive seconds to check the registration status. |
| Maximum Call | Enter the maximum available channel count.<br>If this item is blank or 0, all call using this conference server is rejected.<br>This item can be assigned within SIP Application Channels in external application license. |
| Service List | Select the type of Add-On Conference. |
| A-A Primary Node | Select the Node number of SCM to register.<br>Specify Primary node when Active-Active mode. |
| A-A Dual Registration | Select the Disable.<br>Specify dual registration enable or not when Active-Active mode.<br>If select dual registration enable, Conference Server must be registered to both node. |
| A-A Dual Operation | Select the Disable. |

### 3.2.10.2 Basic Conference Operation

Configure a feature code for Add-On Conference.
Feature code can be changed in [CONFIGURATION/Service/Feature Service/Feature Code] menu.


Feature Code

User Group: UG1

Search | Clear | Reset

| Service Type | Feature Code Digit | Minimum Digit Length | Maximum Digit Length |
|---|---|---|---|
| Barge-In with Tone | *23 | 1 | 30 |
| Barge-In without Tone | *24 | 1 | 30 |
| Call Bridge | | | |
| Call Forward All - Cancel | *610 | 1 | 30 |
| Call Forward All - Set | *611 | 1 | 30 |
| Call Forward Busy - Cancel | *620 | 1 | 30 |
| Call Forward Busy - Set | *621 | 1 | 30 |
| Call Forward Busy/NoAnswer - Cancel | *640 | 1 | 30 |
| Call Forward Busy/NoAnswer - Set | *641 | 1 | 30 |
| Call Forward Clear All | *60 | 1 | 30 |
| Call Forward No Answer - Cancel | *630 | 1 | 30 |
| Call Forward No Answer - Set | *631 | 1 | 30 |
| Call Forward Unreachable - Cancel | *660 | 1 | 30 |
| Call Forward Unreachable - Set | *661 | 1 | 30 |
| Call Intercept | *72 | 1 | 30 |
| Call Park Extension | *11 | 1 | 30 |
| Call Park Orbit | *10 | 1 | 30 |
| Call Recording | *22 | 1 | 30 |
| Call Waiting - Cancel | *180 | 1 | 30 |
| Call Waiting - Set | *181 | 1 | 30 |
| Callback - Cancel | *160 | 1 | 30 |
| Callback - Set | *161 | 1 | 30 |
| Cancel Move To Mobile | *98 | 1 | 30 |
| Change Language | *34 | 1 | 30 |
| Change Password | *33 | 1 | 30 |
| Conference | *89 | 3 | 30 |
| Conference on Answer | *81 | 1 | 30 |
| DND - Cancel | *400 | 1 | 30 |
| DND - Set | *401 | 1 | 30 |
| Multi-Device Conference | | | |
| Direct Call Pickup | *0 | 1 | 30 |

#### Add-On Conference

During a call (including a conference call), the call can be put on hold and a new call is made to another attendee. If the new attendee answers the call, the conference button can be pressed to include the new attendee in the conference.

#### Conference On Answer (COA)

Similar to the Add-On method, a call is made to an attendee and when the called party answers the call, the called party is automatically included in the conference.

### Call Recording

A recording server can receive the mixed RTP data through a conference server or a phone. In case a conference server is used, it is named 'conference recording' service.
If the service permission is set, the call recording feature is available. The call recording feature records a call conversation in voice mail during the call.

### 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 'Override Level' of the user who tries to intrude into the call on conversation should be larger than the 'Privacy Level' of the user who will be intruded the call.

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 'Override Level' of the user who tries to intrude into the call on conversation should be larger than the 'Privacy Level' of the user who will be intruded the call.

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.

### Multi Device Conference

Similar to the Barge-In with tone, you can intrude into a call and establish a three-way conference using multi-device state soft-key.

## 3.2.11 SMS Server

### 3.2.11.1 Configuring SMS Server

You can create Conference server whose [Application type] is External SMS/Route Based SMS on [SCM Administrator/Configuration/Application/Other Application Server] menu. Essential items for SMS server is described on below figure and table.

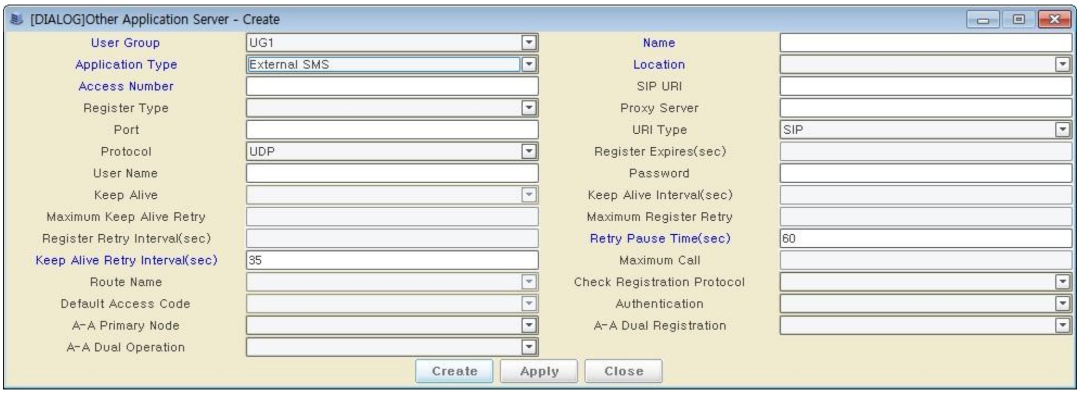



**Figure 3.31 Creation of SMS server**

**Table 3.8 Creation of SMS server**

| Item | Description |
|---|---|
| User Group | Select the User Group. |
| Name | Enter the name of Application. |
| Application Type | Select the type of External SMS. |
| Location | Select one of the configured Location Information. |
| Access Number | Enter the number used for calling the conference system.<br>This is the key data identifying each Conference Server and must not be entered in duplicate with any extension number, any access number, any system speed dial or any extension group number. |
| SIP URI | Enter the form of an e-mail address. Ex) Access Number@IP address |
| Register Type | Select to None type.<br>If Keep Alive is set to Enable, it is used to verify the connection using SIP OPTION message.<br>If Keep Alive is set to Disable, it is not verified the connection with application server when calling to conference server.<br>If you want to verify the connection using REGISTER message, select to Receive REGISTER or Send REGISTER type. |
| Proxy Server | Enter the IP address of Application Server. |
| Port | Enter 5060 port.<br>If Application Server use the other port, enter that port number. |
| URI Type | Keep the default value to SIP. |

**Table 3.8 Creation of SMS server (Continued)**

| Item | Description |
|---|---|
| Protocol | Select a UDP protocol.<br>If the other protocol is used to connecting Application server, enter that protocol. |
| Keep Alive | Select the Enable.<br>This item is activated only if Register Type is selected to None.<br>If this menu is set to Enable, SCM sends the OPTIONS message to the application to check the registration status. |
| Keep Alive Interval (sec) | Keep the default value to 30.<br>This item is activated only if Keep Alive is set to Enable.<br>SCM sends the OPTIONS message to the application sever every Keep Alive seconds to check the registration status. |
| A-A Primary Node | Select the Node number of SCM to register.<br>Specify Primary node when Active-Active mode. |
| A-A Dual Registration | Select the Disable.<br>Specify dual registration enable or not when Active-Active mode.<br>If select dual registration enable, Conference Server must be registered to both node. |
| A-A Dual Operation | Select the Disable. |

## 3.2.12 External Ringback Tone Server

Caller can hear musical ring tone through external ring back tone server when a call is received to user or hunt group of SCME.

### 3.2.12.1 Configuring External Ring back Tone Server

To use musical ring tone service, first SCME has to have the interface of ring back tone server. There are required menu items as follow to create the interface of external ring back tone server.

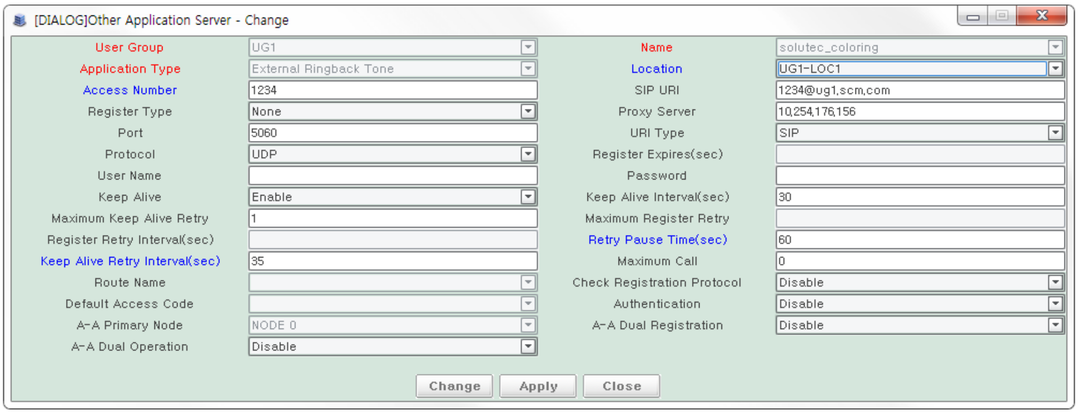



**Figure 3.32 Create external ring back tone server**

**Table 3.9 Create external ring back tone server**

| Item | Description |
|---|---|
| User Group | Select the User Group. |
| Name | Enter the name of Application. |
| Application Type | Select the type of 3rd Party Conference. |
| Location | Select one of the configured Location Information. |
| Access Number | Enter the number used for calling the conference system.<br>This is the key data identifying each Conference Server and must not be entered in duplicate with any extension number, any access number, any system speed dial or any extension group number. |
| SIP URI | Enter the form of an e-mail address. Ex) Access Number@IP address |
| Register Type | Select to None type.<br>If Keep Alive is set to Enable, it is used to verify the connection using SIP OPTION message.<br>If Keep Alive is set to Disable, it is not verified the connection with application server when calling to conference server.<br>If you want to verify the connection using REGISTER message, select to Receive REGISTER or Send REGISTER type. |
| Proxy Server | Enter the IP address of Application Server. |

**Table 3.9 Create external ring back tone server (Continued)**

| Item | Description |
|---|---|
| Port | Enter 5060 port.<br>If Application Server use the other port, enter that port number. |
| URI Type | Keep the default value to SIP. |
| Protocol | Select a UDP protocol.<br>If the other protocol is used to connecting Application server, enter that protocol. |
| Keep Alive | Select the Enable.<br>This item is activated only if Register Type is selected to None.<br>If this menu is set to Enable, SCM sends the OPTIONS message to the application to check the registration status. |
| Keep Alive Interval(sec) | Keep the default value to 30.<br>This item is activated only if Keep Alive is set to Enable.<br>SCM sends the OPTIONS message to the application sever every Keep Alive seconds to check the registration status. |
| Maximum Call | Enter the maximum available channel count.<br>If this item is blank or 0, all call using this conference server is rejected.<br>This item can be assigned within SIP Application Channels in external application license. |
| Service List | Select the type of Paging. |
| A-A Primary Node | Select the Node number of SCM to register.<br>Specify Primary node when Active-Active mode. |
| A-A Dual Registration | Select the Disable.<br>Specify dual registration enable or not when Active-Active mode.<br>If select dual registration enable, Conference Server must be registered to both node. |
| A-A Dual Operation | Select the Disable. |

### 3.2.12.2 External Ring back Tone Server Operation

External ring back tone service has been provided since SCME version 3.2.2.x. Multiple external ring back tone server can be created at SCME. Different ring back tone can be provided by each user. External ring back tone service is provided through Application server service group since SCME version 3.2.4.x. You refer to Ch.3 Application Server Service Group, if you want detail information.

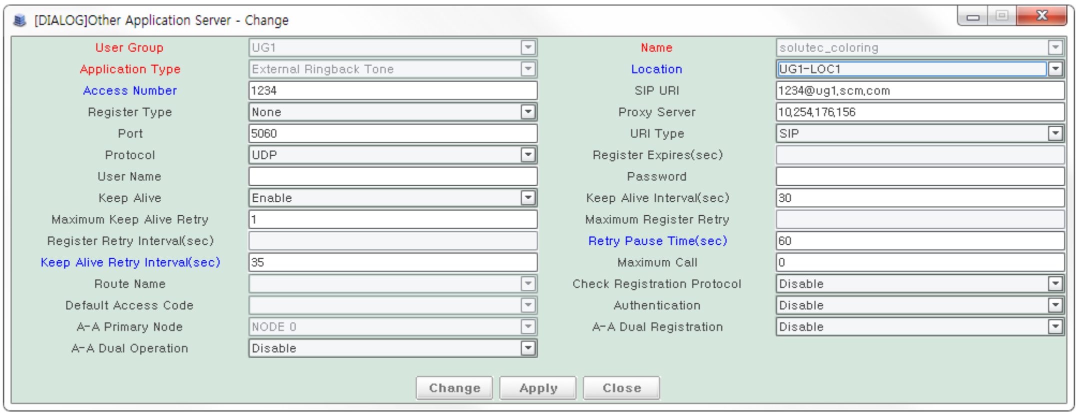



**Figure 3.33 Default Ring back tone server setting**

To create ring back tone server, move [CONFIGURATION → Application → Other Application Server] as follow. Figure 3.28 shows the created display of ring back tone server. And created ring back tone server has to be allocated to Application Server Group. Figure 3.29 shows Application Server Group to create.

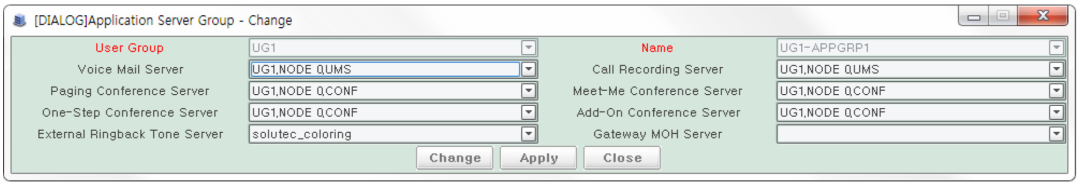



**Figure 3.34 Default Ring back tone server setting**

Also Application Server Group is allocated to user to use ring back tone service. Figure 3 30 shows how to select Application Server Service Group created.

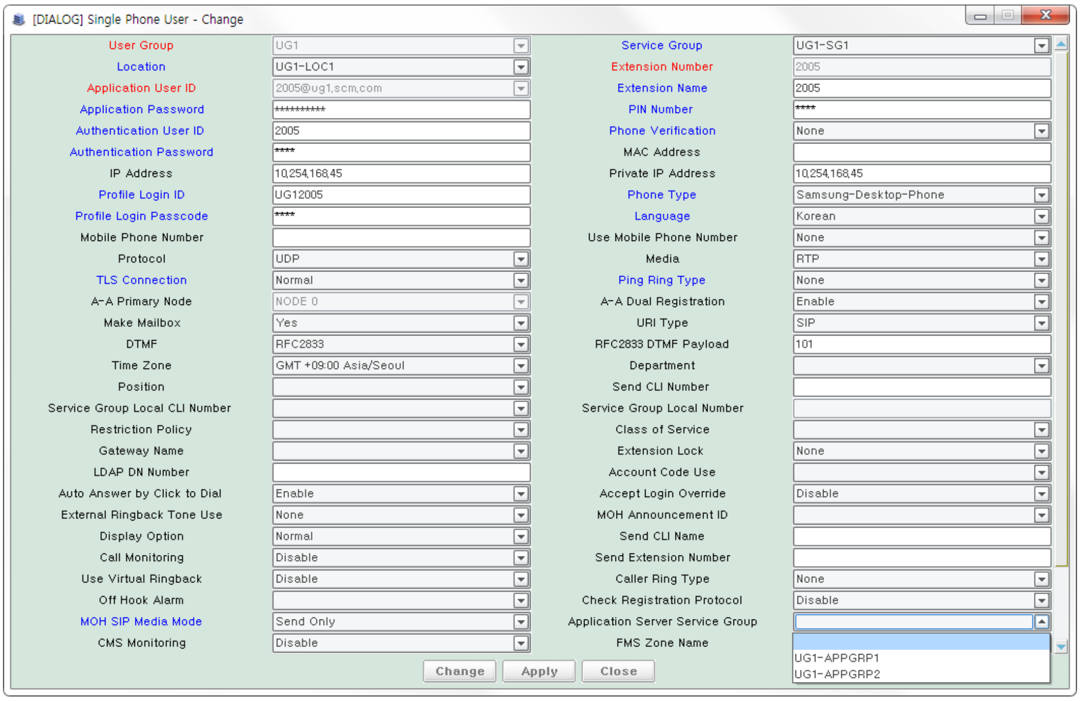



**Figure 3.35 User Application Server Service Group selectiong**

Each hunt group can select External Ringback tone server at menu [SCM Administrator/ CONFIGURATIO/Service/Group Service/Hunt Group]

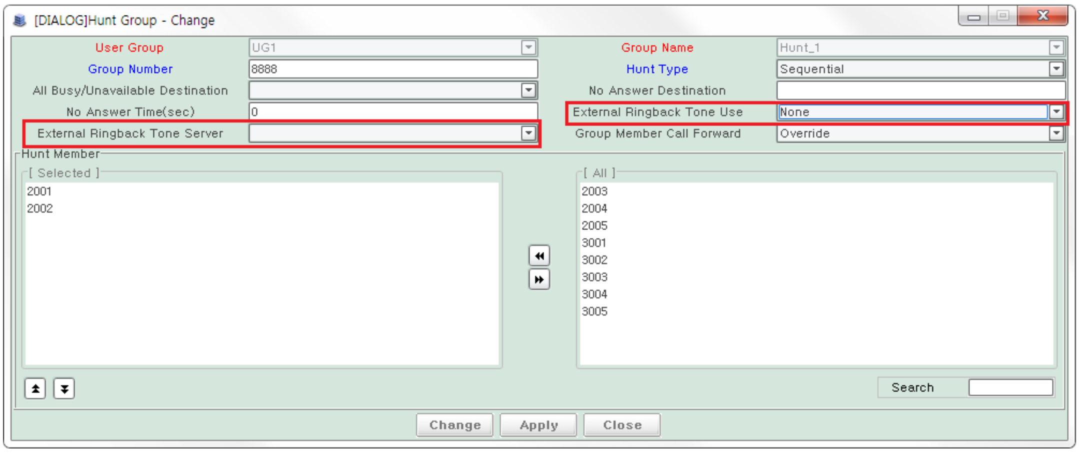



**Figure 3.36 Hunt Group Ring back tone server selection**

# 3.2.13 MS OCS

SCME supports interworking with MS OCS (Microsoft Office Communications Server).

## 3.2.13.1 Configuring SIP Trunk for MS-OCS

SIP route for outgoing to OCS have to be created at
[SCM Administrator/CONFIGURATION/Trunk Routing/Route].

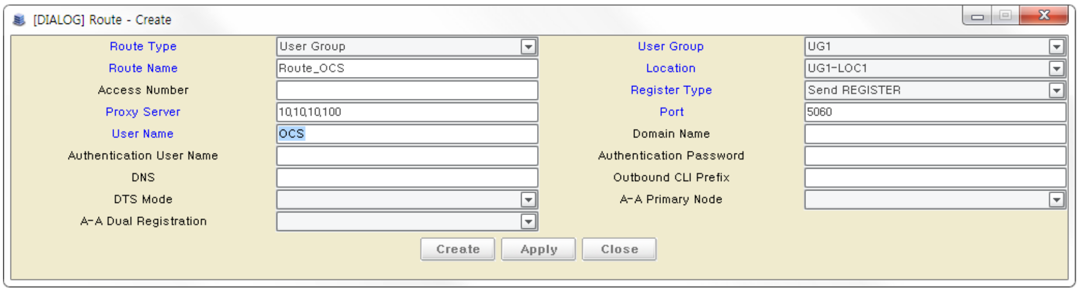



**Figure 3.37 MS OCS outgoing route creation**

If SIP URI must be E.164 format depending on setting of the OCS, ‘Both’ has to be selected at ‘Modify E.164 Format’ item in Route.

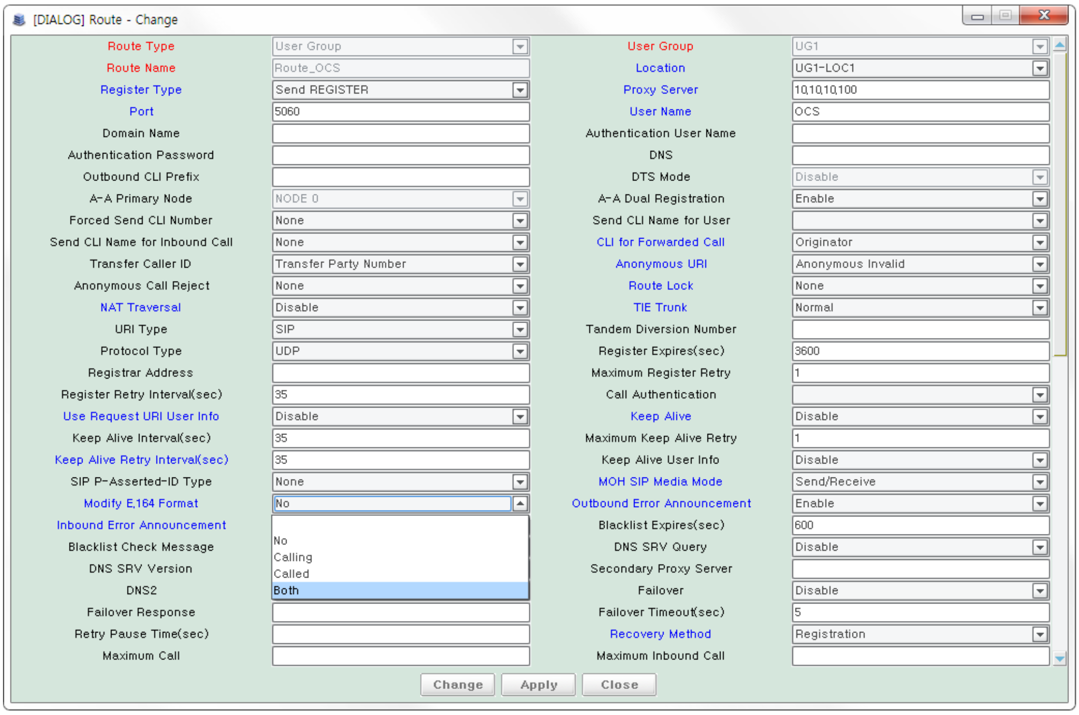



**Figure 3.38 MS OCS SIP URI format setting for interworking**

### 3.2.13.2 Dual Forking

SCME provides forking feature through multi-ring service.
MS OCS client make a call and MS OCS server can fork the call to the SCME. If the user of SCME sets multi-ring service and the member of the user is caller (MS OCS client), it happens double forking (SCME also forks the call to MS OCS client). To prevent this case, MS UC item has to be selected to 'Set'. This setting can be set at [SCME Administrator/ CONFIGURATION/Service/User Service/Multi-ring List].

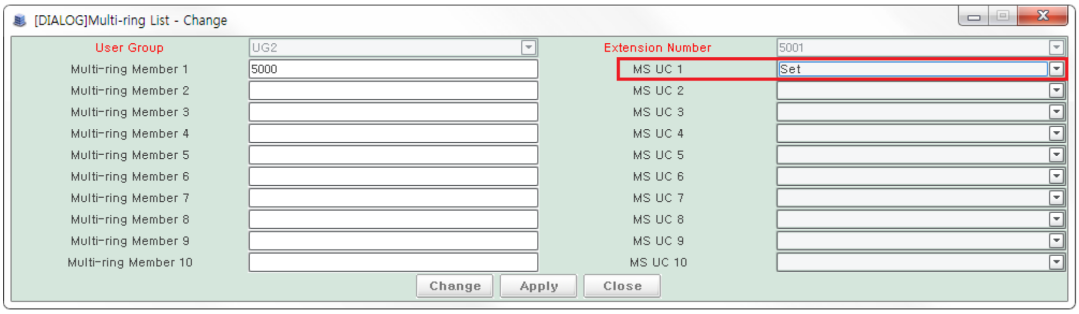



**Figure 3.39 MS OCS Forking feature setting**

# 3.3 Interoperability Specification

## 3.3.1 Basic SIP Usage

This chapter describes SIP Usage that SCM defines to integrate with applications. For more detailed, refer to below chapters that provides the information for each application.

### 3.3.1.1 Headers

#### Outbound Call (SCM to Application)

In case SCM makes a call to application server, SIP Headers should be used as shown below.

- **'From' Header**
  An extension number and a host should be contained in 'From' header.
  *From: <sip:Extension Number@host>*
- 'To' Header /Request URI
  Application access number or feature code should be contained in 'To' header.
  *To: <sip:Application Access Number or Feature Code@host>*
- **'Reason' Header**
  In case of conference calls, the display for master or members is specified in the 'Reason' header.
  *Reason: SIP ;cause=200 ;text='PAGE'*
- **'Contact' Header**
  A conference channel is contained in 'Contact' header. SCM knows whether the call is on conference or not through 'isfocus' of 'Contact' header.
  *Contact: <sip:890009@10.254.176.126:5090>;isfocus*
- **'Subject' Header**
  In case of Recording/Barge-In/Multi-Device conferences, the master number should be contained in 'Subject' header.
  *Subject: <UMS>::[2005]*

#### Inbound Call (Application to SCM)

In case an application server calls members such as Paging or One-Step conference, SIP headers should be used as shown below.

- **'From' Header**
  A conference master and a host should be contained in 'From' header.
  *From: <sip:Conference Master Number@host>*
- **'To' Header /Request URI**
  The number of conference members and a host are specified in the 'TO' header and Request URI.
  *To: <sip:Conference Member Number @host>*

- **‘CallInfo’ Header**
  In case a conference server calls members, there may needs some additional services like Auto Answer. In that case, ‘answer-after’ entry of ‘CallInfo’ header should be set ‘0’.
  SCM can know what kind of conferences is called using ‘purpose’ entry of ‘CallInfo’ header.
  *Call-Info: <http://127.0.0.1>;answer-after=0;purpose=dispatch-conf*
  *Call-Info: <http://127.0.0.1>;purpose=predefined-conf*
- **’Reason’ Header**
  SCM controls the LCD display of conference members through ‘Reason’ header.
  The ‘text’ specified in ‘Reason’ header of INVITE message is displayed through LCD interfaces of a conference member.
  ‘Conference’ will be displayed on the LCD of conference members in the case below.
  *Reason: SIP ;cause=200 ;text=‘Conference’*
- **‘Contact’ Header**
  A conference channel is contained in ‘Contact’ header. SCM knows whether the call is on conference or not through ‘isfocus’ of ‘Contact’ header.
  *Contact: <sip:890009@10.254.176.126:5090>;isfocus*

### 3.3.1.2 Session Description Protocol (SDP) Offer-Answer

#### Outbound-(Re-)INVITE with Offer

If an application server receives an INVITE message which contains SDPs (Session Description Protocol) during a session, one of SDPs offered should be selected and the server should send a 200 OK message containing the selected SDP.

#### Outbound-(Re-)INVITE without Offer

If an application server receives (Re-)INVITE without SDP, all sdp-capabilities should be included in the 200 OK message. It should be done regardless of the media session which was negotiated before.
In this case, a SDP negotiation proceeds between 200 OK (Offer SDP) and ACK (Answer SDP).

#### 3.3.1.3 DTMF

An application server should support three kinds of DTMF types.

- RFC 2833 (It is essential)
- In-band (In-voice)
- INFO: ‘Content-Type’ should be ‘application/dtmf-relay’.

In case below applications are integrated with SCM, the above DTMF types should be supported.

- Voice Mail Server
- Auto Attendant Server
- Meet-me Conference Server
- One-Step Conference Server (Predefined/Progressive Conference)

#### 3.3.1.4 Internal DB Integration between SCM and Application Server

There are several conference types which are supported through conference application server. In order to distinguish these conferences, SCM sends an INVITE message which is containing a feature code. The conference server which received the INVITE message can know the conference type through a feature code.
A conference server should know feature codes matched with conference types in SCM’s database.

An application server should know the member list of conferences that an application server calls members such as Paging or One-Step conference.

## 3.3.2 Voice Mail Server

This chapter describes the SIP usages for 3rd Party Voice Mail Server integration.

### 3.3.2.1 SIP usage

#### Outbound Call (SCM to Voice Mail Server)

- **'From' Header**
  An extension number and a host are contained in 'From' header.
  *From: sip:Extension Number@host>*
- **'To' Header/Request URI**
  The access number and a host of a voice mail server are included in 'To' header and Request URI.
  *To: 'Voice Mail' < sip:Voice Mail Access Number@host>*
- **'Diversion' Header**
  A mailbox number and reason is contained in 'Diversion' header.
  The 'reason' is used to distinguish the requested services, in case one device supports several services such as voice mail, recording, AA and coloring.
  *Diversion: <sip: DID Number (Trunk Incoming) or Extension Number@host>;purpose=record*

### 3.3.2.2 Call Flows

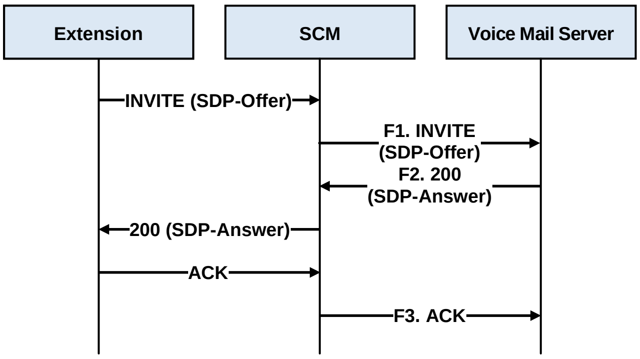



**Figure 3.40 Voice Mail Call Flow**

1) A user sends an INVITE message which is containing a voice mail feature code in 'To' URI.
2) SCM determines a voice mail server by using the feature code and subscriber information.
3) The voice mail server sends a 200 OK message that is including the negotiated SDP capability.
4) SCM sends the received 200 OK message to the user.
5) SCM send the received ACK message to the voice mail server.

### 3.3.2.3 SIP Messages

880880: Voice Mail Access Number
2005: An extension and mailbox number of the user that requests a voice mail service.

In the case below, a 2005 extension dials a voice mail feature code and is connected to a voice mail server.

```
F1.
INVITE sip:880880@ug1.scm.com:5080 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=83b87bb0-b869-4c0b-b147-26c2beeaab9a
To: "Voice Mail"<sip:880880@ug1.scm.com>
Call-ID: 6098cf74-4209-4c8f-af2a-40d75af11047@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7c471-1e575cf4-76893e21
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Diversion: <sip:2005@ug1.scm.com>;reason="intercom-direct-call"
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Max-Forwards: 70
Contact: <sip:2005@10.254.176.126:5060>
User-Agent: SAMSUNG SCM 3.2.400
Content-Type: application/sdp
Content-Length: 339

v=0
o=2005 1340826055 0 IN IP4 165.213.176.85
s=Samsung IP PBX
c=IN IP4 165.213.176.85
t=0 0
a=sendrecv
m=audio 20160 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F2.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=83b87bb0-b869-4c0b-b147-26c2beeaab9a
To: "Voice Mail"<sip:880880@ug1.scm.com>;tag=a716b78-0-13d8-50022-7c471-ced942f-7c471
Call-ID: 6098cf74-4209-4c8f-af2a-40d75af11047@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-7c471-
1e575cf4-76893e21
Contact: <sip:880880@10.254.176.126:5080>
Content-Type: application/sdp
Content-Length: 216

v=0
o=880880 100025 100025 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 14030 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F3.
ACK sip:880880@10.254.176.126:5080 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=83b87bb0-b869-4c0b-b147-26c2beeaab9a
To: "Voice Mail"<sip:880880@ug1.scm.com>;tag=a716b78-0-13d8-50022-7c471-ced942f-7c471
Call-ID: 6098cf74-4209-4c8f-af2a-40d75af11047@ug1.scm.com
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7c472-1e575db8-2871b9e
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Length: 0
```

### 3.3.3 Recording Server

This chapter describes the SIP usages for 3rd Party Recording Server integration.

#### 3.3.3.1 SIP usage

**Outbound Call (SCM to Recording Server)**

- **'From' Header**
  Recording Request user's extension number and a host are contained in 'From' header.
  *From: 'Recording'*<sip:Extension number@host>
- **'To' Header /Request URI**
  Recording Server's extension number and a host are contained in 'To' header.
  *To: <sip:Recording Server's extension number@host>*
- **'Diversion' Header**
  If recording request call is incoming call, DID information is contained in 'Diversion' header.
  Otherwise, recording request call is local call or outgouing call, recording request user's extension number is contained in 'Diversion' header.
  If application server provides voice main/recording/AA/Coloring Service, reason is distinction point.
  *Diversion: <sip:DID Number or Extension Number@host>;reason='record'*
- **'CallInfo' Header**
  Recording target number or calling number are contained in 'CallInfo' header.
  Calling number is contained only external call.
  By using direction field, recording direction is distinguished.
  outgoing call case, field.
  If answered call is outgoing call, 'outgoing' is contained direction field.
  Otherwise answered call is incoming call, 'incoming' is contained direction field.
  *Call-Info: <sip:(Recording target number or calling number)@host>;purpose=record; direction=(outgoing or incoming)*

### 3.3.3.2 Call Flows

If a recording service is activated, the call for a recording server is made through SCM. The call is transferred to a conference server. Then the recording server can receive the mixed RTP data through the conference server.

Following item describes message and flow about between conference server and recording server.

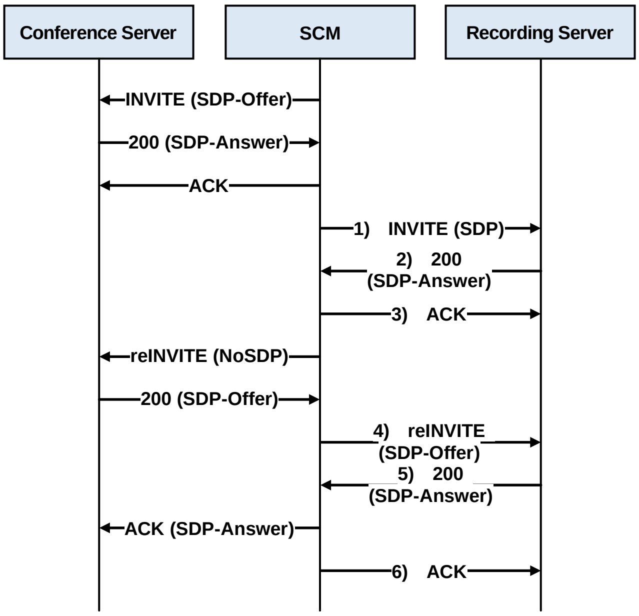



**Figure 3.41 Recording Flow**

1) When subscriber requests recording, SCM sends an INVITE message for conference server.
2) The Conference Server sends a 200 OK messages that is including the negotiated SDP capability.
3) SCM sends the ACK message to the conference server.
4) SCM sends the INVITE message to the recording server, contain conference server's SDP information.
5) The recording server sends a 200 OK message that is including SDP capability.
6) SCM sends the ACK message to the recording server.
7) SCM sends a NoSDP reINVITE message to the conference server.
8) When The conference server sends a 200 OK message that is including SDP capability, SCM sends a reINVITE message that is including received SDP Information to the recording server.
9) When The recording server sends a 200 OK Message that is including SDP capability, SCM sends a ACK Message that is including received SDP Information to the conference server.
10) SCM sends a ACK message to the recording server.

### 3.3.3.3 SIP Messages

#### Recording Message In local call

880880: Recording Server's Extension Number
2005: subscriber's extension number and mailbox number.

In the case below, recording during a call between 2003 extension and 2005 extension.

```
F1.
INVITE sip:880880@ug1.scm.com:5080 SIP/2.0
From: "Recording"<sip:2005@ug1.scm.com>;tag=65a37258-06cb-4215-8b13-e27c47d0193d
To: "Voice Mail"<sip:880880@ug1.scm.com>
Call-ID: 0d1aa3c3-0343-4200-9f1c-ca0f51f6152e@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7df69-1ec0b2bd-309eba65
P-Asserted-Identity: "Recording" <sip:2005@ug1.scm.com>
Diversion: <sip:2005@ug1.scm.com>;reason="call-record"
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 295

v=0
o=2005 100096 100096 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
a=sendrecv
m=audio 35999 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F2.
SIP/2.0 200 OK
From: "Recording"<sip:2005@ug1.scm.com>;tag=65a37258-06cb-4215-8b13-e27c47d0193d
To: "Voice Mail"<sip:880880@ug1.scm.com>;tag=a716b78-0-13d8-50022-7df69-70d440a-7df69
Call-ID: 0d1aa3c3-0343-4200-9f1c-ca0f51f6152e@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-7df69-
1ec0b2bd-309eba65
Contact: <sip:880880@10.254.176.126:5080>
Content-Type: application/sdp
Content-Length: 216

v=0
o=880880 100028 100028 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 14034 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F3.
ACK sip:880880@10.254.176.126:5080 SIP/2.0
From: "Recording"<sip:2005@ug1.scm.com>;tag=65a37258-06cb-4215-8b13-e27c47d0193d
To: "Voice Mail"<sip:880880@ug1.scm.com>;tag=a716b78-0-13d8-50022-7df69-70d440a-7df69
Call-ID: 0d1aa3c3-0343-4200-9f1c-ca0f51f6152e@ug1.scm.com
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7df69-1ec0b309-82de7b2
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Length: 0
```

```
F4.
INVITE sip:880880@10.254.176.126:5080 SIP/2.0
From: "Recording"<sip:2005@ug1.scm.com>;tag=65a37258-06cb-4215-8b13-e27c47d0193d
To: "Voice Mail"<sip:880880@ug1.scm.com>;tag=a716b78-0-13d8-50022-7df69-70d440a-7df69
Call-ID: 0d1aa3c3-0343-4200-9f1c-ca0f51f6152e@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7df69-1ec0b3ff-518cccf6
P-Asserted-Identity: <sip:890022@ug1.scm.com>
Subject: 2005
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:890022@10.254.176.126:5060>;isfocus
Content-Type: application/sdp
Content-Length: 268

v=0
o=890022 0 0 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
a=sendrecv
m=audio 45052 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F5.
SIP/2.0 200 OK
From: "Recording"<sip:2005@ug1.scm.com>;tag=65a37258-06cb-4215-8b13-e27c47d0193d
To: "Voice Mail"<sip:880880@ug1.scm.com>;tag=a716b78-0-13d8-50022-7df69-70d440a-7df69
Call-ID: 0d1aa3c3-0343-4200-9f1c-ca0f51f6152e@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-7df69-1ec0b3ff-518cccf6
Contact: <sip:880880@10.254.176.126:5080>
Content-Type: application/sdp
Content-Length: 216

v=0
o=880880 100029 100029 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 14034 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F6.
ACK sip:880880@10.254.176.126:5080 SIP/2.0
From: "Recording"<sip:2005@ug1.scm.com>;tag=65a37258-06cb-4215-8b13-e27c47d0193d
To: "Voice Mail"<sip:880880@ug1.scm.com>;tag=a716b78-0-13d8-50022-7df69-70d440a-7df69
Call-ID: 0d1aa3c3-0343-4200-9f1c-ca0f51f6152e@ug1.scm.com
CSeq: 2 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7df6b-1ec0bad3-4d445f33
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:880880@10.254.176.126:5060>;isfocus
Content-Length: 0
```

### Recording Message In outbound call

880880: Recording Server’s Extension Number
2005: Subscriber’s extension number and mailbox number.
0312792005: DID number
01095302001: CID number

In the case below, recording during a call between trunk and 2005 extension.
In outbound call case, except 1.INVITE message, message flow is identical in local call case.

```
F1.
INVITE sip:880880@ug1.scm.com:5080 SIP/2.0
From: "Recording"<sip:2005@ug1.scm.com>;tag=ef314d64-7a3a-46b3-b89f-212ad481ca2b
To: "Voice Mail"<sip:880880@ug1.scm.com>
Call-ID: 36d6073e-be81-4ae1-b051-b7482d4b8bb6@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7e307-1eced656-17484231
Diversion: <sip:0312792005@ug1.scm.com>;reason="call-record"
Call-Info: <sip:01095302001@ug1.scm.com>;purpose=record;direction=incoming;
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 295

v=0
o=2005 100101 100101 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
a=sendrecv
m=audio 35999 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

### Recording Message In Trunk Call after Transfer

8888: Recording Server's Extension Number
2000: The extension and mailbox number of the user who tries a recording service.
2000: Transferer Number
0312792001: DID number
0109530xxxx: CID number

In outbound call case, except Diversion Header, message flow is identical in Trunk call case.
Diversion Header contains the recording information and the transfer information.

```
1. INVITE
INVITE sip:8888@ug1.scm.com:5080 SIP/2.0
From: "Recording"<sip:2001@ug1.scm.com>;tag=5223cd38-5a5c-4456-aaef-e68551739e8c
To: "Voice Mail"<sip:8888@ug1.scm.com>
Call-ID: e097e1e2-80ea-4cd5-a682-9b828c5996e6@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.168.85:5060;rport;branch=z9hG4bK-1bc48-6c77cdb-529caca
Diversion: <sip:0312792001@ug1.scm.com>;reason="call-record"
           ,<sip:2000@ug1.scm.com>;reason="unconditional"
Call-Info: <sip:0109530xxxx@ug1.scm.com>;purpose=record;direction=incoming;
Max-Forwards: 70
Contact: <sip:2001@10.254.168.85:5060>
Content-Type: application/sdp
Content-Length: 187

v=0
o=2001 100367 100367 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
a=sendrecv
m=audio 44498 RTP/AVP 0 8 18
a=ptime:20
a=ptime:20
a=ptime:20
a=sendrecv
```

### 3.3.4 IVR/Auto Attendant Server

This chapter describes the SIP usages for AA/IVR Server integration.
In case an application server is registered as a subscriber, it should be controlled through CTI. If SCM receives CTI commands, the 'NOTIFY' including specific event headers is sent to an application server. In that case, the operation of the application server should be different. For more detailed, refer to below.

#### 3.3.4.1 SIP usage

**Outbound Call (SCM to IVR Server)**

- **'From' Header**
  An extension number and a host are contained in 'From' header
  *From: <sip:Extension Number@host>*
- **'To' Header/Request URI**
  An IVR (AA)'s access number and a host are contained in 'To' header
  *To: <sip:Access Number@host>*
- **'Diversion' Header**
  A DID number and a host are included in 'Diversion' header. If one application server provides several services such as Voice Mail/Recording/AA, SCM can distinguish by 'reason' of 'Diversion' header.
  *Diversion: <sip::DID Number @host>;reason='direct-incoming-call'*
- **'Event' Header of NOTIFY Message**
  If an IVR server is registered as a subscriber, it should be controlled through CTI.
  An IVR sever should know specific headers of 'NOTIFY' message and operate as shown below.
  - *Event: hold*
    If an IVR server receives the 'NOTIFY' message including above 'Event', the IVR server should send 'REINVITE' message to the call on conversion. The call will be held.
  - *Event: talk*
    If an IVR server receives the 'NOTIFY' message including above 'Event', the IVR server should send '200' message to the call on ringing.
  - *Event: dial-req*
    If an IVR server receives the 'NOTIFY' message including above 'Event', the IVR server should dial through the designated number referred in 'Body' of the message.

## 3.3.4.2 Call Flows

### AA Basic Call

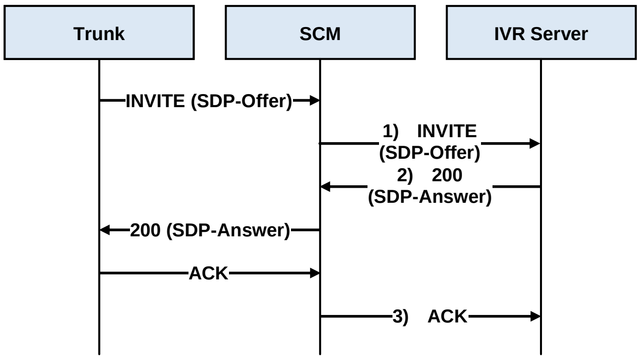



**Figure 3.42 AA Basic Call Flow**

1) SCM receives an incoming call.
2) SCM sends an 'INVITE' message to an IVR server by DID settings

### AA Call Transfer

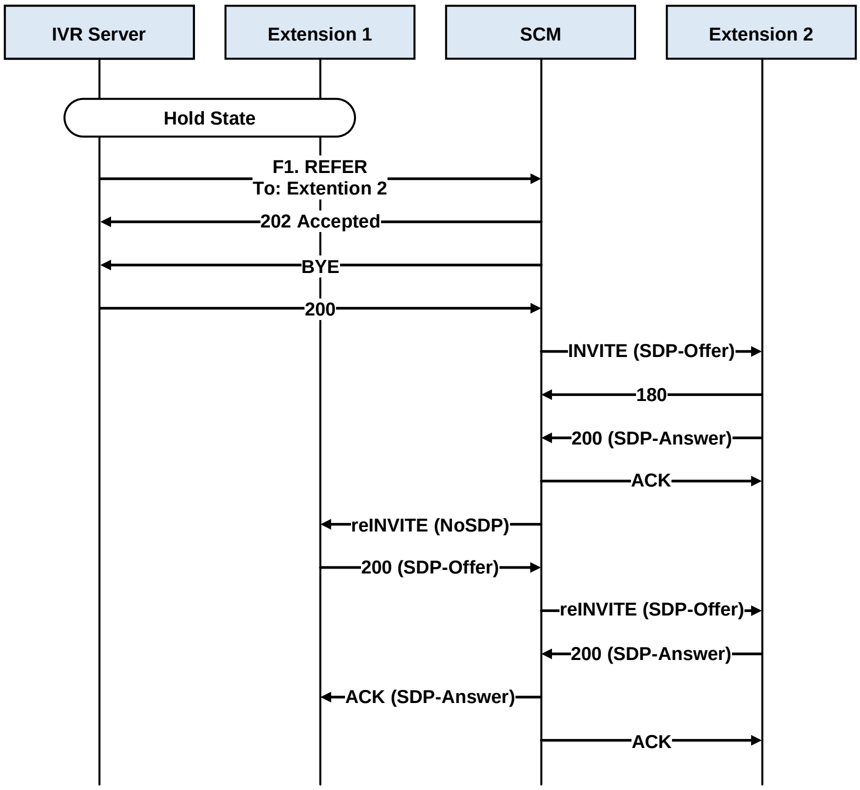



**Figure 3.43 AA Call Transfer Flow**

An IVR server transfers the ‘Extension1’ on conversation to the ‘Extension2’.
The ‘Extension1’ is on conversion with an IVR server.

1) An IVR server sends an ‘REINVITE’ and a ‘REFER’ (to: Extension2) to SCM.
2) If SCM receives the ‘REFER’ message, SCM sends a ‘BYE’ message to an IVR server. ‘Extension1’ is connected to ‘Extension2’.

## IVR CTI DIAL/HOLD

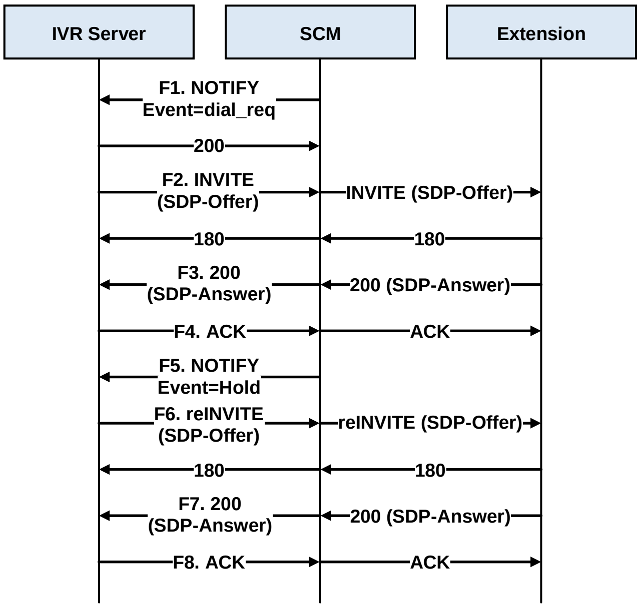



**Figure 3.44 IVR CTI DIAL/HOLD Flow**

If SCM receives ‘Dial’ request through CTI, SCM sends the ‘NOTIFY’ (DIAL_REQ) message to an IVR server. If the IVR server has usable channels, the server should make a call by translating Event headers and XML’s Body.

If SCM receives ‘HOLD’ request through CTI, SCM sends the ‘NOTIFY’ (HOLD) message to an IVR server. The IVR server should send the ‘REINVITE’ message to the call on conversation. The call will be held.

## IVR CTI ANSWER

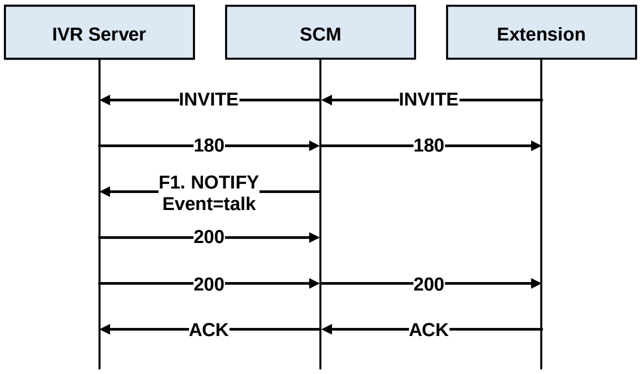



**Figure 3.45 IVR CTI ANSWER Flow**

If SCM receives ‘Answer’ request through CTI, SCM sends the ‘NOTIFY’ (talk) message to an IVR server. The IVR server should send ‘200’ message to the ringing call. The call will be answered.

#### 3.3.4.3 SIP Messages

**AA Basic Call Messages**

882882: AA Server Access Number
01095302001: Outgoing Call Number
0312005555: DID Number

```
F1.
INVITE sip:882882@ug1.scm.com:5060 SIP/2.0
From: "2001"<sip:01095302001@ug1.scm.com>;tag=b2cbcbd3-7899-4721-80ac-e5427f76d6c8
To: "Voice Mail"<sip:882882@ug1.scm.com>
Call-ID: 55113152-166b-4bcf-b731-5fe1f6c490f1@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-8e11d-22af5a5c-2cfe7bca
P-Asserted-Identity: "2001" <sip:01095302001@ug1.scm.com>
Diversion: <sip:0312005555@ug1.scm.com>;reason="direct-incoming-call"
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Max-Forwards: 70
Contact: <sip:01095302001@10.254.176.126:5060>
User-Agent: SAMSUNG SCM 3.2.400
Content-Type: application/sdp
Content-Length: 507

v=0
o=01095302001 1340898898 0 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 165.213.89.183
t=0 0
a=sendrecv
m=audio 20014 RTP/AVP 0 8 18 9 102
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=sendrecv
m=video 25014 RTP/AVP 34 98 99
a=rtpmap:34 H263/90000
a=fmtp:34 CIF=2;QCIF=2
a=rtpmap:98 MP4V-ES/90000
a=fmtp:98 profile-level-id=2
a=rtpmap:99 H264/90000
a=fmtp:99 profile-level-id=42800C;packetization-mode=0
a=sendrecv
```

```
F2.
SIP/2.0 200 OK
From: "2001"<sip:01095302001@ug1.scm.com>;tag=3215936c-35d7-474e-a272-d1c8ccbb1376
To: "Voice Mail"<sip:882882@ug1.scm.com>;tag=a704218-0-13d8-50022-8e086-71a14ef1-8e086
Call-ID: 55113152-166b-4bcf-b731-5fe1f6c490f1@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-8e11d-22af5a5c-2cfe7bca
Contact: <sip:882882@10.254.176.126:5080>
Content-Type: application/sdp
Content-Length: 160

v=0
o=882882100032 100032 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 14038 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendrecv
```

```
F3.
ACK sip:882882@10.254.176.126:5080 SIP/2.0
From: "2001"<sip:01095302001@ug1.scm.com>;tag=3215936c-35d7-474e-a272-d1c8ccbb1376
To: "Voice Mail"<sip:882882@ug1.scm.com>;tag=a704218-0-13d8-50022-8e086-71a14ef1-8e086
Call-ID: 55113152-166b-4bcf-b731-5fe1f6c490f1@ug1.scm.com
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-8e087-22ad103a-739d76ea
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:01095302001@10.254.176.126:5060>
Content-Length: 0
```

## AA Call Transfer Messages

Operator Number: 2005

An operator ‘2005’ is on conversation with an extension ‘2001’. An operator ‘2005’ sends ‘REFER’ message. An extension ‘2001’ is transferred to an extension ‘2003’.

```
F1.
REFER sip:2005@10.254.176.126:5060 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=4e8be8-55b0d5a5-13c4-50022-1a8236-797ca16e-1a8236
To: "2001"<sip:01095302001@ug1.scm.com>;tag=791c68cd-385f-4a1f-aeaa-1518eae718d3
Call-ID: 0343eee1-4190-47d5-aa18-6d9f7b220142@ug1.scm.com
CSeq: 2 REFER
Via: SIP/2.0/UDP 165.213.176.85:5060;rport=5060;branch=z9hG4bK-1a823d-678cbe5c-50f98088
Refer-To: <sip:2003@ug1.scm.com>
Referred-By: <sip:2005@ug1.scm.com>
Max-Forwards: 70
Supported: timer,replaces
Contact: <sip:2005@165.213.176.85>
Content-Length: 0
```

## IVR CTI DIAL/HOLD Messages

IVR Server: 2010

An IVR server sends a 'NOTIFY' (DialReq) message. An extension 2005 makes a call.

```
F1.
NOTIFY sip:2010@165.213.89.182 SIP/2.0
From: <sip:2010@ug1.scm.com>;tag=db5ab088-1fde-48dd-93f1-5cc2108465d4
To: <sip:2010@ug1.scm.com>
Call-ID: 00722a3b-3747-41af-a13a-f62878c8a4a6
CSeq: 98 NOTIFY
Via: SIP/2.0/UDP 165.213.89.65:5060;rport;branch=z9hG4bK-e4784-37c75f50-225a884f
Event: dial-req
Contact: <sip:2010@165.213.89.65:5060>
Max-Forwards: 70
Content-Type: application/vnd.sec-call+xml
Content-Length: 100

<?xml version="1.0"?>
<CallService>
<dial_req call_type="normal" number="2005"/>
</CallService>
```

```
F2.
INVITE sip:2005@ug1.scm.com SIP/2.0
From:       "2010"<sip:2010@ug1.scm.com>;tag=4ca0e0-b659d5a5-13c4-50022-5462b-674ee4d0-
5462b
To: <sip:2005@ug1.scm.com>
Call-ID: 4d33f8-b659d5a5-13c4-50022-5462b-6298d10a-5462b
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.89.182:5060;rport;branch=z9hG4bK-5462b-149a1910-10f53319
Max-Forwards: 70
Supported: timer,replaces
P-Preferred-Identity: "2010"<sip:2010@ug1.scm.com>
Contact: <sip:2010@165.213.89.182>
Session-Expires: 1800;refresher=uas
Min-SE: 90
Allow: INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISH
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/T01.36
Content-Type: application/sdp
Content-Length: 342

v=0
o=SAMSUNG_IP_PHONE 1327939102 0 IN IP4 165.213.89.182
s=SAMSUNG_VOIP_CALL
c=IN IP4 165.213.89.182
t=0 0
m=audio 20100 RTP/AVP 8 0 18 9 102 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

```
F3.
SIP/2.0 200 OK
From:     "2010"<sip:2010@ug1.scm.com>;tag=4ca0e0-b659d5a5-13c4-50022-5462b-674ee4d0-5462b
To: <sip:2005@ug1.scm.com>;tag=756cdd81-5c27-421a-9845-c7766adcf065
Call-ID: 4d33f8-b659d5a5-13c4-50022-5462b-6298d10a-5462b
CSeq: 1 INVITE
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Via: SIP/2.0/UDP 165.213.89.182:5060;rport=5060;branch=z9hG4bK-5462b-149a1910-10f53319
Contact: <sip:2005@165.213.89.65:5060>
Content-Type: application/sdp
Content-Length: 270

v=0
o=2005 1860740927 1090213961 IN IP4 165.213.89.65
s=Samsung IP PBX
c=IN IP4 165.213.66.130
t=0 0
m=audio 3000 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F4.
ACK sip:2005@165.213.89.65:5060 SIP/2.0
From:     "2010"<sip:2010@ug1.scm.com>;tag=4ca0e0-b659d5a5-13c4-50022-5462b-674ee4d0-5462b
To: <sip:2005@ug1.scm.com>;tag=756cdd81-5c27-421a-9845-c7766adcf065
Call-ID: 4d33f8-b659d5a5-13c4-50022-5462b-6298d10a-5462b
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.89.182:5060;rport;branch=z9hG4bK-5462c-149a1f64-2255ec92
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISH
Contact: <sip:2010@165.213.89.182>
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/T01.36
Content-Length: 0
```

```
F5.
ACK sip:2005@165.213.89.65:5060 SIP/2.0
From:       "2010"<sip:2010@ug1.scm.com>;tag=4ca0e0-b659d5a5-13c4-50022-5462b-674ee4d0-5462b
To: <sip:2005@ug1.scm.com>;tag=756cdd81-5c27-421a-9845-c7766adcf065
Call-ID: 4d33f8-b659d5a5-13c4-50022-5462b-6298d10a-5462b
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.89.182:5060;rport;branch=z9hG4bK-5462c-149a1f64-2255ec92
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,INFO,UPDATE,REFER,SUBSCRIBE,NOTIFY,PUBLISH
Contact: <sip:2010@165.213.89.182>
Allow-Events: talk,hold
User-Agent: SAMSUNG-SMT-i5243/T01.36
Content-Length: 0
```

```
F6.
INVITE sip:2010@165.213.89.65:5060 SIP/2.0
Max-Forwards: 70
Content-Length: 265
Via: SIP/2.0/UDP 165.213.66.130:5060;branch=z9hG4bK150252c9f;rport
Call-ID: 6e96a749-162e-4212-bca9-c4f12a4e2cb1@ug1.scm.com
From: "2005" <sip:2005@ug1.scm.com>;tag=686ac946b4627fa
To: "2010" <sip:2010@ug1.scm.com>;tag=105f1a2e-8a8a-409d-a158-9d318e50240f
CSeq: 1112341080 INVITE
Supported: timer,replaces
Allow-Events: talk,hold,conference,check-sync
Content-Type: application/sdp
Contact: <sip:2005@165.213.66.130:5060;transport=udp>
User-Agent: Stonehenge IP270P 1.30.285

v=0
o=2005 1860740927 1860740928 IN IP4 165.213.66.130
s=SIP Call
c=IN IP4 165.213.66.130
t=0 0
m=audio 3000 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendonly
```

```
F7.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=686ac946b4627fa
To: "2010"<sip:2010@ug1.scm.com>;tag=105f1a2e-8a8a-409d-a158-9d318e50240f
Call-ID: 6e96a749-162e-4212-bca9-c4f12a4e2cb1@ug1.scm.com
CSeq: 1112341080 INVITE
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
P-Asserted-Identity: "2010" <sip:2010@ug1.scm.com>
Via: SIP/2.0/UDP 165.213.66.130:5060;rport=5060;branch=z9hG4bK150252c9f
Contact: <sip:2010@165.213.89.65:5060>
Content-Type: application/sdp
Content-Length: 168

v=0
o=2010 1327939102 1 IN IP4 165.213.89.65
s=Samsung IP PBX
c=IN IP4 165.213.89.182
t=0 0
m=audio 20102 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
a=recvonly
```

```
F8.
ACK sip:2010@165.213.89.65:5060 SIP/2.0
Max-Forwards: 70
Content-Length: 0
Via: SIP/2.0/UDP 165.213.66.130:5060;branch=z9hG4bK38c1cf9e2;rport
Call-ID: 6e96a749-162e-4212-bca9-c4f12a4e2cb1@ug1.scm.com
From: "2005" <sip:2005@ug1.scm.com>;tag=686ac946b4627fa
To: "2010" <sip:2010@ug1.scm.com>;tag=105f1a2e-8a8a-409d-a158-9d318e50240f
CSeq: 1112341080 ACK
User-Agent: Stonehenge IP270P 1.30.285
```

## IVR CTI ANSWER Messages

```
F1.
NOTIFY sip:2002@165.213.89.231 SIP/2.0
From: "2001"<sip:2001@10.254.176.178>;tag=c0b1bb0c-c677-46fd-86af-5d0d0530935e
To: "2002"<sip:2002@10.254.176.178>;tag=43f5e0-e759d5a5-13c4-50022-1ba6e3-651d0492-
1ba6e3
Call-ID: f2243b7e-854a-4359-9bef-f701106f0b89@10.254.176.178
CSeq: 2 NOTIFY
Via: SIP/2.0/UDP 10.254.176.178:5060;rport;branch=z9hG4bK-d3a74-33ac5753-4c15ac4
Event: talk
Max-Forwards: 70
Contact: <sip:2001@10.254.176.178:5060>
Content-Length: 0
```

## 3.3.5 Paging Conference Server

SCM supports several types of conferences. (Refer to 3.3.5~3.3.8 chapters). In order to distinguish conference types, SCM refers to the specific headers and features codes included in INVITE messages.

An application server should know below information to support the paging service.

- [Station Paging] feature code
- [Paging Group] Number
- [Paging Member]

In case of a Paging On Answer service, the corresponding information which referred above is needed. If the member of Paging On Answer group should answer automatically, the answer-after contained in 'Call-Info' header should be set '0'.

### 3.3.5.1 SIP usage

#### Outbound Call (SCM to Paging Server)

- **'From' Header**
  A paging master number and a host are contained in 'From' header.
  *From: <sip:Paging Master Number@host>*
- **'To' Header/Request URI**
  The paging feature code (*55)+ paging group number (5000) and a host are included in 'To' header and Request URI.
  *To: <sip:*555000@host>*
- **'Reason' Header**
  SCM controls the lcd display for a paging master and members.
  The lcd of a master will display 'PAGE', if the below reason header is contained in 200 OK message from a paging server.
  *Reason: SIP ;cause=200 ;text='PAGE'*
- **'Contact' Header**
  A paging server sends 200 OK message containing a conference channel. SCM concludes that it is a conference call through 'isfocus'.
  *Contact: <sip:890009@10.254.176.126:5090>;isfocus*

### Inbound Call (Paging Server to SCM)

- **‘From’ Header**
  The extension number and a host of the master who requests paging service are contained in ‘From’ header.
  *From: <sip:Master Extension Number@host>*
- **‘To’ Header/Request URI**
  The extension number and a host of the paging member are included in ‘To’ header and Request URI.
  *To: <sip:Member Extension Number@host>*
- **‘CallInfo’ Header**
  In case a paging server calls members, it is used. If paging members should answer automatically, ‘answer-after’ of ‘CallInfo’ header should be set to ‘0’. SCM can distinguish conference types and control member’s LCD display through ‘purpose’.
  *Call-Info: <http://127.0.0.1>;answer-after=0;purpose=page*
- **’Reason’ Header**
  SCM controls the lcd display of master and members. In case paging server calls paging members and an INVITE message includes the below ‘Reason’ header, ‘PAGE’ will be displayed on the paging member’s phone.
  *Reason: SIP ;cause=200 ;text=‘PAGE’*
- **‘Contact’ Header**
  A conference channel number is contained in ‘Contact’ header. SCM concludes that a conference call is made by ‘isfocus’.
  *Contact: <sip:890009@10.254.176.126:5090>;isfocus*

### 3.3.5.2 Call Flows

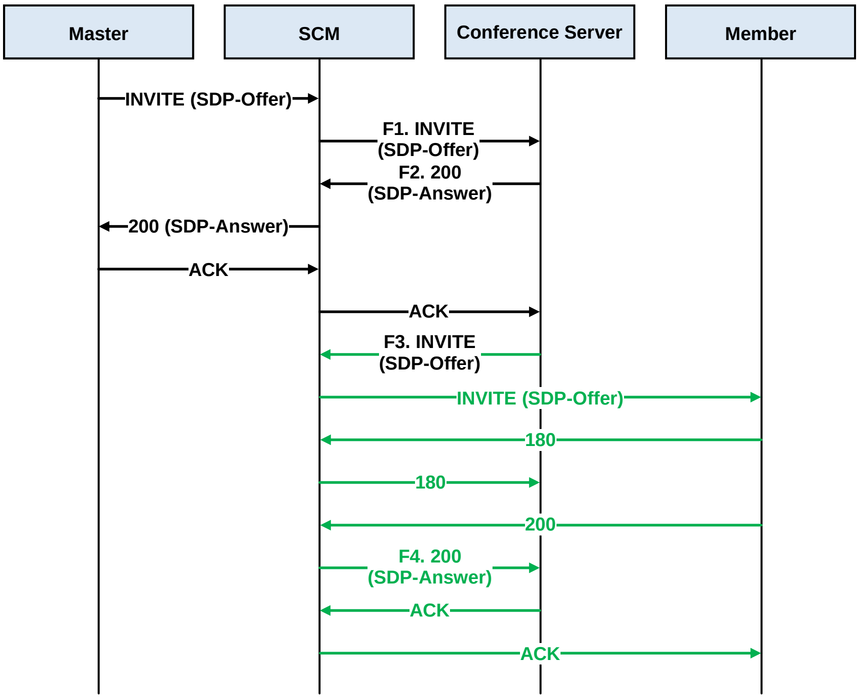



**Figure 3.46 Paging Server Call Flows**

A green arrow means that an application server calls members.

1) Master dials [Station Paging] feature code and paging group number. (The feature code can be omitted).
2) SCM tries to find a paging server through the 'INVITE' message and the master's information. And SCM relays the 'INVITE' message to the paging server.
3) A paging server sends '200 OK' messages and finds paging members through the paging group number. A paging server makes a call to paging members.

#### 3.3.5.3 SIP Messages

*55: [Station Paging] feature code
6000: [Station Paging] group number
2005: Paging Group Master
2004: Paging Group Member

Member 2005 calls paging group by dialing 6000.

```
F1
INVITE sip:*556000@ug1.scm.com:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=aa8458b3-9cee-432f-834e-4eadbb820d77
To: <sip:*556000@ug1.scm.com>
Call-ID: a30db9c0-4947-4350-a32e-ac7444f0f2e0@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-1718-5a37a7-3751e81f
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Max-Forwards: 70
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Contact: <sip:2005@10.254.176.126:5060>
User-Agent: SAMSUNG SCM 3.2.400
Content-Type: application/sdp
Content-Length: 339

v=0
o=2005 1340322894 0 IN IP4 165.213.176.85
s=Samsung IP PBX
c=IN IP4 165.213.176.85
t=0 0
a=sendrecv
m=audio 20048 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F2
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=aa8458b3-9cee-432f-834e-4eadbb820d77
To: <sip:*556000@ug1.scm.com>;tag=b4db7798-0-13e2-50022-1718-201ebc4e-1718
Call-ID: a30db9c0-4947-4350-a32e-ac7444f0f2e0@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-1718-
5a37a7-3751e81f
Contact: <sip:890009@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="PAGE"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890009@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45016 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F3
INVITE sip:2004@ug1.scm.com:5060 SIP/2.0
From:"2005"<sip:890009@ug1.scm.com:5090>;tag=b4db7c60-0-13e2-50022-1718-59c43d80-1718
To: <sip:2004@ug1.scm.com:5060>
Call-ID: b5141298-0-13e2-50022-1718-41c0feba-1718
CSeq: 1 INVITE
Via: SIP/2.0/UDP 192.168.113.50:5090;received=127.0.0.1;branch=z9hG4bK-1718-5a37d1-
56119609
Alert-Info: <http://127.0.0.1>;Bellcore-dr1
Call-Info: <http://127.0.0.1>;answer-after=0;purpose=page
Max-Forwards: 70
Reason: SIP ;cause=200 ;text="PAGE"
Contact: <sip:890009@10.254.176.126:5090>;isfocus
Content-Type: application/sdp
Content-Length: 284

v=0
o=890009@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45016 RTP/AVP 8 0 18 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F4
SIP/2.0 200 OK
From:"2005"<sip:890009@ug1.scm.com:5090>;tag=b4db7c60-0-13e2-50022-1718-59c43d80-1718
To: <sip:2004@ug1.scm.com:5060>;tag=5aa73593-6b1e-408a-b8b3-1902a8a8892f
Call-ID: b5141298-0-13e2-50022-1718-41c0feba-1718
CSeq: 1 INVITE
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
P-Asserted-Identity: "2004" <sip:2004@ug1.scm.com>
User-Agent: SAMSUNG SCM 3.2.400
Via: SIP/2.0/UDP 192.168.113.50:5090;received=127.0.0.1;branch=z9hG4bK-1718-5a37d1-56119609
Contact: <sip:2004@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 224

v=0
o=2004 1168433292 1 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 165.213.80.94
t=0 0
m=audio 20232 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

## 3.3.6 Meet-me Conference Server

SCM is providing several type of conference service (refer to 3.3.5~3.3.8). Conference Server is checking feature code in INVITE message and some specific Header information to distinguish the conference services.

To use Meet-me Conference service, reserving conference is required and server should recognize the feature code for [Meet Me Conference Join] to distinguish conference type.

### 3.3.6.1 SIP usage

#### Outbound Call (SCM to Meet-me Server)

- **'From' Header**
  Information of user and host who request Meet-me conference is included.
  *From: <sip:user who request Meet-me confernece @host>*
- **'To' Header /Request URI**
  Feature code for [Meet Me Conference Join] is included.
  But if feature code for [Meet Me Conference Join] and Meet-me conference ID are included to 'To' URI and Request URI, user can join the specific conference.
  *To: <sip:feature code for Meet-me conference @host>*
- **'Reason' Header**
  Screen display for master and members can be managed by this field.
  If Reason text is included in 200 OK which server response, then it will be displayed on master's screen. For example, if below text is included in 200 OK, then 'Conference' will be displayed on master's screen.
  *Reason: SIP ;cause=200 ;text='Conference'*
- **'Contact' Header**
  A conference channel is contained in 'Contact' header. SCM knows whether the call is on conference or not through 'isfocus' of 'Contact' header.

### 3.3.6.2 Call Flows

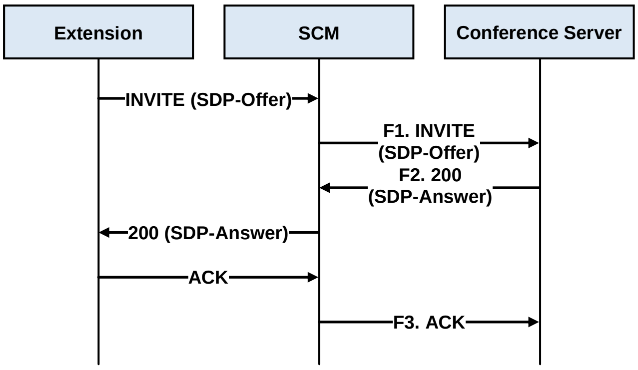



**Figure 3.47 Meet-me Conference Flow**

1) User makes a call by using feature code for [Meet Me Conference join].
2) Conference Server receive and request Meet-me conference ID.
3) If user can press the Meet-me conference ID during the call, then Conference server will recognize the DTMF tone and make him/her join to corresponding conference.

### 3.3.6.3 SIP Messages

*82: feature code for Meet-me conference
2005: member who attends Meet-me conference
890002: Conference Room channel

Member 2005 calls Meet-me conference by dialing *82

```
F1
INVITE sip:*82@ug1.scm.com:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=67f8bf72-68a6-4b8c-b632-3b7fbc796b31
To: <sip:*82@ug1.scm.com>
Call-ID: 677eadf9-1bf9-471f-86d4-6522f2b43e3e@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-137ce-4c1ff32-5e84a44f
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Max-Forwards: 70
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Contact: <sip:2005@10.254.176.126:5060>
User-Agent: SAMSUNG SCM 3.2.400
Content-Type: application/sdp
Content-Length: 339

v=0
o=2005 1340396809 0 IN IP4 165.213.176.85
s=Samsung IP PBX
c=IN IP4 165.213.176.85
t=0 0
a=sendrecv
m=audio 20044 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F2
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=67f8bf72-68a6-4b8c-b632-3b7fbc796b31
To: <sip:*82@ug1.scm.com>;tag=b4fa0a80-0-13e2-50022-137ce-5f15bc-137ce
Call-ID: 677eadf9-1bf9-471f-86d4-6522f2b43e3e@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-137ce-4c1ff32-5e84a44f
Contact: <sip:890002@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 194

v=0
o=890002@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 44000/2 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F3
ACK sip:890002@10.254.176.126:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=67f8bf72-68a6-4b8c-b632-3b7fbc796b31
To: <sip:*82@ug1.scm.com>;tag=b4fa0a80-0-13e2-50022-137ce-5f15bc-137ce
Call-ID: 677eadf9-1bf9-471f-86d4-6522f2b43e3e@ug1.scm.com
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-137cf-4c201d2-ac62224
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Length: 0
```

### 3.3.7 One-Step Conference Server

SCM is providing several type of conference service (refer to 3.3.5~3.3.8). Conference Server is checking feature code in INVITE message and some specific Header information to distinguish the conference services.

One-Step Conference means conference server calls multiple members to conference. There are 3 type of conference.

- Predefined Conference
- Progressive Conference
- Dispatch Conference

Conference server should recognize the feature code which is included in 'To' URI and Request URI and distinguish above 3 type of conference service.

In case of Predefined Conference, Conference server should have group ID and group member information. Conference master can dial the feature code for [Predefined Conference] + group ID to make a Predefined Conference or master can just dial feature code for [Predefined Conference] firstly and after connecting the call he can dial group ID. If conference server receives this call, then conference server will call the group members.

In case of Progressive Conference, Conference server will get the member list by detecting DTMF which master dial after connecting the call from master. And then conference master will call the corresponding members.

In case of Dispatch Conference, SCM gets the conference request through CSTA.
SCM sends message to master to make a call to conference server directly. If SCM receives corresponding response then sends NOTIFY message to send conference member list to conference server. Conference server will call the members which he receives.

There is a conference master for One-step conference. When conference server receives BYE to terminate the call for the master, then it should send BYE to other members to terminate the conference.

### 3.3.7.1 SIP usage

#### Outbound Call (SCM to One-Step Conference Server)

- **'From' Header**
  Information of user and host who request One-step conference is included.
  *From: <sip:user who request One-step conference@host>*
- **'To' Header/Request URI**
  Feature code for One-step conference ([Predefined Conference]/[Progressive Conference]/[Dispatch Conference]) is included. If there is group ID, it can be added after feature code.
  *To: <sip:feature code for One-step conference @host>*
- **'Reason' Header**
  Screen display for master and members can be managed by this field.
  If Reason text is included in 200 OK which server response, then it will be displayed on master's screen. For example, if below text is included in 200 OK, then 'Conference' will be displayed on master's screen.
  *Reason: SIP ;cause=200 ;text='Conference'*
- **'Contact' Header**
  Conference server is including the conference channel in 200 OK message. SCM can conclude conference call by checking isfocus.

#### Inbound Call (One-Step Conference Server to SCM)

- **'From' Header**
  Information of master and host who request conference service is included.
  *From: <sip:master number who request conference@host>*
- **'To' Header/Request URI**
  Information of member number and host is contained in 'To' Header/Request URI.
  *To: <sip: group member number @host>*
- **'CallInfo' Header**
  'CallInfo' header is used when conference server calls conference member
  If you want to make conference member to do Auto Answer, answer-after value set as 0 and purpose value should be added for SCM to recognize which type of service conference server made.
  – Call-Info: <http://127.0.0.1>;answer-after=0;purpose=dispatch-conf
  – Call-Info: <http://127.0.0.1>;purpose=predefined-conf
- **'Reason' Header**
  Screen display for master and members can be managed by this field.
  If Reason text is included in INVITE message which conference server sends to members, then it will be displayed on member's screen. For example, if below text is included in INVITE, then 'Conference' will be displayed on member's screen.
- **'Contact' Header**
  A conference channel is contained in 'Contact' header of INVITE message which is from conference server to call the members. SCM knows whether the call is on conference or not through 'isfocus' of 'Contact' header.

### 3.3.7.2 Call Flows

#### Predefined Conference/Progressive Conference

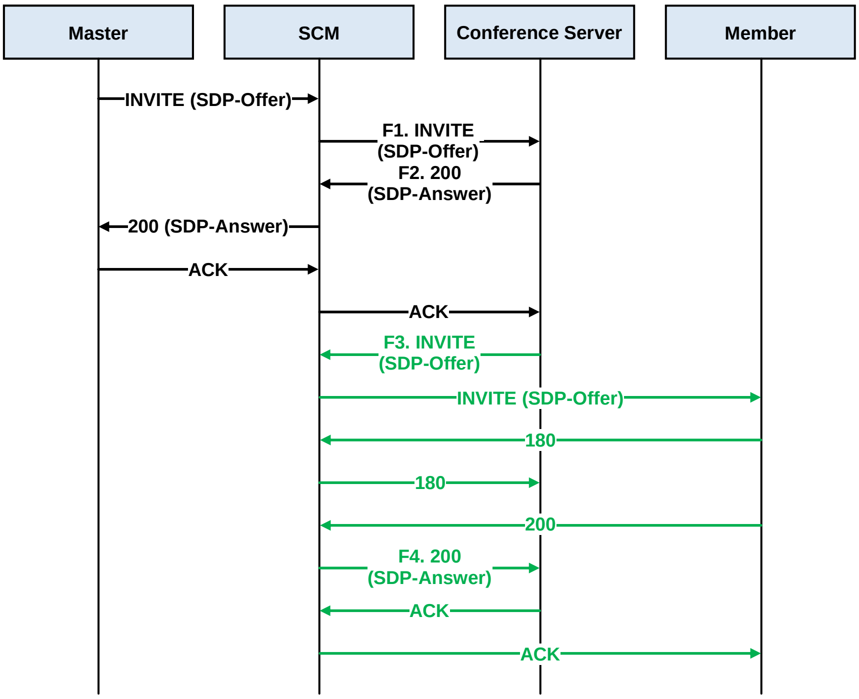



**Figure 3.48 Conference/Progressive Conference Flow**

Green arrow means call flow how conference server calls member.

1) Mater calls [One-step conference] feature code. In case of Predefined conference or Dispatch conference, group number could be included.
2) SCM search the conference server based on received INVITE message and Master information. SCM send the same INVITE message to conference server.
3) Conference server answers with 200 OK and then search and call the conference group members.

## Dispatch Conference

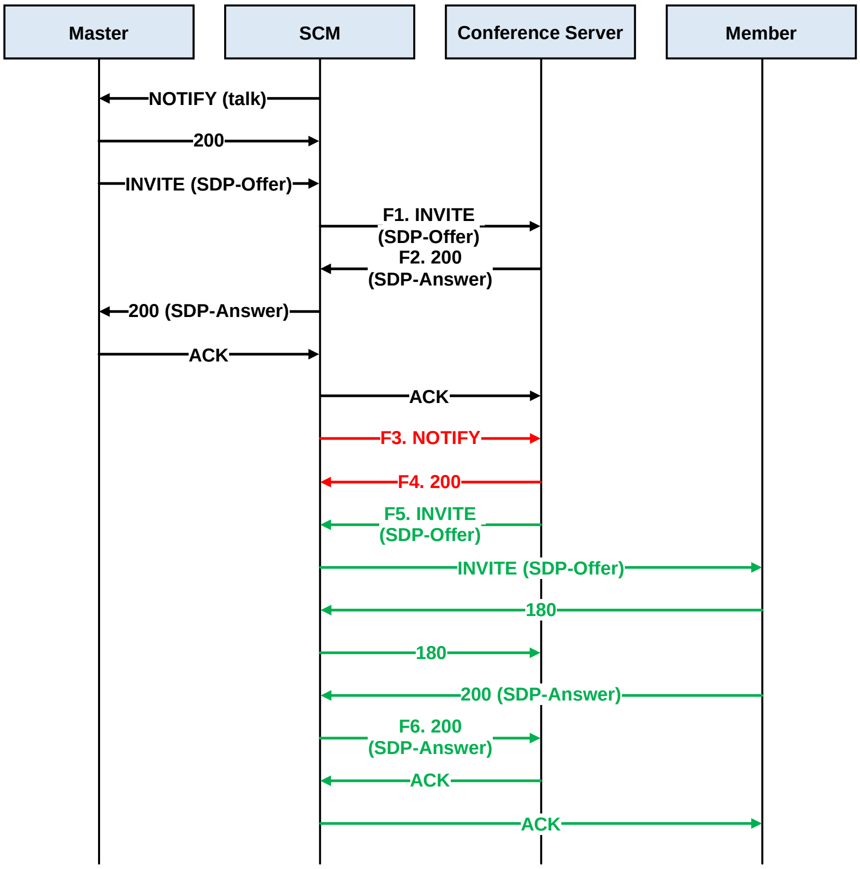



**Figure 3.49 Dispatch Conference Flow**

1) SCM receives Dispatch conference request message through CSTA message.
2) SCM sends NOTIFY message to Master to make a call to conference server.
3) When Master receives NOTIFY from SCM, he makes a call to conference server.
4) If SCM receives 200 OK as a response from Dispatch conference, then he will send NOTIFY message to conference server. This NOTIFY message will contain the Dispatch conference member list. (F3)
5) When conference server receives NOTIFY, he will make calls to corresponding members. (F4)

### 3.3.7.3 SIP Messages

#### Predefined Conference Messages

*11: Predefined Conference feather code
1000: group conference ID
2005: Predefined Conference master
2003: Predefined Conference member

```
F1.
INVITE sip:*111000@ug1.scm.com:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=19926e43-0c70-4037-a69d-830f7526cc4e
To: <sip:*111000@ug1.scm.com>
Call-ID: 63a82819-c9d1-4346-bb60-910a2a466496@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7e978-1ee7ff4e-5a60e864
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Max-Forwards: 70
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Contact: <sip:2005@10.254.176.126:5060>
User-Agent: SAMSUNG SCM 3.2.400
Content-Type: application/sdp
Content-Length: 339

v=0
o=2005 1340835534 0 IN IP4 165.213.176.85
s=Samsung IP PBX
c=IN IP4 165.213.176.85
t=0 0
a=sendrecv
m=audio 20174 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F2.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=19926e43-0c70-4037-a69d-830f7526cc4e
To: <sip:*111000@ug1.scm.com>;tag=b50296b8-0-13e2-50022-7e978-350f509-7e978
Call-ID: 63a82819-c9d1-4346-bb60-910a2a466496@ug1.scm.com
CSeq: 1 INVITE
Via:  SIP/2.0/UDP  10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-7e978-1ee7ff4e-5a60e864
Contact: <sip:890024@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 194

v=0
o=890024@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 44000/2 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F3.
INVITE sip:2003@ug1.scm.com:5060 SIP/2.0
From: "2005"<sip:890024@ug1.scm.com:5090>;tag=b5029b80-0-13e2-50022-7e979-23f0e0bf-7e979
To: <sip:2003@ug1.scm.com:5060>
Call-ID: b4eeaf50-0-13e2-50022-7e979-541f9c09-7e979
CSeq: 1 INVITE
Via: SIP/2.0/UDP 192.168.113.50:5090;received=127.0.0.1;branch=z9hG4bK-7e979-1ee802dd-468a49dc
Call-Info: <http://127.0.0.1>;purpose=predefined-conf
Max-Forwards: 70
Reason: SIP ;cause=200 ;text="Conference"
Contact: <sip:890024@10.254.176.126:5090>;isfocus
Content-Type: application/sdp
Content-Length: 284

v=0
o=890024@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45056 RTP/AVP 8 0 18 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F4.
SIP/2.0 200 OK
From: "2005"<sip:890024@ug1.scm.com:5090>;tag=b5029b80-0-13e2-50022-7e979-23f0e0bf-7e979
To: <sip:2003@ug1.scm.com:5060>;tag=f84ff66e-21fb-40be-ae42-36d32ad3f6c2
Call-ID: b4eeaf50-0-13e2-50022-7e979-541f9c09-7e979
CSeq: 1 INVITE
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
P-Asserted-Identity: "2003" <sip:2003@ug1.scm.com>
User-Agent: SAMSUNG SCM 3.2.400
Via: SIP/2.0/UDP 192.168.113.50:5090;received=127.0.0.1;branch=z9hG4bK-7e979-1ee802dd-468a49dc
Contact: <sip:2003@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 225

v=0
o=2003 1340835546 1 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 165.213.80.225
t=0 0
m=audio 20060 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

## Dispatch Conference Messages

*30: Dispatch Conference Feature Code
2005: Dispatch Conference Master
2003, 2004: Dispatch Conference Members

```
F1.
INVITE sip:*30@ug1.scm.com:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=fcf0e1df-fee8-4627-81eb-19d128703bba
To: <sip:*30@ug1.scm.com>
Call-ID: 9f723833-714c-44ba-bb1e-86dbcfc339b6@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-6380b-184aeed6-7b5aea2
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Contact: <sip:2005@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 339

v=0
o=2005 1340724571 0 IN IP4 165.213.176.85
s=Samsung IP PBX
c=IN IP4 165.213.176.85
t=0 0
a=sendrecv
m=audio 20024 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F2.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=fcf0e1df-fee8-4627-81eb-19d128703bba
To: <sip:*30@ug1.scm.com>;tag=b4fe4fd0-0-13e2-50022-6380b-2a54114d-6380b
Call-ID: 9f723833-714c-44ba-bb1e-86dbcfc339b6@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-6380b-184aeed6-7b5aea2
Contact: <sip:890006@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890006@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45020 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F3.
NOTIFY sip:890006@ug1.scm.com:5090 SIP/2.0
From: <sip:2005@ug1.scm.com>;tag=85e160d6-1eea-4b56-9c2e-aa752d07c94d
To: <sip:890006@ug1.scm.com>;tag=9ec47cb0-c741-46c0-afd6-040b4e6e8e13
Call-ID: fbbc8e6e-46ae-4421-8f19-5e936f0608ee@ug1.scm.com
CSeq: 12 NOTIFY
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-6380c-184af18d-3ab4c460
Event: dispatch-list;type=add;value=2
Call-Info: <sip:890890@ug1.scm.com>,<http://127.0.0.1>;answer-after=0
Max-Forwards: 70
Contact: <sip:2005@10.254.176.126:5060>
User-Agent: SAMSUNG SCM 3.2.400
Content-Type: text/plain
Content-Length: 9

2003/2004
```

```
F4.
SIP/2.0 200 OK
From: <sip:2005@ug1.scm.com>;tag=85e160d6-1eea-4b56-9c2e-aa752d07c94d
To: <sip:890006@ug1.scm.com>;tag=9ec47cb0-c741-46c0-afd6-040b4e6e8e13
Call-ID: fbbc8e6e-46ae-4421-8f19-5e936f0608ee@ug1.scm.com
CSeq: 12 NOTIFY
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-6380c-184af18d-3ab4c460
Content-Length: 0
```

```
F5.
INVITE sip:2003@ug1.scm.com:5060 SIP/2.0
From:     "2005"<sip:890006@ug1.scm.com:5090>;tag=b4fe5630-0-13e2-50022-6380c-12fb8ac0-6380c
To: <sip:2003@ug1.scm.com:5060>
Call-ID: b4ee2508-0-13e2-50022-6380c-685acf02-6380c
CSeq: 1 INVITE
Via: SIP/2.0/UDP 192.168.113.50:5090;branch=z9hG4bK-6380c-184af18f-6ac3d312
Call-Info: <http://127.0.0.1>;purpose=dispatch-conf
Max-Forwards: 70
Reason: SIP ;cause=200 ;text="Conference"
Contact: <sip:890006@10.254.176.126:5090>;isfocus
Content-Type: application/sdp
Content-Length: 284

v=0
o=890006@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45020 RTP/AVP 8 0 18 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F6.
SIP/2.0 200 OK
From:     "2005"<sip:890006@ug1.scm.com:5090>;tag=b4fe5630-0-13e2-50022-6380c-12fb8ac0-6380c
To: <sip:2003@ug1.scm.com:5060>;tag=255cca5b-8050-4ef5-bf06-ee0ec4e30bf7
Call-ID: b4ee2508-0-13e2-50022-6380c-685acf02-6380c
CSeq: 1 INVITE
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
P-Asserted-Identity: "2003" <sip:2003@ug1.scm.com>
User-Agent: SAMSUNG SCM 3.2.400
Via:   SIP/2.0/UDP   192.168.113.50:5090;received=127.0.0.1;branch=z9hG4bK-6380c-184af18f-6ac3d312
Contact: <sip:2003@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 225

v=0
o=2003 1340724579 1 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 165.213.80.225
t=0 0
m=audio 20016 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

## 3.3.8 Basic Conference Server

SCM is providing several type of conference service (refer to 3.3.5~3.3.8).
Conference Server is checking feature code in INVITE message and some specific Header information to distinguish the conference services.

There are 4 types of basic conferences as shown below.

- Adhoc: Holding a conference by using ‘Conference’ key on the SMT-terminal
- COA: Holding a conference by using [COA] soft menu during the conversation
- Record: Holding a conference by using [Call Record] soft menu during the conversation
- Barge-in/Intrusion/Multi-device conference

SCM puts [Conference] feature code in the To URI of first INVITE which is sent to conference server except COA. Therefore SCM should check Subject header to distinguish the service. You can refer to SIP Usage for more details.

Conference server should send 200 OK with SDP which is including his all Capability (RTP/sRTP/video) when he receives not only reINVITE, but also NoSDP INVITE message from SCM to hold a Basic conference. And conference server should be possible to process SDP of ACK message.

If one member terminates the conference during the 3-party conference, then conference server finishes the conference and should send REFER message to keep the conversation between left 2 users.

### 3.3.8.1 SIP usage

#### Outbound Call (SCM to Conference Server)

- **‘From’ Header**
  Information of user and host who request Basic conference are included.
  *From: <sip:user who request Basic conference@host>*
- **‘To’ Header/Request URI**
  Feature code for Basic conference is included. If there is group ID, it can be added after feature code.
  *To: <sip:Basic conference feature code@host>*
- **‘Reason’ Header**
  Screen display for master and members can be managed by this field.
  If Reason text is included in 200 OK which server response, then it will be displayed on master’s screen. For example, if below text is included in 200 OK, then ‘Conference’ will be displayed on master’s screen.
  *Reason: SIP ;cause=200 ;text=‘Conference’*

- **‘Contact’ Header**
  Conference server is including the conference channel in 200 OK message. SCM can conclude conference call by checking isfocus
- **‘Subject’ Header**
  There could be a master’s information in first INVITE to conference server, when master try to record. In case user 2005 try to record, ‘Subject’ header will be like below.
  - *Subject: <UMS>::[2005]*
    In case of Intrusion/Multi-device conference, user number who tries to barge in will be sent to conference server.
  - *Subject: <CINT>::[2005]*
    In case of Barge-in, ‘Subject’ header will be like below.
  - *Subject: <CINT/NOTONE>::[2005]*
    In case of Basic Adhoc conference, Subject Header will not be included.

### 3.3.8.2 Call Flows

#### 3-party Adhoc Conference

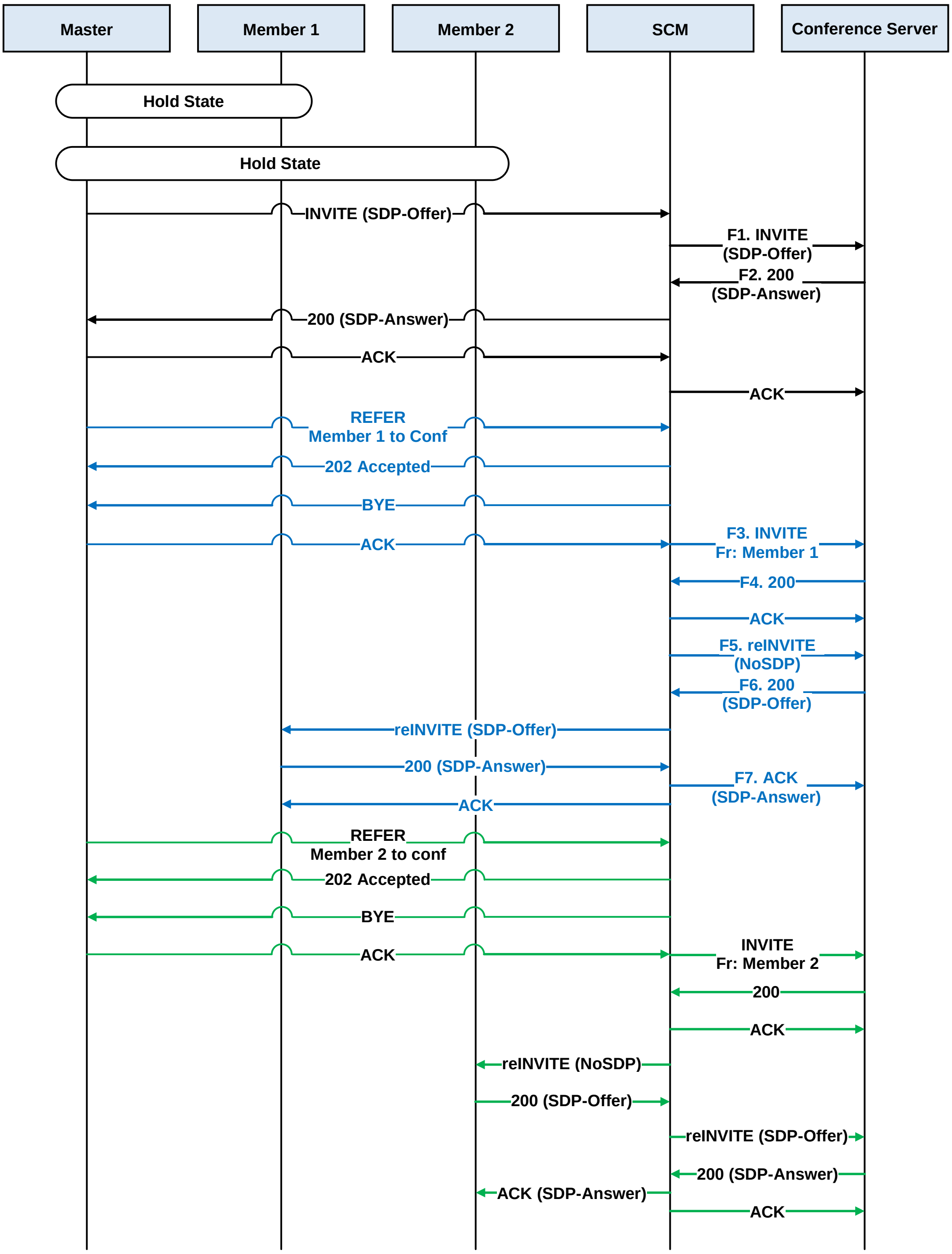



Figure 3.50 3-party Ad-hoc conference Flow

Blue arrows are for the joining flow of Member 1 to conference, and green arrows are for the joining flow of Member 2.

1) Master press ‘Conference’ key to hold a call and make a outgoing call to Member 2.
2) If Master press ‘Conference’ key one more during the conversation with Member 2, then Master send INVITE with [Conference] feature code in To URI.
3) Conference server makes a call with Room channel in Contact.
4) When Master receive 200 OK then, Master will send REFER message to open a conference with Master 1 and Master 2.
5) SCM receives REFER message, and then SCM sends INVITE message with Member number in From and any information in SDP.
6) When SCM receives 200 OK from conference server, SCM sends NoSDP reINVITE to conference server.
7) Conference server should send 200 OK with SDP which is including his all Capability (RTP/sRTP/video).
8) SCM sends reINVITE with SDP information which is from conference server.
9) SCM sends ACK message with SDP which is from SMT terminal.

In case Conference server receives reINVITE to connect conference members, reINVITE could include SDP or NoSDP. Therefore both cases should be possible to contorl.

## Adhoc Conference more than 4-parties

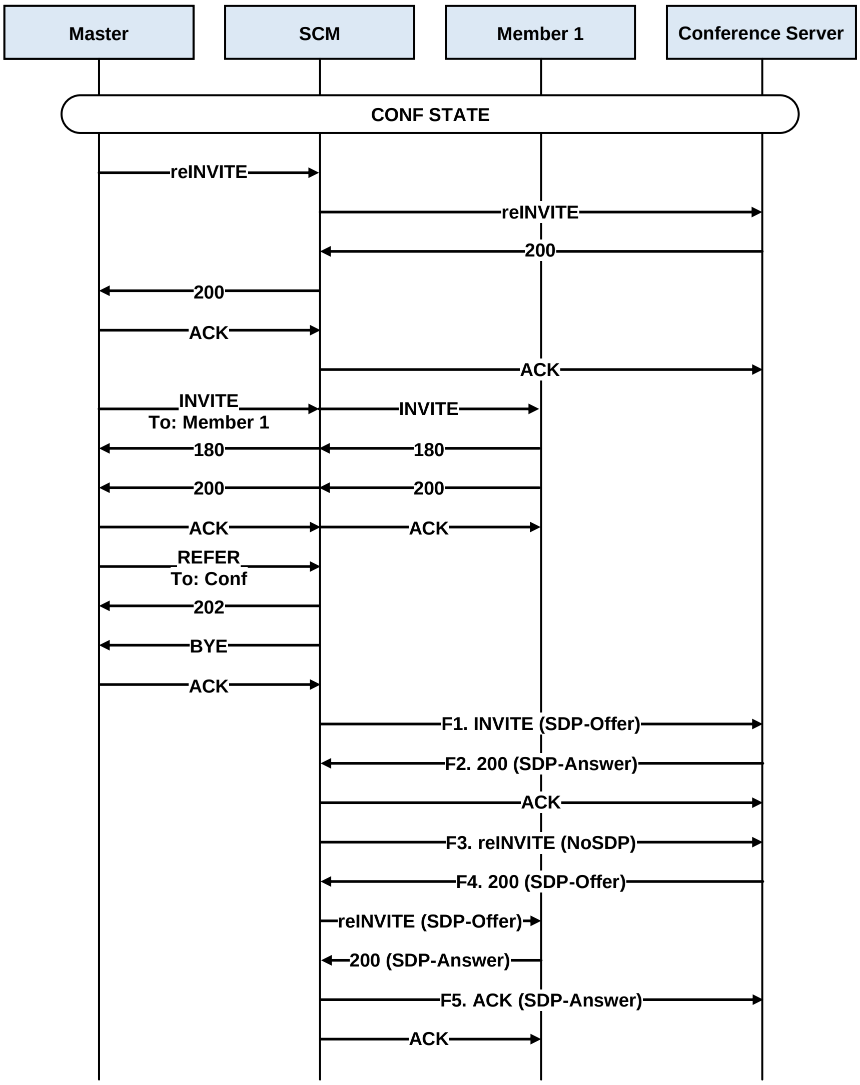



**Figure 3.51 Ad-hoc Conference more than 4-parties**

1) Master press ‘Conference’ key to hold during the conference and call the Member 1.
2) When Member1 response with 200 message then SCM transfer 200 to Master. Master receives 200 and send REFER message with Conference room channel number by pressing ‘Conference’ key.
3) When SCM receives REFER message, then he sends INVITE message to Conference server for joining Member 1.
4) SCM receives 200 from Conference server, and then send NoSDP reINVITE to Conference server.
5) Conference server should send 200 OK with SDP which is including his all Capability (RTP/sRTP/video).
6) SCM sends reINVITE with SDP which is from Conference server to Conference member.
7) SCM sends ACK message with 200 SDP to Conference server.

Conference server should be possible to process SDP INVITE, in case SCM sends NoSDP reINVITE to Member 1.

## 3-party COA Conference

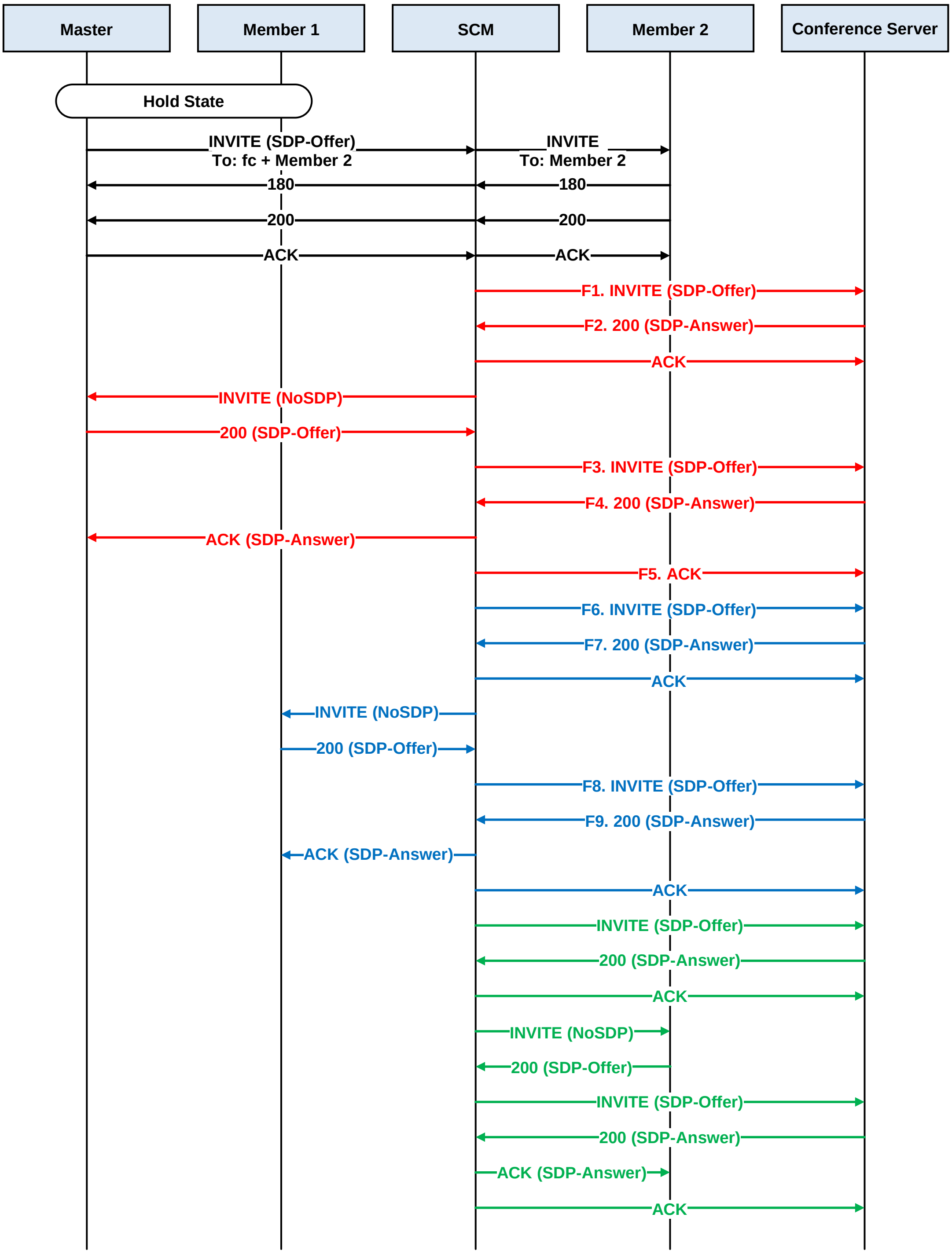



**Figure 3.52 3-party COA Conference**

- Red flow: Master joins in conference
- Blue flow: Member 1 joins in conference
- Green flow: Member 2 joins in conference

1) Master holds a call with Member 1 and then makes a call by pressing [Conference On Answer] feature code + Member 2 number.
2) SCM try to make a call to Member 2. If SCM receives 200, then sends INVITE with [Conference On Answer] feature code as To URI.
3) When SCM receives 200 from Conference server, then he sends reINVITE to Master. If he receives 200 from Mater, then he sends reINVITE to Conference server to make Master to join the conference.
4) When Member 1 and Member 2 receive 200 from Conference server, SCM process the reINVITE and connect to Conference server.

Conference server should be possible to process NoSDP INVITE, in case SCM sends NoSDP reINVITE to Member 1.

### Adhoc Conference more than 4-parties

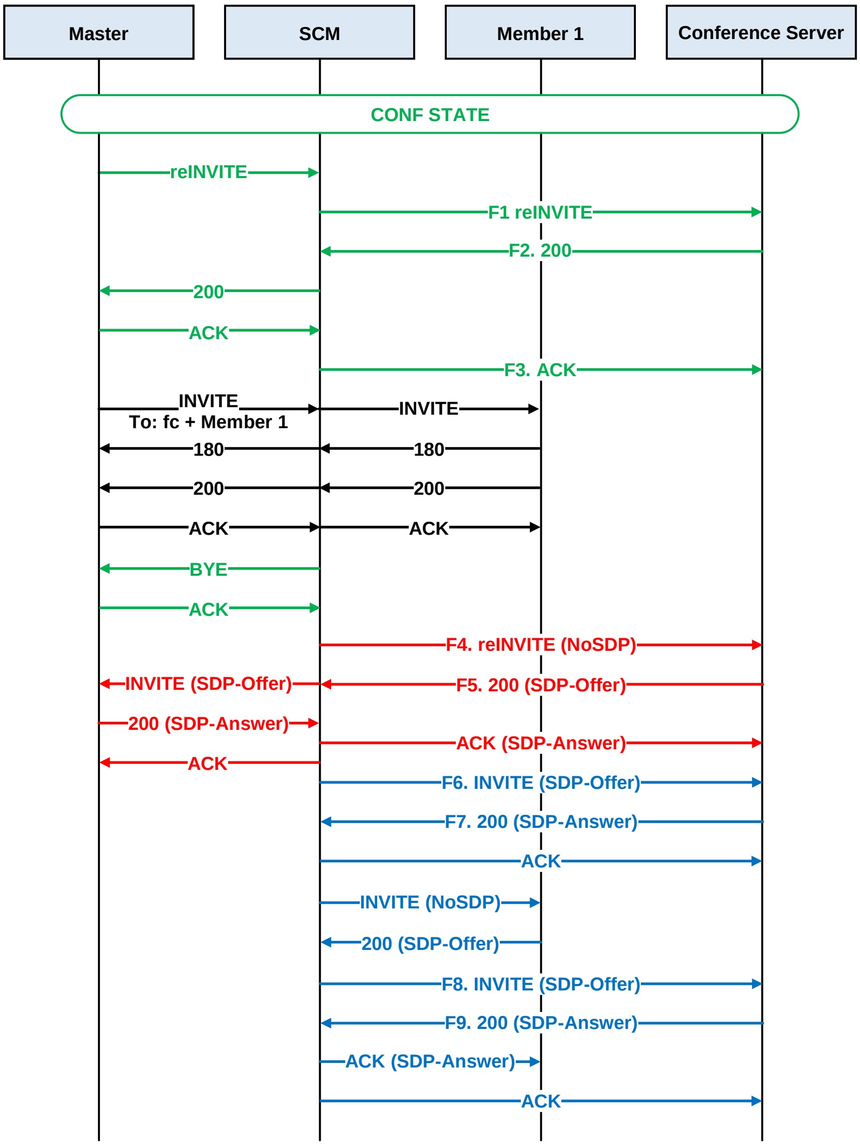



**Figure 3.53 COA Conference more than 4-parties**

- Red flow: Master joins in conference
- Blue flow: Member 1 joins in conference

1) Master holds a call during conference and dials [Conference On Answer] feature code + Member 1 number.
2) When SCM receives 200 from Member 1, then he sends reINVITE to connect current call to conference channel.
3) SCM sends INVITE with SDP to Conference server to connect Member 1 to conference server. When SCM receives 200 from Conference server then he sends reINVITE to Shuffle process.

### Voice Recording Conference

In case of Voice Recording service, Conference server connects to Recording Server to send Mixed RTP after connecting to SCM. Therefore the flow and message between Conference server and Recording server will be explained.

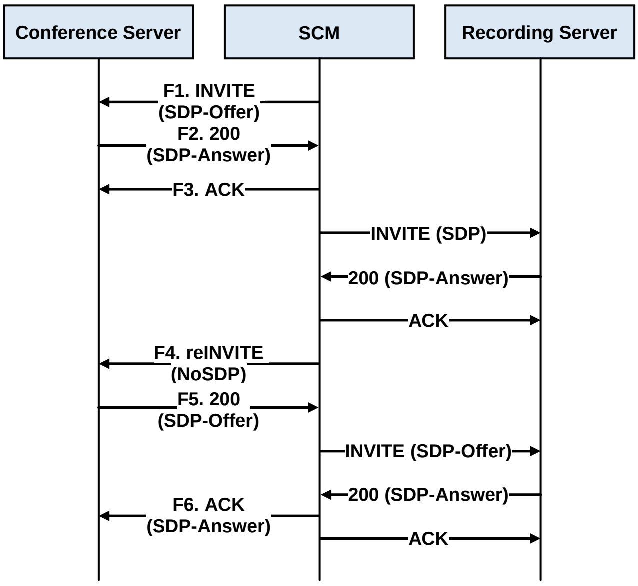



**Figure 3.54 Recording conference Flow**

1) When user tries to record, SCM sends INVITE to Conference server.
2) When Conference server response with 200 OK, then SCM sends ACK to Conference server and INVITE with SDP to Recording server.
3) When Recording server response with 200 OK, then SCM sends ACK to Recording server and NoSDP reINVITE to Conference server.
4) Conference server should send 200 OK with SDP which is including his all Capabilities (RTP/sRTP/video). SCM would send this information to Recording server.
5) When SCM receives 200 OK from Recording server, then he sends ACK with SDP to Conference server and ACK to Recording server.

### Barge-in/Intrusion/Multi-device conference

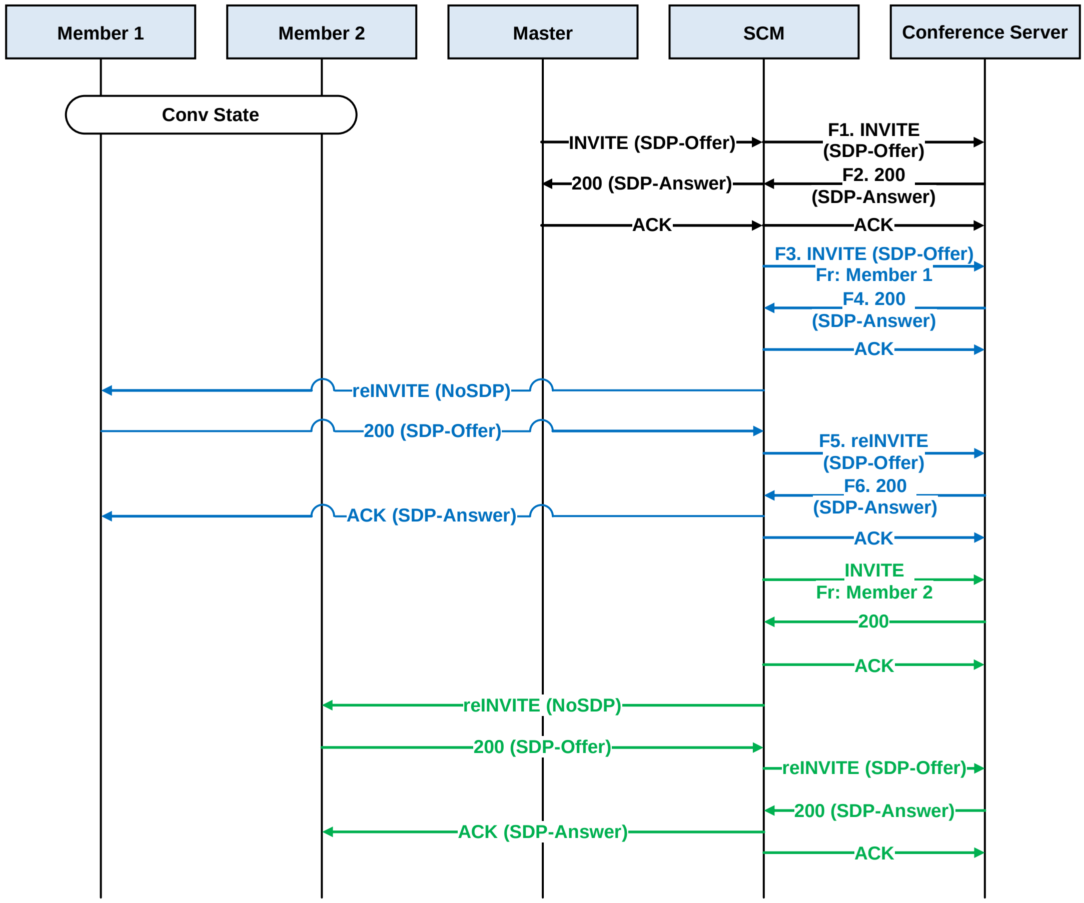



**Figure 3.55 Barge-in/Intrusion/Multi-device conference**

- Blue flow: Member 1 joins in conference
- Green flow: Member 2 joins in conference

Figure 3.50 shows that Barge-in/Intrusion/Multi-device conference during the conversation between Member 1 and 2. If the members are already on the conference status, then the corresponding members will be connected to conference. You can refer to black arrow flow.

To provide Multi-device conference, Conference server should be possible to add conference member which is same number.

1) Master tried to barge-in to Member 1 during the conversation between Member 1 and Member 2.
2) SCM connect the call between Master and Conference server. When SCM receives 200 from Conference server, then he connects the existing call members to Conference server.

## Termination of one member during 3-party basic conference

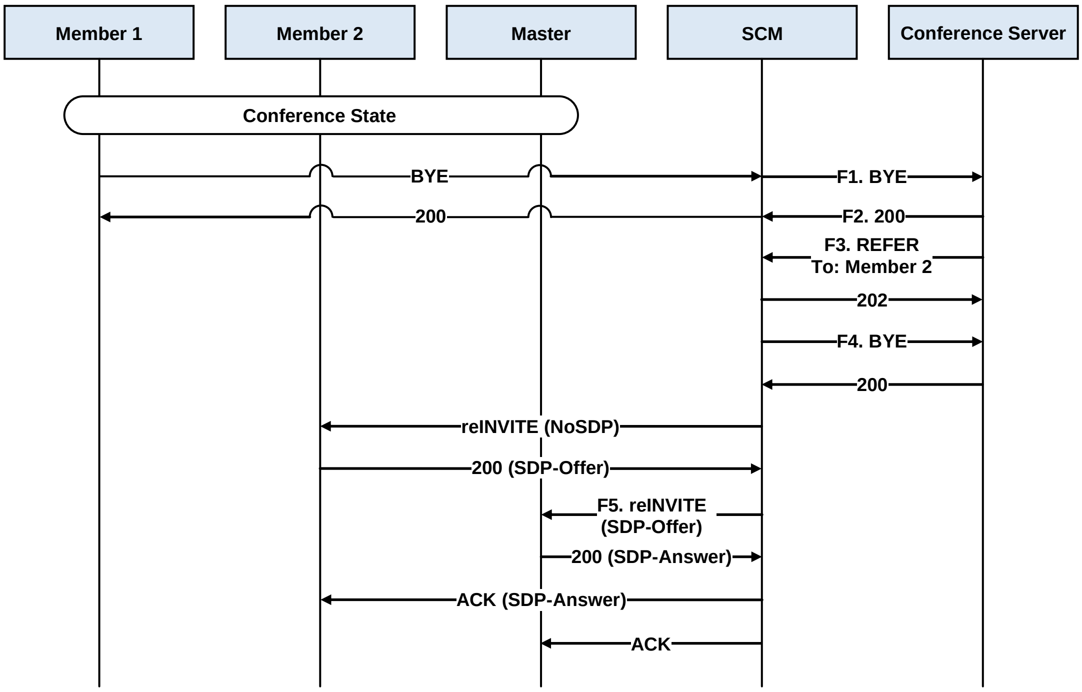



**Figure 3.56 Termination of one member during 3-party basic conference**

If one member terminates the conference during 3-party basic conference then conference resource should be released. Therefore, if Conference server receives BYE message from one member then he should send REFER message to the other members.

1) Member 1 sends BYE message during the conference among Member 1, Member 2, and Master.
2) Conference server receives BYE, then he sends REFER to Member 2. And then he receives the BYE from SCM then conference will be terminated.

### 3.3.8.3 SIP Messages

#### 3-party Adhoc Messages

*89: [Conference] Feature Code
2005: Conference Master
2004, 2001: Conference Members

```
F1.
INVITE sip:*89@ug1.scm.com:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=570d511a-09e1-473b-b479-88679f094d10
To: <sip:*89@ug1.scm.com>
Call-ID: bf7e3695-2bc7-4bed-b366-602e559a7a9a@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-67ae4-19500d70-3ccf767
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Max-Forwards: 70
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Contact: <sip:2005@10.254.176.126:5060>
User-Agent: SAMSUNG SCM 3.2.400
Content-Type: application/sdp
Content-Length: 339

v=0
o=2005 1340741685 0 IN IP4 165.213.176.85
s=Samsung IP PBX
c=IN IP4 165.213.176.85
t=0 0
a=sendrecv
m=audio 20068 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F2.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=570d511a-09e1-473b-b479-88679f094d10
To: <sip:*89@ug1.scm.com>;tag=b5048ff0-0-13e2-50022-67ae4-230f478c-67ae4
Call-ID: bf7e3695-2bc7-4bed-b366-602e559a7a9a@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-67ae4-19500d70-3ccf767
Contact: <sip:890008@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890008@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45024 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F3.
INVITE sip:890008@ug1.scm.com:5090 SIP/2.0
From: <sip:2001@ug1.scm.com>;tag=23093ce8-8f3b-457c-8809-1833275d23e6
To: "890008"<sip:890008@ug1.scm.com>
Call-ID: 47bb0af6-457d-4fef-acf5-f9d5baebb49c@10.254.168.210
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-67ae5-19501027-57a02b53
P-Asserted-Identity: <sip:2001@10.254.168.210>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2001@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 259

v=0
o=2001 100037 100037 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 35999 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F4.
SIP/2.0 200 OK
From: <sip:2001@ug1.scm.com>;tag=23093ce8-8f3b-457c-8809-1833275d23e6
To: "890008"<sip:890008@ug1.scm.com>;tag=b5049980-0-13e2-50022-67ae5-6ef98f43-67ae5
Call-ID: 47bb0af6-457d-4fef-acf5-f9d5baebb49c@10.254.168.210
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-67ae5-19501027-57a02b53
Contact: <sip:890008@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890008@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45024 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F5.
INVITE sip:890008@10.254.176.126:5090 SIP/2.0
From: <sip:2001@ug1.scm.com>;tag=23093ce8-8f3b-457c-8809-1833275d23e6
To: "890008"<sip:890008@ug1.scm.com>;tag=b5049980-0-13e2-50022-67ae5-6ef98f43-67ae5
Call-ID: 47bb0af6-457d-4fef-acf5-f9d5baebb49c@10.254.168.210
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-67ae5-195010a6-193e55a4
P-Asserted-Identity: <sip:2001@10.254.168.210>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2001@10.254.176.126:5060>
Content-Length: 0
```

```
F6.
SIP/2.0 200 OK
From: <sip:2001@ug1.scm.com>;tag=23093ce8-8f3b-457c-8809-1833275d23e6
To: "890008"<sip:890008@ug1.scm.com>;tag=b5049980-0-13e2-50022-67ae5-6ef98f43-67ae5
Call-ID: 47bb0af6-457d-4fef-acf5-f9d5baebb49c@10.254.168.210
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-67ae5-195010a6-193e55a4
Contact: <sip:890008@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 284

v=0
o=890008@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45024 RTP/AVP 8 0 18 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F7.
ACK sip:890008@10.254.176.126:5090 SIP/2.0
From: <sip:2001@ug1.scm.com>;tag=23093ce8-8f3b-457c-8809-1833275d23e6
To: "890008"<sip:890008@ug1.scm.com>;tag=b5049980-0-13e2-50022-67ae5-6ef98f43-67ae5
Call-ID: 47bb0af6-457d-4fef-acf5-f9d5baebb49c@10.254.168.210
CSeq: 2 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-67ae5-1950114f-56e14147
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2001@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 169

v=0
o=2001 1340741638 3 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 165.213.89.183
t=0 0
m=audio 20028 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
a=sendrecv
```

## Adhoc Messages more than 4-parties

*89: [Conference] Feature Code

2005: Conference Master

2004, 2001: Conference Members

```
F1.
INVITE sip:890008@ug1.scm.com:5090 SIP/2.0
From: <sip:2001@ug1.scm.com>;tag=f6982cd8-82b5-4c5d-b99c-7e8526586522
To: "890008"<sip:890008@ug1.scm.com>
Call-ID: 420c5225-10f5-4232-b2cb-401c03950497@10.254.168.210
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-686e3-197ee9cb-5660c438
P-Asserted-Identity: <sip:2001@10.254.168.210>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2001@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 259

v=0
o=2001 100040 100040 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 35999 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F2.
SIP/2.0 200 OK
From: <sip:2001@ug1.scm.com>;tag=f6982cd8-82b5-4c5d-b99c-7e8526586522
To: "890008"<sip:890008@ug1.scm.com>;tag=b505d1e0-0-13e2-50022-686e3-a008086-686e3
Call-ID: 420c5225-10f5-4232-b2cb-401c03950497@10.254.168.210
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-686e3-
197ee9cb-5660c438
Contact: <sip:890008@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890008@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45024 RTP/AVP 0 101
c=IN IP4 10.254.176.126
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F3.
INVITE sip:890008@10.254.176.126:5090 SIP/2.0
From: <sip:2001@ug1.scm.com>;tag=f6982cd8-82b5-4c5d-b99c-7e8526586522
To: "890008"<sip:890008@ug1.scm.com>;tag=b505d1e0-0-13e2-50022-686e3-a008086-686e3
Call-ID: 420c5225-10f5-4232-b2cb-401c03950497@10.254.168.210
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-686e3-197eea15-40b5817
P-Asserted-Identity: <sip:2001@10.254.168.210>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2001@10.254.176.126:5060>
Content-Length: 0
```

```
F4.
SIP/2.0 200 OK
From: <sip:2001@ug1.scm.com>;tag=f6982cd8-82b5-4c5d-b99c-7e8526586522
To: "890008"<sip:890008@ug1.scm.com>;tag=b505d1e0-0-13e2-50022-686e3-a008086-686e3
Call-ID: 420c5225-10f5-4232-b2cb-401c03950497@10.254.168.210
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-686e3-197eea15-40b5817
Contact: <sip:890008@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 284

v=0
o=890008@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45024 RTP/AVP 8 0 18 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F5.
ACK sip:890008@10.254.176.126:5090 SIP/2.0
From: <sip:2001@ug1.scm.com>;tag=f6982cd8-82b5-4c5d-b99c-7e8526586522
To: "890008"<sip:890008@ug1.scm.com>;tag=b505d1e0-0-13e2-50022-686e3-a008086-686e3
Call-ID: 420c5225-10f5-4232-b2cb-401c03950497@10.254.168.210
CSeq: 2 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-686e4-197eeaab-6f783082
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2001@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 169

v=0
o=2001 1340744711 2 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 165.213.89.183
t=0 0
m=audio 20032 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
a=sendrecv
```

### 3-party COA Messages

*81: [Conference On Answer] Feature Code
2005: Master Number
2001: Conference Members

```
F1.
INVITE sip:*81@ug1.scm.com:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=cc48d154-05e8-4261-8cb1-832be3f64f3e
To: <sip:*81@ug1.scm.com>
Call-ID: 4c4cfd75-142d-418a-b972-8e71a9192226@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-68c7a-1994bfae-3ef6559b
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 259

v=0
o=2005 100042 100042 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 35999 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F2.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=cc48d154-05e8-4261-8cb1-832be3f64f3e
To: <sip:*81@ug1.scm.com>;tag=b5066fa8-0-13e2-50022-68c7a-f1f2aa8-68c7a
Call-ID: 4c4cfd75-142d-418a-b972-8e71a9192226@ug1.scm.com
CSeq: 1 INVITE
Via:  SIP/2.0/UDP  10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-68c7a-1994bfae-3ef6559b
Contact: <sip:890009@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890009@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45026 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F3.
INVITE sip:890009@10.254.176.126:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=cc48d154-05e8-4261-8cb1-832be3f64f3e
To: <sip:*81@ug1.scm.com>;tag=b5066fa8-0-13e2-50022-68c7a-f1f2aa8-68c7a
Call-ID: 4c4cfd75-142d-418a-b972-8e71a9192226@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-68c7b-1994c1a9-6020ed1f
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 339

v=0
o=2005 1340746184 1 IN IP4 165.213.176.85
s=Samsung IP PBX
c=IN IP4 165.213.176.85
t=0 0
a=sendrecv
m=audio 20094 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F4.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=cc48d154-05e8-4261-8cb1-832be3f64f3e
To: <sip:*81@ug1.scm.com>;tag=b5066fa8-0-13e2-50022-68c7a-f1f2aa8-68c7a
Call-ID: 4c4cfd75-142d-418a-b972-8e71a9192226@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-68c7b-1994c1a9-6020ed1f
Contact: <sip:890009@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890009@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45026 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F5.
ACK sip:890009@10.254.176.126:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=cc48d154-05e8-4261-8cb1-832be3f64f3e
To: <sip:*81@ug1.scm.com>;tag=b5066fa8-0-13e2-50022-68c7a-f1f2aa8-68c7a
Call-ID: 4c4cfd75-142d-418a-b972-8e71a9192226@ug1.scm.com
CSeq: 2 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-68c7b-1994c1e5-4f3de668
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Length: 0
```

```
F6.
INVITE sip:890009@ug1.scm.com:5090 SIP/2.0
From: "2004"<sip:2004@ug1.scm.com>;tag=d4559d9b-324b-4c11-b948-37508d723c80
To: "890009"<sip:890009@ug1.scm.com>
Call-ID: 6f95078f-34c5-4133-8f3e-3834c0771018@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-68c7a-1994c066-753ea933
P-Asserted-Identity: "2004" <sip:2004@ug1.scm.com>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2004@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 259

v=0
o=2004 100043 100043 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 35999 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F7.
SIP/2.0 200 OK
From: "2004"<sip:2004@ug1.scm.com>;tag=d4559d9b-324b-4c11-b948-37508d723c80
To: "890009"<sip:890009@ug1.scm.com>;tag=b5067470-0-13e2-50022-68c7a-1ef42dfe-68c7a
Call-ID: 6f95078f-34c5-4133-8f3e-3834c0771018@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-68c7a-
1994c066-753ea933
Contact: <sip:890009@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890009@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45026 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F8.
INVITE sip:890009@10.254.176.126:5090 SIP/2.0
From: "2004"<sip:2004@ug1.scm.com>;tag=d4559d9b-324b-4c11-b948-37508d723c80
To: "890009"<sip:890009@ug1.scm.com>;tag=b5067470-0-13e2-50022-68c7a-1ef42dfe-68c7a
Call-ID: 6f95078f-34c5-4133-8f3e-3834c0771018@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-68c7b-1994c14f-4f563379
P-Asserted-Identity: "2004" <sip:2004@ug1.scm.com>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2004@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 285

v=0
o=2004 1340746157 3 IN IP4 165.213.80.94
s=Samsung IP PBX
c=IN IP4 165.213.80.94
t=0 0
a=sendrecv
m=audio 20048 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F9.
SIP/2.0 200 OK
From: "2004"<sip:2004@ug1.scm.com>;tag=d4559d9b-324b-4c11-b948-37508d723c80
To: "890009"<sip:890009@ug1.scm.com>;tag=b5067470-0-13e2-50022-68c7a-1ef42dfe-68c7a
Call-ID: 6f95078f-34c5-4133-8f3e-3834c0771018@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-68c7b-1994c14f-4f563379
Contact: <sip:890009@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890009@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45026 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

## COA Messages more than 4-parties

*81: [Conference On Answer] Feature Code

2005: Master Number

2001: Conference Member

```
F1.
INVITE sip:890010@10.254.176.126:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=de2a7e27-0e53-4bfe-9a0f-9ca6b6898ef7
To: <sip:*81@ug1.scm.com>;tag=b5082de8-0-13e2-50022-76ce0-5dcecaa1-76ce0
Call-ID: fd95da63-3241-4c3f-aebd-4193dcf6d3d4@ug1.scm.com
CSeq: 3 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-76d18-1d022938-4c769d9c
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 339

v=0
o=2005 1340803634 2 IN IP4 165.213.176.85
s=Samsung IP PBX
c=IN IP4 165.213.176.85
t=0 0
a=inactive
m=audio 20104 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=inactive
```

```
F2.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=de2a7e27-0e53-4bfe-9a0f-9ca6b6898ef7
To: <sip:*81@ug1.scm.com>;tag=b5082de8-0-13e2-50022-76ce0-5dcecaa1-76ce0
Call-ID: fd95da63-3241-4c3f-aebd-4193dcf6d3d4@ug1.scm.com
CSeq: 3 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-76d18-1d022938-4c769d9c
Contact: <sip:890010@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890010@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45028 RTP/AVP 0 101
c=IN IP4 10.254.176.126
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F3.
ACK sip:890010@10.254.176.126:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=de2a7e27-0e53-4bfe-9a0f-9ca6b6898ef7
To: <sip:*81@ug1.scm.com>;tag=b5082de8-0-13e2-50022-76ce0-5dcecaa1-76ce0
Call-ID: fd95da63-3241-4c3f-aebd-4193dcf6d3d4@ug1.scm.com
CSeq: 3 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-76d19-1d0229c0-74c7cff0
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Length: 0
```

```
F4.
INVITE sip:890010@10.254.176.126:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=de2a7e27-0e53-4bfe-9a0f-9ca6b6898ef7
To: <sip:*81@ug1.scm.com>;tag=b5082de8-0-13e2-50022-76ce0-5dcecaa1-76ce0
Call-ID: fd95da63-3241-4c3f-aebd-4193dcf6d3d4@ug1.scm.com
CSeq: 4 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-76d22-1d024d36-32e0c94b
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 339

v=0
o=2005 1340803698 1 IN IP4 165.213.176.85
s=Samsung IP PBX
c=IN IP4 165.213.176.85
t=0 0
a=sendrecv
m=audio 20108 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F5.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=de2a7e27-0e53-4bfe-9a0f-9ca6b6898ef7
To: <sip:*81@ug1.scm.com>;tag=b5082de8-0-13e2-50022-76ce0-5dcecaa1-76ce0
Call-ID: fd95da63-3241-4c3f-aebd-4193dcf6d3d4@ug1.scm.com
CSeq: 4 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-76d22-1d024d36-32e0c94b
Contact: <sip:890010@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890010@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45028 RTP/AVP 0 101
c=IN IP4 10.254.176.126
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F6.
INVITE sip:890010@ug1.scm.com:5090 SIP/2.0
From: "2004"<sip:2004@ug1.scm.com>;tag=eaa4aa94-45ab-4685-9e7b-3b66815a21a8
To: "890010"<sip:890010@ug1.scm.com>
Call-ID: bcfbff1c-a0ad-441d-a99f-e502ea6fbd02@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-76d21-1d024abc-f616f4c
P-Asserted-Identity: "2004" <sip:2004@ug1.scm.com>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2004@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 259

v=0
o=2004 100049 100049 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 35999 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F7.
SIP/2.0 200 OK
From: "2004"<sip:2004@ug1.scm.com>;tag=eaa4aa94-45ab-4685-9e7b-3b66815a21a8
To: "890010"<sip:890010@ug1.scm.com>;tag=b5084dc8-0-13e2-50022-76d21-399ddccc-76d21
Call-ID: bcfbff1c-a0ad-441d-a99f-e502ea6fbd02@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-76d21-
1d024abc-f616f4c
Contact: <sip:890010@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890010@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45028 RTP/AVP 0 101
c=IN IP4 10.254.176.126
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F8.
INVITE sip:890010@ug1.scm.com:5090 SIP/2.0
From: "2004"<sip:2004@ug1.scm.com>;tag=eaa4aa94-45ab-4685-9e7b-3b66815a21a8
To: "890010"<sip:890010@ug1.scm.com>
Call-ID: bcfbff1c-a0ad-441d-a99f-e502ea6fbd02@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-76d21-1d024abc-f616f4c
P-Asserted-Identity: "2004" <sip:2004@ug1.scm.com>
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2004@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 259

v=0
o=2004 100049 100049 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 35999 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F9.
SIP/2.0 200 OK
From: "2004"<sip:2004@ug1.scm.com>;tag=eaa4aa94-45ab-4685-9e7b-3b66815a21a8
To: "890010"<sip:890010@ug1.scm.com>;tag=b5084dc8-0-13e2-50022-76d21-399ddccc-76d21
Call-ID: bcfbff1c-a0ad-441d-a99f-e502ea6fbd02@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-76d21-
1d024abc-f616f4c
Contact: <sip:890010@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890010@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45028 RTP/AVP 0 101
c=IN IP4 10.254.176.126
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

## Recording Messages

*89: [Conference] Feature Code

2005: User who wants to record

2004: Opposite side with user who wants to record

```
F1.
INVITE sip:*89@ug1.scm.com:5090 SIP/2.0
From: "Recording"<sip:880880@ug1.scm.com>;tag=8b636ab4-8c30-4b04-a659-a91788cf711a
To: <sip:*89@ug1.scm.com>
Call-ID: a4e91343-058a-49ee-afc4-ccf2aad250b6@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7ad66-1dfd56e1-4c409e38
P-Asserted-Identity: "Recording" <sip:880880@ug1.scm.com>
Subject: <UMS>::[2005]
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:880880@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 258

v=0
o=880880 100085 100085 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
a=sendrecv
m=audio 35999 RTP/AVP 0 8 18
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:18 G729/8000
a=ptime:20
a=sendrecv
```

```
F2.
SIP/2.0 200 OK
From: "Recording"<sip:880880@ug1.scm.com>;tag=8b636ab4-8c30-4b04-a659-a91788cf711a
To: <sip:*89@ug1.scm.com>;tag=b4fc8668-0-13e2-50022-7ad66-31ce9344-7ad66
Call-ID: a4e91343-058a-49ee-afc4-ccf2aad250b6@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-7ad66-1dfd56e1-4c409e38
Contact: <sip:890019@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890019@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45046 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F3.
ACK sip:890019@10.254.176.126:5090 SIP/2.0
From: "Recording"<sip:880880@ug1.scm.com>;tag=8b636ab4-8c30-4b04-a659-a91788cf711a
To: <sip:*89@ug1.scm.com>;tag=b4fc8668-0-13e2-50022-7ad66-31ce9344-7ad66
Call-ID: a4e91343-058a-49ee-afc4-ccf2aad250b6@ug1.scm.com
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7ad66-1dfd575f-32f9efe1
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:880880@10.254.176.126:5060>
Content-Length: 0
```

```
F4.
INVITE sip:890019@10.254.176.126:5090 SIP/2.0
From: "Recording"<sip:880880@ug1.scm.com>;tag=8b636ab4-8c30-4b04-a659-a91788cf711a
To: <sip:*89@ug1.scm.com>;tag=b4fc8668-0-13e2-50022-7ad66-31ce9344-7ad66
Call-ID: a4e91343-058a-49ee-afc4-ccf2aad250b6@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7ad66-1dfd582b-5d9835c8
P-Asserted-Identity: "Voice Mail" <sip:880880@ug1.scm.com>
Subject: <UMS>::[2005]
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:880880@10.254.176.126:5060>
Content-Length: 0
```

```
F5.
SIP/2.0 200 OK
From: "Recording"<sip:880880@ug1.scm.com>;tag=8b636ab4-8c30-4b04-a659-a91788cf711a
To: <sip:*89@ug1.scm.com>;tag=b4fc8668-0-13e2-50022-7ad66-31ce9344-7ad66
Call-ID: a4e91343-058a-49ee-afc4-ccf2aad250b6@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-7ad66-1dfd582b-5d9835c8
Contact: <sip:890019@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 284

v=0
o=890019@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45046 RTP/AVP 8 0 18 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F6.
ACK sip:890019@10.254.176.126:5090 SIP/2.0
From: "Recording"<sip:880880@ug1.scm.com>;tag=8b636ab4-8c30-4b04-a659-a91788cf711a
To: <sip:*89@ug1.scm.com>;tag=b4fc8668-0-13e2-50022-7ad66-31ce9344-7ad66
Call-ID: a4e91343-058a-49ee-afc4-ccf2aad250b6@ug1.scm.com
CSeq: 2 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7ad67-1dfd5d09-e17c703
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:880880@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 216

v=0
o=880880 100023 100023 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 14026 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

### Barge-in/Intrusion/Multi-device Messages

Barge-in message will be displayed here. In case of any other service such as Intrusion and Multi-device, Call processing is same but only difference are feature code and Subject header.

*89: [Conference] Feature Code
2005: User who wants to barge in
2004: User who is barged in

```
F1.
INVITE sip:*89@ug1.scm.com:5090 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=64649a3f-38dd-4cad-8d7e-45b4a99ffd98
To: <sip:*89@ug1.scm.com>
Call-ID: b47212d2-7300-438b-9200-b5551eddd6fa@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7b7e6-1e265b69-19a872ba
Allow: INVITE,ACK,BYE,CANCEL,INFO,REFER,SUBSCRIBE,NOTIFY,UPDATE,PUBLISH
Subject: <CINT>::[2004]
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 339

v=0
o=2005 1340822843 0 IN IP4 165.213.176.85
s=Samsung IP PBX
c=IN IP4 165.213.176.85
t=0 0
a=sendrecv
m=audio 20156 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F2.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=64649a3f-38dd-4cad-8d7e-45b4a99ffd98
To: <sip:*89@ug1.scm.com>;tag=b4fd9bb8-0-13e2-50022-7b7e6-1857738f-7b7e6
Call-ID: b47212d2-7300-438b-9200-b5551eddd6fa@ug1.scm.com
CSeq: 1 INVITE
Via:  SIP/2.0/UDP  10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-7b7e6-1e265b69-19a872ba
Contact: <sip:890020@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890020@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45048 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F3.
INVITE sip:890020@ug1.scm.com:5090 SIP/2.0
From: "2004"<sip:2004@ug1.scm.com>;tag=9603c02c-2dae-486a-a6b1-3b6ca3f167cb
To: "890020"<sip:890020@ug1.scm.com>
Call-ID: 7ba97244-c1a6-410a-a651-649aceb389a1@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7b7e6-1e265e4d-42bed014
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2004@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 259

v=0
o=2004 100089 100089 IN IP4 10.254.176.126
s=Samsung IP PBX
c=IN IP4 10.254.176.126
t=0 0
m=audio 35999 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F4.
SIP/2.0 200 OK
From: "2004"<sip:2004@ug1.scm.com>;tag=9603c02c-2dae-486a-a6b1-3b6ca3f167cb
To: "890020"<sip:890020@ug1.scm.com>;tag=b4fda218-0-13e2-50022-7b7e6-17563722-7b7e6
Call-ID: 7ba97244-c1a6-410a-a651-649aceb389a1@ug1.scm.com
CSeq: 1 INVITE
Via:  SIP/2.0/UDP  10.254.176.126:5060;received=127.0.0.1;rport=5060;branch=z9hG4bK-7b7e6-
1e265e4d-42bed014
Contact: <sip:890020@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890020@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45048 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F5.
INVITE sip:890020@10.254.176.126:5090 SIP/2.0
From: "2004"<sip:2004@ug1.scm.com>;tag=9603c02c-2dae-486a-a6b1-3b6ca3f167cb
To: "890020"<sip:890020@ug1.scm.com>;tag=b4fda218-0-13e2-50022-7b7e6-17563722-7b7e6
Call-ID: 7ba97244-c1a6-410a-a651-649aceb389a1@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7b7e7-1e265f2a-11a3cfb
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2004@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 285

v=0
o=2004 1340776924 1 IN IP4 165.213.80.94
s=Samsung IP PBX
c=IN IP4 165.213.80.94
t=0 0
a=sendrecv
m=audio 20098 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F6.
SIP/2.0 200 OK
From: "2004"<sip:2004@ug1.scm.com>;tag=9603c02c-2dae-486a-a6b1-3b6ca3f167cb
To: "890020"<sip:890020@ug1.scm.com>;tag=b4fda218-0-13e2-50022-7b7e6-17563722-7b7e6
Call-ID: 7ba97244-c1a6-410a-a651-649aceb389a1@ug1.scm.com
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-7b7e7-1e265f2a-11a3cfb
Contact: <sip:890020@10.254.176.126:5090>;isfocus
Reason: SIP ;cause=200 ;text="Conference"
Content-Type: application/sdp
Content-Length: 234

v=0
o=890020@ug1.scm.com 0 0 IN IP4 127.0.0.1
s=Setup Conf
c=IN IP4 10.254.176.126
t=0 0
m=audio 45048 RTP/AVP 8 101
c=IN IP4 10.254.176.126
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

## Termination of one member during 3-party basic conference Messages

In case 2004 terminates the conference during the conference among 2003/2004/2005.

```
F1.
BYE sip:890020@10.254.176.126:5090 SIP/2.0
From: "2004"<sip:2004@ug1.scm.com>;tag=9603c02c-2dae-486a-a6b1-3b6ca3f167cb
To: "890020"<sip:890020@ug1.scm.com>;tag=b4fda218-0-13e2-50022-7b7e6-17563722-7b7e6
Call-ID: 7ba97244-c1a6-410a-a651-649aceb389a1@ug1.scm.com
CSeq: 3 BYE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7be0c-1e3e60c5-6249b283
Reason: Q.850 ;cause=16 ;text="Normal Release"
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Content-Length: 0
```

```
F2.
SIP/2.0 200 OK
From: "2004"<sip:2004@ug1.scm.com>;tag=9603c02c-2dae-486a-a6b1-3b6ca3f167cb
To: "890020"<sip:890020@ug1.scm.com>;tag=b4fda218-0-13e2-50022-7b7e6-17563722-7b7e6
Call-ID: 7ba97244-c1a6-410a-a651-649aceb389a1@ug1.scm.com
CSeq: 3 BYE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-7be0c-1e3e60c5-6249b283
Content-Length: 0
```

```
F3.
REFER sip:2003@10.254.176.126:5060 SIP/2.0
From: <sip:890020@ug1.scm.com>;tag=b4fda548-0-13e2-50022-7b7e7-3eaca28a-7b7e7
To: "2003"<sip:2003@ug1.scm.com>;tag=d17d97ed-02f9-47d3-b3e8-10a594e2f966
Call-ID: d7173c6f-36f1-422a-92bf-5714441f0eb0@ug1.scm.com
CSeq: 1 REFER
Via: SIP/2.0/UDP 192.168.113.50:5090;received=10.254.176.126;branch=z9hG4bK-7be0c-
1e3e60fd-73674ffc
Refer-To: <sip:2005@ug1.scm.com?Replaces=b47212d2-7300-438b-9200-
b5551eddd6fa%40ug1.scm.com%3Bto-tag%3D64649a3f-38dd-4cad-8d7e-
45b4a99ffd98%3Bfrom-tag%3Db4fd9bb8-0-13e2-50022-7b7e6-1857738f-7b7e6>
Referred-By: <sip:890020@ug1.scm.com>
Max-Forwards: 70
Contact: <sip:890020@10.254.176.126:5090>
Date: Wed, 27 Jun 2012 15:30:57 GMT
Content-Length: 0
```

```
F4.
BYE sip:890020@10.254.176.126:5090 SIP/2.0
From: "2003"<sip:2003@ug1.scm.com>;tag=d17d97ed-02f9-47d3-b3e8-10a594e2f966
To: "890020"<sip:890020@ug1.scm.com>;tag=b4fda548-0-13e2-50022-7b7e7-3eaca28a-7b7e7
Call-ID: d7173c6f-36f1-422a-92bf-5714441f0eb0@ug1.scm.com
CSeq: 3 BYE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-7be0c-1e3e612b-7ff6a148
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Content-Length: 0
```

### 3.3.9 SMS Server

#### 3.3.9.1 SIP usage

- **‘From’ Header**
  Extension number and host information is contained.
  *From: sip: extension number@host*
- **‘To’ Header /Request URI**
  Called number and host information is contained.
  *To: ‘Voice Mail’ < sip:called number@host*
- **P-Asserted-Identity**
  Then main number of SMS server and host information is contained.
  *P-Asserted-Identity: sip:main number of SMS server@host*
- **‘Reply-To’ Header**
  Reply number and host information is contained.
  *To: < sip:reply number@host*

### 3.3.9.2 Call Flows

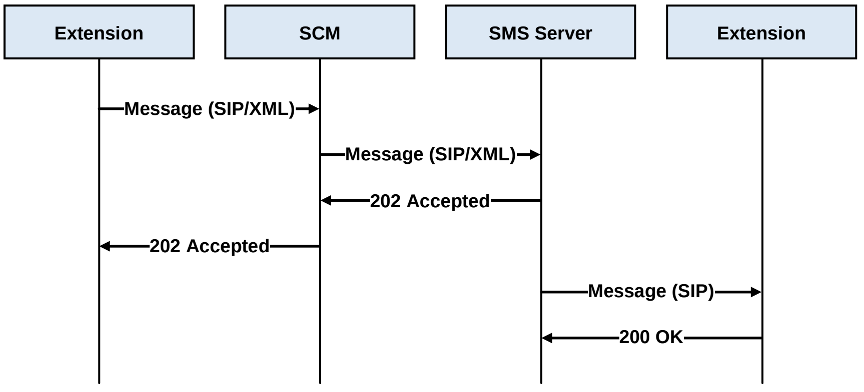



**Figure 3.57 SMS message Flow**

1) Extension input and sends the message, reply number, and called number.
2) SCM recognized and sends the Message to SMS server.
3) SMS server response with '202 accepted', if the Message is including the correct information.
4) SCM sends '202 accepted' to extension.

### 3.3.9.3 SIP Messages

```
1. MESSAGE

Request-Line: MESSAGE sip:1549@ismc.ann.com SIP/2.0
From: <sip:07041615808@kt.co.kr>;tag=10117650-a05a8e7d-13c4-5506-263-
2c44317d-263
To: "5809"<sip:07041615809@ismc.ann.com>
Call-ID: 10117650-a05a8e7d-13c4-5506-263-76a003f7-263
CSeq: 10004 MESSAGE
Via: SIP/2.0/UDP 125.142.90.73:5060;rport;branch=z9hG4bK-16097-
5614f77-7c9f09d5
P-Asserted-Identity: <sip:07041615807@kt.co.kr>
Reply-To: <sip:07041615808@ismc.ann.com>
Max-Forwards: 70
Contact: <sip:07041615808@125.142.90.73:5060>
Content-Type: text/xml
Content-Length: 186

<?xml version="1.0"?><env:Envelope
xmlns:env="http://schemas.xmlsoap.org/soap/envelope/"><env:Body><Deliv
erReq><smsMessage>ABCDEFGHI</smsMessage></DeliverReq></env:Body></env:
Envelope>

2. 202 ACCEPTED

Status-Line: SIP/2.0 202 Accepted
Via: SIP/2.0/UDP 125.142.90.73:5060;rport;branch=z9hG4bK-16097-
5614f77-7c9f09d5
To: "5809"<sip:07041615809@ismc.ann.com>;tag=e9ce0f86Ym
From: <sip:07041615808@kt.co.kr>;tag=10117650-a05a8e7d-13c4-5506-263-
2c44317d-263
Call-ID: 10117650-a05a8e7d-13c4-5506-263-76a003f7-263
CSeq: 10004 MESSAGE
Contact: <sip:ip-smsc@125.144.48.136:5060>
Server: IP-SMSC Server v1.0
User-Agent: dinnoVan SIP Stack 1.1
Organization: SIP Stack By dinnoVan
Content-Length:    0
```

## 3.3.10 External Ringback Tone Server

This chapter is explaining interworking specification of external ringback tone server SCM is providing the coloring service to Held side in case of Transfer. Coloring server should send 200 OK with SDP which is including his all capability (RTP/sRTP/video) when he receives NoSDP INVITE message from SCM. And he could play the coloring source based on SDP information of ACK message.
You can refer to below call flows and SIP message for the more details.

### 3.3.10.1 SIP usage

**Outbound Call (SCM to Coloring Server)**

- **'From' Header**
  Calling side information is contained.
  *From: <sip:calling number@host>*
- **'To' Header /Request URI**
  Called side information is contained.
  *To: sip:called number@host*
- **'Diversion' Header**
  If Voice mail/Recording/AA/Coloring services are supported from the same application server on the same device, you can distinguish those services by using reason field. SCM will mark the reason as coloring in INVITE message.
  *Diversion: <sip:called number@host>;reason='coloring'*

### 3.3.10.2 Call Flows

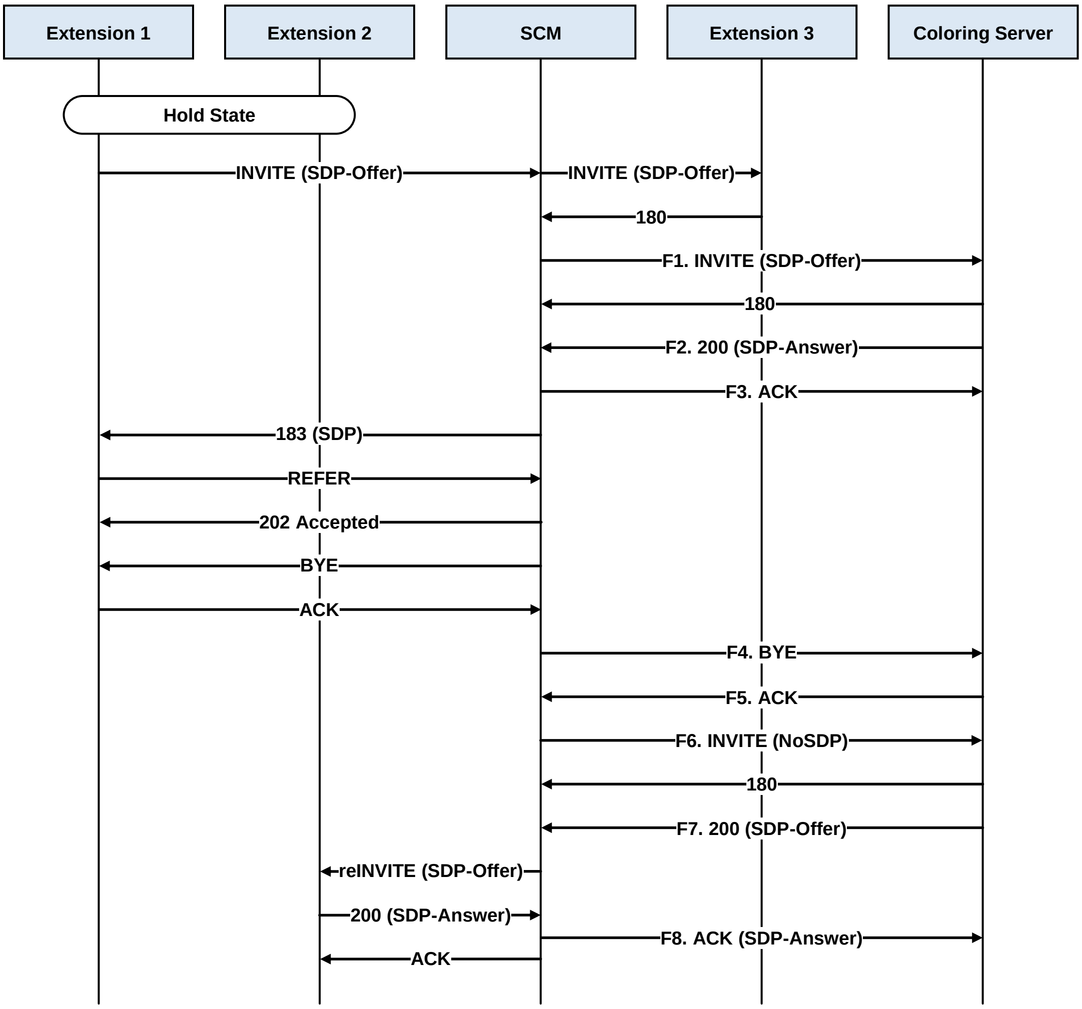



**Figure 3.58 External Ringback Tone Server Call Flow**

1) Extension 1 tries to Semi-consultation Transfer to Extension 3 during the call between Extension 1and Extension 2.
2) When SCM receives 18x from Extension 3 which is getting coloring service, then he tries to make a call to coloring server.
3) When SCM receives 200 OK from coloring server, then he sends 183 with SDP to Extension 1.
4) When Transfer is done for Extension 1 then SCM will send BYE to coloring server to finish the coloring service.
5) In case [Transfer Ringback Tone] is set on SCM, coloring should be provided to Extension 2 which is Held status. SCM sends NoSDP INVITE message to coloring server.
6) When coloring server receives NoSDP INVITE, then he should send his all Capability (RTP/sRTP/video) on SDP of 200 OK
7) SCM will send reINVITE with SDP information which is from coloring server to Extension 2.

8) SCM sends ACK to coloring server. This ACK will include SDP information which was included 200 OK from Extension 2.

Refer to F1~F3 in the figure 3.53 for the Basic coloring flow.

### 3.3.10.3 SIP Messages

```
F1.
INVITE sip:2003@ug1.scm.com:5060 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=f9950899-48df-4195-80fd-216593140b09
To: "2003"<sip:2003@ug1.scm.com>
Call-ID: d5bade0c-d5a6-4c40-81e9-b8157cdb379b@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-14f81-51e9340-5b241dd8
P-Asserted-Identity: "2005" <sip:2005@ug1.scm.com>
Diversion: <sip:2003@ug1.scm.com>;reason="coloring"
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 339

v=0
o=2005 1340402876 0 IN IP4 165.213.176.85
s=Samsung IP PBX
c=IN IP4 165.213.176.85
t=0 0
a=sendrecv
m=audio 20058 RTP/AVP 0 8 18 9 102 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:102 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

```
F2.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=f9950899-48df-4195-80fd-216593140b09
To: "2003"<sip:2003@ug1.scm.com>;tag=47947b8-0-13c4-55013-204411-5e3d5add-204411
Call-ID: d5bade0c-d5a6-4c40-81e9-b8157cdb379b@ug1.scm.com
CSeq: 1 INVITE
Allow: INVITE,ACK,CANCEL,BYE,REFER,INFO
User-Agent: SOLUTECH
Supported: replaces
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-14f81-51e9340-5b241dd8
Contact: <sip:2003@10.254.168.204:5060>
Content-Type: application/sdp
Content-Length: 212

v=0
o=IVR 2890844342 2890844543 IN IP4 10.254.168.204
s=CH004
c=IN IP4 10.254.168.204
t=0 0
m=audio 10206 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F3.
ACK sip:2003@10.254.168.204:5060 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=f9950899-48df-4195-80fd-216593140b09
To: "2003"<sip:2003@ug1.scm.com>;tag=47947b8-0-13c4-55013-204411-5e3d5add-204411
Call-ID: d5bade0c-d5a6-4c40-81e9-b8157cdb379b@ug1.scm.com
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-14f81-51e93cb-2da5a1ba
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2005@10.254.176.126:5060>
Content-Length: 0
```

```
F4.
BYE sip:2003@10.254.168.204:5060 SIP/2.0
From: "2005"<sip:2005@ug1.scm.com>;tag=f9950899-48df-4195-80fd-216593140b09
To: "2003"<sip:2003@ug1.scm.com>;tag=47947b8-0-13c4-55013-204411-5e3d5add-204411
Call-ID: d5bade0c-d5a6-4c40-81e9-b8157cdb379b@ug1.scm.com
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-14f87-51ea8c5-19804286
Reason: Q.850 ;cause=16 ;text="Normal Release"
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Content-Length: 0
```

```
F5.
SIP/2.0 200 OK
From: "2005"<sip:2005@ug1.scm.com>;tag=f9950899-48df-4195-80fd-216593140b09
To: "2003"<sip:2003@ug1.scm.com>;tag=47947b8-0-13c4-55013-204411-5e3d5add-204411
Call-ID: d5bade0c-d5a6-4c40-81e9-b8157cdb379b@ug1.scm.com
CSeq: 2 BYE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-14f87-51ea8c5-19804286
Supported: replaces
Content-Length: 0
```

```
F6.
INVITE sip:2003@ug1.scm.com:5060 SIP/2.0
From: "2004"<sip:2004@ug1.scm.com>;tag=8ff725ea-3ecf-4ff3-a99c-f505e13d9c2d
To: "2003"<sip:2003@ug1.scm.com>
Call-ID: b20d935f-90a7-492c-8a5b-9f8eaf68efbf@ug1.scm.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-14f83-51e9a41-1350e263
P-Asserted-Identity: "2004" <sip:2004@ug1.scm.com>
Diversion: <sip:2003@ug1.scm.com>;reason="coloring"
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2004@10.254.176.126:5060>
Content-Length: 0
```

```
F7.
SIP/2.0 200 OK
From: "2004"<sip:2004@ug1.scm.com>;tag=8ff725ea-3ecf-4ff3-a99c-f505e13d9c2d
To: "2003"<sip:2003@ug1.scm.com>;tag=4797d58-0-13c4-55013-204412-36cfbe41-204412
Call-ID: b20d935f-90a7-492c-8a5b-9f8eaf68efbf@ug1.scm.com
CSeq: 1 INVITE
Allow: INVITE,ACK,CANCEL,BYE,REFER,INFO
User-Agent: SOLUTECH
Supported: replaces
Via: SIP/2.0/UDP 10.254.176.126:5060;rport=5060;branch=z9hG4bK-14f83-51e9a41-1350e263
Contact: <sip:2003@10.254.168.204:5060>
Content-Type: application/sdp
Content-Length: 212

v=0
o=IVR 2890844342 2890844543 IN IP4 10.254.168.204
s=CH005
c=IN IP4 10.254.168.204
t=0 0
m=audio 10208 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

```
F8.
ACK sip:2003@10.254.168.204:5060 SIP/2.0
From: "2004"<sip:2004@ug1.scm.com>;tag=8ff725ea-3ecf-4ff3-a99c-f505e13d9c2d
To: "2003"<sip:2003@ug1.scm.com>;tag=4797d58-0-13c4-55013-204412-36cfbe41-204412
Call-ID: b20d935f-90a7-492c-8a5b-9f8eaf68efbf@ug1.scm.com
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.254.176.126:5060;rport;branch=z9hG4bK-14f83-51e9b2f-38c71b2e
Max-Forwards: 70
User-Agent: SAMSUNG SCM 3.2.400
Contact: <sip:2004@10.254.176.126:5060>
Content-Type: application/sdp
Content-Length: 235

v=0
o=2004 1340402879 3 IN IP4 165.213.80.94
s=Samsung IP PBX
c=IN IP4 165.213.80.94
t=0 0
a=sendrecv
m=audio 20050 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

## 3.3.11 MS OCS

In this chapter basic flow for Forking of SCM is explained. You can refer to 'Interoperability between Microsoft Office Communications Server' document for the more details.

### 3.3.11.1 SIP usage

#### Inbound Call (SCM to Mediation Server)

- 'From' URI
  Mediation Server accepts only SIP URI with user=phone.
  Examples:
  1) SIP URI with E.164 phone number
     sip:+14257123456@x.com;user=phone
  2) SIP URI with a non-E.164 phone number without a phone-context attribute
     sip:123456@x.com;user=phone
  3) SIP URI with a non-E.164 phone number with a phone-context attribute
     sip:123456;phone-context=y@x.com;user=phone
- 'To'/Request URI
  Mediation Server accepts only SIP URI with user=phone.
  Examples:
  1) SIP URI with E.164 phone number (with/without ext parameter)
     sip:+14257123456@x.com;user=phone
  2) SIP URI with a non E.164 phone number without a phone-context attribute
     sip:123456@x.com;user=phone
  3) SIP URI with a non E.164 phone number with a phone-context attribute

#### Outbound Call (Mediation Server to SCM)

- 'From' URI
  Mediation Server sends a SIP URI with/without user=phone.
  Examples:
  1) SIP URI with E.164 phone number
     sip:+14257123456@x.com;user=phone
  2) SIP URI with a non E.164 phone number without a phone-context attribute
     sip:123456@x.com;user=phone
  3) SIP URI with an alias (Note: this SIP URI is used if call is received from a federated user without a phone number or an Office Communications Server conference URI (a call from the A/V Conferencing Server).
     sip:alice@x.com
- 'To'/Request URI
  Mediation Server sends a SIP URI with user=phone.
  Examples:

1) SIP URI with E.164 phone number (with or without the ext parameter)
   sip:+14257123456@x.com;user=phone
2) SIP URI with a non-E.164 phone number without a phone-context attribute
   sip:123456@x.com;user=phone

- 'Reason' Header
  - When the Forking call responses then another side sends CANCEL to clear the call. Below Reason header is added in the CANCEL message.
    Reason: SIP ;cause=200 ;text='Call completed elsewhere'
  - If the incoming call to SCM user is forwarded then the Forking call to MS OCS sends CANCEL to clear the call. Below Reason header is added in the CANCEL message.
    Reason: SIP; cause=181 ;text='Call is being forwarded'
- MS-CALL-SOURCE proprietary Header
  In case Forking is happen from OCS, **ms-call-source:ms-rtc** header is added.

### 3.3.11.2 Call Flows

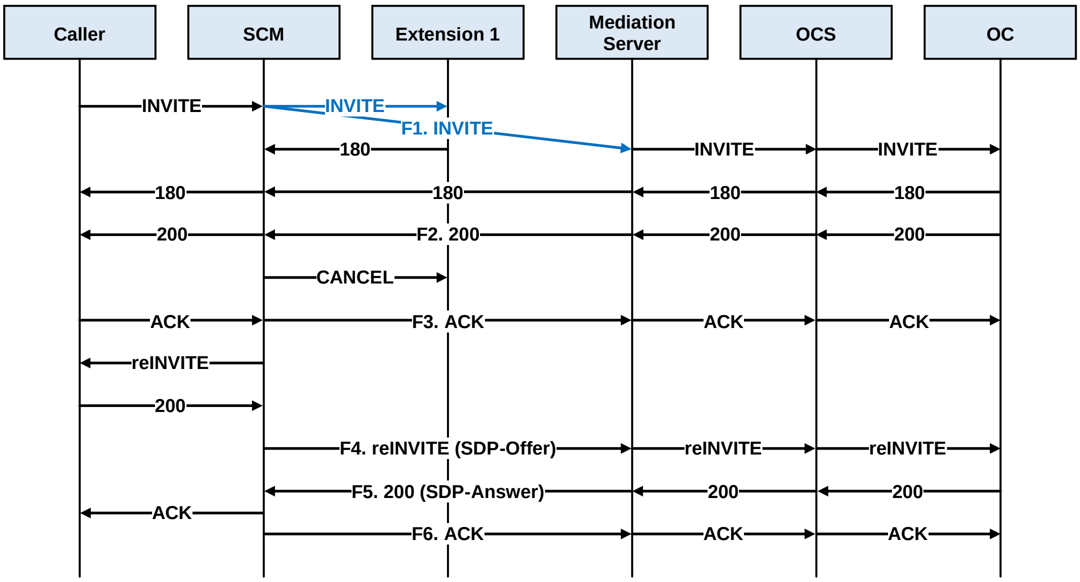



**Figure 3.59 SCM Forking Flow through MS OCS**

Condition: Extension 1 and OC member are set as multi-ring feature and Extension 1 is the master.

1) When SCM receives INVITE, then SCM checks called party and whether multi-ring is set or not.
2) SCM sends INVITE not only Extension 1 which is called party, but also OC member. When OC member response, then SCM sends CANCEL to Extension 1 to clear the call.

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# ABBREVIATION

## C

| | |
|---|---|
| CLI | Command Line Interface |
| CPU | Central Processing Unit |

## D

| | |
|---|---|
| DDR | Double Data Rate |
| DVD | Digital Video Disc |

## F

| | |
|---|---|
| FTP | File Transfer Protocol |

## H

| | |
|---|---|
| HA | High Availability |
| HDD | Hard Disk Drive |

## I

| | |
|---|---|
| IP | Internet Protocol |

## L

| | |
|---|---|
| LAN | Local Area Network |

## O

| | |
|---|---|
| ODD | Optical Disc Drive |

## R

| | |
|---|---|
| RAM | Random Access Memory |
| ROM | Read Only Memory |

## S

| | |
|---|---|
| SCM | Samsung Communication Manager |

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Auto Doc.code Ver. 1.0

SCM Express

# SIP Interoperability Guide