OfficeServ™ 7000 Series

SIP Services Technical Manual

SAMSUNG

Publication Information

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1. Introduction

1.1. Overview

This document is written in order to give guidelines on SIP (Session Initiation Protocol) trunking and station features in OfficeServ system (SIP Server) made by Samsung Telecommunications. SIP may become OfficeServ's main VoIP protocol. Readers of this document are supposed to have at least the minimum knowledge level in programming the OfficeServ system such as basic MMC settings. Cross-referencing *Part 2.* ***SIP Trunking*** and ***Part 3. SIP Station*** may help to better understand SIP trunking and station features described in this document. This may not be necessary if adequate knowledge of SIP is already obtained. As in the case of SIP trunking, the main purpose of this document is not for introducing SIP, rather for informing how to operate SIP station features implemented in OfficeServ system. Therefore, readers who want to have an in-depth knowledge on SIP should refer to RFC3261, which is the base document of SIP.

1.2. Definitions

Following definitions are made to enhance reader's understanding.

- **SIP Trunk Mode:** OfficeServ's SIP operation mode in which OfficeServ acts as a SIP User Agent Client (UAC) and registers to external SIP Server to communicate with other external SIP UACs.

- **SIP Station Mode:** OfficeServ's SIP operation mode in which OfficeServ acts as a SIP Server (or Registrar) so that it can have standard SIP terminals (or UAC) as its internal terminals. However, unlike non-sip terminal interfaces, in this SIP station domain all the signaling messages and protocol follow the standard SIP that IETF recommends.

- **User Agent Client (UAC):** A user agent client is a logical entity that creates a new request. The role of UAC lasts only for the duration of that transaction. In other words, if a piece of software initiates a request, it acts as a UAC for the duration of that transaction. If it receives a request later, it assumes the role of a user agent server for the processing of that transaction.

- **User Agent Server (UAS):** A user agent server is a logical entity that receives a new request from UAC. As in the case of UAC described above, the role of UAS lasts only for the duration of that specific transaction. Therefore, at any time, a UAS can be a UAC according to whether it receives a request or sends a request.

- **Non-Samsung SIP Terminal:** Any SIP terminals (or stations) that were not manufactured by Samsung Electronics. i.e., Cisco 7960, X-Lite and etc.

1.3. Overall Configuration

As shown in Figure 1, the SIP interfaces (remarked as dashed lines in each circle) in each OfficeServ domain are for SIP Station Mode, and external SIP interfaces that are connected to external SIP Servers are for SIP Trunk Mode.

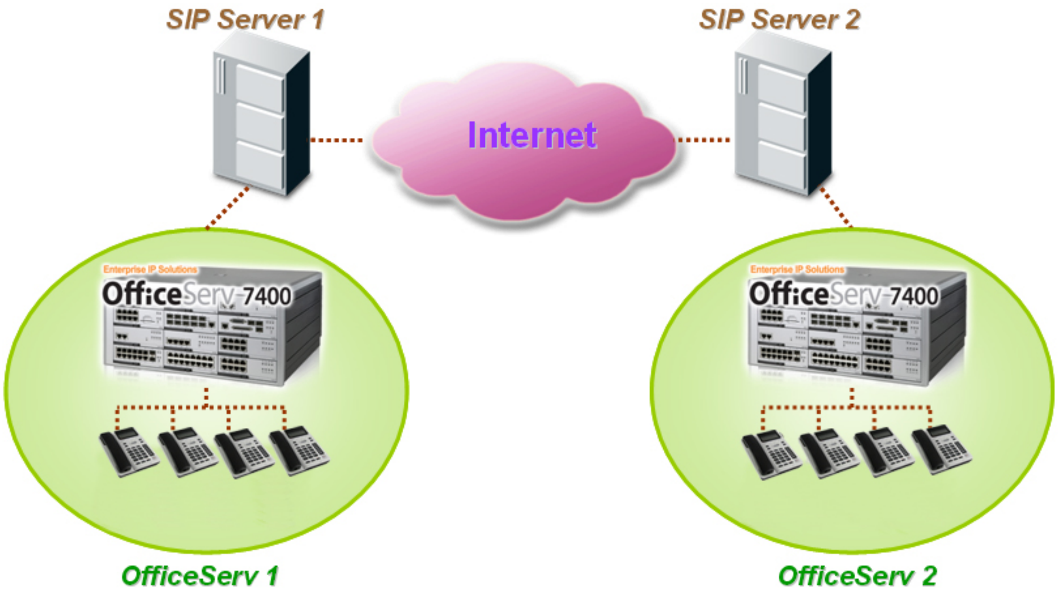

Figure 1. Overall Configuration for SIP trunking mode and SIP station mode

SIP Trunk Capacities			
System	**7100**	**7200**	**7400**
Max Virtual Trunk Slots	**3**	**4**	**4**
Channels per slot	**8**	**8**	**32**
Maximum SIP Trunks	**24**	**32**	**128**
SIP Station Capacities			
System	**7100**	**7200**	**7400**
Max Virtual Station Slots	4	4	4
Channels per slot	8	8	32
Maximum SIP Stations	**32**	**32**	**128**

Figure 2. SIP Trunk and Station Capacities

When using SIP trunks and stations, MGI channel resources are required to convert from analog and TDM signaling to SIP and or H.323 signaling and vice-versa. In some cases, no MGI channels are required. In other cases, one or more MGI channels are required. The table below explains when and how many MGI resources are required to support audio conversations to SIP phones, SIP and H.323 trunks.

Phone	SIP Trunking	Analog/Digital Trunking	H.323 Trunking
IP Phone	2	1	2
Digital Phone	1	0	1
Single Line Telephone	1	0	1
SIP Phone	2	1	2
WiFi Phone	2	1	2

Figure 3. MGI Usage Table

1.4. SIP (Session Initiation Protocol)

SIP is an application-layer control protocol that can establish, modify, and terminate multimedia sessions (conferences) such as Internet telephony calls. SIP can also invite participants into already existing sessions, such as multicast conferences. Media can be added to (and removed from) an existing session.

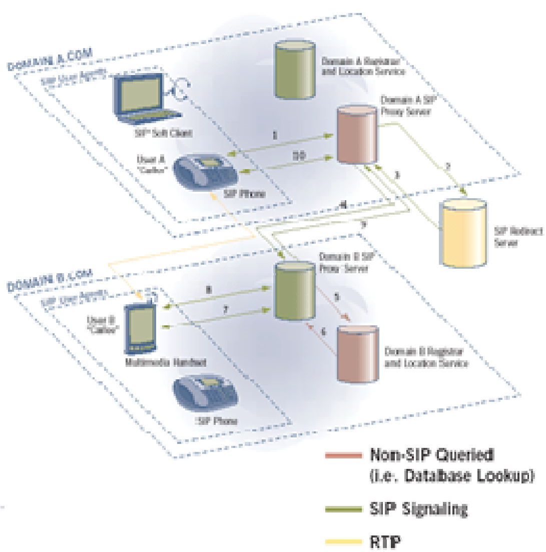

Figure 4. SIP Call Scenario

Standard SIP consists of mainly following 4 elements:

■ User Agent Client (UAC): A user agent client is a logical entity that creates a new request. The role of UAC lasts only for the duration of that transaction. In other words, if a piece of software initiates a request, it acts as a UAC for the duration of that transaction. If it receives a request later, it assumes the role of a user agent server (UAS) for the processing of that transaction.

■ User Agent Server (UAS): A user agent server is a logical entity that generates a response to a SIP request. The response accepts, rejects, or redirects the request. This role lasts only for the duration of that transaction. In other words, if a piece of software responds to a request, it acts as a UAS for the duration of that transaction. If it generates a request later, it assumes the role of a user agent client for the processing of that transaction.

■ Registrar: A registrar is a server that accepts REGISTER requests from UAC and places the information for location services.

■ SIP Server (or Proxy Server): A server is a network element that receives requests in order to service them and sends back responses to those requests. Examples of servers are proxies, user agent servers, redirect servers, and registrars.

1.5. SIP Trunking vs. SIP Peering

OfficeServ system supports both SIP trunking and SIP peering. The main difference is whether to use a SIP server or not. If SIP messages are transmitted via an intermediary SIP server, we call this type of SIP connection, SIP trunking. Meanwhile, if SIP messages are directly transmitted between two end SIP UAs, it is SIP peering.

Table 1. SIP Trunking vs. SIP Peering

	SIP Trunking	SIP Peering
SIP Server Use	Use	No Use
Authentication	REGISTER	OPTIONS
Message Outbound	SIP Server	Peer
DNS	Use	No Use
Related MMC No.	832, 837	832, 833

1.5.1. Locating SIP Server

In SIP trunking mode, OfficeServ can locate an outbound SIP server either by using DNS query or direct IP designation. If a direct IP is designated, OfficeServ sets the address as an outbound SIP Server's IP address. Instead, when an outbound proxy server's domain name is provided with DNS server's IP address, OfficeServ automatically triggers DNS query and fetches the IP addresses of the corresponding domain name from DNS servers. Once outbound server is set, all the SIP outbound messages from OfficeServ will send to the server.

For some SIP carriers that require separate registrar server from outbound SIP server, OfficeServ is able to locate separate registrar server and its mechanism is the same with the case of locating SIP server. If separate registrar is set, OfficeServ sends out all the REGISTER messages to the registrar. Note that other SIP messages are still sent to the outbound server.

DNS query feature is provided in OfficeServ 7100 and 7400 systems but not in OfficeServ 7200.

1.5.2. SIP Trunking Functionalities

SIP trunking functionality on the OfficeServ has two categories: Basic and Supplementary.

Table 2. SIP functionality comparison

Basic Functions	Supplementary Functions
• Registration • Basic Call Setup	• Hold/Resume • Music on Hold • Consultation Call • Transfer (Consultation/Blind) • Call Forward (All/Busy/No-Answer) • DND • MWI • Conference • Call Waiting • Call Pickup • Call Park

Basic SIP trunking functions in the OfficeServ have been implemented based on SIP standard, and they have been tested with various SIP carriers whose SIP servers were manufactured by many different 3rd party vendors. OfficeServ's SIP supplementary service functions, however, were developed and tested mainly using BroadSoft Inc's Soft Switch (a SIP server), and thus there may be some compatibility problems when interoperating with other SIP servers made by different vendors. Another reason why we can not guarantee compatibility of supplementary functions is that each different SIP UA manufacturer can have each different SIP message handling scheme which does not matches with schemes implemented in OfficeServ system. For this reason, some features that are working fine with a certain SIP server may not work properly when interoperating with other servers.

1.5.3. SIP Peering Functionalities

Unlike SIP trunking which normally depends on SIP server's capability, SIP peering functionalities are mostly depending on each participating SIP UA's capability. In many cases, therefore, supplementary features in SIP peering session are comparatively limited due to different SIP specification implemented in each different SIP UA.

1.5.4. Locating SIP Peer

In SIP peering mode, SIP peer's location should be known to OfficeServ in order to send out its SIP messages by setting IP address of the peer.

1.6. Multiple SIP Carriers

Currently OfficeServ system can interacts with **only one SIP carrier at a time**, but it has database frame which is able to contain[1] maximum 4 SIP carrier profile data in its MMC837. Each profile database designates each different SIP carrier and by setting the SIP SERVER field of a certain profile to 'ENABLE' OfficeServ sets the corresponding SIP carrier as its default SIP carrier. Remember that only one SIP carrier can active at a certain time.

1.7. SIP Station

When implementing and testing the supplementary services listed in Table 1 (sec. 1.1), we have tried to adapt the standard SIP call flow and message formats that IETF recommended. At the same time, we also tried to avoid cases where SIP call flows should be certain system or terminal dependent. Thereby, OfficeServ is able to accommodate as many other types of SIP terminals as possible. For this purpose, OfficeServ's SIP call flows were designed to emulate the functionality of BroadWorks Inc's SoftSwitch, one of the most widely used SIP Proxy Servers in the industry.

Although each different manufacturer may have different call flow or message format, interoperability between OfficeServ and 3rd party vendor products can be an issue in some cases. Currently OfficeServ OfficeServ supports the supplementary SIP service features only for SIP terminals, which have actually been tested in Samsung's lab with the OfficeServ platforms. Note: Samsung cannot guarantee the successful operation of 3rd party SIP phones. Use at your own risk.

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[1] In later version, OfficeServ will be able to support multiple active SIP carriers at a time, which means it can decide which SIP carrier's outbound server to send SIP message to without manually changing a default active SIP carrier to another.

For the other SIP terminals not mention in the above table, interoperability testing has not been fully conducted to date. Additional interoperability test will be scheduled and conducted when necessary.

1.7.1. Samsung vs. Non-Samsung

OfficeServ system distinguishes Non-Samsung SIP terminals from Samsung-manufactured SIP terminals (not available in US). From the operator's perspective, the main difference between Non-Samsung SIP terminals and Samsung SIP terminals is that the Non-Samsung SIP terminals require an additional license key verification.

2. Preparation

2.1. Non-Samsung SIP Stations

Unlike using Samsung-manufactured SIP stations in the OfficeServ system, non-Samsung SIP stations need to have an additional license as shown in figure 2. Non-Samsung SIP Phone Count specifies the maximum number of non-Samsung SIP stations (i.e., Cisco 7060) that can be registered to the OfficeServ system as a station, and this number can not be larger than the SIP Stack Count. This means that the total SIP stack license count must be allocated to the services of non-Samsung SIP phones and SIP trunks. The following table is an example of how the SIP stack license count may be allocated in MMC 841.

Table 3. SIP functionality comparison

SIP STACK LICENSE KEY TOTAL	
SIP STACK TOTAL COUNT	10
Non-Samsung SIP	6
SIP Trunks	4
Samsung SIP (not in US)	0
IP-UMS/IVR (not in US)	0

2.1.1. Inserting a License Key

Once a valid license key is generated, operator can insert this license key to OfficeServ system using MMC 841 FEAT LICENSE KEY entry. As the format of license key is composed of 40 cryptic alphabet characters, it is RECOMMENDED to use the OfficeServ Installation Tool. This will make it much easier, faster and less likely to make a mistake when entering the license key.

License Key Status Check

When inserting a license key to OfficeServ system, operator can check the system's total SIP capacity and its assignment status through MMC 841's FEAT LICENSE STS.

Table 4. MMC841 Feat License Status Table

Status	Description
NSIP-S MAX	The maximum number of non-Samsung sip phones
NSIP-S USED	The number of non-Samsung SIP phones currently being registered
NSIP-S CONN	The number of non-Samsung SIP phones currently being connected
SSIP-S MAX	The maximum number of Samsung SIP phones
SSIP-S USED	The number of Samsung SIP phones currently being registered
SSIP-S CONN	The number of Samsung SIP phones currently being connected
SIP STACK	The maximum number of SIP Channel Capabilities

SIPP ALLOW option shows the maximum number of allowed SIP stations, and SIPP USED indicates the number of currently registered SIP stations. (Please refer to Chapter 3. Registration)

MMC 841 FEAT LICENSE STS simply displays the current license key status and each value can not be modified by operators.

2.1.2. Configuring SIP Channels

As mentioned before, operator can assign SIP channels according to system purpose. MMC 841 SIP STACK ALLOW entry provides an interface to SIP channel configuration. First of all, MAX COUNT shows the maximum number of available SIP connection including available non-Samsung SIP stations specified in the license key. These two values are fixed in the license key and can not be changed, however the number of SIP TRUNK and SIP PHONE are configurable within a range of FREE COUNT. The number specified in FREE COUNT entry means total available number that can be assigned to either SIP trunk lines or SIP stations, or both.

MAX COUNT = NON Samsung SIP + FREE COUNT + SIP TRUNK + Samsung SIP PHONE + [2](IP-UMS/IVR)

Note: In order to activate SIP TRUNK and SIP station connections, operator should set desired number for the corresponding entries.

2.2. Virtual Cabinet

OfficeServ system provides virtual cabinet a mechanism to enhance scalability. Using this virtual cabinet system, operator can have flexible system configuration. A virtual cabinet consists of multiple slots and each slot maps to an appropriate virtual cabinet card. OfficeServ system has default mappings for this virtual cabinet card configuration; however, for SIP, the virtual cabinets will need to be reconfigured to support the required number of SIP Services. The following sections show how.

2.2.1. Virtual SIP Trunk Card

The number of virtual SIP trunk card and each card's SIP capacity vary by OfficeServ system type. For example, OfficeServ 7400 can have maximum 4 SIP trunk cards in a system, and each card contains 32 SIP channel capacity (128 channels max). The OfficeServ 7200 has maximum 4 trunk cards and each card has 8 SIP channel capacity (32 channels max). The OfficeServ 7100 has maximum 3 trunk cards and each card has 8 SIP channel capacity (24 channels max). Therefore, the number of SIP trunk card's capacity limits the total SIP channel capacity of the system. Of course, as license key specification has a higher priority than virtual card settings, the total number of SIP channel capacity decided by virtual card setting can not exceed the total number of SIP channel capacity specified in the license key.

To configure virtual SIP trunk card, operator should use MMC857 where each virtual cabinet and its slots are mapped to appropriate virtual cards. Even though operator can make flexible card configuration, there is restriction that a certain types of cards that can be mapped to a certain slot number are limited. That is, a SIP trunk card can not be mapped to all number of slots. As each different OfficeServ system type has different slot configuration, operators should be aware of virtual cabinet and slot number to know where SIP trunk slot can be mapped. Refer to MMC 857 of the OfficeServ 7000 Series Manual for a detailed description of virtual slots configuration.

[2] IP-UMS/IVR (not available in US) also uses the SIP connection and should be counted if active in OfficeServ system, but this documents confines SIP usage to SIP trunk and SIP stations only.

2.2.2. Virtual SIP Station Card

Available SIP station numbers are shown in MMC 842. If virtual SIP station cards are not properly mapped in MMC 857, however, no number will be shown. So, programmer should map virtual SIP station card to an appropriate virtual slots in OfficeServ in order to register SIP station to OfficeServ system. Mapping virtual SIP station cards to slots are done the same way as mapping the SIP trunk card. For example, OfficeServ 7400 can have maximum 4 SIP station cards in a system, and each card contains 32 SIP channel capacity (128 channels max). The OfficeServ 7200 has maximum 4 station cards and each card has 8 SIP channel capacity (32 channels max). The OfficeServ 7100 has maximum 4 station cards and each card has 8 SIP channel capacity (32 channels max). Therefore, the number of SIP station card's capacity limits the total SIP channel capacity of the system.

Table 5. SIP Trunk and SIP Station Capacities

SIP Trunk Capacities			
System	**7100**	**7200**	**7400**
Max Virtual Trunk Slots	**3**	**4**	**4**
Channels per slot	**8**	**8**	**32**
Maximum SIP Trunks	**24**	**32**	**128**
SIP Station Capacities			
System	**7100**	**7200**	**7400**
Max Virtual Station Slots	4	4	4
Channels per slot	8	8	32
Maximum SIP Trunks	**32**	**32**	**128**

3. Registration

As in the case of OfficeServ system in SIP trunk mode, SIP stations also need to register to the SIP Server in order to communicate with other stations attached to an OfficeServ system. In SIP station mode, however, unlike SIP trunk mode in which OfficeServ acts as a UAC, OfficeServ acts as SIP Server (or [3]Registar) for SIP stations. As this registration process also follows SIP standard registrar specification, SIP Stations have to go through standard authorization and authentication process to be registered successfully to an OfficeServ Client. As shown below, Registration flows are all the same between Trunk Mode and Station Mode, except the OfficeServ's is now the Server (Registrar) and the SIP station is the Client. (UAC).

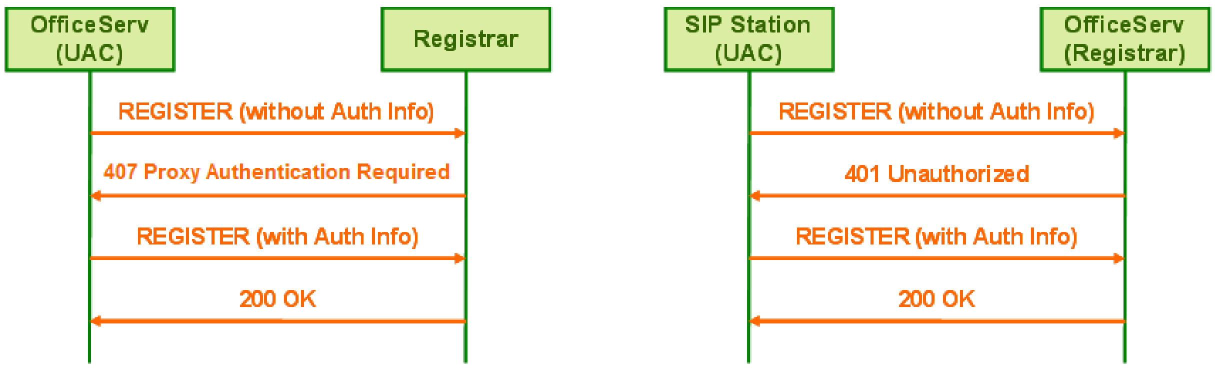

Figure 5. Trunk Mode Registration vs. Station Mode Registration

As shown in above call flow, SIP station registration process goes through 4 steps.

i. UAC sends a Register message without authentication information.
ii. Registrar gives back 401 response having authorization information.
iii. Upon receiving a 401 response, UAC creates a Register message again which contains valid authentication information.
iv. Registrar authorizes UAC's registration after confirming the authentication contained in the Register message.

3.1. Authentication and Authorization

Authentication and authorization are, in brief, about creating encryption value and confirming the value between UAC and SIP server. This encryption value can be made from the composition of username, password, and other authentication values. While UAC and SIP server publicly share the pair of username and password, other authentication values are created only by SIP server side using internal authentication values generating algorithm, and can be known to UAC when a 401 response message is transmitted from SIP server.

After receiving a 401 response message, UAC creates an encryption value, using username, password and other values as encryption entries, and put it into 'response' parameter in the authorization header in the subsequent

[3] A registrar is a server that accepts REGISTER requests from UAC and places the registration data for checking the status and location of UACs. As, more often than not, SIP proxy server and SIP registrar are implemented in a single SIP entity, in this document, we use the two different terms interchangeably according to its context.

REGISTER message. If this response parameter value matches with an encryption value created by SIP server, SIP server finally authorizes the UAC's registration. As both SIP server and UAC have the same encryption seed of username, password, and other authentication values, the encryption value contained in the authorization header should be identical to encryption value made by SIP server.

Among many authentication mechanisms for creating and confirming the encryption values, one of the most widely used is MD5 digest algorithm. This algorithm originated from HTTP's web authentication, which is normally used in logon processes of many web sites. The detailed explanation for the MD5 digest algorithm is beyond the scope of this document.

3.2. Registration Type

Please refer to Part 2. SIP Trunking and Part 3. SIP Stations.

OfficeServ™ 7000 Series

SIP Services

Part 2. SIP Trunking

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1. Registration Types

There are two types of Registration. One is Trunk Registration and the other is Residential Registration. The former is much more widely used in industry and thus when we refer Registration, it means Trunk Registration. OfficeServ system supports both of these registration types and which to use is set using 'REG PER USER' option entry in MMC837 in OfficeServ system; 'Disable' means Trunk Registration and 'Enable' means Residential Registration.

1.1. Trunk Registration

Trunk Registration means that the OfficeServ system does a single registration, whose credential data is shared by all the SIP connections between OfficeServ and an outbound SIP server.

1.1.1. Residential Registration

Residential Registration lets each individual user terminal attached to the OfficeServ have its own registration to a SIP server. This does not mean that each user terminal creates a registration message and directly sends it to the SIP server because many terminals other than 'Standard SIP terminals' can not make SIP register messages. So, the OfficeServ creates each SIP message using pre-assigned registration information, and does the SIP registration process on behalf of each end terminal.

1.1.2. SIP Trunking without Registration

Some SIP Servers do not require UA's registration at all. This type of server authenticates its interacting SIP UAs with their IP addresses and assigned usernames. That is, before starting interoperating with UA, SIP server administrator normally asks SIP UA's IP address and assigns predefined username, and stores the data somewhere in the server. This way, when any SIP message comes from the corresponding SIP UA, the SIP server checks the couple of source IP address and username and when matches with the data pre-stored in the server, it passes and rejects it otherwise.

Sending REGISTER message to a SIP server that does not require registration is meaningless and rather worsening network traffic, and thus it is always better not to send useless REGISTER message. When **leave USER NAME field in MMC837 blank**, **OfficeServ does not send REGISTER** message though SIP SERVER is enabled.

1.2. DNS Query

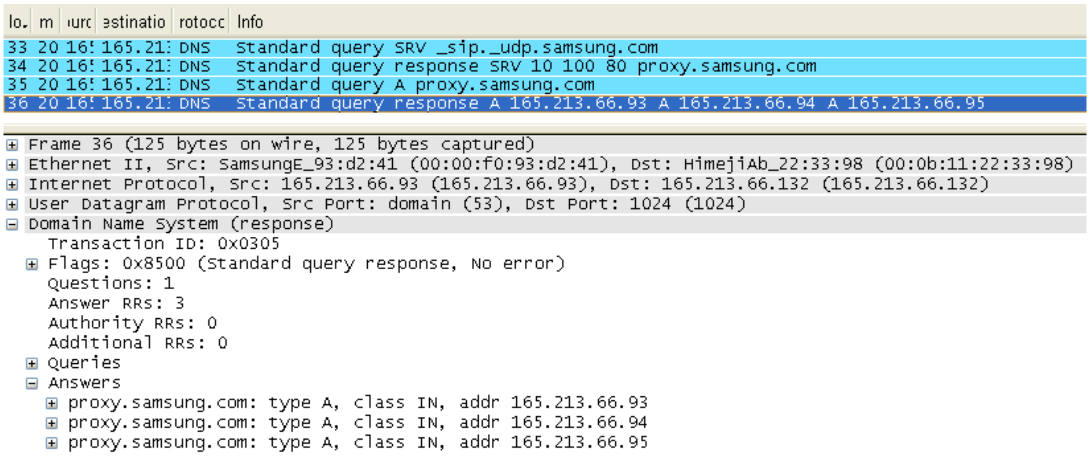


Figure 1. Capture of DNS Query By OfficeServ

OfficeServ is able to determine the location of the outbound SIP Server (registrar or proxy) based on the resolution of SRV and A queries. OfficeServ utilizes DNS servers specified in *DNS SERVER1* & DNS *SERVER2* fields to resolve SIP server names.

Above ethereal capture shows the example of how DNS query for a registrar or an outbound server is made using FQDN of 'samsung.com' from OfficeServ to a DNS server, and 3 IP addresses are fetched; '165.213.66.93', '165.213.66.94', and '165.213.66.95'.

1.3. Registration Example

1.3.1. OfficeServ MMC Settings

MMC837 SIP OPTIONS

ISP1
SIP SERVER: ENABLE
OUT PROXY: samsung.com
DNS SERVER1: 165.213.66.93
USER NAME: 82312794329
AUTH USER: 82312794329
AUTH PSWD: 1234
REG PER USR: DISABLE
TRK REG EXP: 001800

1.3.2. Message Samples

Reg F1

REGISTER sip:samsung.com:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1dd38a8-8442d5a5-13c4-50017-48111d3d-180f5e15-48111d3d
To: <sip:82312794329@samsung.com:5060>
Call-ID: 1dd907c-8442d5a5-13c4-50017-48111d3d-3488e341-48111d3d
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-48111d3d-82da38e6-4839db7a
Max-Forwards: 70
Supported: 100rel,replaces
Expires: 1800
Contact: <sip:82312794329@165.213.66.132:5060>

Reg F2

SIP/2.0 407 Proxy Authentication Required
To: <sip:82312794329@samsung.com:5060>
From: <sip:82312794329@samsung.com:5060>;tag=1dd38a8-8442d5a5-13c4-50017-48111d3d-180f5e15-48111d3d
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-48111d3d-82da38e6-4839db7a
CSeq: 1 REGISTER
Call-ID: 1dd907c-8442d5a5-13c4-50017-48111d3d-3488e341-48111d3d
Proxy-Authenticate: Digest
realm="165.213.66.93",qop="auth",algorithm="MD5",nonce="673d70c8cc47702469cf3aa94277c3df"
Content-Length: 0

Reg F3

```
REGISTER sip:samsung.com:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1dd38a8-8442d5a5-13c4-50017-48111d3d-180f5e15-
48111d3d
To: <sip:82312794329@samsung.com:5060>
Call-ID: 1dd907c-8442d5a5-13c4-50017-48111d3d-3488e341-48111d3d
CSeq: 2 REGISTER
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-48111d3d-82da3922-72cae6b8
Max-Forwards: 70
Supported: 100rel,replaces
Expires: 1800
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="673d70c8cc47702469cf3aa94277c3df",uri="sip:sam
sung.com:5060",response="5df531fe6bc866c82797449b4c1fa2ed",algorithm=MD5,cnonce="82da3922",qop=au
th,nc=00000001\r
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0
```

Reg F4

```
SIP/2.0 200 OK
To: <sip:82312794329@samsung.com:5060>;tag=10322
From: <sip:82312794329@samsung.com:5060>;tag=1dd38a8-8442d5a5-13c4-50017-48111d3d-180f5e15-
48111d3d
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-48111d3d-82da3922-72cae6b8
CSeq: 2 REGISTER
Call-ID: 1dd907c-8442d5a5-13c4-50017-48111d3d-3488e341-48111d3d
Contact: <sip:82312794329@165.213.66.132:5060>;expires=300
Content-Length: 0
```

2. SIP Trunking Services

This Chapter describes the detailed call scenarios involving the SIP service features in OfficeServ 7000 series system. There can be many different scenarios for each service depending on service types and thus, this document does not fully cover all the possible cases but some representative ones are listed for each category.

As mentioned in **section 1.3 SIP trunking vs. SIP peering**, OfficeServ MUST check its registration status first to start SIP trunking services except for the case that outbound SIP server does not require registration process. How to check the registration status is described in chapter **2 Registration**.

2.1. Basic Call Setup

Following call flow shows a typical SIP outbound trunk messages transmitted between OfficeServ system and a SIP server.

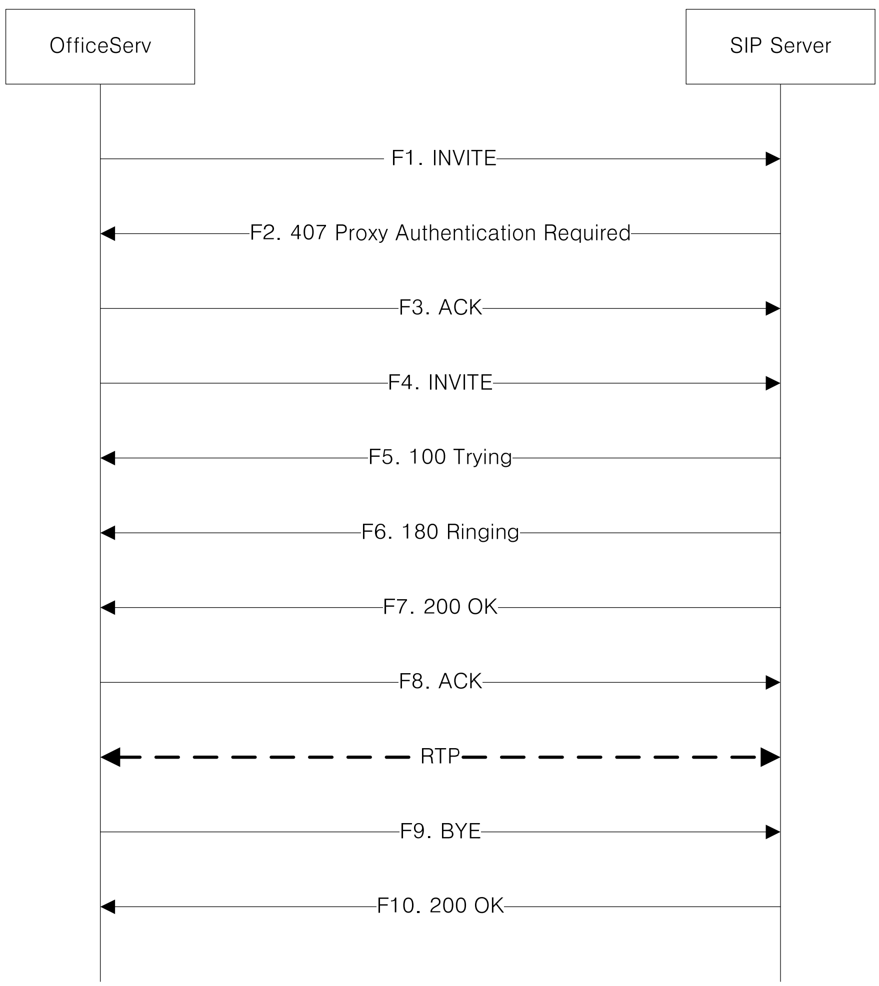


Figure 2. Basic Call Setup

2.1.1. Basic Call MMC Settings

MMC837 SIP OPTIONS

```
ISP1
        SIP SERVER: ENABLE
        OUT PROXY: samsung.com
        DNS SERVER1: 165.213.66.93
        USER NAME: 82312794329
        AUTH USER: 82312794329
        AUTH PSWD: 1234
        REG PER USR: DISABLE
        TRK REG EXP: 001800
```

Above MMC 837 settings are the same with the settings used in registration. Therefore, if you already completed registration, simply skip this.

MMC832 VOIP OUT DGT

```
(O:00)
        ACCESS DGT: 82 (target destination prefix number)
        INSERT DGT:
        DGT LENGTH: 2
        IP TABLE: 1
        IP START: 0
        SERVER USE: YES
        URI TYPE: SIP
```

MMC832 table is used to decide the outbound destination of SIP messages from OfficeServ system. In the previous version of MP S/W, as long as SIP SERVER field in MMC837 is 'ENABLE' and registration is complete, OfficeServ sent all the SIP message to the outbound proxy server. But from software v4.21, OfficeServ checks MMC832 table as well in order to decide the outbound address. Only when SERVER USE field is set to 'YES', OfficeServ sends the SIP message to the outbound server. Otherwise it sends to a designated IP address specified in MMC833 which is used in SIP peering mode. We discuss the usage of MMC833 in more detail in **chapter 4 SIP Peering Services**.

In the above example, ACCESS DGT specifies the digit '82' and DGT LENGTH is '2'. This setting filters out any outbound called number that starts with '82'. i.e., 8231203050.

MMC321 SEND CLIP NO

```
[201] SEND CLIP
1: 82312794329 (same username used for registration)
```

MMC321 table designates the mapping from an internal line number to a registered SIP username (caller ID).

MMC714 DID DIGIT

```
DID DIGIT (xxx)
        DGT: 82312794329 (same username used for registration)
        1: 201
```

MMC714 table designates the mapping from registered SIP username (called ID) to an internal line number.

2.1.2. Message Samples

Inv F1

```
INVITE sip:82312793922@samsung.com:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da7d78-8442d5a5-13c4-50017-48112aa8-33cf5a99-
48112aa8
To: <sip:82312793922@samsung.com:5060>
Call-ID: 1dad610-8442d5a5-13c4-50017-48112aa8-2323234b-48112aa8
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-48112aa8-830ea0fe-491d7077
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2198774014 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30012 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Inv F2

```
SIP/2.0 407 Proxy Authentication Required
To: <sip:82312793922@samsung.com:5060>
From: <sip:82312794329@samsung.com:5060>;tag=1da7d78-8442d5a5-13c4-50017-48112aa8-33cf5a99-
48112aa8
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-48112aa8-830ea0fe-491d7077
CSeq: 1 INVITE
Call-ID: 1dad610-8442d5a5-13c4-50017-48112aa8-2323234b-48112aa8
Proxy-Authenticate: Digest
realm="165.213.66.93",qop="auth",algorithm="MD5",nonce="4059336e99bde1e948ee5e5a6a8245e3"
Content-Length: 0
```

Inv F3

```
ACK sip:82312793922@samsung.com:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da7d78-8442d5a5-13c4-50017-48112aa8-33cf5a99-
48112aa8
To: <sip:82312793922@samsung.com:5060>
Call-ID: 1dad610-8442d5a5-13c4-50017-48112aa8-2323234b-48112aa8
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-48112aa8-830ea0fe-491d7077
Max-Forwards: 70
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0
```

Inv F4

```
INVITE sip:82312793922@samsung.com:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da7d78-8442d5a5-13c4-50017-48112aa8-33cf5a99-
48112aa8
To: <sip:82312793922@samsung.com:5060>
Call-ID: 1dad610-8442d5a5-13c4-50017-48112aa8-2323234b-48112aa8
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-48112aa8-830ea130-13c28e0d
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="4059336e99bde1e948ee5e5a6a8245e3",uri="sip:82
312793922@samsung.com:5060",response="20d0c939567198e48b55c9e53c32b03c",algorithm=MD5,cnonce="
830ea130",qop=auth,n
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2198774014 1 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30012 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Inv F5

```
SIP/2.0 100 Trying
To: <sip:82312793922@samsung.com:5060>
From: <sip:82312794329@samsung.com:5060>;tag=1da7d78-8442d5a5-13c4-50017-48112aa8-33cf5a99-
48112aa8
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-48112aa8-830ea130-13c28e0d
CSeq: 2 INVITE
Call-ID: 1dad610-8442d5a5-13c4-50017-48112aa8-2323234b-48112aa8
Server: ININ-samsung-k1o0rnf-21117695
Content-Length: 0
```

Inv F6

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-48112aa8-830ea130-13c28e0d
Contact: <sip:82312793922@165.213.66.93:23554;rinstance=09b0c09f1dd41754>
To: <sip:82312793922@samsung.com:5060>;tag=5f2bc463
From: <sip:82312794329@samsung.com:5060>;tag=1da7d78-8442d5a5-13c4-50017-48112aa8-33cf5a99-
48112aa8
Call-ID: 1dad610-8442d5a5-13c4-50017-48112aa8-2323234b-48112aa8
CSeq: 2 INVITE
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 0
```

Inv F7

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-48112aa8-830ea130-13c28e0d
Contact: <sip:82312793922@165.213.66.93:23554;rinstance=09b0c09f1dd41754>
To: <sip:82312793922@samsung.com:5060>;tag=5f2bc463
From: <sip:82312794329@samsung.com:5060>;tag=1da7d78-8442d5a5-13c4-50017-48112aa8-33cf5a99-
48112aa8
Call-ID: 1dad610-8442d5a5-13c4-50017-48112aa8-2323234b-48112aa8
CSeq: 2 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 185

v=0
o=- 0 2 IN IP4 165.213.66.93
s=CounterPath X-Lite 3.0
c=IN IP4 165.213.66.93
t=0 0
m=audio 17832 RTP/AVP 8 101
a=fmtp:101 0-15
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Inv F8

```
ACK sip:82312793922@165.213.66.93:23554;rinstance=09b0c09f1dd41754 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da7d78-8442d5a5-13c4-50017-48112aa8-33cf5a99-
48112aa8
To: <sip:82312793922@samsung.com:5060>;tag=5f2bc463
Call-ID: 1dad610-8442d5a5-13c4-50017-48112aa8-2323234b-48112aa8
CSeq: 2 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-48112aab-830ead42-938cbd5
Max-Forwards: 70
Contact: <sip:82312794329@165.213.66.132:5060>
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="4059336e99bde1e948ee5e5a6a8245e3",uri="sip:82
312793922@samsung.com:5060",response="20d0c939567198e48b55c9e53c32b03c",algorithm=MD5,cnonce="
830ea130",qop=auth,n
Content-Length: 0
```

Inv F9

```
BYE sip:82312793922@165.213.66.93:23554;rinstance=09b0c09f1dd41754 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da7d78-8442d5a5-13c4-50017-48112aa8-33cf5a99-
48112aa8
To: <sip:82312793922@samsung.com:5060>;tag=5f2bc463
Call-ID: 1dad610-8442d5a5-13c4-50017-48112aa8-2323234b-48112aa8
CSeq: 3 BYE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-48112aba-830ee97e-791801c3
Max-Forwards: 70
Supported: 100rel,replaces
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="4059336e99bde1e948ee5e5a6a8245e3",uri="sip:82
312793922@165.213.66.93:23554;rinstance=09b0c09f1dd41754",response="832b2e6abc70863441055099cf10
3367",algorithm=MD
Content-Length: 0
```

Inv F10

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-48112aba-830ee97e-791801c3
Contact: <sip:82312793922@165.213.66.93:23554;rinstance=09b0c09f1dd41754>
To: <sip:82312793922@samsung.com:5060>;tag=5f2bc463
From: <sip:82312794329@samsung.com:5060>;tag=1da7d78-8442d5a5-13c4-50017-48112aa8-33cf5a99-
48112aa8
Call-ID: 1dad610-8442d5a5-13c4-50017-48112aa8-2323234b-48112aa8
CSeq: 3 BYE
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 0
```

2.2. Hold & Resume

Hold and Resume are bases of all the other SIP supplementary services. As many SIP services consist of a combination of Hold and Resume functions, it is essential to understand the internal mechanism of them in order to understand the mechanisms of more complicated services.

According to the SIP standard, the Hold/Resume service can be implemented by either an UPDATE method or Re-INVITE method. The basic mechanism that lies in both of the two methods is the same although messages have different names. Currently the OfficeServ supports Re-INVITE message as its default Hold/Resume method. The Re-INVITE is a normal INVITE message except it is sent within an active session. By sending an INVITE message which contains different [1]SDP (Session Description Protocol) during a session, the SIP session mode can be switched to one of sendrecv, sendonly and recvonly according to the session mode attribute value designated in the SDP.

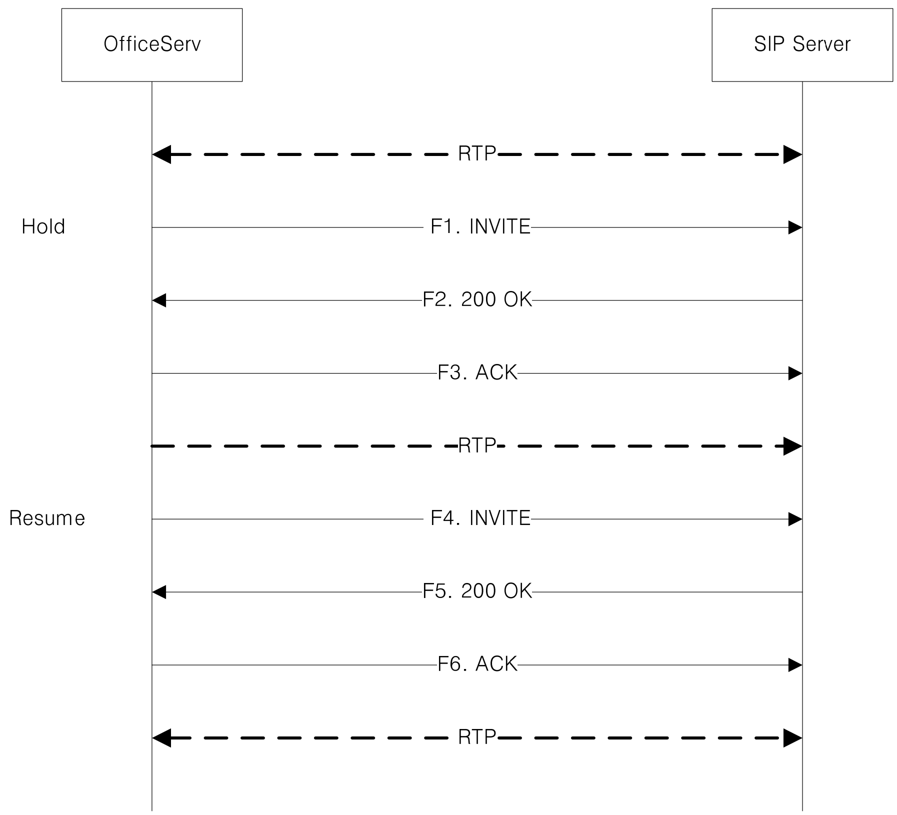


Figure 3. Hold and Resume

[1] SDP specifies the session attributes such as codec types, RTP port, RTP IP address etc. For more detailed information, please refer to RFC2327.

Hold F1

```
INVITE sip:82312793922@165.213.66.93:23554;rinstance=9c39f4fb86603c5e SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da8da0-8442d5a5-13c4-50017-48123208-33639043-
48123208
To: <sip:82312793922@samsung.com:5060>;tag=74757e1e
Call-ID: 1db8a88-8442d5a5-13c4-50017-48123208-5cd79b4d-48123208
CSeq: 3 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-48123211-8713936c-137dbf65
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="e6d451a0e7558d12317347ea5f24cd80",uri="sip:82
312793922@165.213.66.93:23554;rinstance=9c39f4fb86603c5e",response="8e8cc30372699fb57043ffec820dd2
f7",algorithm=MD
Content-Type: application/sdp
Content-Length: 198

v=0
o=SAMSUNG_SIP_GATEWAY 2266198364 2 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 0.0.0.0
t=0 0
m=audio 30000 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendonly
```

Hold F2

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-48123211-8713936c-137dbf65
Contact: <sip:82312793922@165.213.66.93:23554;rinstance=9c39f4fb86603c5e>
To: <sip:82312793922@samsung.com:5060>;tag=74757e1e
From: <sip:82312794329@samsung.com:5060>;tag=1da8da0-8442d5a5-13c4-50017-48123208-33639043-
48123208
Call-ID: 1db8a88-8442d5a5-13c4-50017-48123208-5cd79b4d-48123208
CSeq: 3 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 185

v=0
o=- 4 3 IN IP4 165.213.66.93
s=CounterPath X-Lite 3.0
c=IN IP4 165.213.66.93
t=0 0
m=audio 17674 RTP/AVP 8 101
a=fmtp:101 0-15
a=recvonly
a=rtpmap:101 telephone-event/8000
```

Hold F3

```
ACK sip:82312793922@165.213.66.93:23554;rinstance=9c39f4fb86603c5e SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da8da0-8442d5a5-13c4-50017-48123208-33639043-
48123208
To: <sip:82312793922@samsung.com:5060>;tag=74757e1e
Call-ID: 1db8a88-8442d5a5-13c4-50017-48123208-5cd79b4d-48123208
CSeq: 3 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-48123211-87139416-68c74f0d
Max-Forwards: 70
Contact: <sip:82312794329@165.213.66.132:5060>
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="e6d451a0e7558d12317347ea5f24cd80",uri="sip:82
312793922@165.213.66.93:23554;rinstance=9c39f4fb86603c5e",response="8e8cc30372699fb57043ffec820dd2
f7",algorithm=MD
Content-Length: 0
```

Resume F4

```
INVITE sip:82312793922@165.213.66.93:23554;rinstance=9c39f4fb86603c5e SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da8da0-8442d5a5-13c4-50017-48123208-33639043-
48123208
To: <sip:82312793922@samsung.com:5060>;tag=74757e1e
Call-ID: 1db8a88-8442d5a5-13c4-50017-48123208-5cd79b4d-48123208
CSeq: 4 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-48123214-8713a172-27f7807b
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="e6d451a0e7558d12317347ea5f24cd80",uri="sip:82
312793922@165.213.66.93:23554;rinstance=9c39f4fb86603c5e",response="096ab005bf073eb35c30e9b9a89f9
50d",algorithm=MD
Content-Type: application/sdp
Content-Length: 205

v=0
o=SAMSUNG_SIP_GATEWAY 2266198364 3 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30000 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Hold F5

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-48123214-8713a172-27f7807b
Contact: <sip:82312793922@165.213.66.93:23554;rinstance=9c39f4fb86603c5e>
To: <sip:82312793922@samsung.com:5060>;tag=74757e1e
From: <sip:82312794329@samsung.com:5060>;tag=1da8da0-8442d5a5-13c4-50017-48123208-33639043-
48123208
Call-ID: 1db8a88-8442d5a5-13c4-50017-48123208-5cd79b4d-48123208
CSeq: 4 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 185

v=0
o=- 4 4 IN IP4 165.213.66.93
s=CounterPath X-Lite 3.0
c=IN IP4 165.213.66.93
t=0 0
m=audio 17674 RTP/AVP 8 101
a=fmtp:101 0-15
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Hold F6

```
ACK sip:82312793922@165.213.66.93:23554;rinstance=9c39f4fb86603c5e SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da8da0-8442d5a5-13c4-50017-48123208-33639043-
48123208
To: <sip:82312793922@samsung.com:5060>;tag=74757e1e
Call-ID: 1db8a88-8442d5a5-13c4-50017-48123208-5cd79b4d-48123208
CSeq: 4 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-48123215-8713a208-54281d40
Max-Forwards: 70
Contact: <sip:82312794329@165.213.66.132:5060>
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="e6d451a0e7558d12317347ea5f24cd80",uri="sip:82
312793922@165.213.66.93:23554;rinstance=9c39f4fb86603c5e",response="096ab005bf073eb35c30e9b9a89f9
50d",algorithm=MD
Content-Length: 0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-48123211-8713936c-137dbf65
Contact: <sip:82312793922@165.213.66.93:23554;rinstance=9c39f4fb86603c5e>
To: <sip:82312793922@samsung.com:5060>;tag=74757e1e
From: <sip:82312794329@samsung.com:5060>;tag=1da8da0-8442d5a5-13c4-50017-48123208-33639043-
48123208
Call-ID: 1db8a88-8442d5a5-13c4-50017-48123208-5cd79b4d-48123208
CSeq: 3 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 185

v=0
o=- 4 3 IN IP4 165.213.66.93
s=CounterPath X-Lite 3.0
c=IN IP4 165.213.66.93
t=0 0
m=audio 17674 RTP/AVP 8 101
a=fmtp:101 0-15
a=recvonly
```

In a normal dialogue state, the active session mode is sendrecv which allows both way RTP transmissions. When a Re-INVITE message is sent which designates its RTP transmission to sendonly mode, it informs the called party that it wants to only send RTP and will not receive. After receiving the Re-Invite message, the called party knows that the caller wants to put the session into hold mode and stops sending RTP packets, giving a 200 OK response back. The 200 OK response, like the Re-Invite message, contains a SDP and its session mode attribute is set to recvonly. Meanwhile, the caller can either provide music-on-hold or mute the session by sending no RTP at all, shutting down its listening port. Whether to send MOH or not during the hold time is station dependent. To resume the held session, the caller sends a Re-INVITE message again designating the RTP transmission back to sendrecv.

Remember that only the caller can resume the held session, which means that even if the called party sends a Re-INVITE message specifying sendrecv, the session will remain on hold and caller’s mode will not change.

2.2.1. Another Way of Specifying Sendonly Mode

Some SIP UAs use another, slightly older way of specifying the sendonly mode in its Re-INVITE message. It sets the connection parameter value in the SDP to all zeros, which tells the message receiver (the called party in this context) not to send any RTP packets because there is no destination IP address to which it can send RTP packets to. The OfficeServ supports this connection-allzero-specified hold method for backward compatibility purpose.

2.2.2. Hold Re-Invite

“HOLD Re-invite” in MMC837 for handling MOH issue.

DISABLE: OfficeServ will not send any re-INVITE message for Hold case, which satisfies some ITSP’s requirement.

ENABLE: OfficeServ will send re-INVITE message for Hold case.

This option can be used on Centrix and PBX MNGD type only, not Samsung type in “SS Type. When SS type is set to Samsung, OfficeServ will not send any re-INVITE messages no matter what. It simply switches Voice RTP to Music RTP and vice versa. However, when SS type is either Centrix or PBX MNGD, OfficeServ checks this HOLD RE-INV option in order to decide whehter to send re-INVITE message or not.

2.3. Transfer

2.3.1. Supervised Transfer

There are two types of transfer: One is to use REFER method and the other is to use Re-Invite. When using REFER, OfficeServ gets out of the session when transfer is made. In case of Re-Invite, however, OfficeServ waits till the two transferee finish talking.

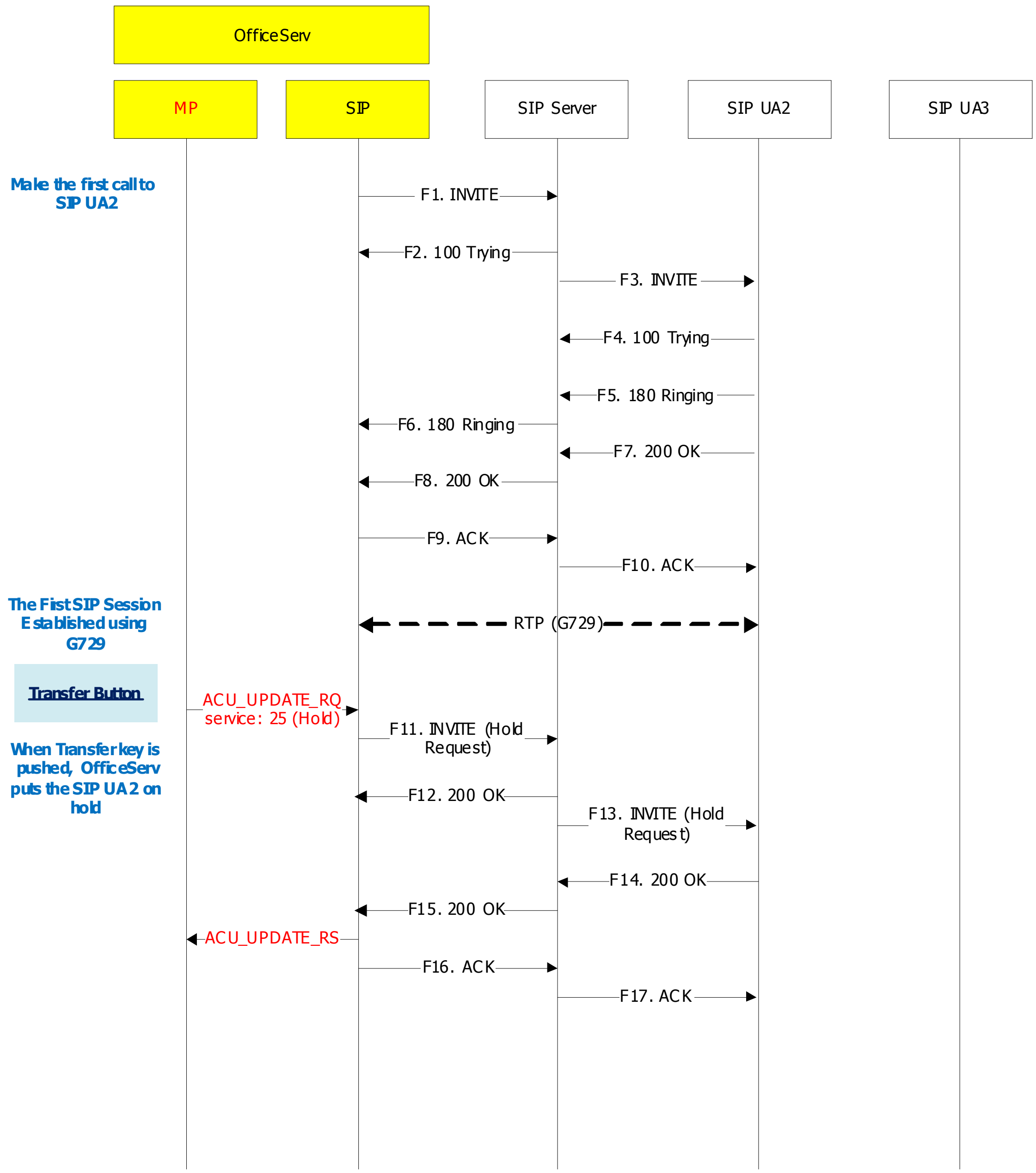


Figure 4. Supervised Transfer #1

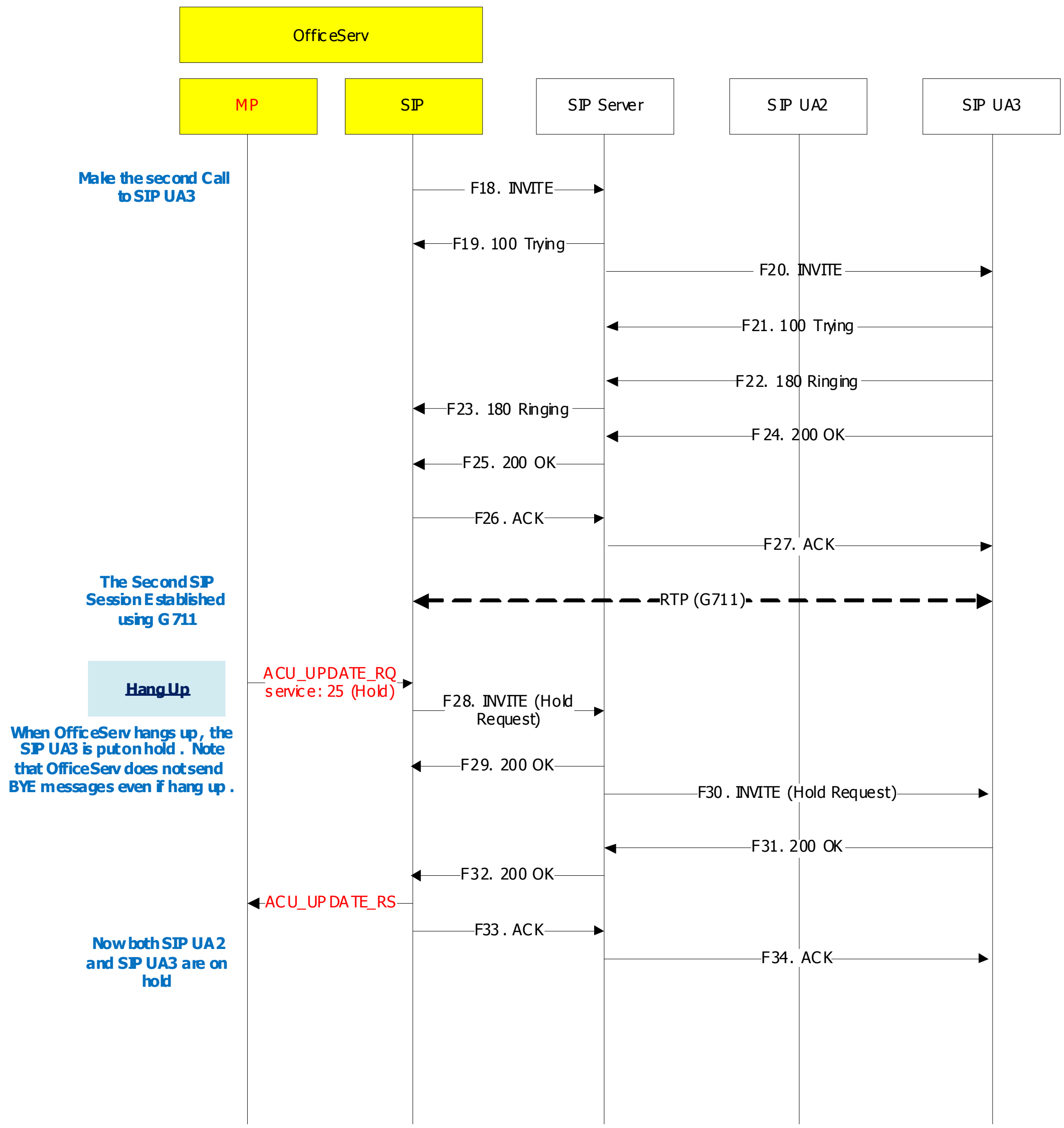


Figure 5. Supervised Transfer #2

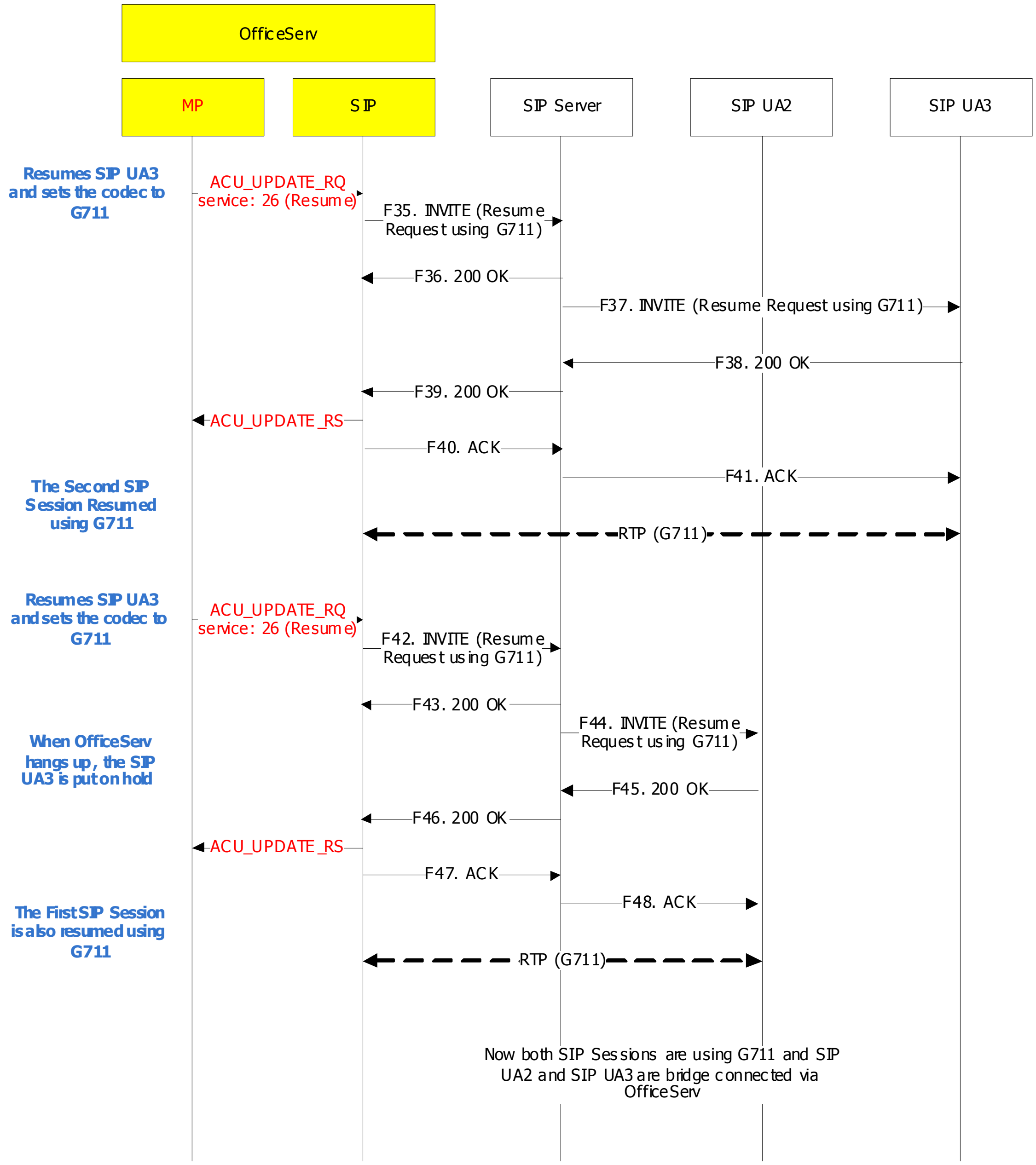


Figure 6. Supervised Transfer #3

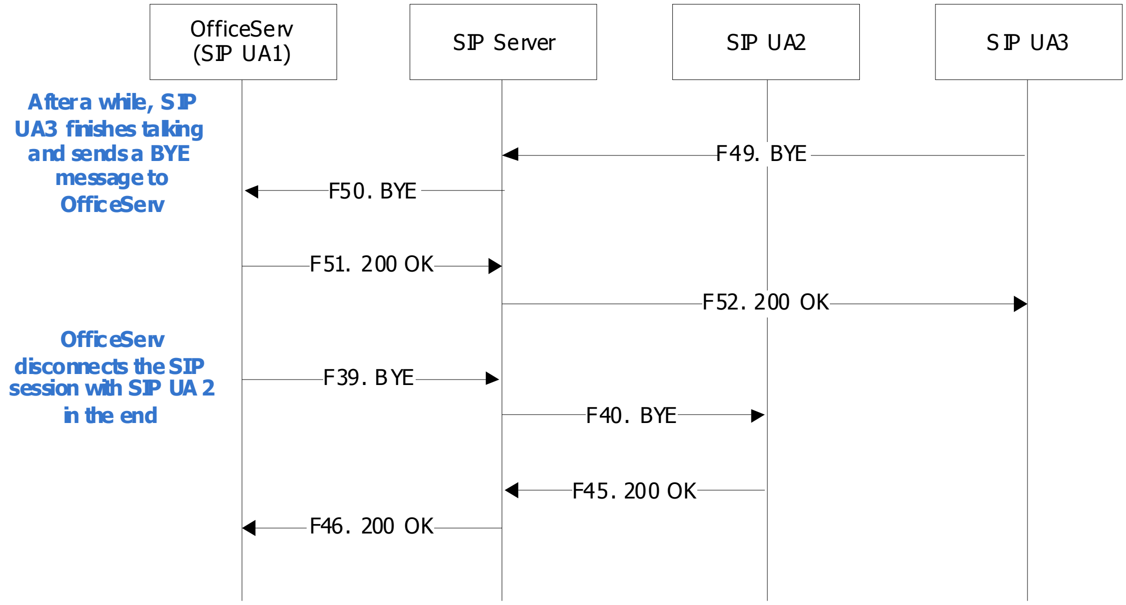


Figure 7. Supervised Transfer #4

2.3.2. Brief Supervised Transfer

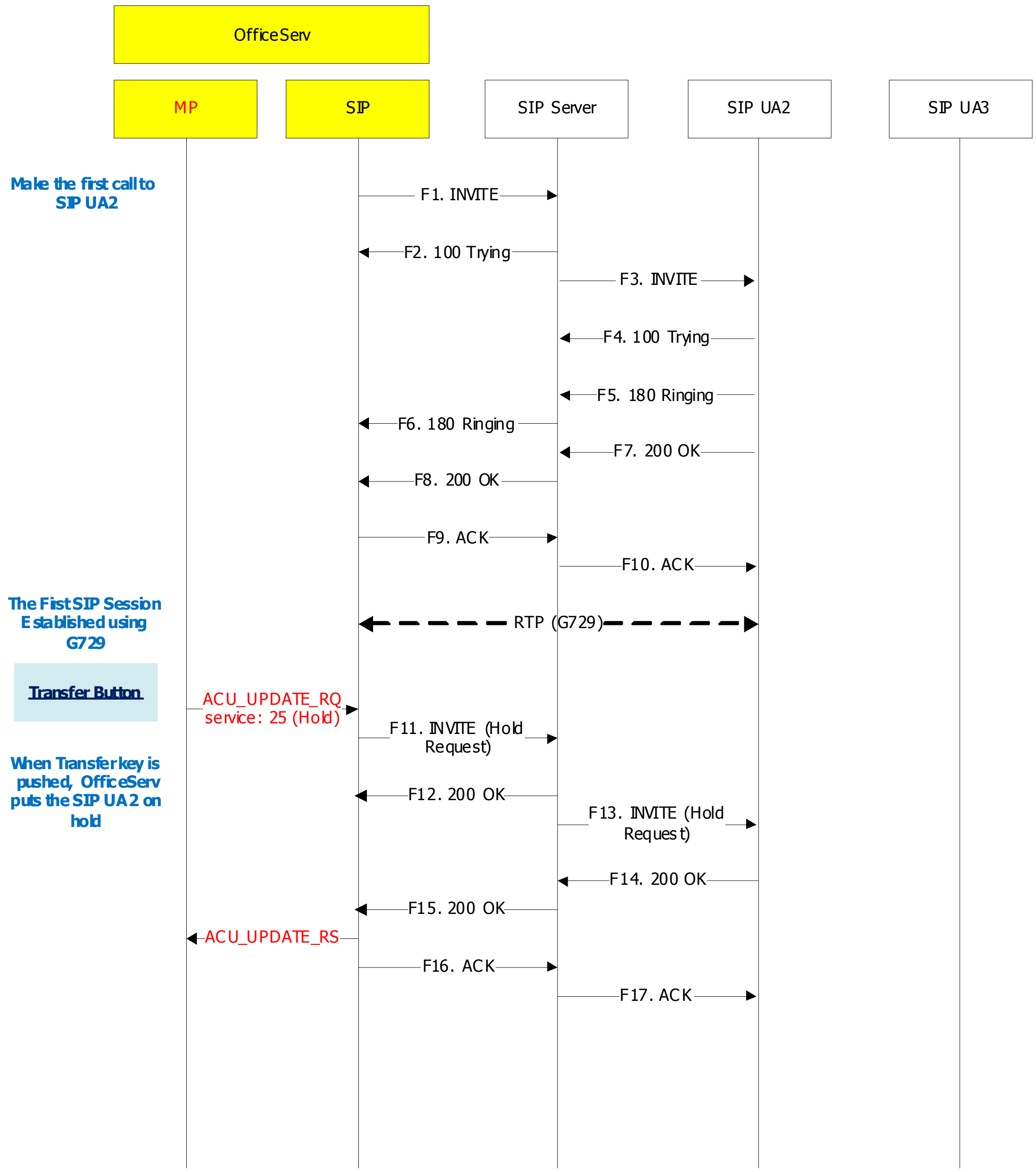


Figure 8. Brief Supervised Transfer #1

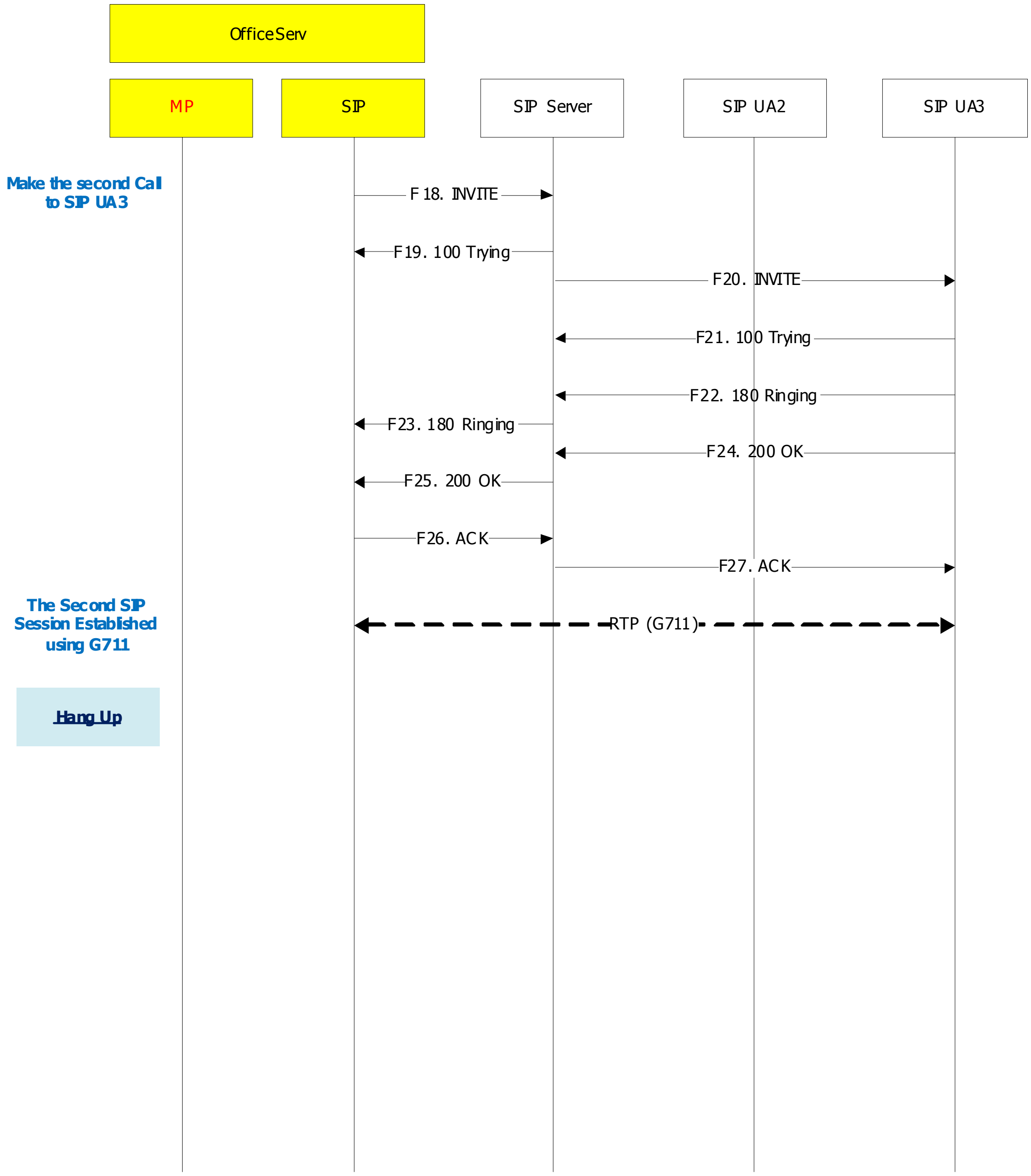


Figure 9. Brief Supervised Transfer #2

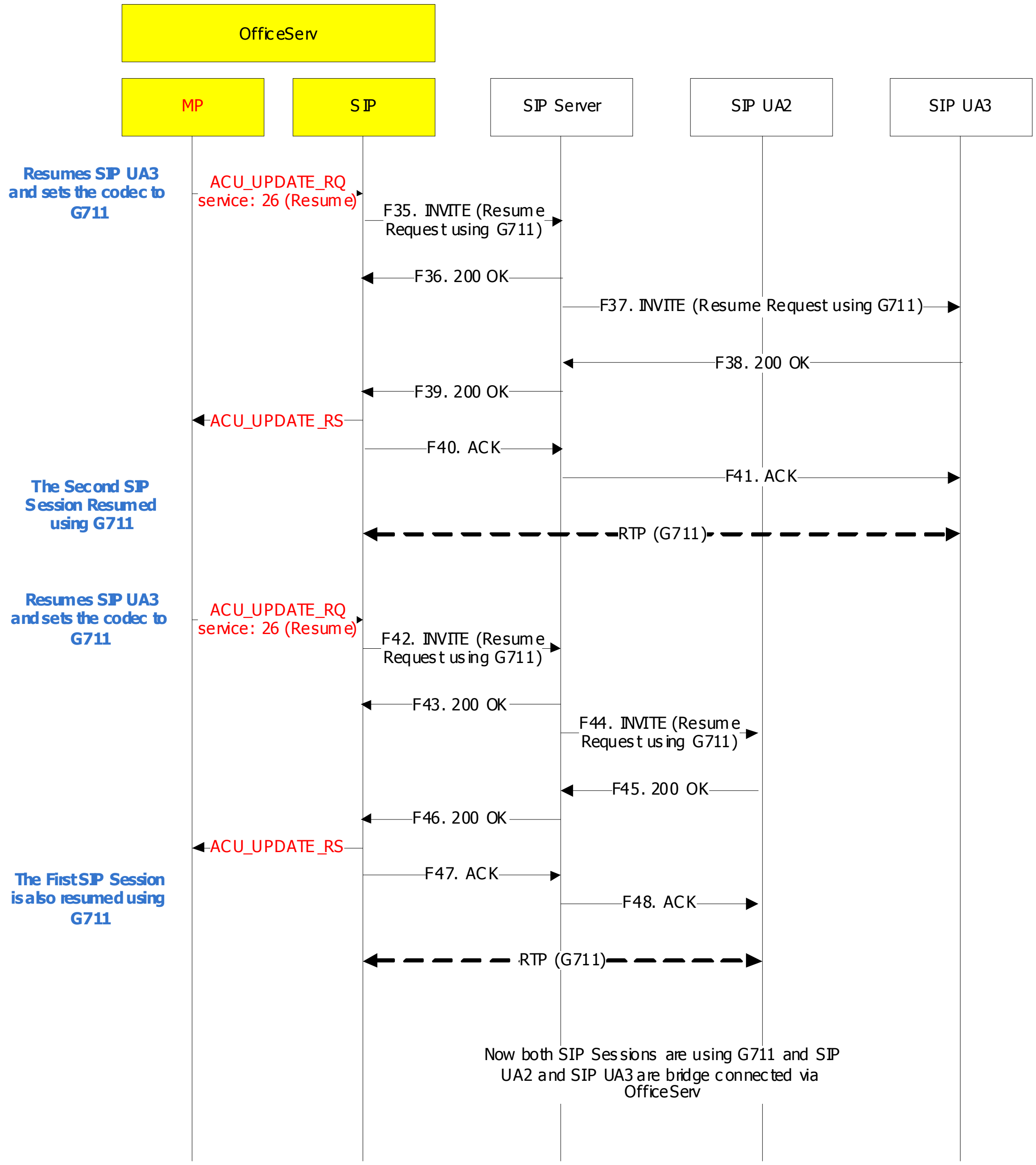


Figure 10. Brief Supervised Transfer #3

2.3.3. Consultation Transfer

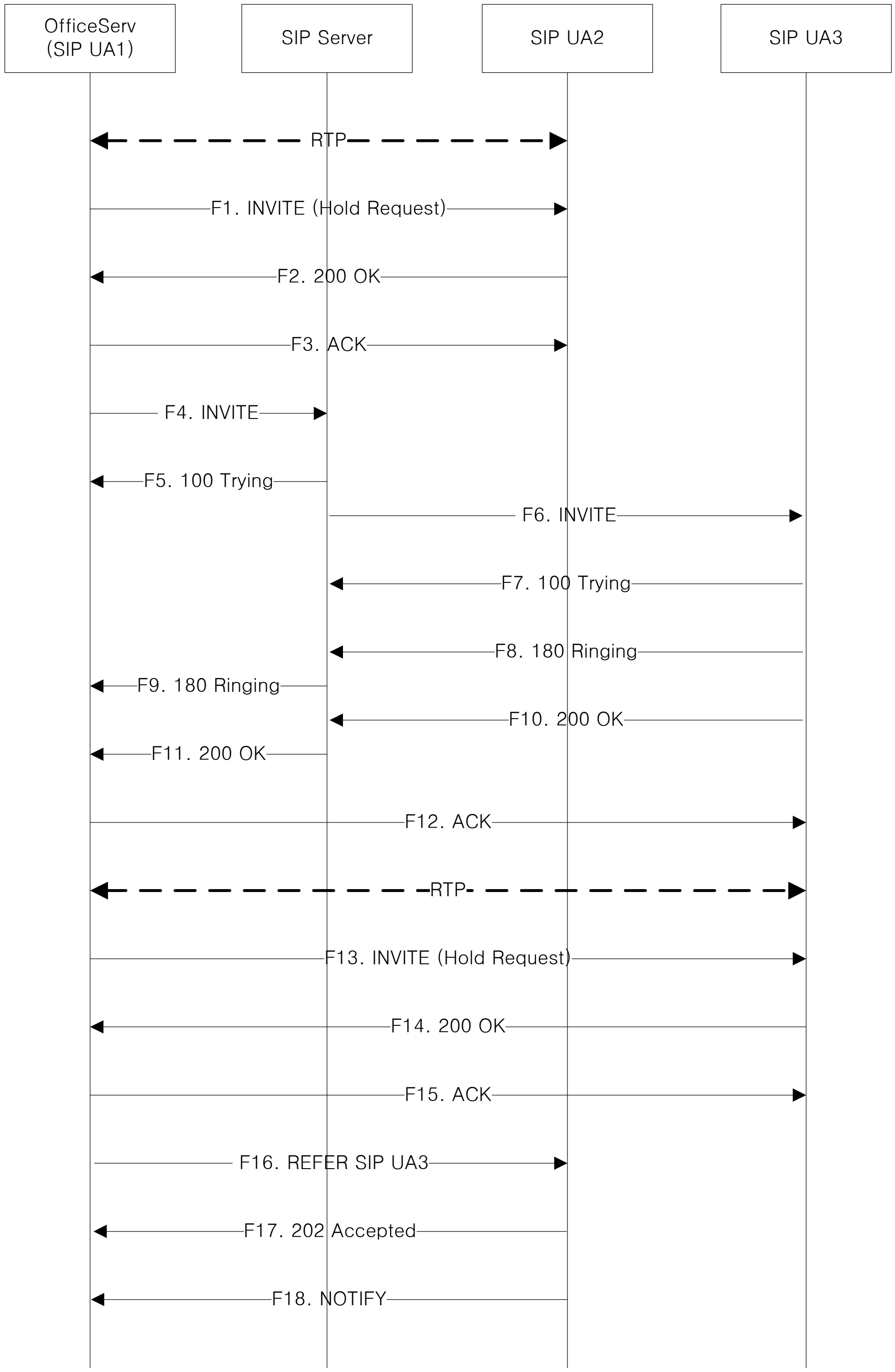


Figure 11. Consultation Transfer #1

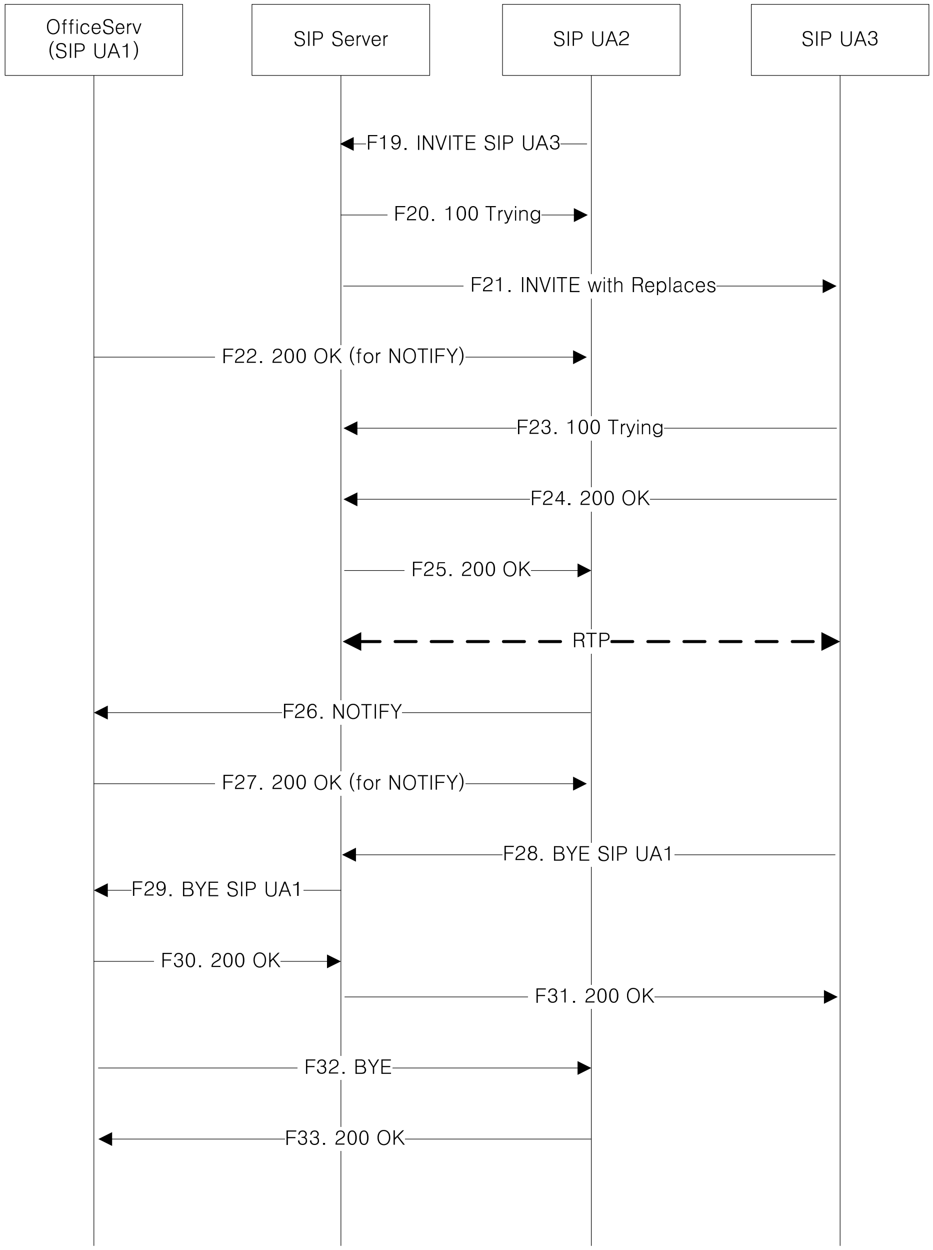


Figure 12. Consultation Transfer #2

Cons_Xfer F1

```
INVITE sip:82312793922@165.213.66.94:35925;rinstance=bbe52bb8ca87498e SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da74a8-8442d5a5-13c4-50017-4812cf15-53ca178a-
4812cf15
To: <sip:82312793922@samsung.com:5060>;tag=5362901f
Call-ID: 1dad2a0-8442d5a5-13c4-50017-4812cf15-5b4c6c24-4812cf15
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4812cf27-897933c8-761d0576
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 198

v=0
o=SAMSUNG_SIP_GATEWAY 2306403644 1 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 0.0.0.0
t=0 0
m=audio 30008 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendonly
```

Cons_Xfer F2

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4812cf27-897933c8-761d0576
Contact: <sip:82312793922@165.213.66.94:35925;rinstance=bbe52bb8ca87498e>
To: <sip:82312793922@samsung.com:5060>;tag=5362901f
From: <sip:82312794329@samsung.com:5060>;tag=1da74a8-8442d5a5-13c4-50017-4812cf15-53ca178a-
4812cf15
Call-ID: 1dad2a0-8442d5a5-13c4-50017-4812cf15-5b4c6c24-4812cf15
CSeq: 2 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 185

v=0
o=- 8 3 IN IP4 165.213.66.94
s=CounterPath X-Lite 3.0
c=IN IP4 165.213.66.94
t=0 0
m=audio 40356 RTP/AVP 8 101
a=fmtp:101 0-15
a=recvonly
a=rtpmap:101 telephone-event/8000
```

Cons_Xfer F3

ACK sip:82312793922@165.213.66.94:35925;rinstance=bbe52bb8ca87498e SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da74a8-8442d5a5-13c4-50017-4812cf15-53ca178a-4812cf15
To: <sip:82312793922@samsung.com:5060>;tag=5362901f
Call-ID: 1dad2a0-8442d5a5-13c4-50017-4812cf15-5b4c6c24-4812cf15
CSeq: 2 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4812cf28-89793454-51255512
Max-Forwards: 70
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0

Cons_Xfer F4

INVITE sip:82312794630@samsung.com:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
To: <sip:82312794630@samsung.com:5060>
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4812cf2e-89794f0c-1ffada59
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2306428684 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30010 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv

Cons_Xfer F5

SIP/2.0 100 Trying
To: <sip:82312794630@samsung.com:5060>
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4812cf2e-89794f0c-1ffada59
CSeq: 1 INVITE
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
Server: ININ-samsung-k1o0rnf-21119919
Content-Length: 0

Cons_Xfer F6

INVITE sip:82312794630@165.213.66.56:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
To: <sip:82312794630@samsung.com:5060>
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk32d96b6ac39e1c59db0fbef76, SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4812cf2e-89794f0c-1ffada59
Max-Forwards: 69
Supported: 100rel, replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2306428684 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30010 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv

Cons_Xfer F7

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk32d96b6ac39e1c59db0fbef76, SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4812cf2e-89794f0c-1ffada59
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
To: <sip:82312794630@samsung.com:5060>
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
Date: Tue, 29 Apr 2008 11:19:13 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Content-Length: 0

Cons_Xfer F8

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk32d96b6ac39e1c59db0fbef76, SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4812cf2e-89794f0c-1ffada59
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
Date: Tue, 29 Apr 2008 11:19:13 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Content-Length: 0

Cons_Xfer F9

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4812cf2e-89794f0c-1ffada59
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
Date: Tue, 29 Apr 2008 11:19:13 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Content-Length: 0

Cons_Xfer F10

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk32d96b6ac39e1c59db0fbef76, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-4812cf2e-89794f0c-1ffada59
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-
4812cf2e
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
Date: Tue, 29 Apr 2008 11:19:14 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Supported: replaces,join,norefersub
Content-Length: 207
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 19103 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 29160 RTP/AVP 8 101
c=IN IP4 165.213.66.56
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Cons_Xfer F11

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4812cf2e-89794f0c-1ffada59
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-
4812cf2e
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
Date: Tue, 29 Apr 2008 11:19:14 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Supported: replaces, join, norefersub
Content-Length: 207
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 19103 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 29160 RTP/AVP 8 101
c=IN IP4 165.213.66.56
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Cons_Xfer F12

```
ACK sip:82312794630@165.213.66.56:5060;transport=udp SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-
4812cf2e
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4812cf30-89795574-462a8d91
Max-Forwards: 70
Contact: <sip:82312794329@165.213.66.132:5060>
```

Cons_Xfer F13

```
INVITE sip:82312794630@165.213.66.56:5060;transport=udp SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-
4812cf2e
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4812cf34-897964f6-46aa233f
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 198

v=0
o=SAMSUNG_SIP_GATEWAY 2306428684 1 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 0.0.0.0
t=0 0
m=audio 30010 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendonly
```

Cons_Xfer F14

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4812cf34-897964f6-46aa233f
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-
4812cf2e
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
Date: Tue, 29 Apr 2008 11:19:18 GMT
CSeq: 2 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Supported: replaces,join,norefersub
Content-Length: 207
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 19103 1 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 29160 RTP/AVP 8 101
c=IN IP4 165.213.66.56
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=inactive
```

Cons_Xfer F15

ACK sip:82312794630@165.213.66.56:5060;transport=udp SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
CSeq: 2 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4812cf34-89796622-7d83433b
Max-Forwards: 70
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0

Cons_Xfer F16

REFER sip:82312793922@165.213.66.94:35925;rinstance=bbe52bb8ca87498e SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da74a8-8442d5a5-13c4-50017-4812cf15-53ca178a-4812cf15
To: <sip:82312793922@samsung.com:5060>;tag=5362901f
Call-ID: 1dad2a0-8442d5a5-13c4-50017-4812cf15-5b4c6c24-4812cf15
CSeq: 3 REFER
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4812cf34-89796528-2cb90d9b
Refer-To: <sip:82312794630@samsung.com:5060?replaces=1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e%3Bto-tag%3D00141ca537d4005749c9b6e0-046b20b9%3Bfrom-tag%3D1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e>
Referred-By: <sip:82312794329@samsung.com>
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0

Cons_Xfer F17

SIP/2.0 202 Accepted
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4812cf34-89796528-2cb90d9b
Contact: <sip:82312793922@165.213.66.94:35925;rinstance=bbe52bb8ca87498e>
To: <sip:82312793922@samsung.com:5060>;tag=5362901f
From: <sip:82312794329@samsung.com:5060>;tag=1da74a8-8442d5a5-13c4-50017-4812cf15-53ca178a-4812cf15
Call-ID: 1dad2a0-8442d5a5-13c4-50017-4812cf15-5b4c6c24-4812cf15
CSeq: 3 REFER
Expires: 60
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 0

Cons_Xfer F18

```
NOTIFY sip:82312794329@165.213.66.132:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.94:35925;branch=z9hG4bK-d87543-022ef3211751ed2e-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:82312793922@165.213.66.94:35925;rinstance=bbe52bb8ca87498e>
To: <sip:82312794329@samsung.com:5060>;tag=1da74a8-8442d5a5-13c4-50017-4812cf15-53ca178a-4812cf15
From: <sip:82312793922@samsung.com:5060>;tag=5362901f
Call-ID: 1dad2a0-8442d5a5-13c4-50017-4812cf15-5b4c6c24-4812cf15
CSeq: 2 NOTIFY
Content-Type: message/sipfrag
User-Agent: X-Lite release 1011s stamp 41150
Subscription-State: active;expires=56
Event: refer
Content-Length: 22

SIP/2.0 100 Trying
```

Cons_Xfer F19

```
INVITE sip:82312794630@samsung.com:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.94:35925;branch=z9hG4bK-d87543-600a501a2a61d867-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:82312793922@165.213.66.94:35925>
To: <sip:82312794630@samsung.com:5060>
From: "82312793922"<sip:82312793922@samsung.com>;tag=4c6c4a7f
Call-ID: YThiZjIzYTlmOTdkM2RkZTc3ZDdkNjhmYWI3ZDJlN2E.
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: X-Lite release 1011s stamp 41150
Referred-By: <sip:82312794329@samsung.com>
Replaces: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e;to-tag=00141ca537d4005749c9b6e0-
046b20b9;from-tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
Content-Length: 423

v=0
o=- 4 2 IN IP4 165.213.66.94
s=CounterPath X-Lite 3.0
c=IN IP4 165.213.66.94
t=0 0
m=audio 58466 RTP/AVP 107 119 100 106 0 105 98 8 101
a=alt:1 1 : 4nKLQGQu qvGEv654 165.213.66.94 58466
a=fmtp:101 0-15
a=rtpmap:107 BV32/16000
a=rtpmap:119 BV32-FEC/16000
a=rtpmap:100 SPEEX/16000
a=rtpmap:106 SPEEX-FEC/16000
a=rtpmap:105 SPEEX-FEC/8000
a=rtpmap:98 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Cons_Xfer F20

```
SIP/2.0 100 Trying
To: <sip:82312794630@samsung.com:5060>
From: "82312793922" <sip:82312793922@samsung.com>;tag=4c6c4a7f
Via: SIP/2.0/UDP 165.213.66.94:35925;branch=z9hG4bK-d87543-600a501a2a61d867-1--d87543-;rport=35925
CSeq: 1 INVITE
Call-ID: YThiZjIzYTlmOTdkM2RkZTc3ZDdkNjhmYWI3ZDJlN2E.
Server: ININ-samsung-k1o0rnf-21119919
Content-Length: 0
```

Cons_Xfer F21

```
INVITE sip:82312794630@165.213.66.56:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk0d899cacc69019481eed90b4c, SIP/2.0/UDP
165.213.66.94:35925;rport=35925;branch=z9hG4bK-d87543-600a501a2a61d867-1--d87543-
Max-Forwards: 69
Contact: <sip:82312793922@165.213.66.94:35925>
To: <sip:82312794630@samsung.com:5060>
From: "82312793922" <sip:82312793922@samsung.com>;tag=4c6c4a7f
Call-ID: YThiZjIzYTlmOTdkM2RkZTc3ZDdkNjhmYWI3ZDJlN2E.
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: X-Lite release 1011s stamp 41150
Referred-By: <sip:82312794329@samsung.com>
Replaces: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e;to-tag=00141ca537d4005749c9b6e0-
046b20b9;from-tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
Content-Length: 423

v=0
o=- 4 2 IN IP4 165.213.66.94
s=CounterPath X-Lite 3.0
c=IN IP4 165.213.66.94
t=0 0
m=audio 58466 RTP/AVP 107 119 100 106 0 105 98 8 101
a=alt:1 1 : 4nKLQGQu qvGEv654 165.213.66.94 58466
a=fmtp:101 0-15
a=rtpmap:107 BV32/16000
a=rtpmap:119 BV32-FEC/16000
a=rtpmap:100 SPEEX/16000
a=rtpmap:106 SPEEX-FEC/16000
a=rtpmap:105 SPEEX-FEC/8000
a=rtpmap:98 iLBC/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Cons_Xfer F22

SIP/2.0 200 OK
From: <sip:82312793922@samsung.com:5060>;tag=5362901f
To: <sip:82312794329@samsung.com:5060>;tag=1da74a8-8442d5a5-13c4-50017-4812cf15-53ca178a-4812cf15
Call-ID: 1dad2a0-8442d5a5-13c4-50017-4812cf15-5b4c6c24-4812cf15
CSeq: 2 NOTIFY
Via: SIP/2.0/UDP 165.213.66.94:35925;rport=35925;branch=z9hG4bK-d87543-022ef3211751ed2e-1--d87543-
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0

Cons_Xfer F23

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk0d899cacc69019481eed90b4c, SIP/2.0/UDP
165.213.66.94:35925;rport=35925;branch=z9hG4bK-d87543-600a501a2a61d867-1--d87543-
From: "82312793922" <sip:82312793922@samsung.com>;tag=4c6c4a7f
To: <sip:82312794630@samsung.com:5060>
Call-ID: YThiZjIzYTlmOTdkM2RkZTc3ZDdkNjhmYWI3ZDJlN2E.
Date: Tue, 29 Apr 2008 11:19:22 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Content-Length: 0

Cons_Xfer F24

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk0d899cacc69019481eed90b4c, SIP/2.0/UDP
165.213.66.94:35925;rport=35925;branch=z9hG4bK-d87543-600a501a2a61d867-1--d87543-
From: "82312793922" <sip:82312793922@samsung.com>;tag=4c6c4a7f
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d400580fa5f094-12127f11
Call-ID: YThiZjIzYTlmOTdkM2RkZTc3ZDdkNjhmYWI3ZDJlN2E.
Date: Tue, 29 Apr 2008 11:19:23 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Supported: replaces,join,norefersub
Content-Length: 206
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 2021 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 31716 RTP/AVP 0 101
c=IN IP4 165.213.66.56
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Cons_Xfer F25

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.94:35925;rport=35925;branch=z9hG4bK-d87543-600a501a2a61d867-1--d87543-
From: "82312793922" <sip:82312793922@samsung.com>;tag=4c6c4a7f
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d400580fa5f094-12127f11
Call-ID: YThiZjIzYTlmOTdkM2RkZTc3ZDdkNjhmYWI3ZDJlN2E.
Date: Tue, 29 Apr 2008 11:19:23 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Supported: replaces, join, norefersub
Content-Length: 206
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 2021 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 31716 RTP/AVP 0 101
c=IN IP4 165.213.66.56
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Cons_Xfer F26

```
NOTIFY sip:82312794329@165.213.66.132:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.94:35925;branch=z9hG4bK-d87543-84315f3df71c2826-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:82312793922@165.213.66.94:35925;rinstance=bbe52bb8ca87498e>
To: <sip:82312794329@samsung.com:5060>;tag=1da74a8-8442d5a5-13c4-50017-4812cf15-53ca178a-4812cf15
From: <sip:82312793922@samsung.com:5060>;tag=5362901f
Call-ID: 1dad2a0-8442d5a5-13c4-50017-4812cf15-5b4c6c24-4812cf15
CSeq: 3 NOTIFY
Content-Type: message/sipfrag
User-Agent: X-Lite release 1011s stamp 41150
Subscription-State: terminated;reason=noResource
Event: refer
Content-Length: 18

SIP/2.0 200 OK
```

Cons_Xfer F27

SIP/2.0 200 OK
From: <sip:82312793922@samsung.com:5060>;tag=5362901f
To: <sip:82312794329@samsung.com:5060>;tag=1da74a8-8442d5a5-13c4-50017-4812cf15-53ca178a-4812cf15
Call-ID: 1dad2a0-8442d5a5-13c4-50017-4812cf15-5b4c6c24-4812cf15
CSeq: 3 NOTIFY
Via: SIP/2.0/UDP 165.213.66.94:35925;rport=35925;branch=z9hG4bK-d87543-84315f3df71c2826-1--d87543-
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0

Cons_Xfer F28

BYE sip:82312794329@165.213.66.132:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK77330deb
From: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
To: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
Max-Forwards: 70
Date: Tue, 29 Apr 2008 11:19:23 GMT
CSeq: 101 BYE
User-Agent: Cisco-CP7960G/8.0

Cons_Xfer F29

BYE sip:82312794329@165.213.66.132:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bkad2bb3d29dd6764738530c3f6, SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK77330deb
From: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
To: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
Max-Forwards: 69
Date: Tue, 29 Apr 2008 11:19:23 GMT
CSeq: 101 BYE
User-Agent: Cisco-CP7960G/8.0
Content-Length: 0

Cons_Xfer F30

SIP/2.0 200 OK
From: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
To: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
CSeq: 101 BYE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bkad2bb3d29dd6764738530c3f6
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK77330deb
Supported: 100rel,replaces
Content-Length: 0

Cons_Xfer F31

SIP/2.0 200 OK
From: <sip:82312794630@samsung.com:5060>;tag=00141ca537d4005749c9b6e0-046b20b9
To: <sip:82312794329@samsung.com:5060>;tag=1da7798-8442d5a5-13c4-50017-4812cf2e-59ae96c3-4812cf2e
Call-ID: 1dad458-8442d5a5-13c4-50017-4812cf2e-22010d5-4812cf2e
CSeq: 101 BYE
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK77330deb
Supported: 100rel, replaces
Content-Length: 0

Cons_Xfer F32

BYE sip:82312793922@165.213.66.94:35925;rinstance=bbe52bb8ca87498e SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da74a8-8442d5a5-13c4-50017-4812cf15-53ca178a-4812cf15
To: <sip:82312793922@samsung.com:5060>;tag=5362901f
Call-ID: 1dad2a0-8442d5a5-13c4-50017-4812cf15-5b4c6c24-4812cf15
CSeq: 4 BYE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4812cf39-89797860-151f6133
Max-Forwards: 70
Supported: 100rel,replaces
Content-Length: 0

Cons_Xfer F33

SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4812cf39-89797860-151f6133
Contact: <sip:82312793922@165.213.66.94:35925;rinstance=bbe52bb8ca87498e>
To: <sip:82312793922@samsung.com:5060>;tag=5362901f
From: <sip:82312794329@samsung.com:5060>;tag=1da74a8-8442d5a5-13c4-50017-4812cf15-53ca178a-4812cf15
Call-ID: 1dad2a0-8442d5a5-13c4-50017-4812cf15-5b4c6c24-4812cf15
CSeq: 4 BYE
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 0

2.3.4. Blind Transfer

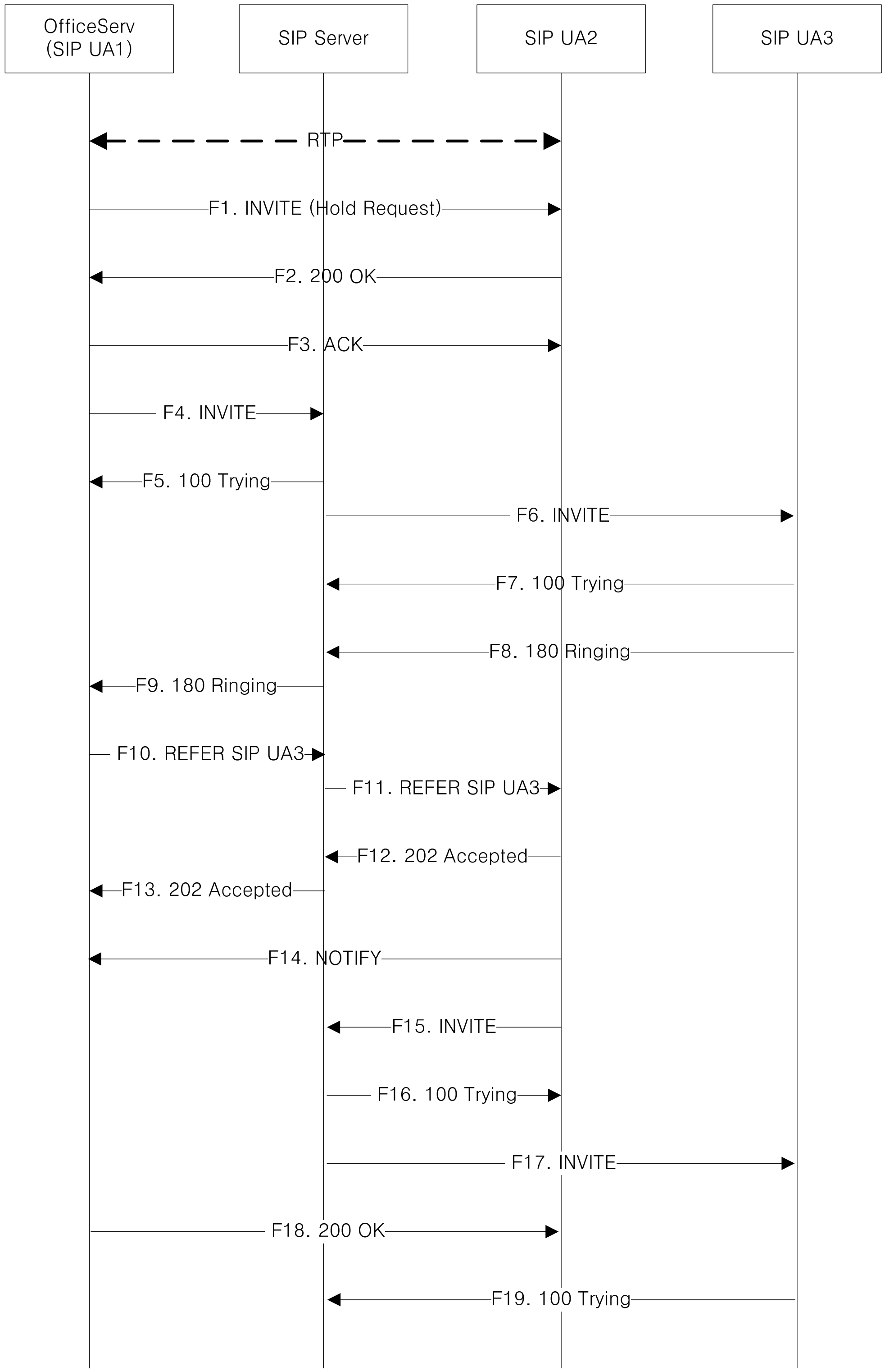


Figure 13. Blind Transfer #1

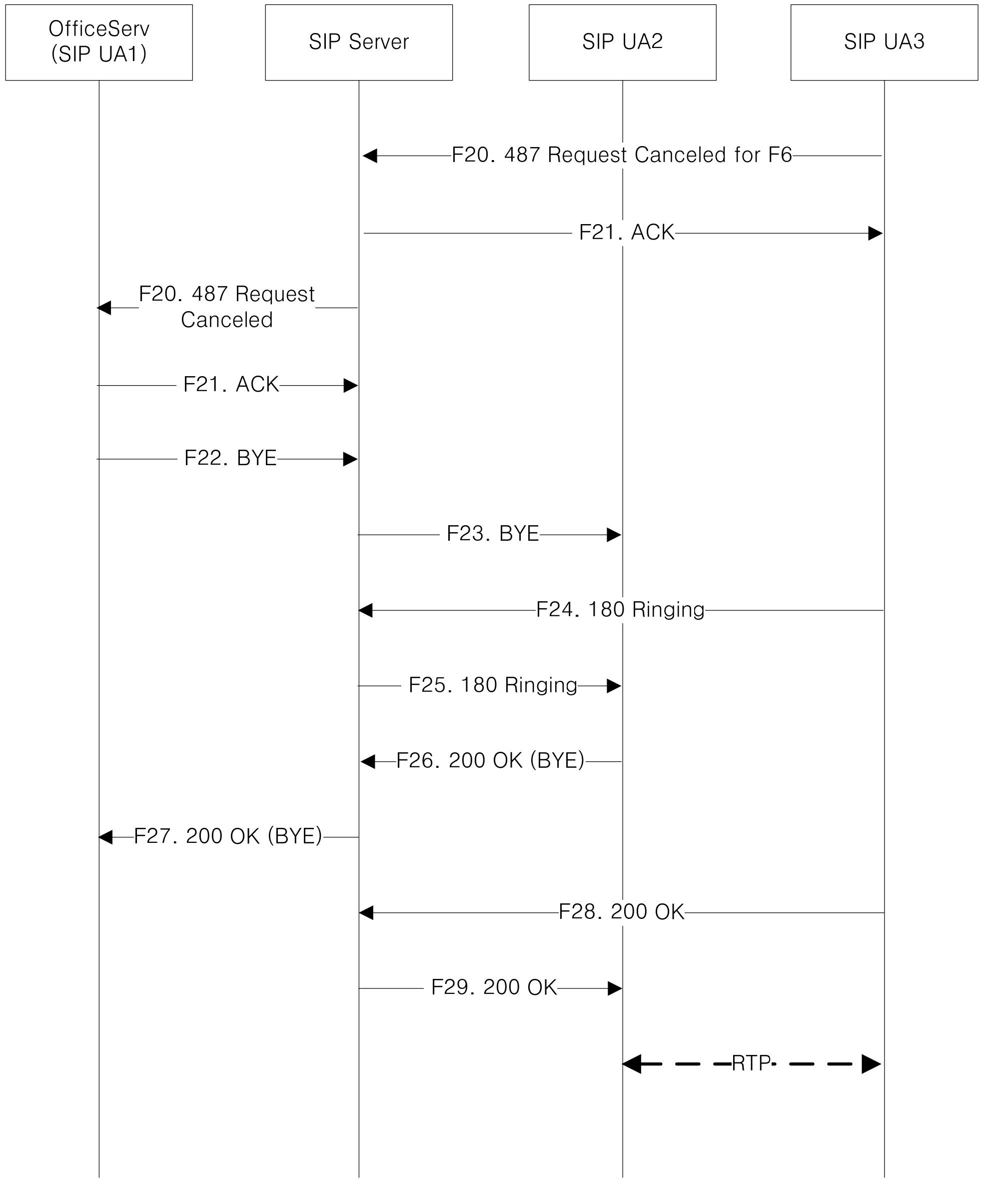


Figure 14. Blind Transfer #2

2.4. Call Forward

Call Forward feature is to redirect a call to an original recipient to the other recipient. According to who is redirector, OfficeServ should behave differently. When redirecting a call, the original recipient should respond with a 302 REDIRECTED response. Therefore, if OfficeServ is the one who redirects the call, it should answer with 302 REDIRECTED against received INVITE message and if it is the opposite case, OfficeServ will receive the 302 response. This chapter shows how OfficeServ reacts on these two different cases.

2.4.1. Call Forward by a SIP Server

Call Forward feature is set on either a SIP server or a recipient. In whichever case, OfficeServ is supposed to receive a 302 response and should re-send the original INVITE message to the forwarded recipient.

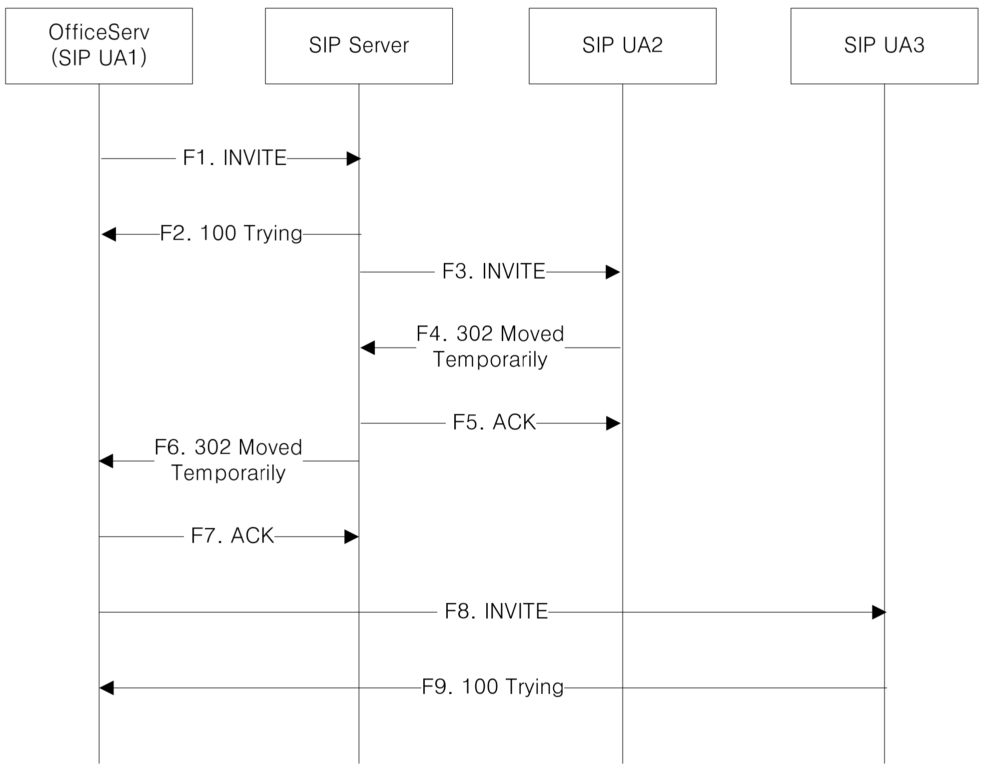


Figure 15. 302 Moved Temporarily Received

302_rcvd F1

INVITE sip:82312794630@samsung.com:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da8ab0-8442d5a5-13c4-50017-4813d39b-1988c5e6-4813d39b
To: <sip:82312794630@samsung.com:5060>
Call-ID: 1dadcf0-8442d5a5-13c4-50017-4813d39b-f9a83c8-4813d39b
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4813d39b-8d72971c-224f9640
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2373097244 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30012 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv

302_rcvd F2

SIP/2.0 100 Trying
To: <sip:82312794630@samsung.com:5060>
From: <sip:82312794329@samsung.com:5060>;tag=1da8ab0-8442d5a5-13c4-50017-4813d39b-1988c5e6-4813d39b
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4813d39b-8d72971c-224f9640
CSeq: 1 INVITE
Call-ID: 1dadcf0-8442d5a5-13c4-50017-4813d39b-f9a83c8-4813d39b
Server: ININ-samsung-k1o0rnf-21119919
Content-Length: 0

302_rcvd F3

```
INVITE sip:82312794630@165.213.66.56:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da8ab0-8442d5a5-13c4-50017-4813d39b-1988c5e6-
4813d39b
To: <sip:82312794630@samsung.com:5060>
Call-ID: 1dadcf0-8442d5a5-13c4-50017-4813d39b-f9a83c8-4813d39b
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk3815f9ba38f7f9ac638fd3efa, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-4813d39b-8d72971c-224f9640
Max-Forwards: 69
Supported: 100rel, replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2373097244 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30012 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

302_rcvd F4

```
SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk3815f9ba38f7f9ac638fd3efa, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-4813d39b-8d72971c-224f9640
From: <sip:82312794329@samsung.com:5060>;tag=1da8ab0-8442d5a5-13c4-50017-4813d39b-1988c5e6-
4813d39b
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d401f860ff7663-010e59e4
Call-ID: 1dadcf0-8442d5a5-13c4-50017-4813d39b-f9a83c8-4813d39b
Date: Wed, 30 Apr 2008 05:50:42 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312793922@165.213.66.94:5060>
Diversion: "82312794630" <sip:82312794630@165.213.66.56>;reason=unconditional;privacy=off;screen=yes
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Content-Length: 0
```

302_rcvd F5

ACK sip:82312794630@165.213.66.56:5060 SIP/2.0
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d401f860ff7663-010e59e4
From: <sip:82312794329@samsung.com:5060>;tag=1da8ab0-8442d5a5-13c4-50017-4813d39b-1988c5e6-4813d39b
Call-ID: 1dadcf0-8442d5a5-13c4-50017-4813d39b-f9a83c8-4813d39b
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk3815f9ba38f7f9ac638fd3efa
CSeq: 1 ACK
Content-Length: 0

302_rcvd F6

SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4813d39b-8d72971c-224f9640
From: <sip:82312794329@samsung.com:5060>;tag=1da8ab0-8442d5a5-13c4-50017-4813d39b-1988c5e6-4813d39b
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d401f860ff7663-010e59e4
Call-ID: 1dadcf0-8442d5a5-13c4-50017-4813d39b-f9a83c8-4813d39b
Date: Wed, 30 Apr 2008 05:50:42 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312793922@165.213.66.94:5060>
Diversion: "82312794630" <sip:82312794630@165.213.66.56>;reason=unconditional;privacy=off;screen=yes
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Content-Length: 0

302_rcvd F7

ACK sip:82312794630@samsung.com:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da8ab0-8442d5a5-13c4-50017-4813d39b-1988c5e6-4813d39b
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d401f860ff7663-010e59e4
Call-ID: 1dadcf0-8442d5a5-13c4-50017-4813d39b-f9a83c8-4813d39b
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4813d39b-8d72971c-224f9640
Max-Forwards: 70
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0

302_rcvd F8

```
INVITE sip:82312793922@165.213.66.94:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1da8ab0-8442d5a5-13c4-50017-4813d39b-1988c5e6-
4813d39b
To: <sip:82312794630@samsung.com:5060>
Call-ID: 1dadcf0-8442d5a5-13c4-50017-4813d39b-f9a83c8-4813d39b
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4813d39c-8d729bfe-218bc766
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2373097244 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30012 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

302_rcvd F9

```
SIP/2.0 100 Trying
To: <sip:82312794630@samsung.com:5060>
From: <sip:82312794329@samsung.com:5060>;tag=1da8ab0-8442d5a5-13c4-50017-4813d39b-1988c5e6-
4813d39b
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4813d39c-8d729bfe-218bc766
CSeq: 2 INVITE
Call-ID: 1dadcf0-8442d5a5-13c4-50017-4813d39b-f9a83c8-4813d39b
Server: ININ-retail_-12796967
Content-Length: 0
```

2.4.2. Call Forward by OfficeServ

If OfficeServ 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 OfficeServ, asks the original caller to redirect the call to designated destination, and the latter is OfficeServ itself redirects the call by sending an INVITE to the 3rd destination.

This is how to set call forward in OfficeServ using MMC102. Let's assume that OfficeServ has been assigned a primary number of '82312794329' and when it receives an INVITE message, whose called number in TO header is the primary number, it sends the call to a station 201.

MMC714 SEND CLIP NO

```
DID DIGIT (xxx)
        DGT: 82312794329 (same username used for registration)
        1: 201
```

MMC102 CALL FORWARD

```
[201] FORWARD
        1. FWD ALL: 80582312793922
```

When a call is received to station 201, the call is forwarded to the designated number in FWD ALL field. In example, staring digit '805' is SIP trunk group number, and thus we can see that incoming call is to be forwarded to a number of '82312793922' using SIP trunk line.

2.4.2.1. Sending 302 Response

MMC837 SIP OPTIONS

```
ISP1
        SIP SERVER: ENABLE
        OUT PROXY: samsung.com
        DNS SERVER1: 165.213.66.93
        USER NAME: 82312794329
        AUTH USER: 82312794329
        AUTH PSWD: 1234
        REG PER USR: DISABLE
        TRK REG EXP: 001800
        302 RESPONSE: ENABLE
```

Above MMC 837 settings are the same with the settings used in registration except for the last item. In order to respond with '[2]302 Moved Temporarily', OfficeServ needs to set '302 RESP' field to ENABLE.

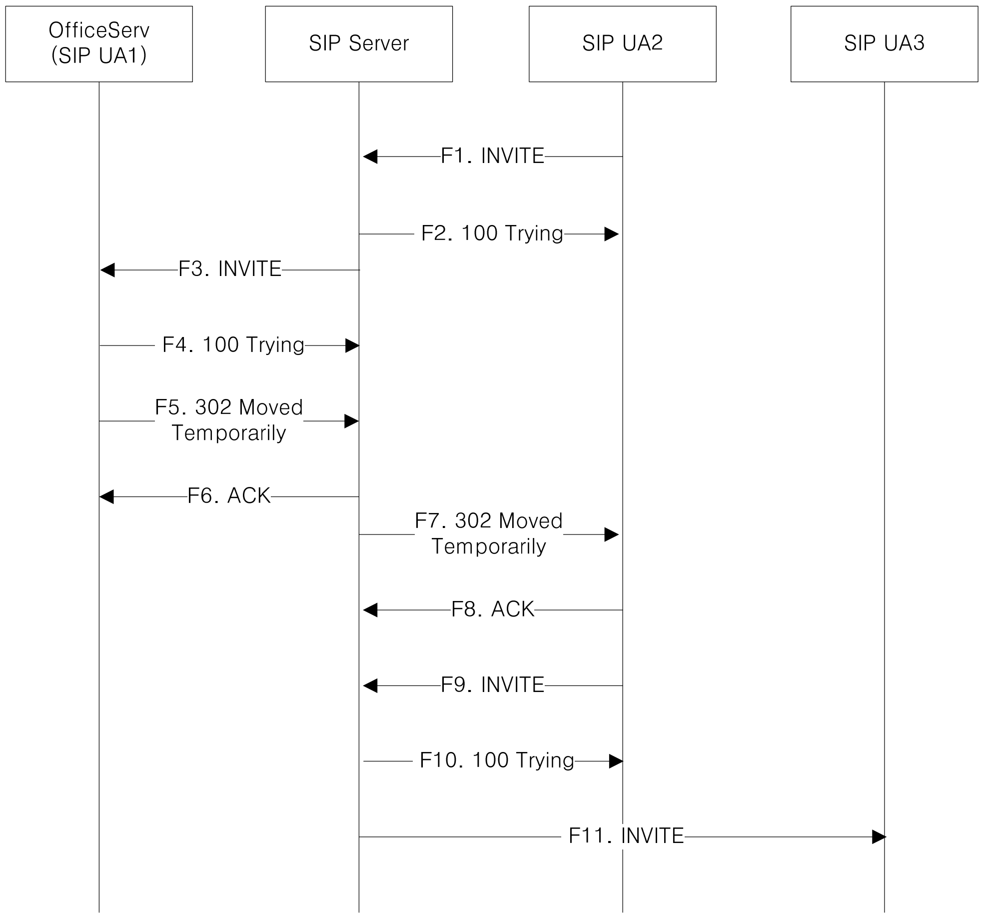


Figure 16. 302 Moved Temporarily Sent

[2] Note that sending a 302 Moved Temporarily response is possible only when set ALL CALL FORWARD.

302_send F1

```
INVITE sip:82312794329@samsung.com SIP/2.0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK2e3be9dc
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4040a3e4a561a-3e735954
To: <sip:82312794329@samsung.com>
Call-ID: 00141ca5-37d40014-35683d54-165be706@165.213.66.56
Max-Forwards: 70
Date: Thu, 01 May 2008 05:35:33 GMT
CSeq: 101 INVITE
User-Agent: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Expires: 180
Accept: application/sdp
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Supported: replaces,join,norefersub
Content-Length: 278
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 15464 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 18056 RTP/AVP 0 8 18 101
c=IN IP4 165.213.66.56
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

302_ send F2

```
SIP/2.0 100 Trying
To: <sip:82312794329@samsung.com>
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4040a3e4a561a-3e735954
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK2e3be9dc
CSeq: 101 INVITE
Call-ID: 00141ca5-37d40014-35683d54-165be706@165.213.66.56
Server: ININ-samsung-k1o0rnf-21119919
Content-Length: 0
```

302_send F3

INVITE sip:82312794329@165.213.66.132:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk2d926da1868c1e3f081574098, SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK2e3be9dc
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4040a3e4a561a-3e735954
To: <sip:82312794329@samsung.com>
Call-ID: 00141ca5-37d40014-35683d54-165be706@165.213.66.56
Max-Forwards: 69
Date: Thu, 01 May 2008 05:35:33 GMT
CSeq: 101 INVITE
User-Agent: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Expires: 180
Accept: application/sdp
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Supported: replaces, join, norefersub
Content-Length: 278
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 15464 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 18056 RTP/AVP 0 8 18 101
c=IN IP4 165.213.66.56
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=sendrecv

302_ send F4

SIP/2.0 100 Trying
From: "82312794630"<sip:82312794630@samsung.com>;tag=00141ca537d4040a3e4a561a-3e735954
To: <sip:82312794329@samsung.com>
Call-ID: 00141ca5-37d40014-35683d54-165be706@165.213.66.56
CSeq: 101 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk2d926da1868c1e3f081574098
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK2e3be9dc
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0

302_ send F5

SIP/2.0 302 Moved Temporarily
From: "82312794630"<sip:82312794630@samsung.com>;tag=00141ca537d4040a3e4a561a-3e735954
To: <sip:82312794329@samsung.com>;tag=1da9c50-8442d5a5-13c4-50017-48152188-22ffc852-48152188
Call-ID: 00141ca5-37d40014-35683d54-165be706@165.213.66.56
CSeq: 101 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk2d926da1868c1e3f081574098
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK2e3be9dc
Supported: 100rel,replaces
Contact: <sip:82312793922@samsung.com>

302_ send F6

ACK sip:82312794329@165.213.66.132:5060 SIP/2.0
To: <sip:82312794329@samsung.com>;tag=1da9c50-8442d5a5-13c4-50017-48152188-22ffc852-48152188
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4040a3e4a561a-3e735954
Call-ID: 00141ca5-37d40014-35683d54-165be706@165.213.66.56
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk2d926da1868c1e3f081574098
CSeq: 101 ACK
Content-Length: 0

302_ send F7

SIP/2.0 302 Moved Temporarily
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4040a3e4a561a-3e735954
To: <sip:82312794329@samsung.com>;tag=1da9c50-8442d5a5-13c4-50017-48152188-22ffc852-48152188
Call-ID: 00141ca5-37d40014-35683d54-165be706@165.213.66.56
CSeq: 101 INVITE
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK2e3be9dc
Supported: 100rel, replaces
Contact: <sip:82312793922@samsung.com>
Content-Length: 0

302_ send F8

ACK sip:82312794329@samsung.com SIP/2.0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK2e3be9dc
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4040a3e4a561a-3e735954
To: <sip:82312794329@samsung.com>;tag=1da9c50-8442d5a5-13c4-50017-48152188-22ffc852-48152188
Call-ID: 00141ca5-37d40014-35683d54-165be706@165.213.66.56
Date: Thu, 01 May 2008 05:35:34 GMT
CSeq: 101 ACK
Content-Length: 0

302_send F9

```
INVITE sip:82312793922@samsung.com:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK592e280d
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4040b3b27c831-24b59245
To: <sip:82312794329@samsung.com>
Call-ID: 00141ca5-37d40014-35683d54-165be706@165.213.66.56
Max-Forwards: 70
Date: Thu, 01 May 2008 05:35:34 GMT
CSeq: 102 INVITE
User-Agent: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Expires: 180
Accept: application/sdp
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Supported: replaces,join,norefersub
Content-Length: 278
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 15464 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 18056 RTP/AVP 0 8 18 101
c=IN IP4 165.213.66.56
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

302_ send F10

```
SIP/2.0 100 Trying
To: <sip:82312794329@samsung.com>
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4040b3b27c831-24b59245
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK592e280d
CSeq: 102 INVITE
Call-ID: 00141ca5-37d40014-35683d54-165be706@165.213.66.56
Server: ININ-samsung-k1o0rnf-21119919
Content-Length: 0
```

302_send F9

```
INVITE sip:82312793922@samsung.com:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk9558339ebf9a4939e16ce8d16, SIP/2.0/UDP
165.213.66.56:5060;branch=z9hG4bK592e280d
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4040b3b27c831-24b59245
To: <sip:82312794329@samsung.com>
Call-ID: 00141ca5-37d40014-35683d54-165be706@165.213.66.56
Max-Forwards: 69
Date: Thu, 01 May 2008 05:35:34 GMT
CSeq: 102 INVITE
User-Agent: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Expires: 180
Accept: application/sdp
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Supported: replaces, join, norefersub
Content-Length: 278
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 15464 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 18056 RTP/AVP 0 8 18 101
c=IN IP4 165.213.66.56
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=sendrecv
```

2.4.2.2. Forwarding Received INVITE

Forwarding a received INVITE message to a designated number is another way of doing Call Forward. There is only one difference in MMC837 setting from the case of sending 302 response: setting '302 RESP' to DISABLE.

MMC837 SIP OPTIONS

ISP1

SIP SERVER: *ENABLE*
OUT PROXY: *samsung.com*
DNS SERVER1: 165.213.66.93
USER NAME: *82312794329*
AUTH USER: *82312794329*
AUTH PSWD: *1234*
REG PER USR: *DISABLE*
TRK REG EXP: *001800*

As seen in sample messages, OfficeServ does not forward the received INVITE as it is, rather it sends its own INVITE message to the designated destination. That is, in this way, one SIP session is made between the original caller and OfficeServ, and the second SIP session is additionally made between OfficeServ and the 3rd destination. Each SIP session consumes each MGI resource in OfficeServ and the RTP packets should go through 4 steps of encoding and decoding procedure, which degrades voice quality significantly. **Thus this INVITE message forwarding is not a good way to do call forward**.

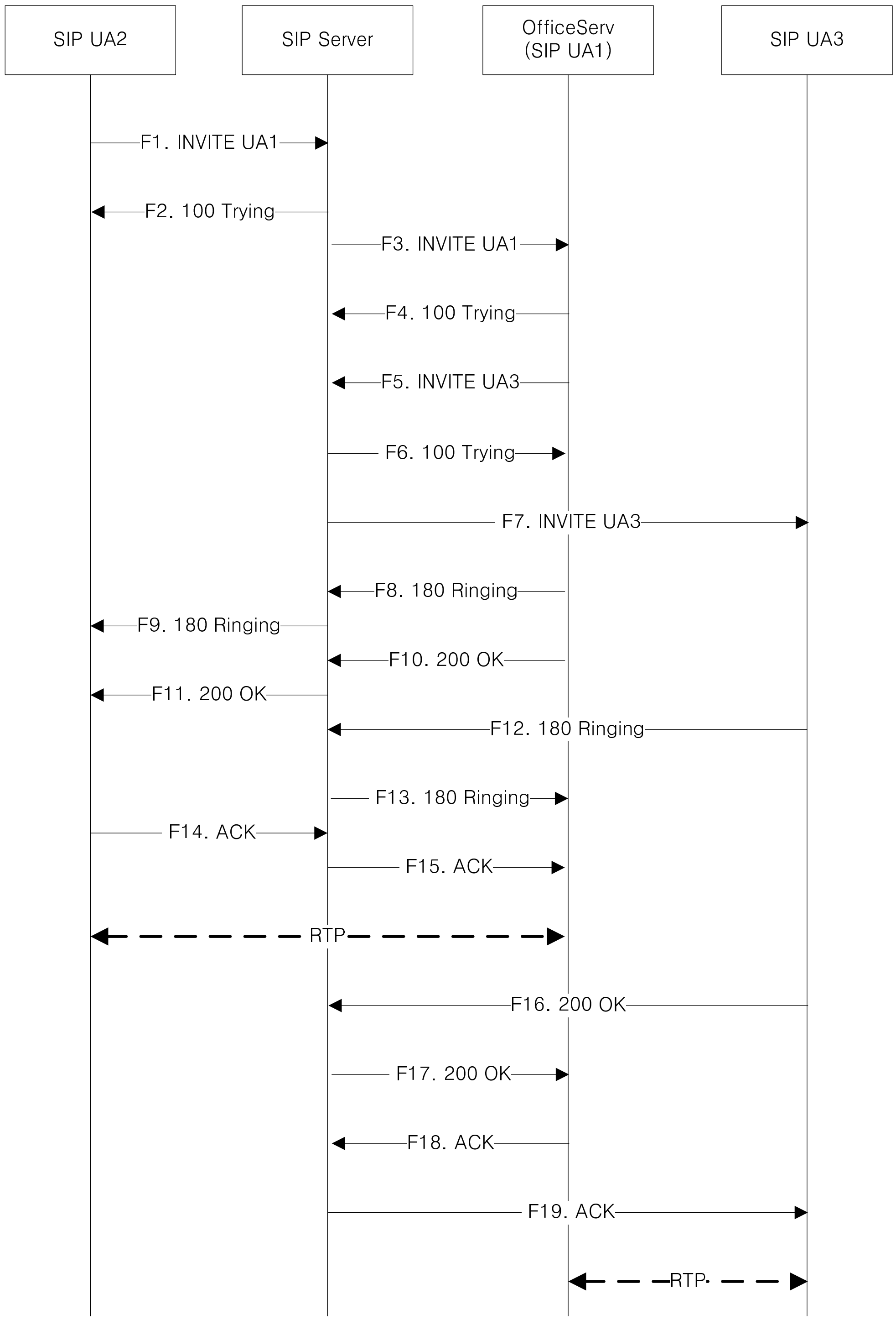


Figure 17. Forwarding Received INVITE

Fwd_inv F1

```
INVITE sip:82312794329@samsung.com SIP/2.0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK378c08e1
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4102504950a1c-5b2da26a
To: <sip:82312794329@samsung.com>
Call-ID: 00141ca5-37d40018-76a485c7-3a0c34e8@165.213.66.56
Max-Forwards: 70
Date: Fri, 02 May 2008 11:22:33 GMT
CSeq: 101 INVITE
User-Agent: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Expires: 180
Accept: application/sdp
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Supported: replaces,join,norefersub
Content-Length: 278
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 11869 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 32260 RTP/AVP 0 8 18 101
c=IN IP4 165.213.66.56
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Fwd_inv F2

```
SIP/2.0 100 Trying
To: <sip:82312794329@samsung.com>
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4102504950a1c-5b2da26a
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK378c08e1
CSeq: 101 INVITE
Call-ID: 00141ca5-37d40018-76a485c7-3a0c34e8@165.213.66.56
Server: ININ-samsung-k1o0rnf-21113719
Content-Length: 0
```

Fwd_inv F3

INVITE sip:82312794329@165.213.66.132:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk21691eeeb42916368ff6422ac, SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK378c08e1
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4102504950a1c-5b2da26a
To: <sip:82312794329@samsung.com>
Call-ID: 00141ca5-37d40018-76a485c7-3a0c34e8@165.213.66.56
Max-Forwards: 69
Date: Fri, 02 May 2008 11:22:33 GMT
CSeq: 101 INVITE
User-Agent: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Expires: 180
Accept: application/sdp
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Supported: replaces, join, norefersub
Content-Length: 278
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 11869 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 32260 RTP/AVP 0 8 18 101
c=IN IP4 165.213.66.56
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=sendrecv

Fwd_inv F4

SIP/2.0 100 Trying
From: "82312794630"<sip:82312794630@samsung.com>;tag=00141ca537d4102504950a1c-5b2da26a
To: <sip:82312794329@samsung.com>
Call-ID: 00141ca5-37d40018-76a485c7-3a0c34e8@165.213.66.56
CSeq: 101 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk21691eeeb42916368ff6422ac
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK378c08e1
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0

Fwd_inv F5

INVITE sip:82312793922@samsung.com:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1dd4350-8442d5a5-13c4-50017-4816c456-41f71b1e-4816c456
To: <sip:82312793922@samsung.com:5060>
Call-ID: 1e006c8-8442d5a5-13c4-50017-4816c456-2b7f7580-4816c456
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4816c456-98eef248-1890ef18
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2565796424 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30010 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv

Fwd_inv F6

SIP/2.0 100 Trying
To: <sip:82312793922@samsung.com:5060>
From: <sip:82312794329@samsung.com:5060>;tag=1dd4350-8442d5a5-13c4-50017-4816c456-41f71b1e-4816c456
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4816c456-98eef248-1890ef18
CSeq: 1 INVITE
Call-ID: 1e006c8-8442d5a5-13c4-50017-4816c456-2b7f7580-4816c456
Server: ININ-samsung-k1o0rnf-21113719
Content-Length: 0

Fwd_inv F7

```
INVITE sip:82312793922@165.213.66.94:9298;rinstance=aa040136f1a83a06 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1dd4350-8442d5a5-13c4-50017-4816c456-41f71b1e-
4816c456
To: <sip:82312793922@samsung.com:5060>
Call-ID: 1e006c8-8442d5a5-13c4-50017-4816c456-2b7f7580-4816c456
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk85d108316089ac74bc804eecc, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-4816c456-98eef248-1890ef18
Max-Forwards: 69
Supported: 100rel, replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2565796424 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30010 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Fwd_inv F8

```
SIP/2.0 180 Ringing
From: "82312794630"<sip:82312794630@samsung.com>;tag=00141ca537d4102504950a1c-5b2da26a
To: <sip:82312794329@samsung.com>;tag=1dd41d8-8442d5a5-13c4-50017-4816c456-64e583c1-4816c456
Call-ID: 00141ca5-37d40018-76a485c7-3a0c34e8@165.213.66.56
CSeq: 101 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk21691eeeb42916368ff6422ac
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK378c08e1
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0
```

Fwd_inv F9

```
SIP/2.0 180 Ringing
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4102504950a1c-5b2da26a
To: <sip:82312794329@samsung.com>;tag=1dd41d8-8442d5a5-13c4-50017-4816c456-64e583c1-4816c456
Call-ID: 00141ca5-37d40018-76a485c7-3a0c34e8@165.213.66.56
CSeq: 101 INVITE
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK378c08e1
Supported: 100rel, replaces
Contact: <sip:82312794329@165.213.66.132:5060>
```

Fwd_inv F10

```
SIP/2.0 200 OK
From: "82312794630"<sip:82312794630@samsung.com>;tag=00141ca537d4102504950a1c-5b2da26a
To: <sip:82312794329@samsung.com>;tag=1dd41d8-8442d5a5-13c4-50017-4816c456-64e583c1-4816c456
Call-ID: 00141ca5-37d40018-76a485c7-3a0c34e8@165.213.66.56
CSeq: 101 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk21691eeeb42916368ff6422ac
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK378c08e1
Supported: 100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 205

v=0
o=SAMSUNG_SIP_GATEWAY 2565796504 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30008 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Fwd_inv F11

SIP/2.0 200 OK
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4102504950a1c-5b2da26a
To: <sip:82312794329@samsung.com>;tag=1dd41d8-8442d5a5-13c4-50017-4816c456-64e583c1-4816c456
Call-ID: 00141ca5-37d40018-76a485c7-3a0c34e8@165.213.66.56
CSeq: 101 INVITE
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK378c08e1
Supported: 100rel, replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 205

v=0
o=SAMSUNG_SIP_GATEWAY 2565796504 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30008 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv

Fwd_inv F12

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk85d108316089ac74bc804eecc
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4816c456-98eef248-1890ef18
Contact: <sip:82312793922@165.213.66.94:9298;rinstance=aa040136f1a83a06>
To: <sip:82312793922@samsung.com:5060>;tag=1e3cf677
From: <sip:82312794329@samsung.com:5060>;tag=1dd4350-8442d5a5-13c4-50017-4816c456-41f71b1e-4816c456
Call-ID: 1e006c8-8442d5a5-13c4-50017-4816c456-2b7f7580-4816c456
CSeq: 1 INVITE
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 0

Fwd_inv F13

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4816c456-98eef248-1890ef18
Contact: <sip:82312793922@165.213.66.94:9298;rinstance=aa040136f1a83a06>
To: <sip:82312793922@samsung.com:5060>;tag=1e3cf677
From: <sip:82312794329@samsung.com:5060>;tag=1dd4350-8442d5a5-13c4-50017-4816c456-41f71b1e-4816c456
Call-ID: 1e006c8-8442d5a5-13c4-50017-4816c456-2b7f7580-4816c456
CSeq: 1 INVITE
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 0

Fwd_inv F14

ACK sip:82312794329@165.213.66.132:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK4cd44590
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4102504950a1c-5b2da26a
To: <sip:82312794329@samsung.com>;tag=1dd41d8-8442d5a5-13c4-50017-4816c456-64e583c1-4816c456
Call-ID: 00141ca5-37d40018-76a485c7-3a0c34e8@165.213.66.56
Max-Forwards: 70
Date: Fri, 02 May 2008 11:22:35 GMT
CSeq: 101 ACK
User-Agent: Cisco-CP7960G/8.0
Content-Length: 0

Fwd_inv F15

ACK sip:82312794329@165.213.66.132:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk5c55e719a63eb221e19a75945, SIP/2.0/UDP
165.213.66.56:5060;branch=z9hG4bK4cd44590
From: "82312794630" <sip:82312794630@samsung.com>;tag=00141ca537d4102504950a1c-5b2da26a
To: <sip:82312794329@samsung.com>;tag=1dd41d8-8442d5a5-13c4-50017-4816c456-64e583c1-4816c456
Call-ID: 00141ca5-37d40018-76a485c7-3a0c34e8@165.213.66.56
Max-Forwards: 69
Date: Fri, 02 May 2008 11:22:35 GMT
CSeq: 101 ACK
User-Agent: Cisco-CP7960G/8.0
Content-Length: 0

Fwd_inv F16

SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk85d108316089ac74bc804eecc
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4816c456-98eef248-1890ef18
Contact: <sip:82312793922@165.213.66.94:9298;rinstance=aa040136f1a83a06>
To: <sip:82312793922@samsung.com:5060>;tag=1e3cf677
From: <sip:82312794329@samsung.com:5060>;tag=1dd4350-8442d5a5-13c4-50017-4816c456-41f71b1e-
4816c456
Call-ID: 1e006c8-8442d5a5-13c4-50017-4816c456-2b7f7580-4816c456
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 185

v=0
o=- 5 2 IN IP4 165.213.66.94
s=CounterPath X-Lite 3.0
c=IN IP4 165.213.66.94
t=0 0
m=audio 35568 RTP/AVP 8 101
a=fmtp:101 0-15
a=rtpmap:101 telephone-event/8000
a=sendrecv

Fwd_inv F17

SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4816c456-98eef248-1890ef18
Contact: <sip:82312793922@165.213.66.94:9298;rinstance=aa040136f1a83a06>
To: <sip:82312793922@samsung.com:5060>;tag=1e3cf677
From: <sip:82312794329@samsung.com:5060>;tag=1dd4350-8442d5a5-13c4-50017-4816c456-41f71b1e-4816c456
Call-ID: 1e006c8-8442d5a5-13c4-50017-4816c456-2b7f7580-4816c456
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 185

v=0
o=- 5 2 IN IP4 165.213.66.94
s=CounterPath X-Lite 3.0
c=IN IP4 165.213.66.94
t=0 0
m=audio 35568 RTP/AVP 8 101
a=fmtp:101 0-15
a=rtpmap:101 telephone-event/8000
a=sendrecv

Fwd_inv F18

ACK sip:82312793922@165.213.66.94:9298;rinstance=aa040136f1a83a06 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1dd4350-8442d5a5-13c4-50017-4816c456-41f71b1e-4816c456
To: <sip:82312793922@samsung.com:5060>;tag=1e3cf677
Call-ID: 1e006c8-8442d5a5-13c4-50017-4816c456-2b7f7580-4816c456
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-4816c45e-98ef1264-4c05d5be
Max-Forwards: 70
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0

Fwd_inv F19

ACK sip:82312793922@165.213.66.94:9298;rinstance=aa040136f1a83a06 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1dd4350-8442d5a5-13c4-50017-4816c456-41f71b1e-4816c456
To: <sip:82312793922@samsung.com:5060>;tag=1e3cf677
Call-ID: 1e006c8-8442d5a5-13c4-50017-4816c456-2b7f7580-4816c456
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk7fc6ab028c30a30ea83dec70e, SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-4816c45e-98ef1264-4c05d5be
Max-Forwards: 69
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0

2.5. Alphanumeric Username

SIP UA may use alphabets as well as digits in its SIP-URI. In order to use alphanumeric username in SIP trunking mode, OfficeServ first has to register its alphanumeric username to a registrar. Once successfully registered, OfficeServ can send and receive SIP messages using the alphanumeric username contained in FROM/TO headers. From registrar's perspective, whichever is used for registration, there is no difference between handling an alphabetic username and handling a digit-only username.

2.5.1. Registering Alphanumeric Username

MMC837 SIP OPTIONS

ISP1
SIP SERVER: *ENABLE*
OUT PROXY: *samsung.com*
DNS SERVER1: *165.213.66.93*
USER NAME: *sungwoo1769*
AUTH USER: *82312794329*
AUTH PSWD: *1234*
REG PER USR: *DISABLE*
TRK REG EXP: *001800*

As shown above, username field in MMC837 is set to an alphanumeric value of 'sungwoo1769', and OfficeServ registers to a registrar as 'sungwoo1769'.

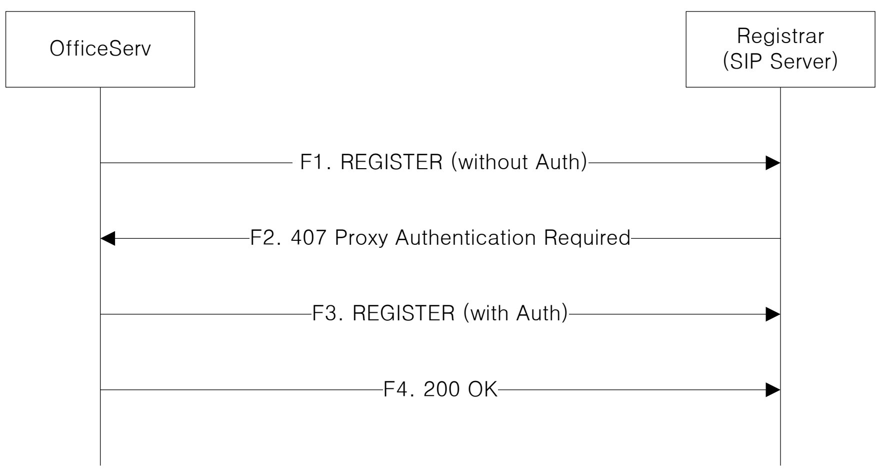


Figure 18. Register using Alphanumeric Username

Alpha_reg F1

REGISTER sip:samsung.com:5060 SIP/2.0
From: <sip:sungwoo1769@samsung.com:5060>;tag=1da1388-8442d5a5-13c4-50017-481bf8b3-303fd0b7-481bf8b3
To: <sip:sungwoo1769@samsung.com:5060>
Call-ID: 1dc2504-8442d5a5-13c4-50017-481bf8b1-5e78c88b-481bf8b1
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-481bf8b3-ad437d9a-23140674
Max-Forwards: 70
Supported: 100rel,replaces
Expires: 1800
Contact: <sip:sungwoo1769@165.213.66.132:5060>
Content-Length: 0

Alpha_reg F2

SIP/2.0 407 Proxy Authentication Required
To: <sip:sungwoo1769@samsung.com:5060>
From: <sip:sungwoo1769@samsung.com:5060>;tag=1da1388-8442d5a5-13c4-50017-481bf8b3-303fd0b7-481bf8b3
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481bf8b3-ad437d9a-23140674
CSeq: 1 REGISTER
Call-ID: 1dc2504-8442d5a5-13c4-50017-481bf8b1-5e78c88b-481bf8b1
Proxy-Authenticate: Digest
realm="165.213.66.93",qop="auth",algorithm="MD5",nonce="79393552c6f14bd7b7943cd86f1e5da7"
Content-Length: 0

Alpha_reg F3

REGISTER sip:samsung.com:5060 SIP/2.0
From: <sip:sungwoo1769@samsung.com:5060>;tag=1da1388-8442d5a5-13c4-50017-481bf8b3-303fd0b7-481bf8b3
To: <sip:sungwoo1769@samsung.com:5060>
Call-ID: 1dc2504-8442d5a5-13c4-50017-481bf8b1-5e78c88b-481bf8b1
CSeq: 2 REGISTER
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-481bf8b3-ad437dd6-2475c752
Max-Forwards: 70
Supported: 100rel,replaces
Expires: 1800
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="79393552c6f14bd7b7943cd86f1e5da7",uri="sip:samsung.com:5060",response="8f4607037930a33cd6c782b386b7c606",algorithm=MD5,cnonce="ad437dd6",qop=auth,nc=00000001\r
Contact: <sip:sungwoo1769@165.213.66.132:5060>
Content-Length: 0

Alpha_reg F4

```
SIP/2.0 200 OK
To: <sip:sungwoo1769@samsung.com:5060>;tag=26827
From: <sip:sungwoo1769@samsung.com:5060>;tag=1da1388-8442d5a5-13c4-50017-481bf8b3-303fd0b7-
481bf8b3
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481bf8b3-ad437dd6-2475c752
CSeq: 2 REGISTER
Call-ID: 1dc2504-8442d5a5-13c4-50017-481bf8b1-5e78c88b-481bf8b1
Contact: <sip:sungwoo1769@165.213.66.132:5060>;expires=300
Content-Length: 0
```

2.5.2. Outgoing Alphanumeric Username

When sending out an INVITE message to alphanumeric destination, we have to consider following two points:

- How to call alphanumeric destination from legacy station in OfficeServ system?
- How to match a specific station with a registered alphanumeric username?

DID and DOD numbers can be set using MMC321 Send CLIP table and MMC714 DID Digit table. When using digit-only caller and called number, there will be no problem. If OfficeServ has to use alphanumeric values, however, for the caller and called info, we need some intermediary storage to convert alphanumeric values into generic digit-only values.

MMC839 contains mapping information which is used to convert digit number to alphanumeric called username. As we cannot dial alphanumeric called name from a legacy station in OfficeServ we dial this intermediary digit instead, and let OfficeServ convert the digit into a designated alphanumeric called name.

MMC839 SIP USER

```
SP1-001
        USERNAME:
        AUTH UID:
        AUTH PWD:
        TEL NO:
OPP0001
        SITE URL: miyoung4692
        TEL NO: 4692
        CLI NAME: sungwoo1769
```

In above example, OfficeServ will convert a station-dialed digit ‘4692’ (TEL NO) to designated called name of ‘miyoung4692’ (SITE URL), which is finally put into To Header. CLI NAME field specifies the value which should be put into FROM header. **Note that value in CLI NAME field should be the same value that is registered to registrar**.

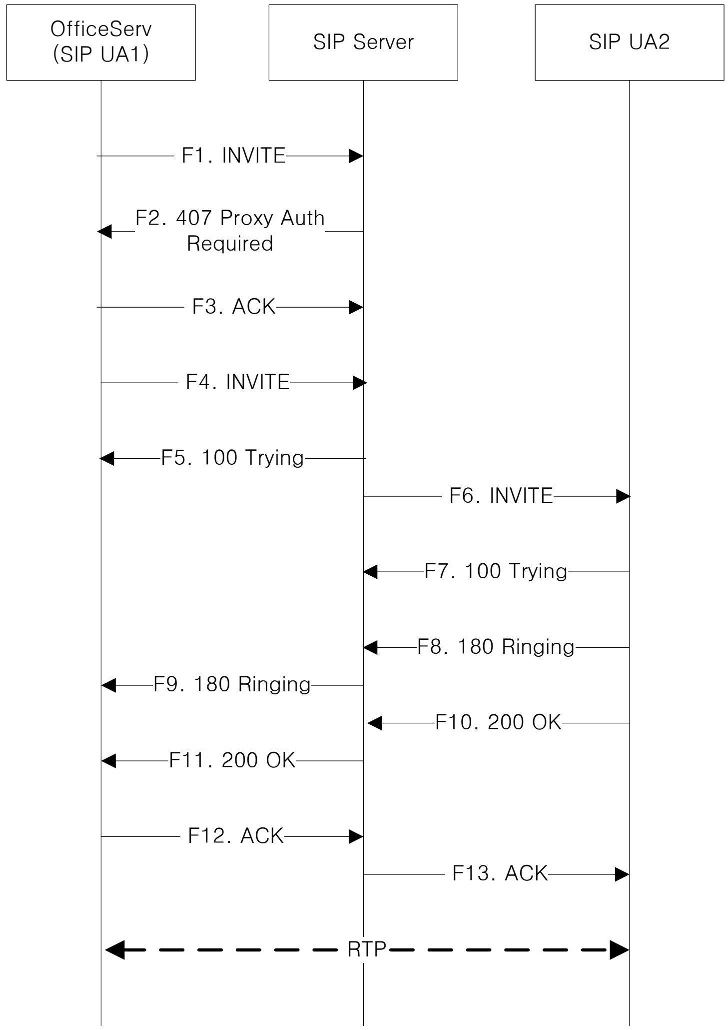


Figure 19. Basic Outbound Call using Alphanumeric Username

Alpha_Outbound F1

```
INVITE sip:miyoung4692@samsung.com:5060 SIP/2.0
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-
481bf69f
To: <sip:miyoung4692@samsung.com:5060>
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-481bf69f-ad3b5fe8-4dbd4596
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:201@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2906349544 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30002 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Alpha_Outbound F2

```
SIP/2.0 407 Proxy Authentication Required
To: <sip:miyoung4692@samsung.com:5060>
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-
481bf69f
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481bf69f-ad3b5fe8-4dbd4596
CSeq: 1 INVITE
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
Proxy-Authenticate: Digest
realm="165.213.66.93",qop="auth",algorithm="MD5",nonce="50f4a13551178946b5da191d8889563a"
Content-Length: 0
```

Alpha_Outbound F3

ACK sip:miyoung4692@samsung.com:5060 SIP/2.0
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-481bf69f
To: <sip:miyoung4692@samsung.com:5060>
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-481bf69f-ad3b5fe8-4dbd4596
Max-Forwards: 70
Contact: <sip:201@165.213.66.132:5060>
Content-Length: 0

Alpha_Outbound F4

INVITE sip:miyoung4692@samsung.com:5060 SIP/2.0
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-481bf69f
To: <sip:miyoung4692@samsung.com:5060>
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-481bf69f-ad3b601a-7dd4161c
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:201@165.213.66.132:5060>
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="50f4a13551178946b5da191d8889563a",uri="sip:miyoung4692@samsung.com:5060",response="cbe5b4057c50a1a7afdfed1b779a9207",algorithm=MD5,cnonce="ad3b601a",qop=auth,n
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2906349544 1 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30002 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv

Alpha_Outbound F5

```
SIP/2.0 100 Trying
To: <sip:miyoung4692@samsung.com:5060>
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-
481bf69f
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481bf69f-ad3b601a-7dd4161c
CSeq: 2 INVITE
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
Server: ININ-samsung-k1o0rnf-20847703
Content-Length: 0
```

Alpha_Outbound F6

```
INVITE sip:miyoung4692@165.213.66.56:5060 SIP/2.0
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-
481bf69f
To: <sip:miyoung4692@samsung.com:5060>
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk30593106abba9703b6b6b4448, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-481bf69f-ad3b601a-7dd4161c
Max-Forwards: 69
Supported: 100rel, replaces
Contact: <sip:201@165.213.66.132:5060>
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="50f4a13551178946b5da191d8889563a",uri="sip:mi
young4692@samsung.com:5060",response="cbe5b4057c50a1a7afdfed1b779a9207",algorithm=MD5,cnonce="a
d3b601a",qop=auth,n
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 2906349544 1 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30002 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Alpha_Outbound F7

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk30593106abba9703b6b6b4448, SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481bf69f-ad3b601a-7dd4161c
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-481bf69f
To: <sip:miyoung4692@samsung.com:5060>
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
Date: Tue, 06 May 2008 09:57:48 GMT
CSeq: 2 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:miyoung4692@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Content-Length: 0

Alpha_Outbound F8

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk30593106abba9703b6b6b4448, SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481bf69f-ad3b601a-7dd4161c
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-481bf69f
To: <sip:miyoung4692@samsung.com:5060>;tag=00141ca537d4180737ec0e9b-76545d39
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
Date: Tue, 06 May 2008 09:57:49 GMT
CSeq: 2 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:miyoung4692@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Content-Length: 0

Alpha_Outbound F9

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481bf69f-ad3b601a-7dd4161c
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-481bf69f
To: <sip:miyoung4692@samsung.com:5060>;tag=00141ca537d4180737ec0e9b-76545d39
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
Date: Tue, 06 May 2008 09:57:49 GMT
CSeq: 2 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:miyoung4692@165.213.66.56:5060;transport=udp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Content-Length: 0

Alpha_Outbound F10

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk30593106abba9703b6b6b4448, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-481bf69f-ad3b601a-7dd4161c
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-
481bf69f
To: <sip:miyoung4692@samsung.com:5060>;tag=00141ca537d4180737ec0e9b-76545d39
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
Date: Tue, 06 May 2008 09:58:03 GMT
CSeq: 2 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:miyoung4692@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Supported: replaces,join,norefersub
Content-Length: 206
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 9402 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 29762 RTP/AVP 8 101
c=IN IP4 165.213.66.56
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Alpha_Outbound F11

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481bf69f-ad3b601a-7dd4161c
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-
481bf69f
To: <sip:miyoung4692@samsung.com:5060>;tag=00141ca537d4180737ec0e9b-76545d39
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
Date: Tue, 06 May 2008 09:58:03 GMT
CSeq: 2 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:miyoung4692@165.213.66.56:5060;transport=udp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Supported: replaces, join, norefersub
Content-Length: 206
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 9402 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 29762 RTP/AVP 8 101
c=IN IP4 165.213.66.56
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Alpha_Outbound F12

```
ACK sip:miyoung4692@165.213.66.56:5060;transport=udp SIP/2.0
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-
481bf69f
To: <sip:miyoung4692@samsung.com:5060>;tag=00141ca537d4180737ec0e9b-76545d39
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
CSeq: 2 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-481bf6ad-ad3b94f4-56dc0270
Max-Forwards: 70
Contact: <sip:201@165.213.66.132:5060>
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="50f4a13551178946b5da191d8889563a",uri="sip:mi
young4692@samsung.com:5060",response="cbe5b4057c50a1a7afdfed1b779a9207",algorithm=MD5,cnonce="a
d3b601a",qop=auth,n
Content-Length: 0
```

Alpha_Outbound F13

```
ACK sip:miyoung4692@165.213.66.56:5060 SIP/2.0
From: <sip:sungwoo1769@samsung.com:5060>;tag=1daa550-8442d5a5-13c4-50017-481bf69f-14e8e4d4-
481bf69f
To: <sip:miyoung4692@samsung.com:5060>;tag=00141ca537d4180737ec0e9b-76545d39
Call-ID: 1dad198-8442d5a5-13c4-50017-481bf69f-2664efd6-481bf69f
CSeq: 2 ACK
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bkf826b940a89e2bf9c9ccb63b8, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-481bf6ad-ad3b94f4-56dc0270
Max-Forwards: 69
Contact: <sip:201@165.213.66.132:5060>
Proxy-Authorization: Digest
username="82312794329",realm="165.213.66.93",nonce="50f4a13551178946b5da191d8889563a",uri="sip:mi
young4692@samsung.com:5060",response="cbe5b4057c50a1a7afdfed1b779a9207",algorithm=MD5,cnonce="a
d3b601a",qop=auth,n
Content-Length: 0
```

2.5.3. Incoming Alphanumeric Username

As mentioned before, receiving an INVITE message which contains digit-only called number in its To Header and mapping it to digit-only station number can be done by setting MMC714 table alone. When OfficeServ, however, is receiving an INVITE message which contains alphanumeric called number, it has to have additional table which maps the alphanumeric value to digit-only station number in order to decide which station to receive the call because generic MMC714 table only accepts digit value. As in the case of outgoing alphanumeric username, OfficeServ has this alphanumeric-to-digit conversion mechanism in MMC839 table.

MMC839 SIP USER

SP1-001
    USERNAME: *sungwoo1769*
    AUTH UID:
    AUTH PWD:
    TEL NO: *201*
OPP0001
    SITE URL: *miyoung4692*
    TEL NO: *4692*
    CLI NAME: *sungwoo1769*

In above example, OfficeServ converts the alphanumeric value (sungwoo1769) in USERNAME field into digit value (201) specified in TEL NO field. The TEL NO value can be some other value which is different from actual station number because this number will be mapped to a value in generic MMC714 DID table, which originally has a role of mapping the called number to station number. To eliminate confusion, however, I recommend using station number directly in MMC839 and set the same value in MMC714 DID table as well.

Following is the MMC714 table setting example. (Set by Default)

MMC714 DID DIGIT

DID DIGIT (001)
DGT: *2***
1: *B*
2: *B*

As we can see in the example, the alphanumeric value (sungwoo1769) is converted into a digit number (201) in MMC839 and the digit value is mapped to station DID number in MMC714.

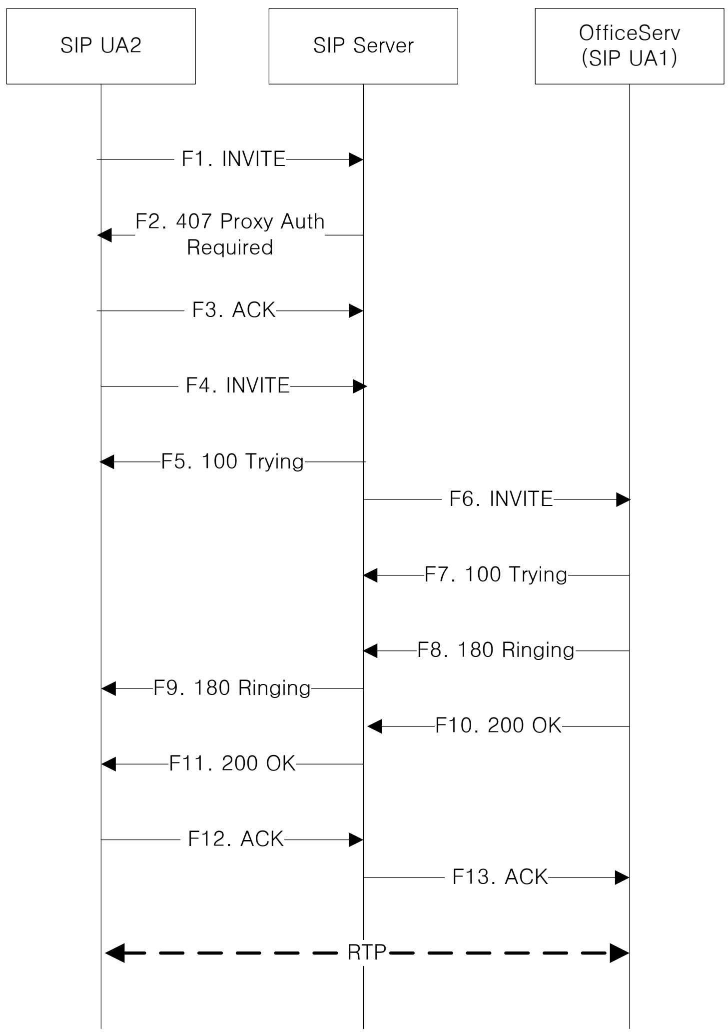


Figure 20. Basic Inbound Call using Alphanumeric Username

Alpha_Inbound F1

```
INVITE sip:sungwoo1769@samsung.com SIP/2.0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK4cca955e
From: "miyoung4692" <sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
To: <sip:sungwoo1769@samsung.com>
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
Max-Forwards: 70
Date: Tue, 06 May 2008 23:31:12 GMT
CSeq: 101 INVITE
User-Agent: Cisco-CP7960G/8.0
Contact: <sip:miyoung4692@165.213.66.56:5060;transport=udp>
Expires: 180
Accept: application/sdp
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Supported: replaces,join,norefersub
Content-Length: 277
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 2307 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 16884 RTP/AVP 0 8 18 101
c=IN IP4 165.213.66.56
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Alpha_ Inbound F2

```
SIP/2.0 407 Proxy Authentication Required
To: <sip:sungwoo1769@samsung.com>
From: "miyoung4692" <sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK4cca955e
CSeq: 101 INVITE
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
Proxy-Authenticate: Digest
realm="165.213.66.93",qop="auth",algorithm="MD5",nonce="f4dbd55bb4c39efaca9a82f27e5dc61f"
Content-Length: 0
```

Alpha_ Inbound F3

ACK sip:sungwoo1769@samsung.com SIP/2.0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK4cca955e
From: "miyoung4692" <sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
To: <sip:sungwoo1769@samsung.com>
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
Date: Tue, 06 May 2008 23:31:12 GMT
CSeq: 101 ACK
Content-Length: 0

Alpha_Inbound F4

INVITE sip:sungwoo1769@samsung.com SIP/2.0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK70bea949
From: "miyoung4692" <sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
To: <sip:sungwoo1769@samsung.com>
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
Max-Forwards: 70
Date: Tue, 06 May 2008 23:31:12 GMT
CSeq: 102 INVITE
User-Agent: Cisco-CP7960G/8.0
Contact: <sip:miyoung4692@165.213.66.56:5060;transport=udp>
Proxy-Authorization: Digest
username="82312794630",realm="165.213.66.93",uri="sip:sungwoo1769@samsung.com",response="20b39d6
bd8d06efb2e7000c21a1dd36d",nonce="f4dbd55bb4c39efaca9a82f27e5dc61f",cnonce="7f9905e7",qop="auth",
nc=00000001,algori
Expires: 180
Accept: application/sdp
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Supported: replaces,join,norefersub
Content-Length: 277
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 2307 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 16884 RTP/AVP 0 8 18 101
c=IN IP4 165.213.66.56
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

Alpha_ Inbound F5

```
SIP/2.0 100 Trying
To: <sip:sungwoo1769@samsung.com>
From: "miyoung4692" <sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK70bea949
CSeq: 102 INVITE
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
Server: ININ-samsung-k1o0rnf-20847703
Content-Length: 0
```

Alpha_Inbound F6

```
INVITE sip:sungwoo1769@165.213.66.132:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bkdd303660bac68ac5b4ab29589, SIP/2.0/UDP
165.213.66.56:5060;branch=z9hG4bK70bea949
From: "miyoung4692" <sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
To: <sip:sungwoo1769@samsung.com>
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
Max-Forwards: 69
Date: Tue, 06 May 2008 23:31:12 GMT
CSeq: 102 INVITE
User-Agent: Cisco-CP7960G/8.0
Contact: <sip:miyoung4692@165.213.66.56:5060;transport=udp>
Proxy-Authorization: Digest
username="82312794630",realm="165.213.66.93",uri="sip:sungwoo1769@samsung.com",response="20b39d6
bd8d06efb2e7000c21a1dd36d",nonce="f4dbd55bb4c39efaca9a82f27e5dc61f",cnonce="7f9905e7",qop="auth",
nc=00000001,algori
Expires: 180
Accept: application/sdp
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Supported: replaces, join, norefersub
Content-Length: 277
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 2307 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 16884 RTP/AVP 0 8 18 101
c=IN IP4 165.213.66.56
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Alpha_ Inbound F7

SIP/2.0 100 Trying
From: "miyoung4692"<sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
To: <sip:sungwoo1769@samsung.com>
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
CSeq: 102 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bkdd303660bac68ac5b4ab29589
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK70bea949
Supported: 100rel,replaces
Contact: <sip:sungwoo1769@165.213.66.132:5060>
Content-Length: 0

Alpha_ Inbound F8

SIP/2.0 180 Ringing
From: "miyoung4692"<sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
To: <sip:sungwoo1769@samsung.com>;tag=1da5a08-8442d5a5-13c4-50017-481cb53f-61851fc-481cb53f
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
CSeq: 102 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bkdd303660bac68ac5b4ab29589
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK70bea949
Supported: 100rel,replaces
Contact: <sip:sungwoo1769@165.213.66.132:5060>
Content-Length: 0

Alpha_ Inbound F9

SIP/2.0 180 Ringing
From: "miyoung4692" <sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
To: <sip:sungwoo1769@samsung.com>;tag=1da5a08-8442d5a5-13c4-50017-481cb53f-61851fc-481cb53f
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
CSeq: 102 INVITE
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK70bea949
Supported: 100rel, replaces
Contact: <sip:sungwoo1769@165.213.66.132:5060>
Content-Length: 0

Alpha_Inbound F10

```
SIP/2.0 200 OK
From: "miyoung4692"<sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
To: <sip:sungwoo1769@samsung.com>;tag=1da5a08-8442d5a5-13c4-50017-481cb53f-61851fc-481cb53f
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
CSeq: 102 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bkdd303660bac68ac5b4ab29589
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK70bea949
Supported: 100rel,replaces
Contact: <sip:sungwoo1769@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 205

v=0
o=SAMSUNG_SIP_GATEWAY 2955152194 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30010 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Alpha_Inbound F11

```
SIP/2.0 200 OK
From: "miyoung4692" <sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
To: <sip:sungwoo1769@samsung.com>;tag=1da5a08-8442d5a5-13c4-50017-481cb53f-61851fc-481cb53f
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
CSeq: 102 INVITE
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK70bea949
Supported: 100rel, replaces
Contact: <sip:sungwoo1769@165.213.66.132:5060>
Content-Type: application/sdp
Content-Length: 205

v=0
o=SAMSUNG_SIP_GATEWAY 2955152194 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30010 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Alpha_ Inbound F9

```
ACK sip:sungwoo1769@165.213.66.132:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.56:5060;branch=z9hG4bK186572f3
From: "miyoung4692" <sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
To: <sip:sungwoo1769@samsung.com>;tag=1da5a08-8442d5a5-13c4-50017-481cb53f-61851fc-481cb53f
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
Max-Forwards: 70
Date: Tue, 06 May 2008 23:31:15 GMT
CSeq: 102 ACK
User-Agent: Cisco-CP7960G/8.0
Proxy-Authorization: Digest
username="82312794630",realm="165.213.66.93",uri="sip:sungwoo1769@samsung.com",response="20b39d6
bd8d06efb2e7000c21a1dd36d",nonce="f4dbd55bb4c39efaca9a82f27e5dc61f",cnonce="7f9905e7",qop="auth",
nc=00000001,algori
Content-Length: 0
```

Alpha_ Inbound F9

```
ACK sip:sungwoo1769@165.213.66.132:5060 SIP/2.0
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bkd070bf63bde8d9a295b33b3c2, SIP/2.0/UDP
165.213.66.56:5060;branch=z9hG4bK186572f3
From: "miyoung4692" <sip:miyoung4692@samsung.com>;tag=00141ca537d419223d1941c5-446f68e0
To: <sip:sungwoo1769@samsung.com>;tag=1da5a08-8442d5a5-13c4-50017-481cb53f-61851fc-481cb53f
Call-ID: 00141ca5-37d4001c-4e49091c-778516b0@165.213.66.56
Max-Forwards: 69
Date: Tue, 06 May 2008 23:31:15 GMT
CSeq: 102 ACK
User-Agent: Cisco-CP7960G/8.0
Proxy-Authorization: Digest
username="82312794630",realm="165.213.66.93",uri="sip:sungwoo1769@samsung.com",response="20b39d6
bd8d06efb2e7000c21a1dd36d",nonce="f4dbd55bb4c39efaca9a82f27e5dc61f",cnonce="7f9905e7",qop="auth",
nc=00000001,algori
Content-Length: 0
```

2.5.4. Multiple Alphanumeric Usernames

Previous sections so far has described how to set OfficeServ's MMC databases in order to support an alphanumeric username. Then, it is high time to talk about how to make OfficeServ support multiple alphanumeric usernames.

If understood previous sections, it is relatively easy to set multiple alphanumeric usernames in MMC database. Let's assume that OfficeServ has been assigned one primary alphanumeric username (sungwoo1769) and one secondary alphanumeric username (tigerwoods). In this scenario, a registrar server requires authentication credential based on the primary alphanumeric username for both primary and secondary usernames. And OfficeServ system has two legacy stations (201 and 202) whose number will be mapped to each of the alphanumeric SIP username. Following shows how to setup MMC databases.

MMC839 SIP USER

```
SP1-001
          USERNAME: sungwoo1769
          AUTH UID:
          AUTH PWD:
          TEL NO: 201
SP1-002
          USERNAME: tigerwoods
          AUTH UID:
          AUTH PWD:
          TEL NO: 202
OPP0001
          SITE URL: miyoung4692
          TEL NO: 4692
```

SP-1 means 'Service Provider #1' and currently OfficeServ supports only one SIP Carrier at a time, therefore it should always be SP-1. From whichever station we make an outbound call dialing '4692', OfficeServ will put 'miyoung4692' in To Header and 'sungwoo1769' in From Header of the outgoing INVITE message.

As to call receiving case, OfficeServ first checks value in To Header of a incoming INVITE message and converts the alphanumeric value to a digit value specified in TEL NO field, which finally decides a station to receive the call.

2.6. SIP Trunking Related MMC837 Options

This section describes miscellaneous MMC837 database options which are related to SIP trunking message formats or call flows. As different SIP servers in different SIP carriers may require each different message specification or call flows, OfficeServ operator should adjust following MMC837 options in accordance with the server's request.

2.6.1. Proxy Name field

Values in this field will override the URL part in FROM and TO header of OfficeServ's SIP messages. If some SIP carrier may want to receive SIP messages whose TO and FROM headers contain a value that is different from its outbound server domain name. In this case, we need to put the designated value into this PROXY NAME field. Unless designated, its value will remain as NULL and a value specified in OUT PROXY field will be used.

MMC837 SIP OPTIONS

```
ISP1
          SIP SERVER: ENABLE
          OUT PROXY: samsung.com
          PROXY NAME: sec.samsung.com
          DNS SERVER1: 165.213.66.93
          USER NAME: sungwoo1769
          AUTH USER: 82312794329
          AUTH PSWD: 1234
          REG PER USR: DISABLE
```

Note that changing a value in PROXY NAME field simply changes the URL part of SIP messages and does not affect the messages' outbound address nor DNS query result for outbound server.

When PROXY NAME is set to sec.samsung.com

```
INVITE sip:miyoung4692@sec.samsung.com:5060 SIP/2.0
From: <sip:sungwoo1769@sec.samsung.com:5060>;tag=1da1f48-8442d5a5-13c4-50017-481e0956-7c9fbda1-481e0956
To: <sip:miyoung4692@sec.samsung.com:5060>
Call-ID: 1da7dc0-8442d5a5-13c4-50017-481e0956-3f15e5d3-481e0956
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-481e0956-b5547a48-294c66f5
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:201@165.213.66.132:5060>
```

When PROXY NAME is set to NULL

```
INVITE sip:miyoung4692@samsung.com:5060 SIP/2.0
From: <sip:sungwoo1769@samsung.com:5060>;tag=1da1f48-8442d5a5-13c4-50017-481e0956-7c9fbda1-481e0956
To: <sip:miyoung4692@samsung.com:5060>
Call-ID: 1da7dc0-8442d5a5-13c4-50017-481e0956-3f15e5d3-481e0956
CSeq: 2 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-481e0956-b5547a48-294c66f5
Max-Forwards: 70
Supported: 100rel,replaces
Contact: <sip:201@165.213.66.132:5060>
```

2.6.2. Session TMR

If this SESSION TMR option is set to 'UPDATE' or 'REINVITE', OfficeServ system puts a 'Session Expires' header into its outbound INVITE messages. Session Timer is used to refresh an active SIP session by sending a SIP request message to the other peer. The SIP request messages can be either UPDATE or re-INVITE, and the request messages are sent at each time period whose interval is specified in SESSION EXP field. If the refresher never gets the answer (200 OK) for the refresh request, it sends a BYE message to disconnect the SIP session. For more detailed, please refer to RFC4028.

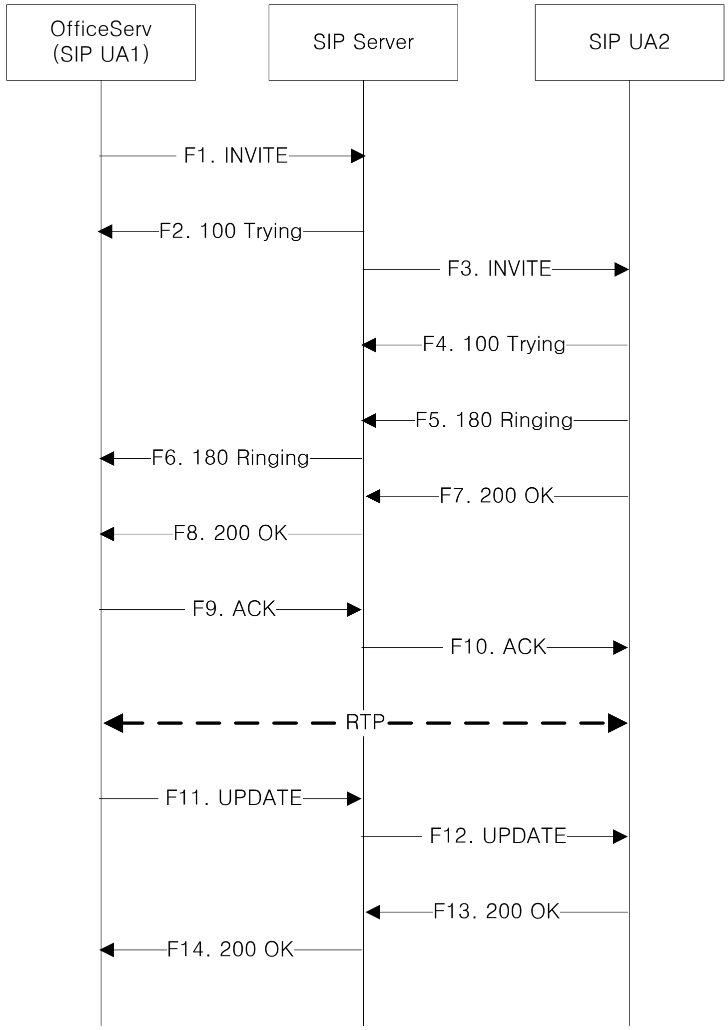


Figure 21. Session Refreshed by OfficeServ

In following example, as SESSION TMR is set to UPDATE and SESSION EXP is set to 90 (sec), OfficeServ system sends UPDATE message at every 45 seconds which is the half of the value in Session Expires header.

MMC837 SIP OPTIONS

ISP1 SIP SERVER: *ENABLE* OUT PROXY: *samsung.com* DNS SERVER1: *165.213.66.93* USER NAME: *82312794329* AUTH USER: *82312794329* AUTH PSWD: *1234* REG PER USR: *DISABLE* SESSION TMR: *DUPATE* SESSION EXP: *000090*

Session_exp F1

```
INVITE sip:82312794630@samsung.com:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-
481ea4a6
To: <sip:82312794630@samsung.com:5060>
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-481ea4a6-b7b32bb8-5fd6cf50
Max-Forwards: 70
Supported: timer,100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Session-Expires: 90;refresher=uac
Min-SE: 45
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 3081972664 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30002 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Session_exp F2

```
SIP/2.0 100 Trying
To: <sip:82312794630@samsung.com:5060>
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-
481ea4a6
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481ea4a6-b7b32bb8-5fd6cf50
CSeq: 1 INVITE
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
Server: ININ-samsung-k1o0rnf-20847703
Content-Length: 0
```

Session_exp F3

```
INVITE sip:82312794630@165.213.66.56:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-
481ea4a6
To: <sip:82312794630@samsung.com:5060>
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
CSeq: 1 INVITE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bke1adb56ad8083a9d7275eaf32, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-481ea4a6-b7b32bb8-5fd6cf50
Max-Forwards: 69
Supported: timer, 100rel, replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Session-Expires: 90;refresher=uac
Min-SE: 45
Content-Type: application/sdp
Content-Length: 255

v=0
o=SAMSUNG_SIP_GATEWAY 3081972664 0 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30002 RTP/AVP 18 4 8 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Session_exp F4

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bke1adb56ad8083a9d7275eaf32, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-481ea4a6-b7b32bb8-5fd6cf50
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-
481ea4a6
To: <sip:82312794630@samsung.com:5060>
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
Date: Thu, 08 May 2008 10:45:18 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Content-Length: 0
```

Session_exp F5

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bke1adb56ad8083a9d7275eaf32, SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481ea4a6-b7b32bb8-5fd6cf50
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-481ea4a6
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d41c0d7959f2e9-622db6fd
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
Date: Thu, 08 May 2008 10:45:18 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Content-Length: 0

Session_exp F6

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481ea4a6-b7b32bb8-5fd6cf50
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-481ea4a6
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d41c0d7959f2e9-622db6fd
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
Date: Thu, 08 May 2008 10:45:18 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Content-Length: 0

Session_exp F7

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bke1adb56ad8083a9d7275eaf32, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-481ea4a6-b7b32bb8-5fd6cf50
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-
481ea4a6
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d41c0d7959f2e9-622db6fd
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
Date: Thu, 08 May 2008 10:45:19 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Supported: replaces,join,norefersub
Content-Length: 207
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 21377 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 29472 RTP/AVP 8 101
c=IN IP4 165.213.66.56
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Session_exp F8

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481ea4a6-b7b32bb8-5fd6cf50
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-
481ea4a6
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d41c0d7959f2e9-622db6fd
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
Date: Thu, 08 May 2008 10:45:19 GMT
CSeq: 1 INVITE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Supported: replaces, join, norefersub
Content-Length: 207
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 21377 0 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 29472 RTP/AVP 8 101
c=IN IP4 165.213.66.56
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Session_exp F9

```
ACK sip:82312794630@165.213.66.56:5060;transport=udp SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-
481ea4a6
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d41c0d7959f2e9-622db6fd
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-481ea4a8-b7b332e8-112f2434
Max-Forwards: 70
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0
```

Session_exp F10

```
ACK sip:82312794630@165.213.66.56:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-
481ea4a6
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d41c0d7959f2e9-622db6fd
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
CSeq: 1 ACK
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk57281bfc7a05dfda822819e38, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-481ea4a8-b7b332e8-112f2434
Max-Forwards: 69
Contact: <sip:82312794329@165.213.66.132:5060>
Content-Length: 0
```

Session_exp F11

```
UPDATE sip:82312794630@165.213.66.56:5060;transport=udp SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-
481ea4a6
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d41c0d7959f2e9-622db6fd
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
CSeq: 2 UPDATE
Via: SIP/2.0/UDP 165.213.66.132:5060;rport;branch=z9hG4bK-481ea4d5-b7b3e2d8-1a850112
Max-Forwards: 70
Supported: timer,100rel,replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Session-Expires: 1800;refresher=uac
Min-SE: 100
Content-Type: application/sdp
Content-Length: 205

v=0
o=SAMSUNG_SIP_GATEWAY 3081972664 1 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30002 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Session_exp F12

```
UPDATE sip:82312794630@165.213.66.56:5060 SIP/2.0
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-
481ea4a6
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d41c0d7959f2e9-622db6fd
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
CSeq: 2 UPDATE
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk878219c19d30a08d79327a2e9, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-481ea4d5-b7b3e2d8-1a850112
Max-Forwards: 69
Supported: timer, 100rel, replaces
Contact: <sip:82312794329@165.213.66.132:5060>
Session-Expires: 1800;refresher=uac
Min-SE: 100
Content-Type: application/sdp
Content-Length: 205

v=0
o=SAMSUNG_SIP_GATEWAY 3081972664 1 IN IP4 165.213.66.132
s=SIP_CALL
c=IN IP4 165.213.66.132
t=0 0
m=audio 30002 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=sendrecv
```

Session_exp F13

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.93;branch=z9hG4bk878219c19d30a08d79327a2e9, SIP/2.0/UDP
165.213.66.132:5060;rport=5060;branch=z9hG4bK-481ea4d5-b7b3e2d8-1a850112
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-
481ea4a6
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d41c0d7959f2e9-622db6fd
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
Date: Thu, 08 May 2008 10:46:05 GMT
CSeq: 2 UPDATE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
Content-Length: 207
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 21377 1 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 29472 RTP/AVP 8 101
c=IN IP4 165.213.66.56
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Session_exp F14

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 165.213.66.132:5060;rport=5060;branch=z9hG4bK-481ea4d5-b7b3e2d8-1a850112
From: <sip:82312794329@samsung.com:5060>;tag=1d8d8e0-8442d5a5-13c4-50017-481ea4a6-1dd3f936-481ea4a6
To: <sip:82312794630@samsung.com:5060>;tag=00141ca537d41c0d7959f2e9-622db6fd
Call-ID: 1d931f8-8442d5a5-13c4-50017-481ea4a6-10022218-481ea4a6
Date: Thu, 08 May 2008 10:46:05 GMT
CSeq: 2 UPDATE
Server: Cisco-CP7960G/8.0
Contact: <sip:82312794630@165.213.66.56:5060;transport=udp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE
Content-Length: 207
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 21377 1 IN IP4 165.213.66.56
s=SIP Call
t=0 0
m=audio 29472 RTP/AVP 8 101
c=IN IP4 165.213.66.56
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

2.6.3. Asserted ID

ASSERTED ID item in MMC 837. According to SIP connect 1.0 standard, SIP servers are expecting to receive SIP messages whose FROM header contains secondary number and P-ASSERTED-ID header contains primary header. However, some SIP servers want opposite way: primary number in FROM header and secondary number in P-ASSERTED-ID header, which does not match the SIP connect 1.0. The main reason we put ALTERNATE option field is to provide more flexibility to our customers who will use our OfficeServ system in various environments. When SIP server does not care P-ASSERTED-ID header, you simply leave this option value to NONE by default.

3. SIP Peering Services

SIP peering is relatively simple compared to SIP trunking in that it does not have to concern about registration nor outbound SIP server's behavior. On the other hand, SIP peering's functionalities are more depending on SIP UAs that are being involved in a SIP session and thus it has relatively limited functionalities.

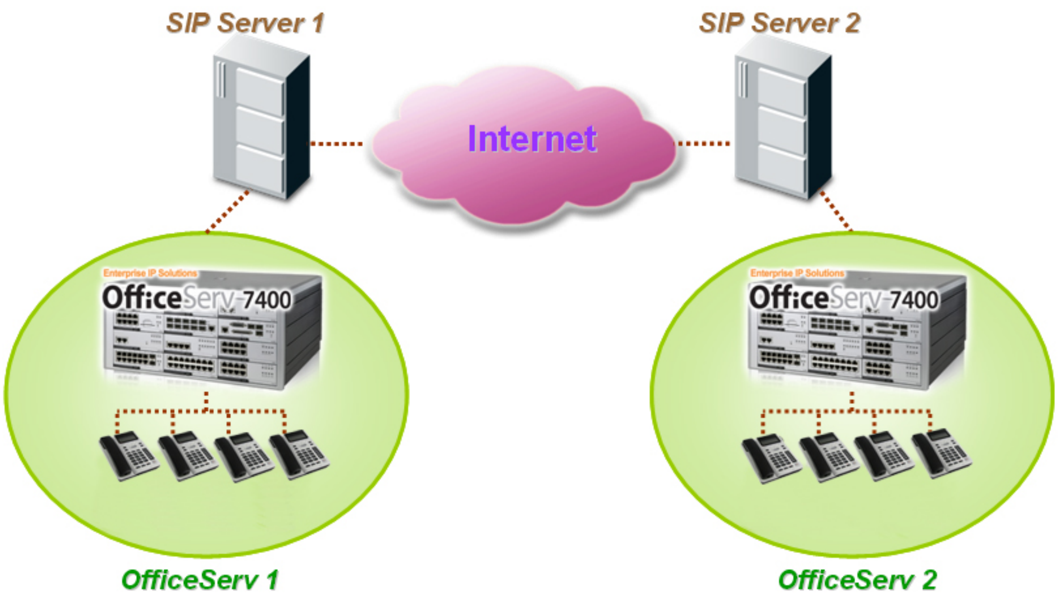


Figure 22. Overall Configuration for SIP Peering mode and SIP Station mode

SIP peer in this context means SIP UA and SIP peering does not need any intermediary SIP server in between two SIP peers. In SIP peering, all the SIP messages are out-bounded toward each other, therefore understanding outbound address setting is essential.

3.1. Basic Call Setup

As mentioned above, to make an outbound call, OfficeServ first needs to know where to send the INVITE message. Once destination is set, OfficeServ can send INVITE message and make a SIP session with the other peer.

MMC832 and MMC833 table contains dialed number-outbound IP address mapping mechanism. Let's look at following MMC example.

MMC832 VOIP OUT DGT

(O:00)
ACCESS DGT: *2* (target destination prefix number)
INSERT DGT:
DGT LENGTH: *1*
IP TABLE: *0*
IP START: *0*
SERVER USE: *NO*
URI TYPE: *SIP*

SERVER USE field is set to 'NO' and this makes OfficeServ set outgoing INVITE message's outbound address to an IP address specified in MMC833 (IP TABLE:0 and IP START index: 0). Note that if the SERVER USE field is set and OfficeServ is legitimately registered to a registrar, it will set the outbound address to an address specified in MMC837 OUT PROXY.

MMC833 VOIP IP ADDR

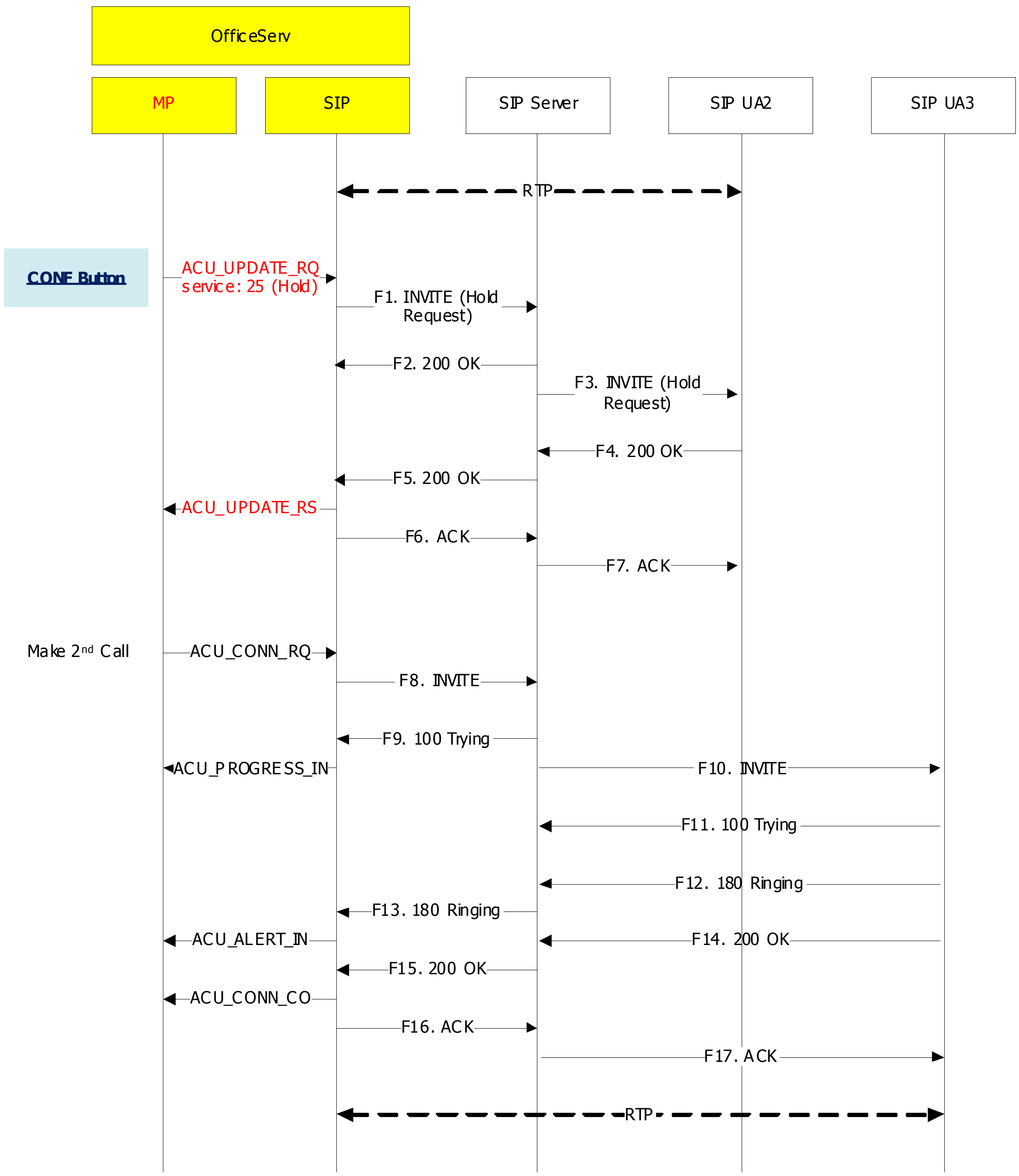
TB (00) ENTRY (00): *165.213.66.91* (target destination ip address)
TB (00) ENTRY (01):
TB (00) ENTRY (02):
.
.
.
TB (01) ENTRY (00):
TB (01) ENTRY (01):

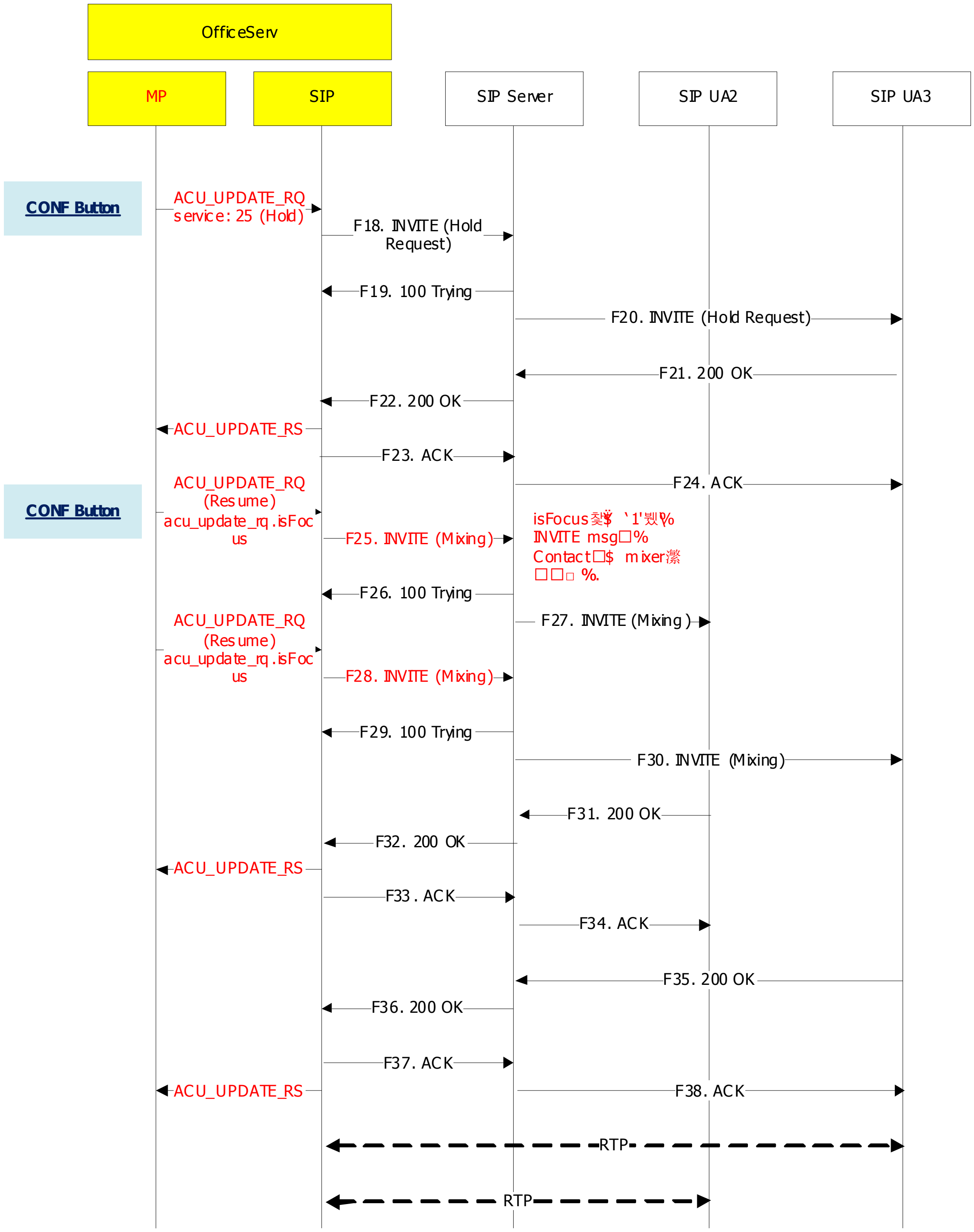
MMC833 table contains IP address list which will be specified as an outbound IP address of outgoing message by MMC832 setting. In the above example, MMC832 specifies IP TABLE '0' and IP START '0' which is mapped to TB '0' and ENTRY '00' in MMC833 and finally designates an IP address '165.213.66.91' as an outbound IP address.

4. SIP Conference Call Flows

4.1. Conference Call Flow

OfficeServ supports supervised conference, which means that OfficeServ itself become a conference mixer and distribute RTP packets to each of conference member. When conference initiator goes out of the conference, the conference ends.




OfficeServ
MP
SIP
SIP Server
SIP UA2
SIP UA3
CONF Button
ACU_UPDATE_RQ
service: 25 (Hold)
F18. INVITE (Hold Request)
F19. 100 Trying
F20. INVITE (Hold Request)
F21. 200 OK
F22. 200 OK
ACU_UPDATE_RS
F23. ACK
F24. ACK
CONF Button
ACU_UPDATE_RQ
(Resume)
acu_update_rq.isFocus
F25. INVITE (Mixing)
F26. 100 Trying
ACU_UPDATE_RQ
(Resume)
acu_update_rq.isFocus
F27. INVITE (Mixing)
F28. INVITE (Mixing)
F29. 100 Trying
F30. INVITE (Mixing)
F31. 200 OK
F32. 200 OK
ACU_UPDATE_RS
F33. ACK
F34. ACK
F35. 200 OK
F36. 200 OK
F37. ACK
ACU_UPDATE_RS
F38. ACK
RTP
RTP

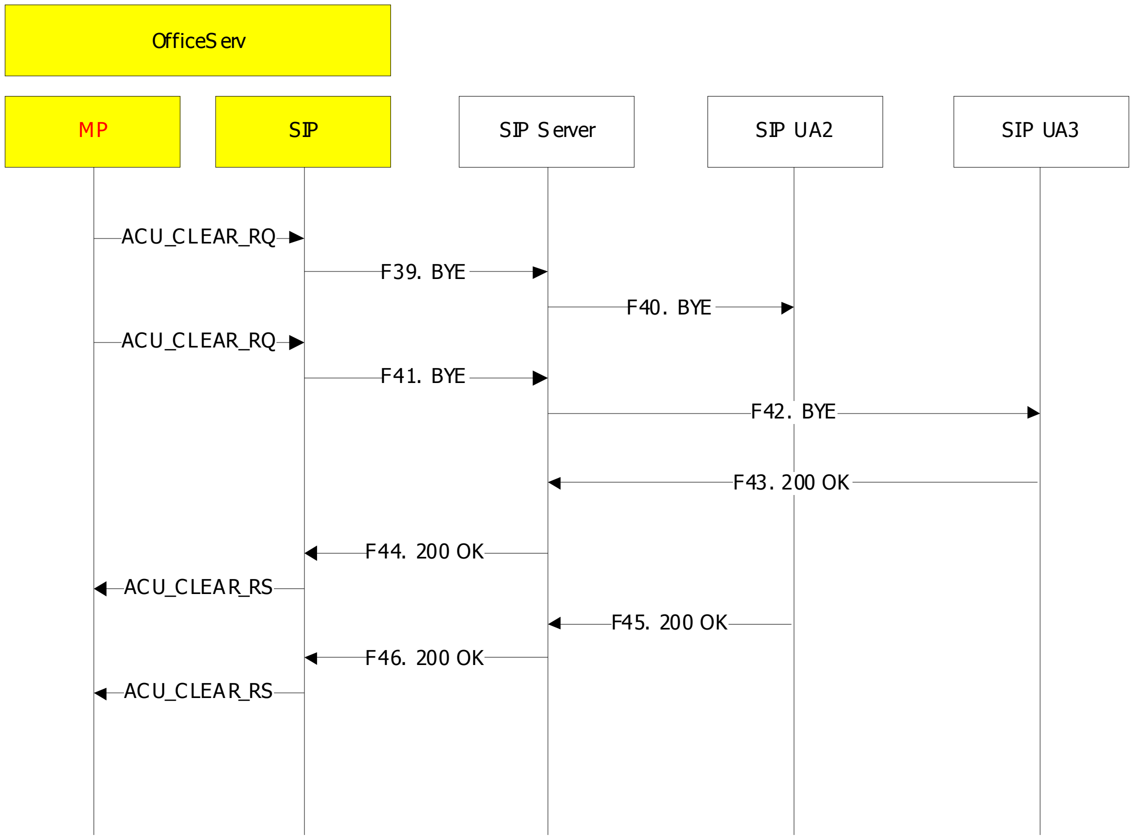


Figure 23. Conference Call Flow

4.2. Brief Conference Call Flow #1

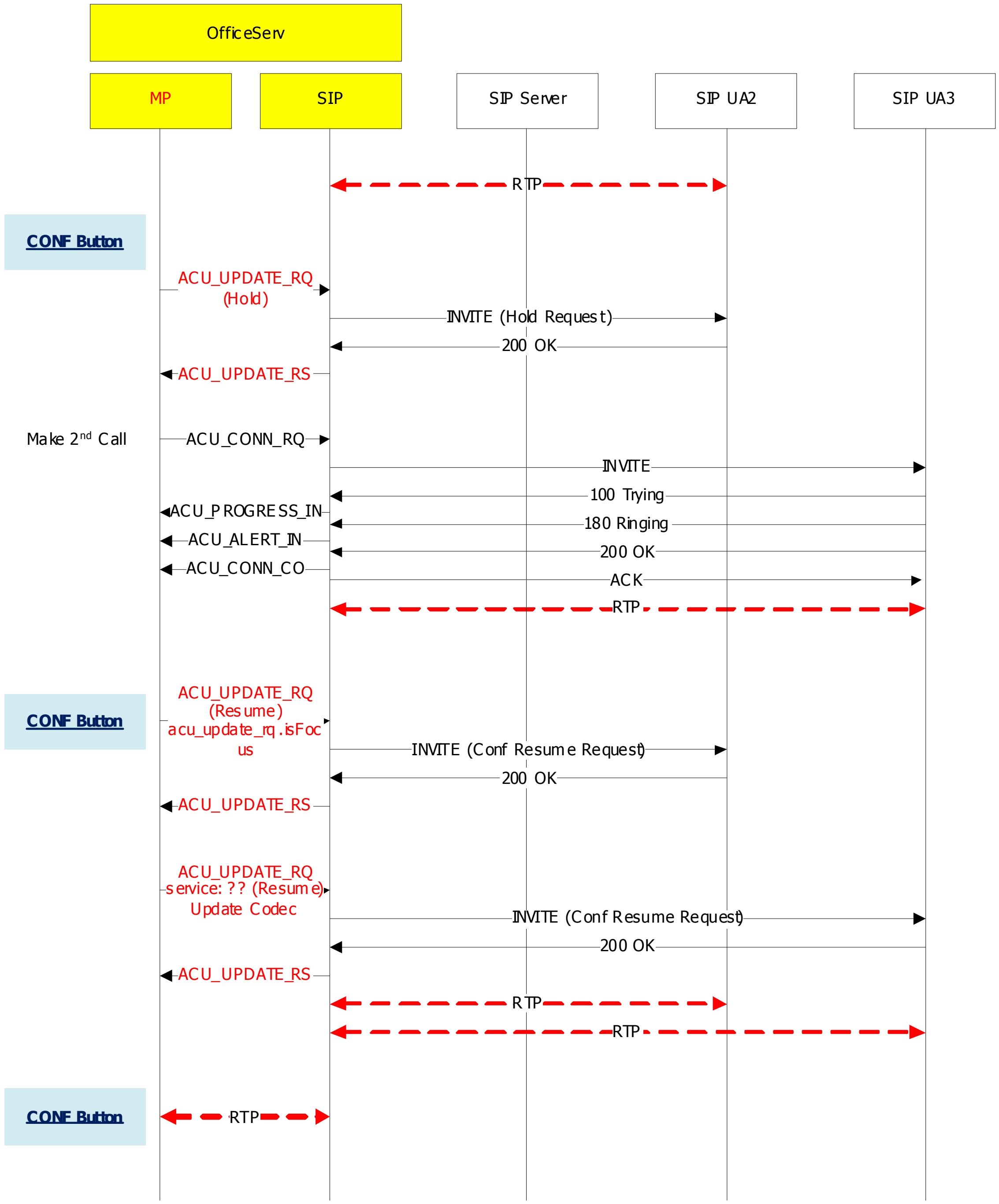


Figure 24. Brief Call Conference Call Flow 1

4.3. Brief Conference Call Flow #2

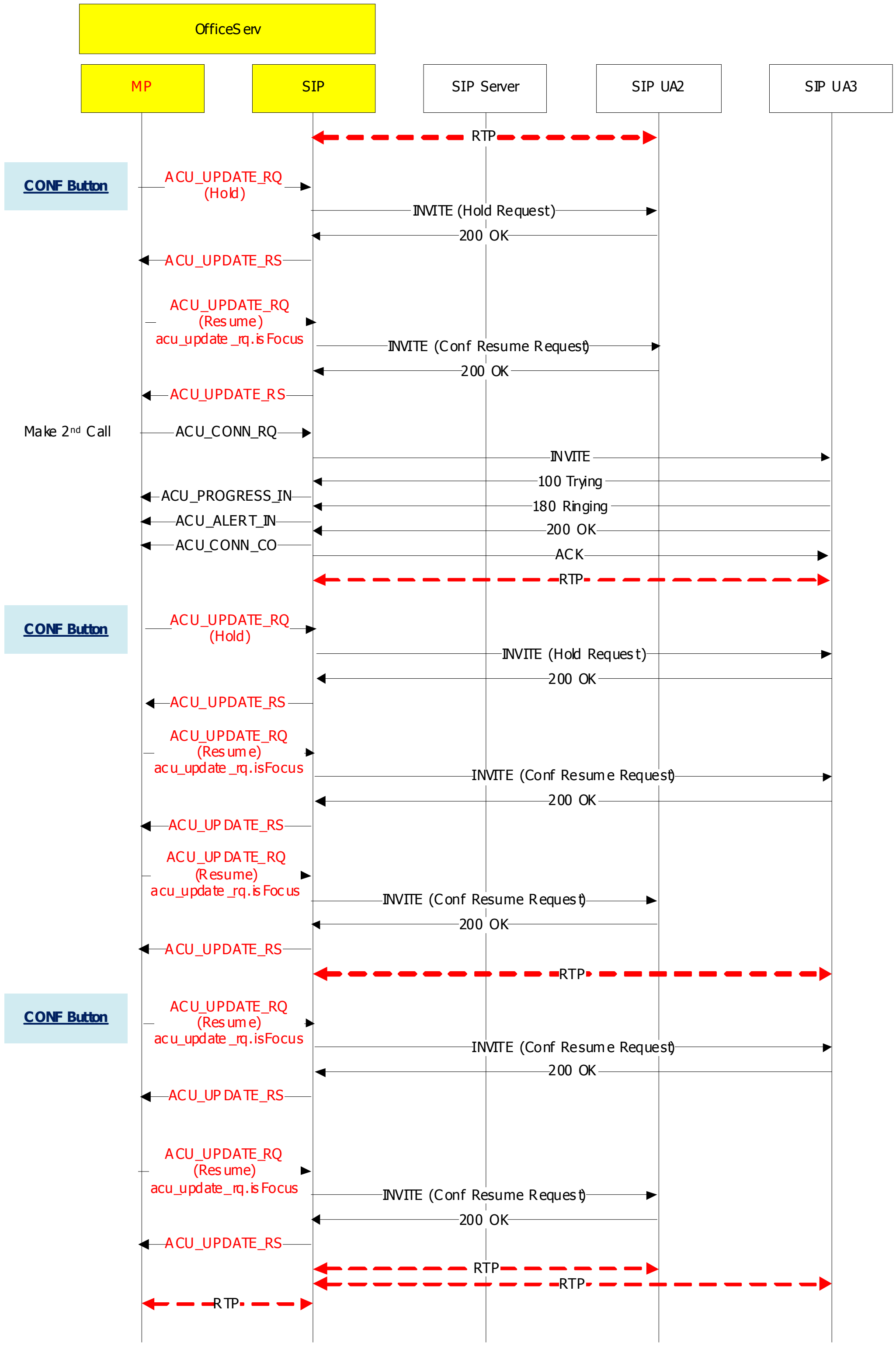


Figure 25. Brief Conference Call Flow 2

OfficeServ™ 7000 Series

SIP Services

Part 3. SIP Station

Part 3. Table of Contents

Part 3. Figures

Part 3. Tables

1. Introduction

1.1. Feature Code

OfficeServ system provides various feature codes in order to support SIP supplementary services. Those feature codes are specified in MMC724 FEAT DIAL NUMBER entry and each different supplementary service has its unique feature code. OfficeServ system has default feature code values but users can change them if necessary.

Table 1. MMC724 Feature Code List for SIP Supplementary Services

Supplementary Services		Feature Codes
Call Forward	Call Forward Cancel	600
	All Call Forward	601
	Busy Forward	602
	No Answer Forward	603
	Busy / No Answer Forward	604
	Forward Query	60
Call Waiting	Call Waiting Set	771
	Call Waiting Cancel	770
Call Back	Call Back	44
DND	DND Set	401
	DND Cancel	400
	DND Query	40
Call Park	Call Park	11
Conference	Conference	46
Call Pickup	Call Pickup	NONE

1.1.1. MMC724 Feature Code

Setting feature codes for SIP stations in order to use SIP supplementary services is the same as setting the feature codes for [1]legacy stations in OfficeServ. SIP station users first should know what the feature code is for the selected supplementary service. And then they can set the code for the target supplementary services for SIP stations using the corresponding feature code.

Unlike the case of setting a feature code using legacy stations, registering (or setting) a SIP supplementary service using SIP station is done in a form of a normal SIP outbound call from the SIP station to OfficeServ system. That is, a SIP station informs the OfficeServ system by sending an INVITE message inserting the target service feature code instead of the called number. This way, OfficeServ will know that the specific SIP station (specified as caller number in INVITE message's from header) wants to set the specific feature code.

[1] Legacy stations refer to any terminals that are using proprietary protocols or messages to communicate with OfficeServ system. i.e., Analog Phones, Digital Phones, and IP Phones.

For example, if a SIP station wants to set DND service, and the feature code is '40', the SIP station user simply calls to '401' (401 to set, and 400 for cancel) in a form of normal INVITE message. But as you may guess, because this is not an actual outbound call, nor are there a specific station whose number is '401', OfficeServ responds back with a 480 message (meaning temporarily not available) message and terminates the call. Therefore, dialing 401 does not mean anything but notifies the OfficeServ system that the SIP station wants to set the DND service. And, of course, the 480 response from OfficeServ to SIP station is simply for terminating the incoming call.

What happens if a station actually exists with the number 401? To avoid this kind confusion, OfficeServ strictly checks feature code numbers and prohibits assigning the same feature code with existing station number. As this procedure is done automatically in OfficeServ, users do not have to be concerned about this feature code conflicting.

1.2. Software Requirements

SIP station features in this manual are provided from MP/MCP S/W v4.22 or higher. Earlier versions of MCP S/W will not have the added SIP MMC options, or they may display some different names for the same functions.

1.3. Hardware Requirements

Currently SIP station features are implemented in OfficeServ 7100, 7200 and OfficeServ 7400 systems. **OfficeServ 100 and 500 systems do not support SIP Services.**

2. Registration Types

OfficeServ classifies SIP stations by two categories; Samsung SIP Phones (not available in US) and Non-Samsung SIP Phones (i.e., Cisco 79XX, Linsys, Aastra, X-Lite, etc.). This is for allowing Samsung SIP stations to be registered on OfficeServ without any further restriction, whereas non-Samsung SIP stations need a SIP license key for registration.

2.1. Registering SIP Station

In order to register non-Samsung SIP phones to OfficeServ system, inserting a valid license key to the system and proper virtual cabinet setting should be done in advance. (If these two conditions are not met, please refer to Chapter 2. Preparation). If virtual SIP station card is properly set, we can see available SIP station numbers in MMC 842.

First of all, MMC 842 STND SIPP Table contains a default SIP station numbering plan which is shown in the 'Station No' column. This number range differs according to license key and the system's virtual cabinet setting. If OfficeServ system sets a large number of virtual SIP station card in MMC 857, the number range will be also be large. If no SIP station card is set, MMC 842 will display none as well. The starting number sequence of the list depends on the virtual cabinet's slot number that is mapped to SIP station card.

Table 2. MMC842 STND SIPP Table

Station No	Registered	IP Address	User ID	Password	SIPP Tone	Call Wait
3301	YES	192.168.89.7	3301	0000	Use System Tone	Disable
3302	NO				Use System Tone	Disable
. .	. .	. .	. .	. .	. .	. .
3361	NO		3361	0000	Use System Tone	Disable
3362	NO		3359	0000	Use System Tone	Disable

Table 5 shows a sample SIP station number ranges from 3301 to 3362. OfficeServ system operator may change the default numbers in MMC 724, but for now, using the default numbering plan in this list to SIP stations is recommended to make things easier.

For successful registration, each SIP station has to have the same authentication data with data listed in corresponding row in MMC842 table. OfficeServ system requires standard MD5 registration process for each register request from SIP stations, and for this reason, OfficeServ and SIP station should share 2 data in common: User ID and Password.

User id and password combination should be the same with values set in actual SIP station because MD5 registration encryption key will be made based on the pair of these two values, if different, SIP station's registration will fail. By default, all password values are set to '0000', the programmer, however, can change the values if necessary as long as corresponding SIP station has the same changed password value.

MMC 842 will display valid IP addresses when the respective SIP station is officially registered to OfficeServ system. By default, it displays blank IP address.

2.1.1. SIP Phone Registration Procedure (Example)

In order to register non-samsung SIP phones to an OfficeServ 7000 series platform setup as the SIP server, some programming MMCs (724, 841, 842, 857,) will have to be set correctly Please follow the below procedure:

<u>Note:</u> MGI channels are required with the OS7000 series Server to support SIP Phones. Confirm that all programming, licensing and hardware is setup to insure the MGI channels are working within the Server.

1. The first thing the technician will have to do is enter the SIP stack license (including the non-samsung SIP phones) into **<u>MMC 841</u>** under "FEATURE LICENSE KEY" option. When the license is entered correctly, this MMC will show:

EXAMPLE

NSIP-S(non Samsung SIP stations)	MAX :	6
	USED :	0
	CONN :	0

When Samsung SIP phones are used: Enter the SIP Stack license key in MMC 841. The technician will have to assign part of the Samsung SIP STACK licenses as Samsung SIP phones (SSIP-S) in this MMC 841.

SSIP-S (Samsung SIP stations)	MAX :	6
	USED :	0
	CONN :	0

Note: Samsung SIP Phones not available in US

---

2. Next, the technician will have to enter **<u>MMC 857</u>** (VIRTUAL SLOTS) and assign 1 or more virtual slots as "SIP STN" depending on how many SIP phones will be supported.

EXAMPLE

[C4-S1 :WLAN ITP]
[SIP STN]

Table 3. SIP Station Capacities

SIP Station Capacities			
OfficeServ System	**7100**	**7200**	**7400**
Max Virtual Station Slots	4	4	4
Channels per slot	8	8	32
Maximum SIP Stations	**32**	**32**	**128**

Note: After assigning virtual(s) as "SIP STN", the extension numbers 3301 through 33XX will be assigned as the default numbering plan for SIP stations. This can be checked in MMC 724 under Station numbering plan [STN NUMB.].

Now the Samsung SIP server is setup and ready to permit Non-samsung SIP phones to register to the server.

2.1.2. Non-Samsung SIP Phone Registration Procedure (Example)

There are many different manufacturers of SIP phones on the US marketplace today. Each SIP phone will be different for how you will access the setup wizard or web server page used to set the necessary parameters for the SIP phone to register and work. Once you have entered the setup wizard for the particular phone, the following options should be set correctly:

EXAMPLE

ENTER SETUP WIZARD:

Phone Settings >

- (Set as: **Static** or DHCP IP**)**
- IP address: **192.168.9.196**
- Subnet: **255.255.255.0**
- Gateway: **192.168.9.1**
- User ID, User Name, Auth Name, Auth User Name, or Display Name: **3301**
- Password: **0000**

Network Service Settings > SIP Server >

- Domain: **192.168.9.200** (This is IP address of the MP in MMC 830)
- Outbound Proxy: **192.168.9.200** (This is the IP address of the MP in MMC 830)

When the SIP Phone has registered correctly, go offhook from the SIP phone and make an internal call to test for proper operation. Also, the Technician can enter MMC 842 and check. A SIP phone that is register correctly will display the following in MMC 842:

3301 [REGISTERED]
YES

3301 [IP ADDRESS]
192.168.9.190

3301 [USER ID]
YES

3301 [PASSWORD]
****

3301 [TONE SRC]
USE SYSTEM TONE

3301 [CALL WAIT]
DISABLE

3301 [PHONE TYPE]
NON-SAMSUNG PHONE

3. SIP Station Services

This Chapter describes the detailed call scenarios regarding SIP service features in which SIP stations are involved. There can be too many different scenarios for each service depending on what terminals are used and thus, this document does not fully cover all the cases but some representative ones for each category.

For readers who want to follow each scenario step by step for self training, detailed terminal types and the test sequences are mentioned with the scenario description. Sample SIP station numbers in scenarios do not need to be the same with your test cases.

3.1. Basic Call Setup

Once registration is done, OfficeServ is now ready to handle SIP station calls. If SIP station's valid registration data is not seen in the MMC 842, please go back to section 3.3 and make sure the SIP station successfully registers first.

Like using the other internal terminals such as DGP and ITP, users can make outbound calls by simply dialing an extension number (called party). The called party will not notice the difference between using the legacy internal terminals and SIP station terminals. However note that when using SIP station, its signaling protocol is SIP, and not the Samsung proprietary protocol.

3.1.1. Basic Call Setup between SIP Stations

When [2]SIPP1 (SIP Phone 1) calls to SIPP2 (SIP Phone 2), SIP signaling is done as shown in following figure 4. Here we have to consider two important points;

- The reason the OfficeServ system is shown in between the two SIP stations is that all the SIP signaling messages have to go through OfficeServ system because OfficeServ acts as a SIP proxy server for all the SIP stations registered to it.

- Even though SIP signaling messages are passed through the OfficeServ system, its RTP packets are transmitted directly between two SIP stations. This means SIP stations do not use MGI resources when SIP to SIP calls, and direct RTP transmission enhances the Voice Quality by avoiding unnecessary encoding/decoding of RTP data by MGI. However, when [3]non-IP terminals (i.e., DGP) are involved in the session, MGI resources has to be used by the terminals.

[2] SIPP stands for SIP Phone in this document.
[3] Non-IP terminal means DGP and other legacy phones. Though ITP is not SIP terminal, it works the same in that it does not use MGI either. That is, a session between SIP terminal and ITP does not use MGI as in the case of SIP-to-SIP.

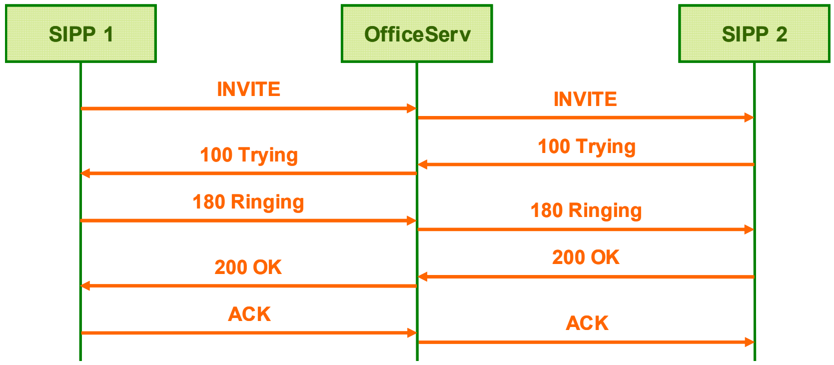


Figure 1. Basic Call Setup Flow between SIP Stations

3.1.2. Basic Call Setup between SIP Station and Non-SIP Terminal

Talking about SIP calls in the case which involves a SIP station and DGP or ITP. From the SIP station's perspective, it simply interacts with OfficeServ system using SIP protocol. It does not have to know whether the other party can understand the SIP or not. Interpreting the SIP message is the function of the OfficeServ Server. When the other party is a Non-SIP terminal, the OfficeServ server translates the received SIP message to DGP or ITP Samsung-proprietary message protocol, and vice versa. Therefore, we can abbreviate the SIP-based call flow as shown in the following figure, letting DGP represent the OfficeServ server.

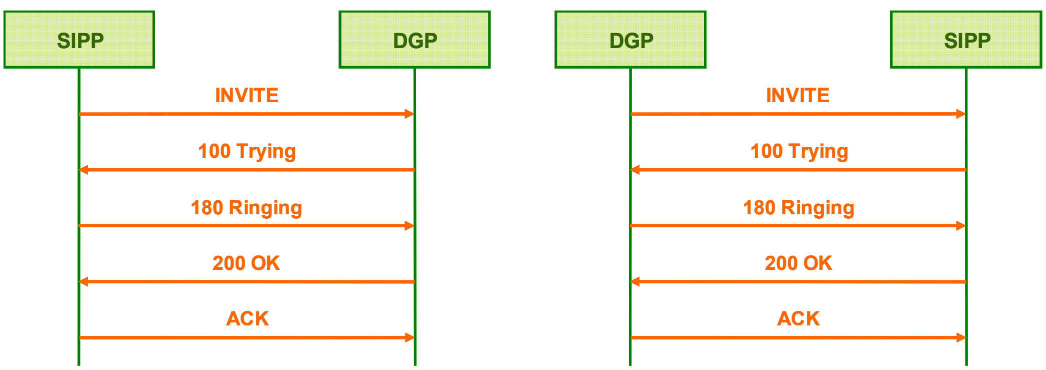


Figure 2. Basic Call Setup between SIP Station and DGP

3.2. Hold/Resume

Hold/Resume is a start point of supplementary services because Hold/Resume itself is one of supplementary services, and many other supplementary services in OfficeServ system are composed of multiple Hold/Resumes. That is, if Hold/Resume is not working properly, other supplementary services may not, either.

According to SIP standard, Hold/Resume feature can be implemented by either UPDATE method or Re-INVITE method. Basic mechanism that lies in both methods is all the same although the messages have different names. Currently OfficeServ systems supports Re-INVITE message as its default Hold/Resume method. Re-INVITE is a normal INVITE message except it is sent while already in the invite session. By sending an INVITE message which contains different [4]SDP (Session Description Protocol) during a session, the SIP session mode can be switched to one of **sendrecv**, **sendonly** and **recvonly** according to the session mode attribute value designated in the SDP.

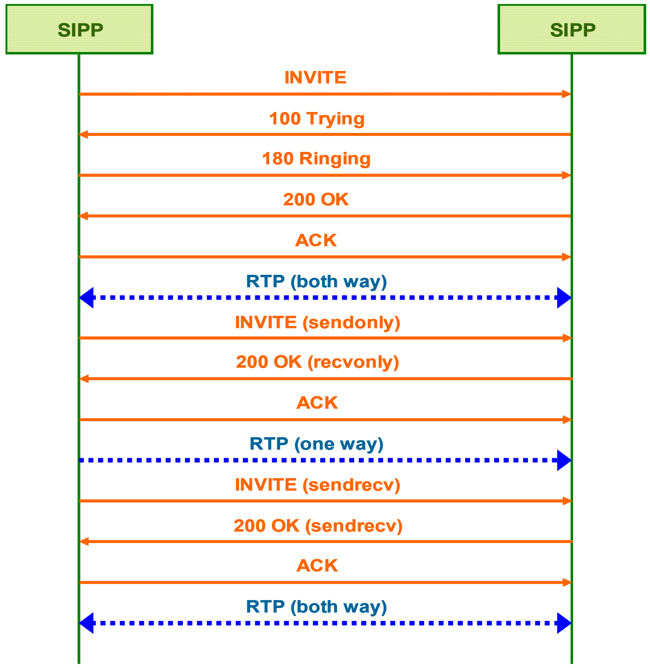


Figure 3. Hold/Resume between SIP Terminals

In a normal dialog state, the session mode is sendrecv which allows both way RTP transmissions. When holdee (party placed on hold) sends a Re-INVITE message which designates its RTP transmission to sendonly mode, it informs the holdee that it wants to only send RTP and will not receive. After receiving the Re-Invite message, the holdee knows that the other party wants to put the session in hold mode and stops sending RTP packets, giving a 200 OK response back. The 200 OK response, like the Re-Invite message, contains a SDP and its session mode attribute is set to recvonly. Mean while, holder (party initiating the hold condition) can either only send music tone (called 'MOH: Music On Hold') or set the session mode mute by sending no RTP at all, shutting down its listening port. Whether to send music tone or not during hold time is station dependent. To resume the held session, holder sends Re-INVITE message again designating the RTP transmission back to sendrecv.

Remember that only the holder can resume the held session, which means no matter holdee (holding party) sends Re-INVITE message specifying sendrecv, the session will remain in hold mode and holder will not change its mode.

3.2.1. Another Way of Specifying Hold Mode

Some SIP terminals use another, probably old-fashioned, way of specifying sendonly mode in its Re-INVITE message. It is to set the connection parameter value in SDP to all zero, which tells the message receiver (maybe holdee in this

[4] SDP specifies the session attributes such as codec types, rtp port, rtp ip address and etc. For more detailed information, please refer to RFC2327. A good example of SDP for the readers who want super-simple explanation, is the second half of a SIP message which is divided by a blank line.

context) not to send any RTP packets because there is no destination ip address to which it can send RTP packets. OfficeServ also supports this connection-all zero-specified Hold method for backward compatibility.

3.3. Call Transfer

There are two kinds of Call Transfers: Consultation Transfer and Blind Transfer. TRSF in Call Transfer call flow means transfer key. Most SIP terminals have physical (fixed) transfer key as normal keys but some others provide softkey-typed transfer key interface through their LCD interfaces.

3.3.1. Consultation Transfer

Party A (DGP) is in conversation with party B (SIPP 2), Party A then presses transfer, (Party B now is hold state) then dials party C (SIPP 1) and waits for party C to answer. When party C answers, Party A hangs up. Party B and C are now connected.

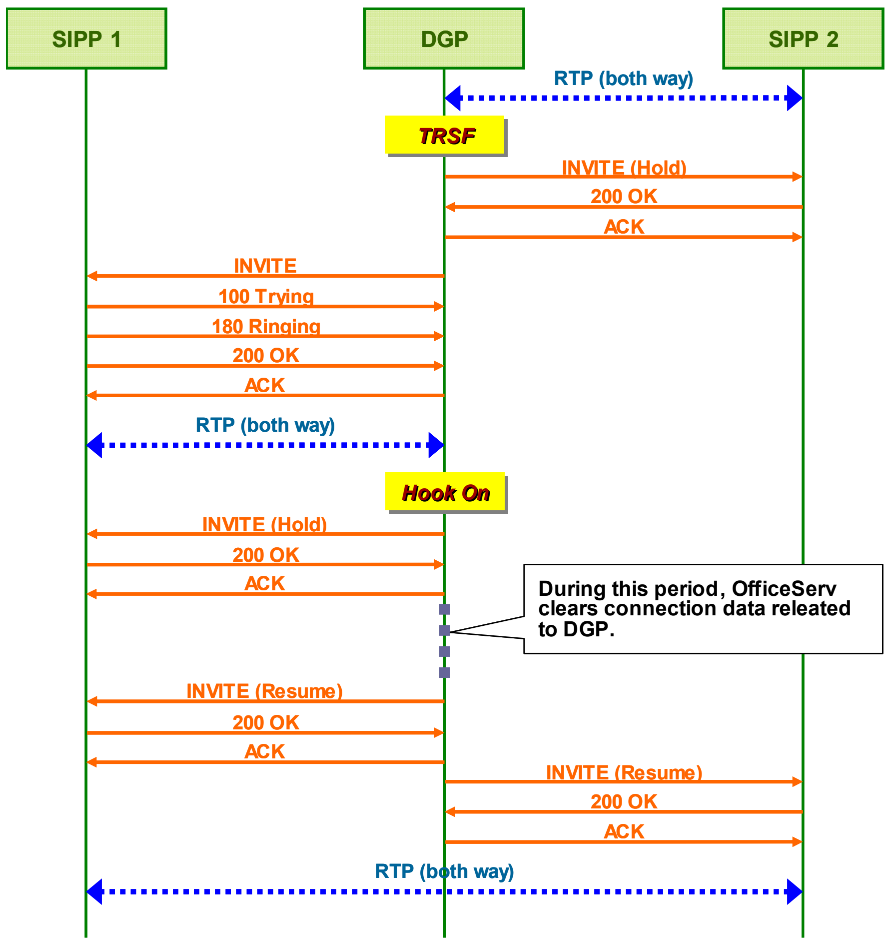


Figure 4. Consultation Transfer by DGP as a Transferor

This scenario starts from the state where DGP and SIPP2 are already in an active dialog session and followings are the steps thereafter.

i. DGP initiates transfer process by pushing TRSF button, and it puts SIPP 2 on hold state.
ii. DGP makes the second session with SIPP1.

iii. After the DGP hangs up, OfficeServ clears all the connection data that is related to DGP because DGP is now released from the connection.
iv. As shown in the call flow above, OfficeServ resumes each held SIP session, changing SIP terminal's RTP destination so that SIPP 1 and SIPP2 can now communicate directly each other without using MGI resources.

Most SIP terminals have their own Transfer Key either in a hard-form type or a soft-form type. Therefore, additional feature codes for Transfer are not required.

3.3.2. Blind Transfer

Party A (DGP) is in conversation with party B (SIPP 2), Party A then presses transfer, (Party B now is hold state) then dials party C (SIPP 1) and hangs up before party C can answer the call.(Party A hangs up during in ringing state). Now party B is connected to party C in the ringing state. When party C answers the call, party B and C is connected.

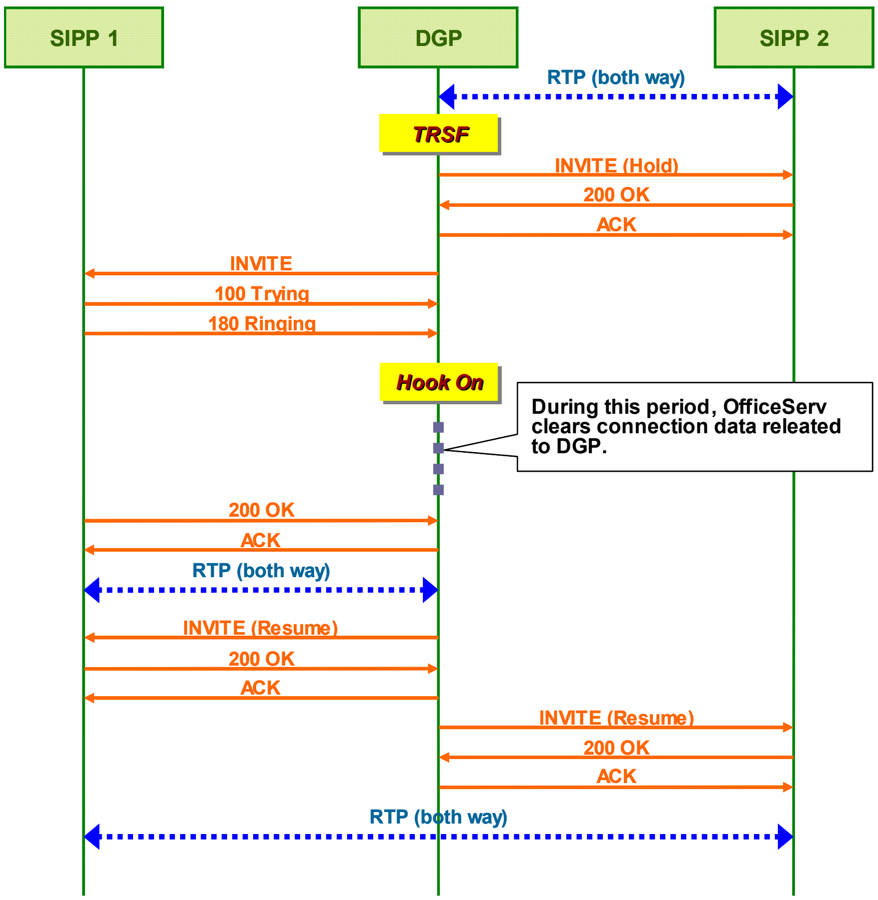


Figure 5. Blind Transfer by DGP as a Transferor

The main difference between Blind Transfer and Consultation Transfer is that in Blind Transfer mode, the initiating party will hang up during the ringback state. Followings are the steps.

- DGP initiates transfer process by pushing TRSF button, and it puts SIPP 2 on hold state.
- DGP makes the second session.
- After the DGP hangs up, OfficeServ clears all the connection data related to DGP because DGP is now

disconnected.

- As shown in the call flow above, OfficeServ resumes each held SIP session, changing SIP terminal's RTP destination so that SIPP 1 and SIPP2 can communicate directly with each other without using MGI resources.

- There is one important point in Blind Transfer. Whichever station initiates the transfer (i.e., standing in the middle in the call flow), will put the first party (i.e., SIPP2 in the call flow) in hold state before it sends the INVITE message to the called station (i.e., SIPP1 in the call flow). This is necessary because OfficeServ can not process subsequent Blind Transfer properly unless this condition is met.

- According to Lab test results, some SIP terminals do not send the hold message that the OfficeServ is supposed to receive in order to put the transferee (party being transferred) in hold state, and this will cause the OfficeServ to fail to execute Blind Transfer. Currently this is the current specification of the OfficeServ's Blind Transfer feature.

3.3.3. Call Forward

There are 4 kinds of Call Forwards: All Call Forward, Busy Forward, No Answer Forward, and Busy/No Answer Forward. Though conceptions are slightly different among those 4, the main mechanism and MMC settings are almost same.

To execute call forward, each SIP terminal first should know desired call forward feature code and set the call forward using the code. OfficeServ operator can check the Call Forward feature codes in MMC 724 FEAT DIAL NUMBER. Like other feature codes, Call Forward feature codes can be changed is required.

3.3.4. Using MMC 724 Call Forward Feature Codes

Registering Call Forward supplementary service from a SIP station is done in a form of normal INVITE message from the SIP station to OfficeServ. (For more detailed, please refer to 1.1 MMC724 Feature Code)

As far as No Answer Forward, OfficeServ forwards an incoming call to a designated number when the original party does not answer the call for a certain period of time which is specified in MMC 502 No Answer Forward Time.

Table 4. MMC724 Call Forward Feature Code

Call Forward	Feature Codes
Call Forward Cancel	600
All Call Forward	601
Busy Forward	602
No Answer Forward	603
Busy/No Answer Forward	604

Following is a sample scenario of Call Forward. (2001/2002: DGP, 3337:Cisco 7960)

- If operator wants to forward calls incoming to a SIP station (3337) to some other DGP (2002) when 3337 is busy.

- From the 3337, dial '6022002' to set Busy Forward to '2002'.

- Once Call Forward is set for 3337, another SIP station (3334) calls 3337 and make a connection.

- DGP (2001) makes a call to 3337 which is already in a session with 3334.

- The call is forwarded to 2002.
- To clear Call Forward setting, dial '600' from 3337.

3.4. Call Waiting

Call waiting feature can be enabled at a SIP station. When a station is in an active conversation with another terminal, the caller can accept another incoming call, instead of sending a busy signal (a 486 busy response). To execute call waiting, each SIP terminal first should know desired Call Waiting feature code and set the call waiting using the code. OfficeServ programmer can check the Call Waiting feature code specified in SIP CW in MMC 724 FEAT DIAL NUMBER. Like other feature codes, Call Waiting feature code is operator-definable and thus changeable as well.

3.4.1. Using MMC 724 Call Waiting Feature Codes

By default this SIP CW value is set to NONE, and operator can define any available feature code. When Call Waiting feature code is set, a SIP station can register Call Waiting supplementary service. Registering Call Waiting supplementary service from a SIP station is done in a form of normal INVITE message from the SIP station to the OfficeServ server. (For more detailed, please refer to 1.1 MMC 724 Feature Code) If Call Waiting service for a certain SIP station is set, the corresponding SIP station's MMC 842 CALL WAIT value will also changed to ENABLE. The following table shows the example feature code set for Call Waiting feature in MMC 724. If the Call Waiting feature code is set as '77', use '771' for set and '770' for cancel.

Table 5. MMC724 Call Waiting Feature Code

Call Waiting	Feature Codes
Call Waiting	77
Call Waiting Set	771
Call Waiting Cancel	770

The usage of Call Waiting feature for SIP station is pretty simple. The following figure shows the internal call flow for Call Waiting setting.

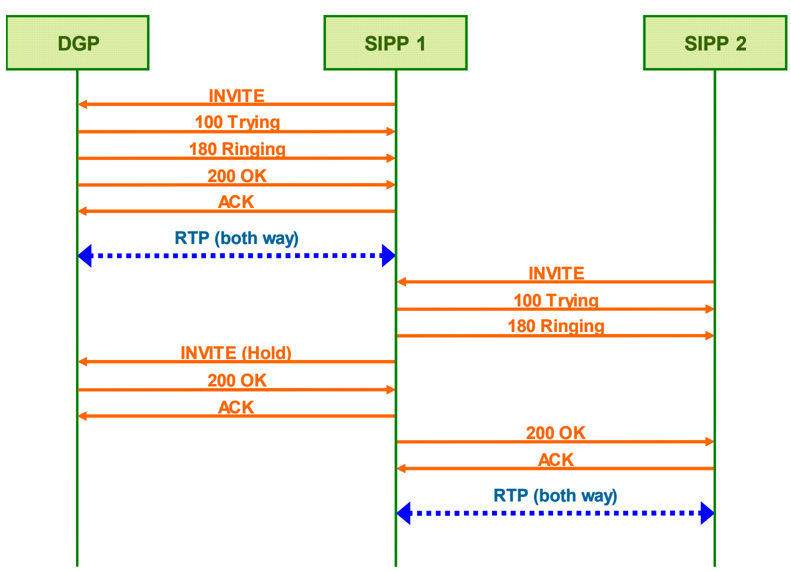


Figure 6. Call Waiting at SIP Station

To use Call Waiting feature, go through following steps. (2002:DGP, 3334:WIP6000, 3336:Cisco 7960)

- Set the Call Waiting feature for a SIP station (3334) by simply dialing '771'.
- From the SIP station (3334), make an outbound call to another terminal (2002)
- From the third SIP station (3337), make a call to 3334 SIP station.
- With station 3334 now set to ON for Call Waiting mode, it does not reject the newly incoming call from 3337 but notifies the use (call waiting tone) of the new incoming call.
- When accepting the newly incoming call from 3337, 3334 puts the first connection with 2002 on hold. (Of course, 3334 can reject the second call by selecting 'REJECT' button)

Note: In case of DGP's requesting Call Waiting for a SIP station that is already busy in a session, select the "CAMP" option in DGP's LCD. This means that, unlike SIP station, DGP does not need to register Call Waiting service to OfficeServ. The next scenario shows this.

Here is a sample Call Waiting scenario. (2001/2002:DGP, 3332:Cisco 7960)

- Set the SIP CW value to 77 in MMC 724 FEAT DIAL NUMBER option.
- From a SIP station (3332), dial '771' and send to set Call Forward option enabled in OfficeServ. MMC 867 shows the Call Waiting service is set for the SIP station.
- From another DGP terminal (2001), make a call to the SIP station (3332).
- From another DGP (2002), make a call to the SIP station (3332) which is already in a session with 2001.
- 2002's LCD shows that the called party (3332) is busy and 2002's LCD displays 3 options: CBK, MSG, and CAMP. Select CAMP.
- The SIP station (3332) informs that the new call is incoming (call waiting tone) and called party accepts the call.
- Automatically the first call is placed on hold, and the new connection between 3332 and 2002 becomes an active session.

3.5. Call Pickup

Call Pick can be done by any terminal attached to OfficeServ as an internal station. To execute Call Pickup, each SIP terminal first should know desired Call Pickup feature codes and set Call Pickup using the codes. The OfficeServ operator can check the Call Pickup feature codes specified in MMC 724 FEAT DIAL NUMBER. Like other feature codes, Call Pickup feature codes are operator-definable and thus changeable as well.

3.5.1. Using MMC 724 Call Pickup Feature Codes

When performing Call Pickup using MMC 724 feature code, users can use 3 different pickup services which were originally provided by OfficeServ as a Key Phone feature. Those are 1) my pickup, 2) group pickup and 3) direct pickup. In this document, only my group pickup (MYGPIK) service is mentioned because all SIP stations are tied in a single group by default. And default value for MYGPIK feature code is 'NONE'. For this example * is used as the My Pickup access code.

Following call flow shows the detailed story.

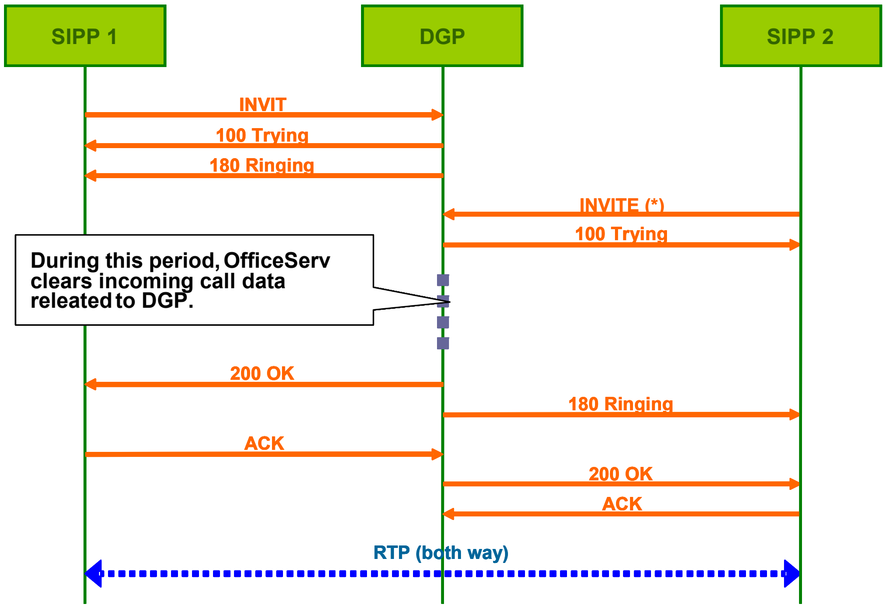


Figure 7. Call Pickup from DGP by SIP Station

- When picking up a call from a SIP station, only the feature code is needed as the scenario described below. (2001:DGP, 3332:Cisco 7960, 3334:SMT-W5100E)
- From a SIP station (3332), dial a DGP number (2001).
- DGP (2001) rings and the SIP station (3332) is in a Ringback state,
- From another SIP station (3334), simply dial Call Pickup feature code of '*'.
- The original call is picked up by 3334, and a connection is made between 3332 and 3334.

3.6. Conference

Conference feature may be the most complicated feature among SIP supplementary services implemented in OfficeServ both in usage and in its how it works. When conferencing, all the stations including SIP stations should use MGI instead of using their own DSP for handling RTP packets, because OfficeServ gathers all the RTP packets transmitted in the conference into a single conference chip (MGI) using Hold/Resumes repeatedly. Maximum 5 members are allowed in a conference.

3.6.1. Using MMC 724 Conference Feature Codes

By default, the feature code for Conference specified in MMC 724 FEAT DIAL NUMBER is '46' and of course this can be changed by operator like other supplementary feature codes.

The key point to understand how the SIP station's Conference in OfficeServ system is used, is to know what actions are taking place by OfficeServ system when the Conference feature code is given. In short, the usage of Conference feature code is similar to the case of using the 'CONF' soft key in legacy DGPs. Following is an example of conferencing internal calls:

Conference can be made by a conference owner station using the conference feature code while it is in conversation. Whenever the conference owner terminal wants to make a new conference or add a new member to the conference group, it first sends a Re-Invite message to make the current session a hold state and then it sends off the conference feature code in a form of a normal INVITE message. As in the case of handling other feature codes, OfficeServ responds back with a 480 message and then the owner terminal comes back to IDLE state, and is ready to take the next action.

As to the conference member stations, when conference feature code is received, OfficeServ makes stations in the session connect in a single conference session so that they can send/receive RTP packets to each other via MGI, instead of direct transmission.

At this point, the conference owner station has two options: 1) make another call to invite another member into the conference or 2) end adding additional member. If the owner station chooses option #1, then it can invite another member by sending out INVITE message to the station, and it goes back to step 1 again. However, when choosing option #2, it simply needs to send the conference feature code again to OfficeServ system without taking any other action.

When OfficeServ receives consecutive conference feature code from the owner station, it now knows that the owner station does not want to invite additional conference members, and OfficeServ resumes with all stations in the conference so that they can communicate with each other via MGI.

Finally, the conference owner station must come out of hold and actually join in the conference and communicate with other stations. This is because while making the conference, the owner station has set its session into hold mode, which disabled its RTP transmission. Therefore to resume the session, it should give a final Re-Invite and OfficeServ connects the owner station's RTP into the conference giving the 200 OK response designating the RTP destination to MGI IP address. Without resuming the hold session, the owner station would not listen nor speak to the conference even though it is a member of the conference call.

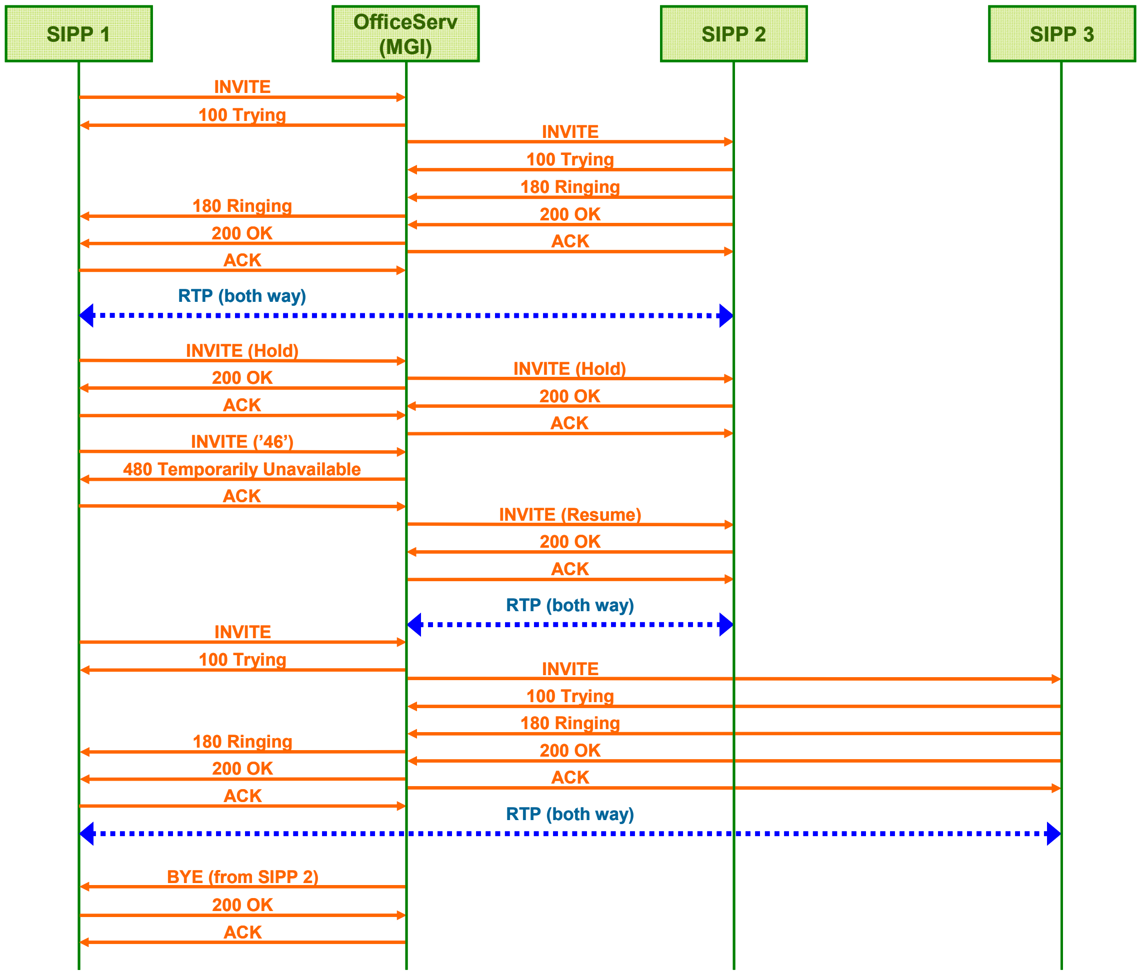


Figure 8. Conference made by a SIP station (#1)

As shown in figure 11, conference owner SIPP 1 starts making a conference session using '46' feature code after making an initial session with SIPP 2. Note that there must be a Re-Invite (remove hold) right before 'Invite 46' message. This means OfficeServ always has to put the target conference member into hold state before actually adding it into the conference by removing it from hold. Some terminals (i.e., WIP6000) automatically set the current session to hold mode before it sends out an Invite message '46', which is not a Re-Invite, but some others do not. In this case, the manual hold should be preceded before the conference feature code.

The conference feature code of '46' gives a conference signal to OfficeServ so that OfficeServ can take necessary actions to make a conference such as modifying RTP transmissions using Re-INVITE messages. Right after this conference feature code, the session in which SIPP 2 was involved is resumed by OfficeServ and the RTP packets are transmitted between the SIPP 2 and MGI not SIPP 1. On the other hand, the conference owner SIPP 1 is now in an IDLE state and able to make another call to invite the 2nd member, SIPP 3.

When the 2nd session is made with the SIPP 3, OfficeServ disconnects the session between SIPP 1 and SIPP 2 because this session is no longer useful. And the most important thing is that the maximum SIP terminals supported is 2 concurrent connections only. Leaving the original connection will block SIPP 1's next attempt to send out the code '46' conference feature code, which may lead to failure of further conference processing.

Now SIPP 1 is having an active session with SIPP 3, and the session with SIPP 2 is gone. Mean while SIPP 2 is put in a session with OfficeServ (MGI).

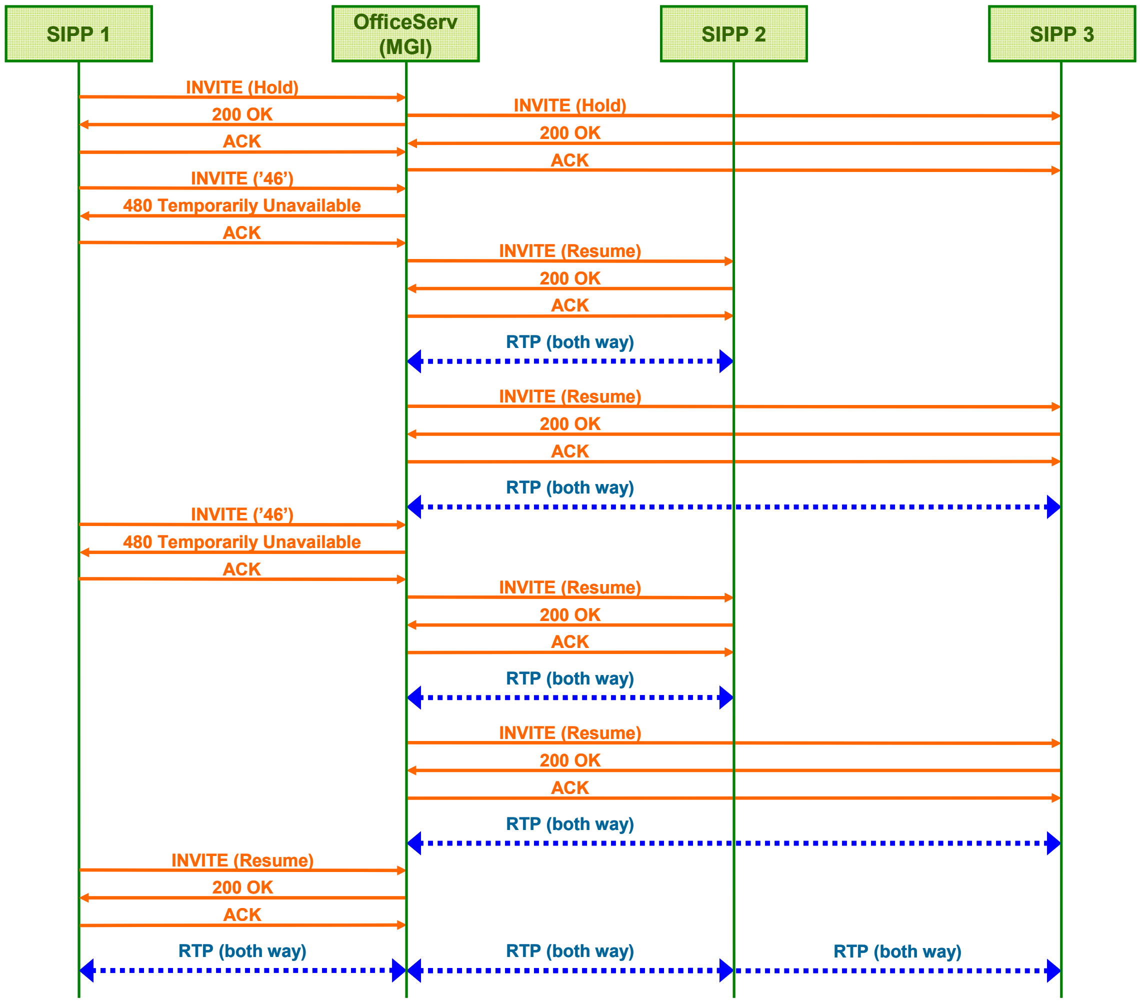


Figure 9. Conference made by a SIP station (#2)

To put SIPP 3 in the conference session which SIPP 2 is already in, SIPP 1 sends out the conference feature code. Then, OfficeServ modifies the SIPP 3's RTP destination to MGI so that it can be joined in the conference. Note that before sending out the Conference feature code, SIPP puts itself in the session with SIPP 3 in the hold mode again. After this, SIPP 1 goes back to the IDLE state so that it can choose either to end adding conference member or to add additional member.

As SIPP 1 chooses to end adding additional members according to this scenario, it simply enters the conference feature code again. As mentioned before in 4.7.1, because OfficeServ has received the consecutive conference feature code from conference owner station, it sends Re-INVITE (resume) messages toward all the terminals including the conference owner station to change RTP packets destination into MGI. For some stations that are already in the conference session directing their RTPs to MGI, this Re-INVITE message may be redundant, but this will not cause a problem.

Finally, the conference owner station MUST resume the hold session which was made right before sending the 2nd Conference feature code in order to actually communicate (join) in the conference. In many cases, users are likely to miss this last process of resuming the conference owner station, therefore it is important not to forget to do this.

3.7. Call Park (System Hold)

Using Call Park feature, a SIP station can park the call in an active session, and later on, the parked call can be picked up by the SIP station itself or some other. Call Park feature in OfficeServ can be practiced using Call Park feature code. As in the case of call pick up feature, a Call Park feature code is applied only for a single call.

3.7.1. Using MMC 724 Call Park Feature Codes

To execute Call Park feature in OfficeServ, operator should be aware of operator-definable feature codes that are specified in HOLD/HLDPK in MMC 724 FEAT DIAL NUMBER option. By default, the feature codes are set to 11 and 12 respectively, and of course programmer can change these values if necessary. **To enable retrieving a parked call using MMC 724 feature code, programmer should set MMC option of STNHOLD PICK in MMC210**.

Table 6. MMC 724 Call Park Feature Code

Call Park	Feature Codes
HOLD	11
HLDPK	12

A parked call can be retrieved by either the original call parker station or the 3rd station in the OfficeServ. When retrieving the parked call, use the feature code as following format: HLDPK code + the originating station's extension.

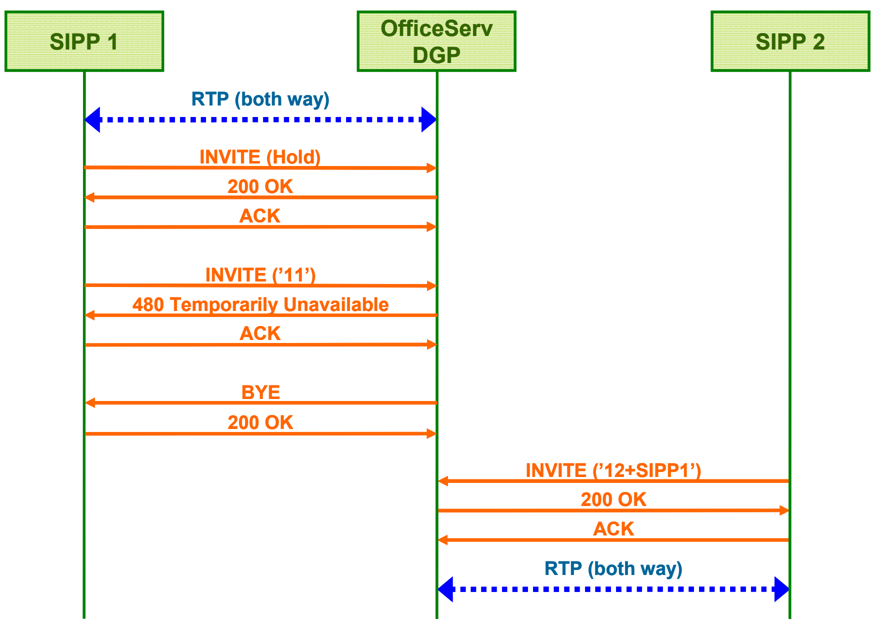


Figure 10. Call Park from SIP Station

Following is the sample scenario for Call Park. (2001/2002:DGP, 3332:Cisco 7960)

- From a SIP station (3332), dial a DGP number (2001).
- DGP (2001) answers and an active connection is made between 3332 and 2001.
- To park the current call, the SIP station (3332) first puts the call on Hold by pressing Hold button. (When using

SMT-W5100E, this process is unnecessary because it is automatically done right before sending the call park feature code)

- Then, the SIP station (3332), parks the call by dialing the Call Park feature code of '11'. (The feature code name of Call Park (System Hold).
- The call is parked onto OfficeServ system, and the SIP station (3332) is now in an IDLE state.
- From another DGP (2002), retrieve the parked call by dialing '12 3332'. (not 12 2001)
- The original call is now retrieved by 2001 and the connection is made between 2001 and 2002.
- If there is no pickup action on the parked call for a certain period of time, OfficeServ will recall the original station that parked to call to make the connection again.

3.8. Call Back

Call Back feature enables a caller to receive a call from its original called station when the called station becomes IDLE, which was in a BUSY state at the time of calling party initiated the call.

The biggest difference between case of using SIP station and case of using legacy Key Phone station is that SIP station must register Call Back service by an Invite message form, like other SIP supplementary service registrations, when it comes back to IDLE state after receiving the 486 BUSY response from the original called station, mean while legacy the Key Phone station can ask the Call Back service directly through LCD interface (CBK) while it is in ringback state or in busy state. Of course a SIP station would not ask for Call Back service if it does not receive any busy response from the called party.

Call Back service is provided on a call-by-call basis, so even though the Call Back feature is set for a certain station at a certain point, it will be cancelled automatically once the Call Back service is performed.

3.8.1. Using MMC 724 Call Back Feature Codes

When initiating Call Back service from DGP, select the CBK soft key from LCD after receiving busy response from the called terminal. Therefore, here it is not necessary to mention how to use MMC 724 Call Back feature code when using other standard SIP terminals.

Default Call Back feature code specified in CBK in MMC 724 FEAT DIAL NUMBER option is '44'. So, use this code + target station number when performing a Call Back from a SIP terminal.

Table 7. Call Back Feature Code List

Call Back	Feature Codes
Call Back	44

Following test scenario is for Call Back feature. (3332, 3334, 3340: Cisco 7960)

- From a SIP station (3332), dial another SIP station (3340).
- 3340 answers and an active connection is made between 3332 and 3340
- From the other SIP station (3334), try to make a call to 3332 who is already in a session in step ii, then it

receives 486 busy response.

- 3334 request the Call Back service to OfficeServ dialing Call Back feature code + target station number.(i.e., 44 3332')

- 3332 hangs up to end the connection with 3340.

- When the SIP station (3332) becomes IDLE state, the OfficeServ calls 3334 who set the Call Back service, and then subsequently calls 3332 sending INVITE message which specifies 3334's IP address and port number as destination RTP information in the message's SDP.

- After receiving successful 200 OK response from 3332, OfficeServ sends Re-INVITE message to inform 3334 the 3332's RTP port number.

- Now the call is made between 3334 and 3332.

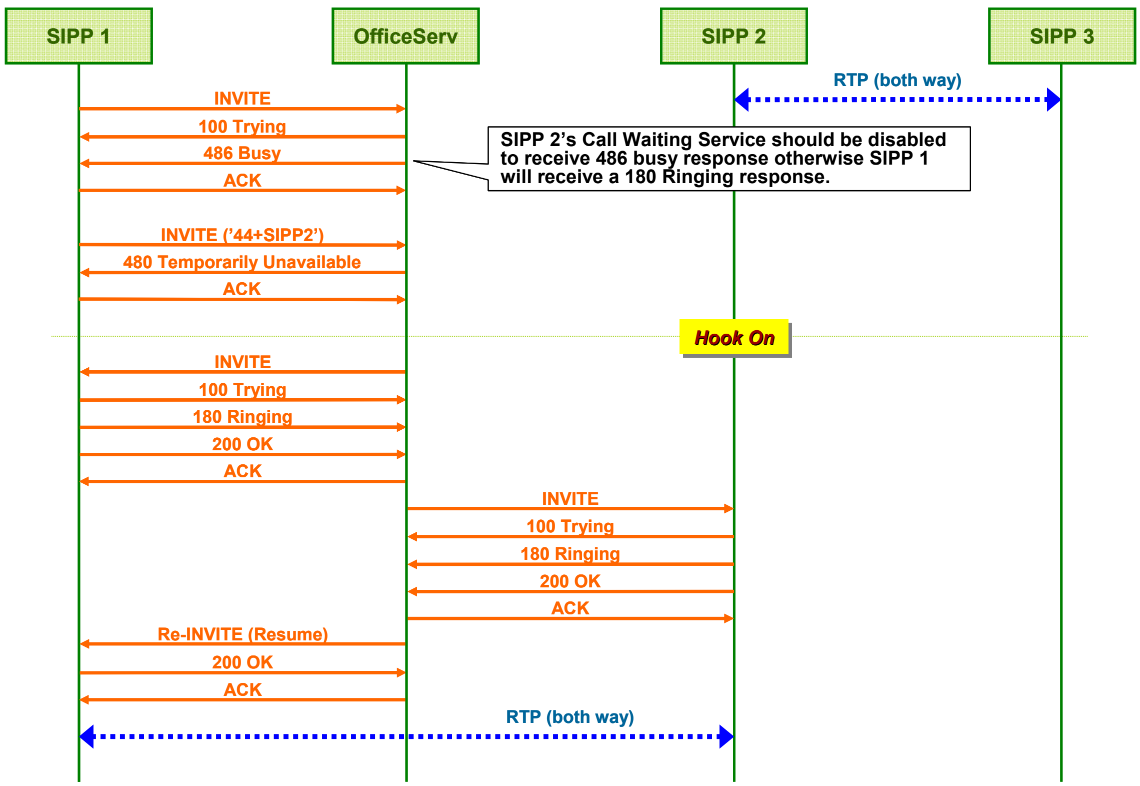


Figure 11. Call Back from SIP Station

There are two points to be noticed; 1) There is no automatic Hold before Call Back feature code because SIPP 1 does not have any active connection when it sends the message. 2) OfficeServ sends INVITE message again to SIPP 1 to let SIPP 1 change its RTP destination port number so that RTPs can be transmitted directly between two SIP stations of SIPP1 and SIPP 2. This is essential because there is no way for SIPP 1 to know SIPP 2's RTP port in advance.

3.9. DND (Do Not Disturb)

When DND is set for a terminal attached to OfficeServ, OfficeServ blocks the incoming call toward the terminal and the terminal does not receive any message and does not need to give any response. DND setting stays active until it is manually cancelled, so be careful not to forget to cancel DND when the service is not necessary anymore otherwise the terminal will not receive any incoming call ever.

3.9.1. Using MMC 724 DND Feature Codes

To practice DND feature, operator should use DND feature code specified in DND in MMC 724 FEAT DIAL NUMBER option. By default, OfficeServ has DND MMC 724 feature code as '40' and this value can be changed. The usage is all the same with other feature code except the feature code: 401 for DND set and 400 for DND cancel.

Table 8. Call Back Feature Code List

DND	Feature Codes
DND	40
DND Set	401
DND Cancel	400

Following is the scenario for DND setup. (3332:Cisco 7960, 3340:SMT-W5100E)

- From a SIP station (3340), set DND service by dialing DND feature code of '401'.
- From another SIP station (3332), try to make a call to 3340 who has set DND service to on, then OfficeServ sends 486 busy response back to 3332 on behalf of 3340.
- From a SIP station (3340), cancel DND service by the dialing feature code of '400'.

3.10. MWI (Message Waiting Indication)

MWI feature is implemented mainly for VMS (Voice Mail System) in OfficeServ. As indicated by the service name itself, MWI gives notification message to a SIP station when its voice mail box receives new voice mail. As this MWI service is provided by OfficeServ as a default function, no feature code is required.

MWI message internally uses standard SIP's NOTIFY method and its format is also compatible with MWI message format used in other standard SIP terminals (i.e., CISCO 7960).

The indications varies according to SIP stations, normally SIP stations which received the MWI message blink their LEDs to indicate new messages are arrived. Some stations that do not support any indication method may not work on the MWI message.