SIP Station Manual

For Samsung OfficeServ

Nov 23, 2007 doc v1.11

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1. Introduction

1.1. Overview

This document is written in order to give guidelines on SIP (Session Initiation Protocol) station features in OfficeServ system made by Samsung Electronics. OfficeServ system first launched the [1]SIP station features in MCP S/W (v2.69 for OS7200 and V3.34 for OS7400) released in Jan 2007, and along with its SIP trunking features, SIP now has become an OfficeServ's main VoIP protocol. Readers of this document are supposed to have at least the minimum degree of knowledge in manipulating OfficeServ system such as basic MMC settings, how to configure OfficeServ system and etc. Advanced reading or cross-referencing to ***SIP Trunking Manual for Samsung OfficeServ*** may help better understanding on SIP station features described in this document, but not mandatory if adequate knowledge on general SIP is already obtained. As in the case of SIP trunking manual, the main purpose of this document is not for introducing SIP general, rather for informing how to operate SIP station features implemented in OfficeServ system. Therefore, readers who want to have in-depth knowledge on SIP general should refer to RFC3261, which is the base document of SIP.

In supporting SIP stations, OfficeServ system provides not only basic features such as Registration and Basic Call Setups, but also 12 other widely used SIP supplementary features shown in the following tables.

Table 1. SIP Station Features in OfficeServ System

Basic Features	Supplementary Features
• Registration (MD5) • Basic Call	• Hold/Resume • Music on Hold • Consultation Call • Transfer (Consultation/Blind) • Call Waiting • Call Pickup • Call Forward (All/Busy/No-Answer) • Conference • Call Park (System Hold) • DND • Call Back • Voice Mail Indication

As target readers of this document are S/W developers, system engineers, and field engineers who are working for Samsung Electronics or its affiliates, exposure of this document to end users is not welcomed.

[1] SIP station refers to SIP UAC (User Agent Client) or simply SIP phone (terminal) that supports SIP protocol for IP telephony. OfficeServ system classifies SIP stations by two kinds; Samsung SIP Phones and Non-Samsung SIP Phones.

1.2. 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 heterogeneous SIP terminals as possible. For this purpose, OfficeServ's SIP call flows were designed after that of BroadWorks Inc's SoftSwitch, one of the most widely used SIP Proxy Servers in industry.

But still, as each different manufacturer may have each different call flow or message format, interoperability between OfficeServ and 3rd party vendor products can be an issue in some cases. Currently OfficeServ system guarantees the supplementary SIP service features only for the following SIP terminals, which have been actually gone through rigorous SEC's lab-testing and adapted by OfficeServ.

Table 2. SIP Services Compatibility Table

	Cisco 7960	SMT-i2015	X-Lite	WIP-5000	D-Link DPH-540	Linksys(SPA3102)	Linksys(SPA941)	Thomson IP Phone	GrandStream N/W
Registration	√	√	√	√	√	√	√	√	√
Basic Call Setup	√	√	√	√	√	√	√	√	√
Hold/Resume	√	√	√	√	√	√	√	√	√
Music On Hold	√	√	√	√	√	√	√	√	√
Consultation Call	√	√	√	√	√			√	√
Transfer (Consultation)	√	√	√	√	√			√	√
Transfer (Blind)	√	√	√	√	√			√	√
Call Waiting	√	√	√	√	√			√	√
Call Pickup	√	√	√	√				√	√
Call Forward (All)	√	√	√	√	√			√	√
Call Forward (Busy)	√	√	√	√	√			√	√
Call Forward (No Answer)	√	√	√	√	√			√	√
Conference	√	√	√	√				√	√
Call Park (System Hold)	√	√	√	√				√	√
DND	√	√	√	√				√	√
Call Back	√	√	√	√				√	√
Voice Mail Indication		√			√	√	√		

For the other SIP terminals, interoperability tests have not been fully conducted as of the moment when this document is being written. Additional interoperability test will be scheduled and conducted when necessary.

1.2.1. Samsung vs. Non-Samsung

OfficeServ system distinguishes Non-Samsung SIP terminals from Samsung-manufactured SIP terminals. From the operator's perspective, the main difference between Non-Samsung SIP terminals and Samsung SIP terminals is that the former needs additional license key verification.

From the SIP station user's perspective, however, SIP station's operating process is all the same. More technical stuffs will be mentioned in Chapter 2 SIP License Key.

1.3. 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 they want to.

Table 3. MMC724 Feature Code List for SIP Supplementary Services

Supplementary Services		Feature Codes
Call Forward	Call Forward Clear	600
	All Call Forward	601
	Busy Forward	602
	No Answer Forward	603
	Busy / No Answer Forward	604
	Forward Query	
Call Waiting	Call Waiting On	
	Call Waiting Off	
Call Back	Call Back Clear	
	Call Back Set	
	Call Back Query	
DND	DND Off	
	DND On	
	DND Query	
Call Park	Call Park	
Conference	Conference	
Call Pickup	Call Pickup	

Note: OfficeServ's default feature codes can vary according to different country setting.

1.3.1. MMC724 Feature Code

How to set feature codes for SIP stations in order to use SIP supplementary services is the same with how to set the feature codes for [2]legacy stations in OfficeServ. SIP station users first should know what feature code matches to which supplementary service. And then they can set 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 callee number. This way, OfficeServ can know that the specific SIP station (specified as caller number in INVITE message's from header) wants to set the specific feature code.

[2] 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 for itself and the feature code is '40', then SIP station simply calls to '401' (401 is for setting, and 400 for unsetting) in a form of normal INVITE message. But as you may guess, because this is not a actual outbound call, nor are there a specific station whose number is '401', OfficeServ responds back with a 480 (means temporary not available) message and terminates the call. Therefore, calling to 401 does not mean anything but letting the OfficeServ know 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 if there actually exists a station whose number is 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 worry about this feature code confliction.

Note that if OfficeServ's country code is set to Korea, there may needs some additional settings in USABLE FEATURES in MMC701. But readers of this document are not likely to be Korean and thus the last sentence can be ignored.

1.4. License Key Policy

One of the biggest changes from the previous version of MCP S/W and v4.10 is that this new version uses a license key to define SIP channel capabilities. We have used a license key in older versions of MCP S/W, but it was only for restricting the number of Soft Phones that can be attached to OfficeServ. This new license key scheme defines SIP channel capabilities including the number SIP trunk channels and the number SIP station channels. Therefore, if there were not for a valid license key, OfficeServ can not make or receive even a single SIP call.

License keys are issued only by a license server that is managed by license server manager. To obtain a valid license key, OfficeServ operator should consult the license key manager and let him know the MAC address of the corresponding OfficeServ system's MCP card. That means that use of the license key issued for a specific MCP card is restricted only for the card, and can not be used for any other MCP card which has a different MAC address.

As the number of SIP channels is defined and fixed into the license key when creating the key, you should let the license key manager know the desired number of SIP channels in advance along with the MCP's MAC address.

Note that a license key explicitly specifies the number of Non-Samsung SIP stations, mean while the number of Samsung SIP stations can be configurable freely within the total number of SIP channels specified in the license key. (Please refer to 2.2.2 Non-Samsung SIP Stations)

1.5. Definitions

Following definitions are made to enhance reader's understandability.

- SIP Trunk Mode: OfficeServ's SIP operation mode in which OfficeServ acts as a SIP 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.

1.6. 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.

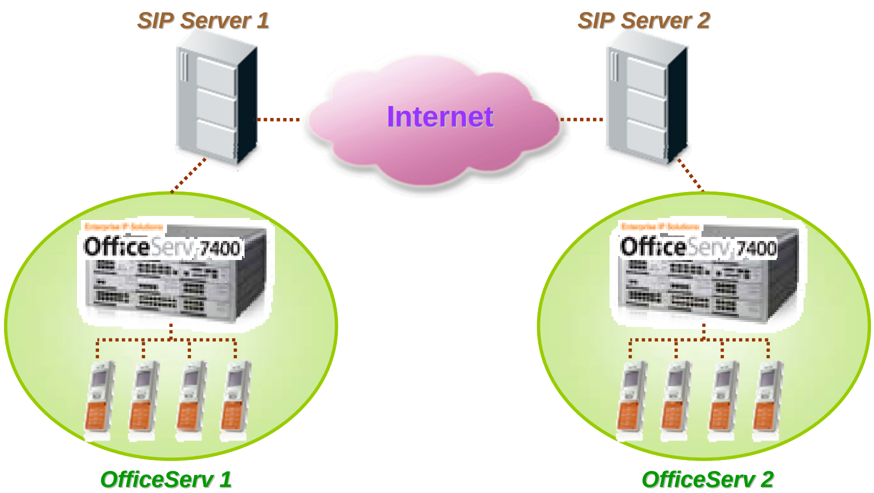

Figure 1. Overall Configuration for SIP trunking mode and SIP station mode

1.7. Software Requirements

SIP station features dealt in this manual are fully provided from MCP S/W v4.10 or later, therefore earlier versions of MCP S/W may not have these newly added SIP MMC options, or they may display some different names for the same functions if any.

1.8. Hardware Requirements

Currently SIP station features are implemented in OfficeServ 7200 and OfficeServ 7400 systems only. Samsung will broaden this system range based on customers' requests. But OfficeServ 500 system and under are excluded from this plan.

2. Preparation

2.1. License Key

As mentioned in 1.4 License Key Policy, from v4.10 of OfficeServ MCP S/W, a valid license key is a pre-requisite to use any SIP features in OfficeServ system (both for SIP trunk and SIP station). This license key defines the maximum number of SIP connections in OfficeServ and operators can assign the proper number of SIP channels for SIP trunk and SIP station within the range of possible channel capabilities specified in the license key. Let's say that an OfficeServ has a license whose capacity is a total of 100 SIP connections. If the corresponding OfficeServ system wants to have all the SIP connections as SIP trunk lines, operator can assign all the connections only for SIP trunk uses. Later on, if the same OfficeServ system should use both SIP trunk and SIP station at the same time, operator may change the original configuration and divide the SIP capabilities such as 50 for SIP trunk connections and 50 for SIP stations according to situation.

2.1.1. Generating a License Key

License keys are obtained from the license key server. Therefore, OfficeServ users first should contact the license key manager and get a valid license key in order to use SIP capabilities. Following figure is from the actual screen shot of license key server's web application and shows how a new license key is generated, specifying each different SIP capabilities.

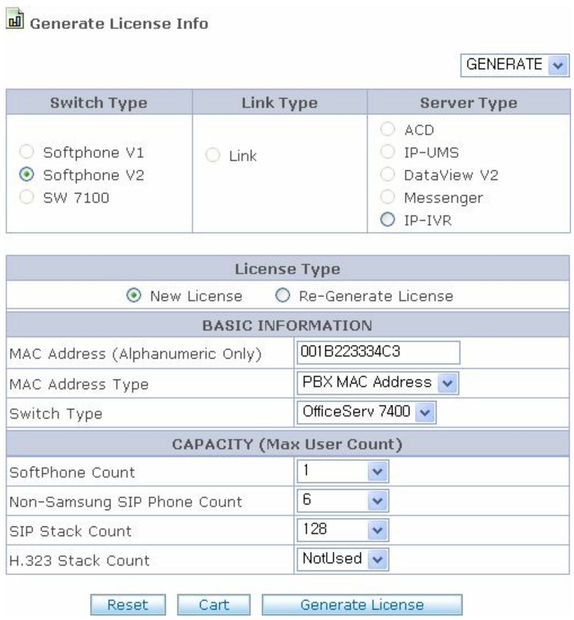

Figure 2. Generating a License Key

As shown above, to generate a license key, the license key server needs basic hardware information which specifies a MAC address of MCP card and the OfficeServ system type. This means that a specific license key should be used only for the corresponding OfficeServ system, and can not be shared with other systems. The capacity part consists of 4 parts: SoftPhone count, Non-Samsung SIP Phone Count, SIP Stack Count, and H.323 Stack Count. Among them, as SoftPhone Count and H.323 Stack Count are not mentioned here because they have nothing to do with SIP. Simply note that a single license key contains all the maximum capacity values for each different function that needs usage limit.

SIP Stack Count specifies the maximum number of SIP connections, which include both SIP trunk lines and SIP stations in OfficeServ system. Within this range, operator can freely assign SIP channels according to the OfficeServ system purpose. This assignment should be done using MMC because the license key specifies the maximum SIP connection only. When the maximum SIP usage in OfficeServ reaches this limit, the system will automatically drops the additional SIP connection request.

2.1.2. Non-Samsung SIP Stations

Unlike using Samsung-manufactured SIP stations in OfficeServ, non-Samsung SIP stations need to have 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 its stations, and this number can not be larger than the SIP Stack Count. This means that Non-Samsung SIP Phone Count should be included in the assigned number of SIP stations from total SIP Stack Count capacity.

2.1.3. Inserting a License Key

Once a valid license key is generated, operator can insert this license key to OfficeServ using MMC841 FEAT LICENSE KEY entry. As the format of license key is composition of 40 cryptic alphabet characters, it is RECOMMENDED to use PCMMC or installer S/W. If you think you are patient enough or want to practice inserting alphabet character using KMMC, Be My Guest!

2.1.4. 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 MMC841'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)

MMC841 FEAT LICENSE STS simply displays the current license key status and each value can not be modified by operators.

2.1.5. Configuring SIP channels

As mentioned before, operator can assign SIP channels according to system purpose. MMC841 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 SEC SIP + FREE COUNT + SIP TRUNK + SIP PHONE + [3](IP-UMS/IVR)

Note that 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 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, SIP needs a certain modification on this default configuration, and following sections show how.

2.2.1. 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. Whereas, OfficeServ 7100 has maximum 3 cards and each card has 8 SIP channel capacity. Therefore, the number of SIP trunk card and the 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.

2.2.2. SIP Station Card

Available SIP station numbers are shown in MMC842. If virtual SIP station cards are not properly mapped in MMC857, however, no number will be shown. So, operator should map virtual SIP

[3] IP-UMS/IVR 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.

station card to an appropriate virtual slots in OfficeServ in order to register SIP station to OfficeServ system. How to map virtual SIP station card to slots are all the same with the case of mapping the SIP trunk card.

3. Registration

As in the case of OfficeServ system in SIP trunk mode, SIP stations also need to register to a 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 [4]Registrar (or SIP Server) 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. As shown below, Registration flows are all the same between Trunk Mode and Station Mode, except for OfficeServ's role in each mode: UAC or Registrar.

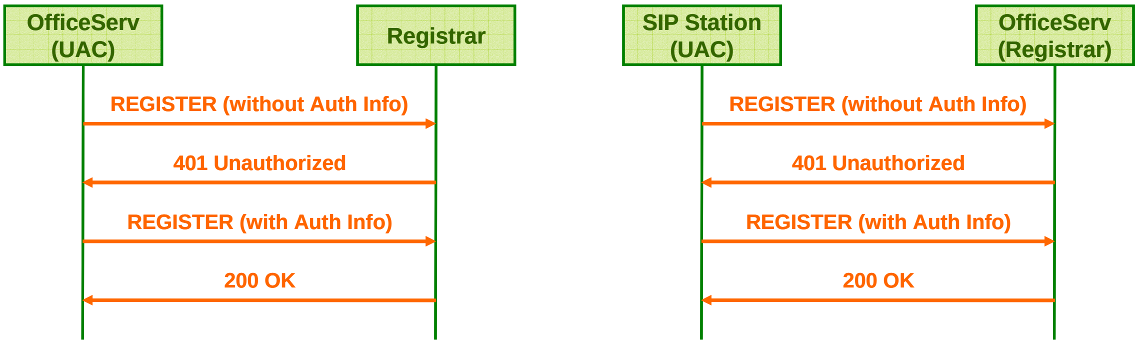

Figure 3. Trunk Mode Registration vs. Station Mode Registration

As shown in above call flow, SIP 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, nonce values. While UAC and SIP server publicly share the pair of username and password, nonce value is created only by SIP server side using internal nonce value generating algorithm, and can be known to UAC when a 401 response message is transmitted from SIP server.

[4] 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.

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

OfficeServ classifies SIP stations by two categories; 1) Samsung SIP Phones (i.e., Simple IP Phone, WIP6000, and any other SIP terminal manufactured by Samsung Electronics) and 2) Non-Samsung SIP Phones (i.e., Cisco 79XX series and X-Lite). This is for allowing Samsung SIP stations to be registered on OfficeServ without any further restriction, whereas non-Samsung SIP stations need special license quota for registration.

3.3. Registering SIP station

In order to register 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 MMC842.

First of all, MMC842 STND SIPP Table contains a pre-designated SIP station number list 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 MMC857, the number range will be also large. If no SIP station card is set, MMC842 will display none as well. Starting number of the list depends on the virtual cabinet's slot number that is mapped to SIP station card.

Table 5. MMC842 STND SIPP Table

Station No	Registered	IP Address	User ID	Password	SIPP Tone	Call Wait
3331	YES	192.168.89.7	3331	0000	Use System Tone	Disable
3332	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 3331 to 3362. OfficeServ system operator may change this pre-designated numbers to some others at MMC724 but, for now, assigning numbers 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', operator, however, can change the values if necessary as long as corresponding SIP station has the same changed password value.

MMC842 will display valid IP addresses when the respective SIP station is legally registered to OfficeServ system. By default, it displays blank IP address.

3.3.1. Non-Samsung SIP station

Procedure for registering Non-Samsung SIP station into OfficeServ is all the same with that of Samsung SIP station provided proper license key is placed in OfficeServ system. This license key prevents illegal registration of unauthorized SIP station. So, OfficeServ operator should insert legally issued license key into MMC841 prior to register Non-Samsung SIP station. If non-Samsung SIP terminal fails to register to OfficeServ, please check whether the license key has a valid license quota for non-Samsung SIP station.

4. 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 scenario cases for each services 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 follow each scenario steps 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.

4.1. 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 MMC842, please go back to section 3.3 and make it sure that SIP station is registered first.

Like using the other internal terminals such as DGP and ITP, users can make outbound calls by simply using destination callee numbers. That way, end users can 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, not Samsung proprietary protocol.

4.1.1. Basic Call Setup between SIP Stations

When [5]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 why we see OfficeServ stands in-between those two SIP stations is that all the SIP signaling messages have to go through OfficeServ because OfficeServ acts as a SIP proxy server for all the SIP stations registered to it.

- Even though SIP signaling messages are passed through OfficeServ, its RTP packets are transmitted directly between two SIP stations. This means SIP stations do not use MGI 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 [6]non-IP terminals (i.e., DGP) are involved in the session, MGI has to be used by the terminals.

[5] SIPP stands for SIP Phone in this document.

[6] 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.

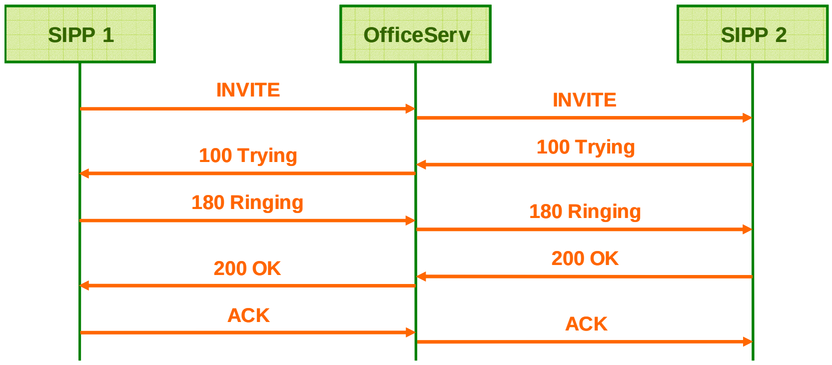

Figure 4. Basic Call Setup Flow between SIP Stations

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

Now, let's talk about more general cases which involve 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 OfficeServ's job. When the other party is Non-SIP terminal, OfficeServ translates the received SIP message to DGP or ITP understandable Samsung-proprietary message, and vice versa. Therefore, we can abbreviate the SIP-based call flow as a following figure, letting DGP stand for OfficeServ.

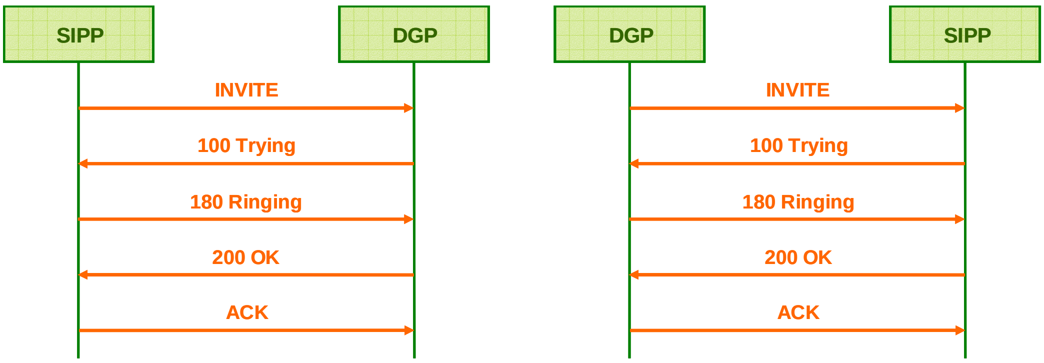

Figure 5. Basic Call Setup between SIP Station and DGP

4.2. Hold/Resume

Hold and 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 (You can see why later on). 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 two methods is all the same though messages have different names. Currently OfficeServ supports Re-INVITE message as its default Hold/Resume method. Re-INVITE is a normal INVITE message except it is sent within already-made active session. By sending an INVITE message which contains different [7]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.

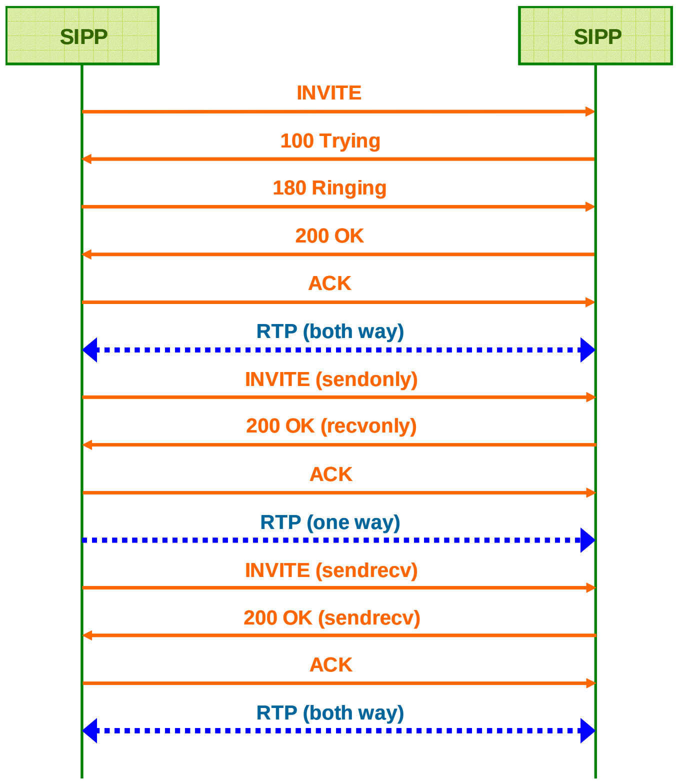

Figure 6. Hold/Resume between SIP Terminals

In a normal dialog state, the session mode is sendrecv which allows both way RTP transmissions. When holder 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

[7] 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.

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 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 sends Re-INVITE message specifying sendrecv, the session will remain in hold mode and holder will not change its mode.

4.2.1. Another way of specifying sendonly 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 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-allzero-specified Hold method for backward compatibility.

4.3. 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 transfer key as normal keys but some others provide softkey-typed transfer key interface through their LCD interfaces.

4.3.1. Consultation Transfer

Transferor executes call transfer while it is in the consultation state, having one of the callee terminal in active session (both way RTP state) and the other in hold state.

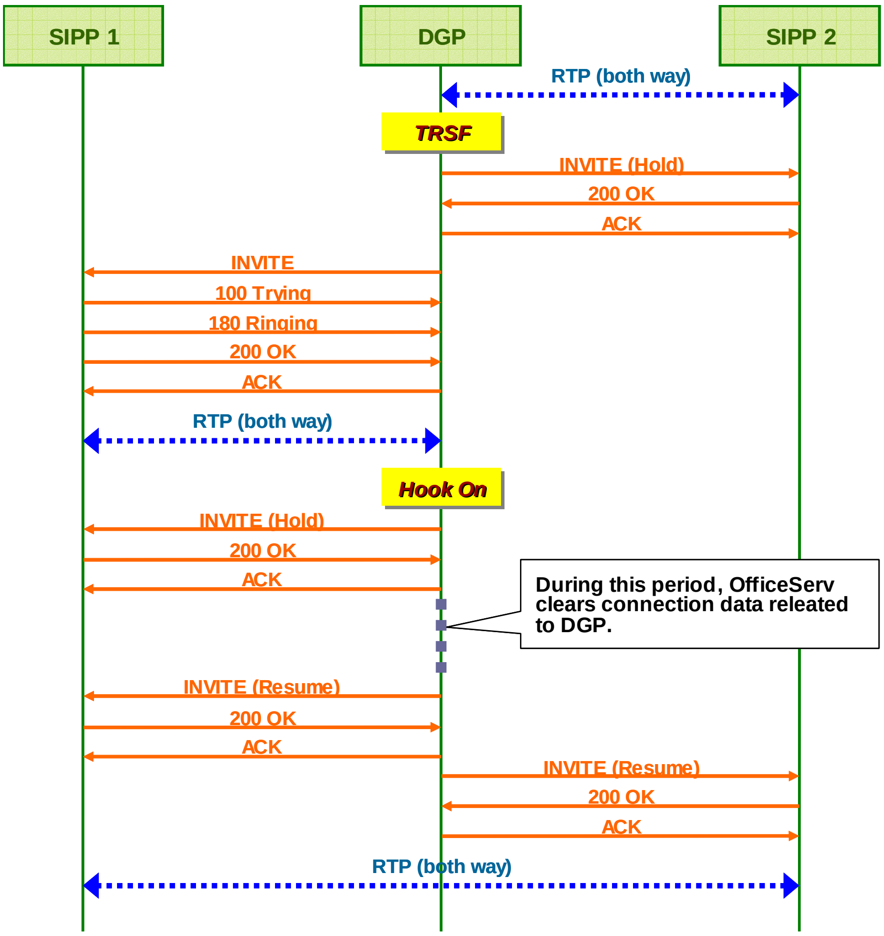

Figure 7. Consultation Transfer by DGP as a Transferor

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 Hook On by DGP, OfficeServ clears all the connection data that is related to DGP because DGP is now going out from the game.

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 communicate directly each other without using MGI.

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.

4.3.2. Blind Transfer

Transferor executes call transfer while it is in ringback state (180 ringing response received) against the second callee, putting the first callee terminal in hold state.

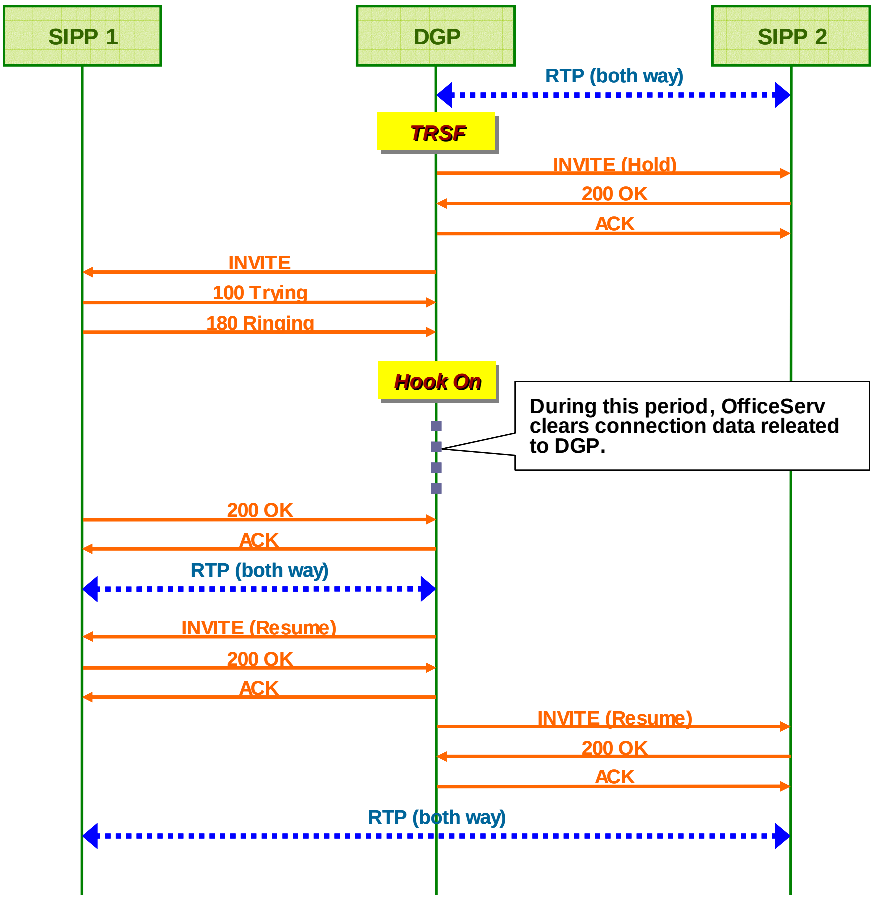

Figure 8. Blind Transfer by DGP as a Transferor

The main difference between Blind Transfer and Consultation Transfer is that in Blind Transfer, transferor hooks on when it is in ringback state. Except for it, from SIP station user's persfective, the main flow and its mechanism are almost same with the Consultation Transfer. Followings are the steps.

i. DGP initiates transfer process by pushing TRSF button, and it puts SIPP 2 on hold state.

ii. DGP makes the second session.

iii. After Hook On by DGP, OfficeServ clears all the connection data related to DGP because DGP is now going out from the game.

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 communicate directly each other without using MGI.

There is one important point in Blind Transfer. Whoever becomes Blind Transferor (i.e., standing in the middle in the call flow), it should put the transferee (i.e., SIPP2 in the call flow) in hold state before it sends the INVITE message to the 3rd station (i.e., SIPP1 in the call flow). This is a MUST 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 a hold message OfficeServ is supposed to receive in order to put the transferee in hold state, and this makes OfficeServ fail to execute Blind Transfer. Currently this is OfficeServ's Blind Transfer specification, R&D team, however, will consider revising the call flow in later release so that OfficeServ is able to process the Blind Transfer without hold message.

4.4. 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 MMC724 FEAT DIAL NUMBER. Like other feature codes, Call Forward feature codes are operator-definable and thus changeable as well.

4.4.1. Using MMC724 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 1.3.1 MMC724 Feature Code)

As to No Answer Forward, OfficeServ forwards an incoming call to a designated number when the original callee does not answer for a certain period of time which is specified in MMC502 No Answer Forward Time.

Table 6. MMC724 Call Forward Feature Code

Call Forward	Feature Codes
Call Forward Clear	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)

i. If operator wants to forward calls incoming to a SIP station (3337) to some other DGP (2002) when 3337 is busy.
ii. From the 3337, dial '6022002' to set Busy Forward to '2002'.
iii. Once Call Forward is set for 3337, another SIP station (3334) calls 3337 and make a connection.
iv. DGP (2001) makes a call to 3337 which is already in a session with 3334.
v. The call is forwarded to 2002.
vi. To clear Call Forward setting, dial '600' from 3337.

4.5. Call Waiting

Call waiting feature enables a SIP station, while it is in an active dialog session with some other terminal, to accept incoming calls, not simply rejecting it with 486 busy responses. To execute call waiting, each SIP terminal first should know desired Call Waiting feature code and set the call waiting using the code. OfficeServ operator can check the Call Waiting feature code specified in SIP CW in MMC724 FEAT DIAL NUMBER. Like other feature codes, Call Waiting feature code is operator-definable and thus changeable as well.

4.5.1. Using MMC724 Call Waiting Feature Codes

By default this SIP CW value is set to NONE, and operator can define any available feature code he/she wants. 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 OfficeServ. (For more detailed, please refer to 1.3.1 MMC724 Feature Code) If Call Waiting service for a certain SIP station is set, the corresponding SIP station's MMC842 CALL WAIT value is also changed to ENABLE. Following table shows the example feature code set for Call Waiting feature in MMC724. If the Call Waiting feature code is set as '77', use '771' for set and '770' for unset.

Table 7. MMC724 Call Waiting Feature Code

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

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

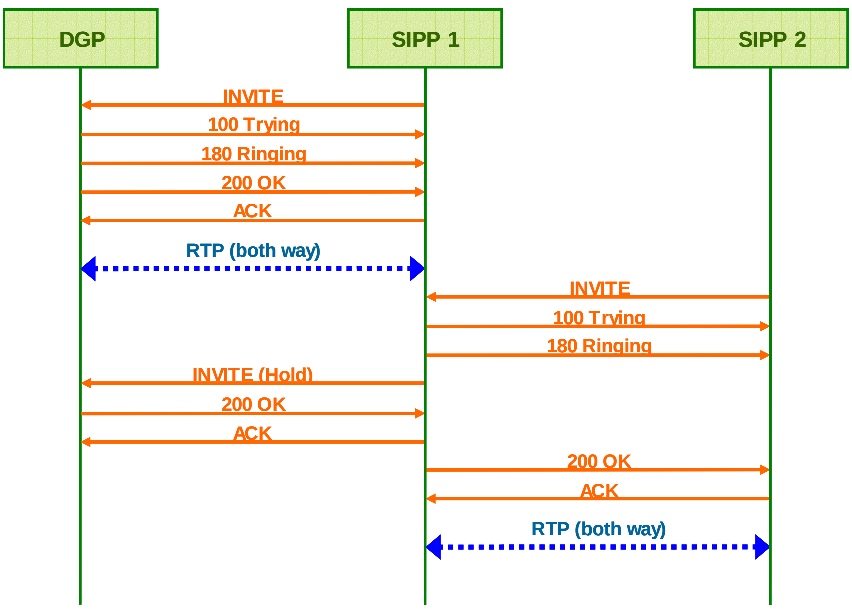

Figure 9. Call Waiting at SIP Station

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

i. Sets the Call Waiting feature for a SIP station (3334) by simply dialing ‘771’.

ii. From the SIP station (3334), make an outbound call to another terminal (2002)

iii. From the third SIP station (3337), make a call to 3334 SIP station.

iv. As 3334 now is set on Call Waiting mode, it does not reject the newly incoming call from 3337 and instead, informs the new call is incoming..

v. 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 that the in case of DGP’s requesting Call Waiting for a SIP station that is already busy in a session, select CAMP option in DGP’s LCD. This means that, unlike SIP station, DGP does not need to register Call Waiting service to OfficeServ. Next scenario shows this.

Here is a sample Call Waiting scenario. (2001/2002:DGP, 3332:SMT-i3015)

i. Sets the SIP CW value to 77 in MMC724 FEAT DIAL NUMBER option.

ii. From a SIP station (3332), dial ‘771’ and send to set Call Forward option enabled in OfficeServ. MMC867 shows the Call Waiting service is set for the SIP station.

iii. From other DGP terminal (2001), make a call to the SIP station (3332).

iv. From another DGP (2002), make a call to the SIP station (3332) which is already in a session with 2001.

v. 2002’s LCD shows that the callee (3332) is busy and displays 3 options: CBK, MSG, and CAMP. Select CAMP.

vi. The SIP station (3332) informs that the new call is incoming and operator accepts the call.

vii. Automatically the first call is set to hold mode, and the new connection between 3332 and 2002 becomes an active session.

4.6. Call Pickup

Call Pick can be done by any terminal attached to OfficeServ as its internal stations. To execute Call Pickup, each SIP terminal first should know desired Call Pickup feature codes and set the Call Pickup using the codes. OfficeServ operator can check the Call Pickup feature codes specified in MMC724 FEAT DIAL NUMBER. Like other feature codes, Call Waiting feature codes are operator-definable and thus changeable as well.

4.6.1. Using MMC724 Call Pickup Feature Codes

When doing Call Pickup using MMC724 feature code, users can use 3 different pickup services which were originally provided by OfficeServ as a K/P feature. Those are 1) my group 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 '*'.

Following call flow shows the detailed story.

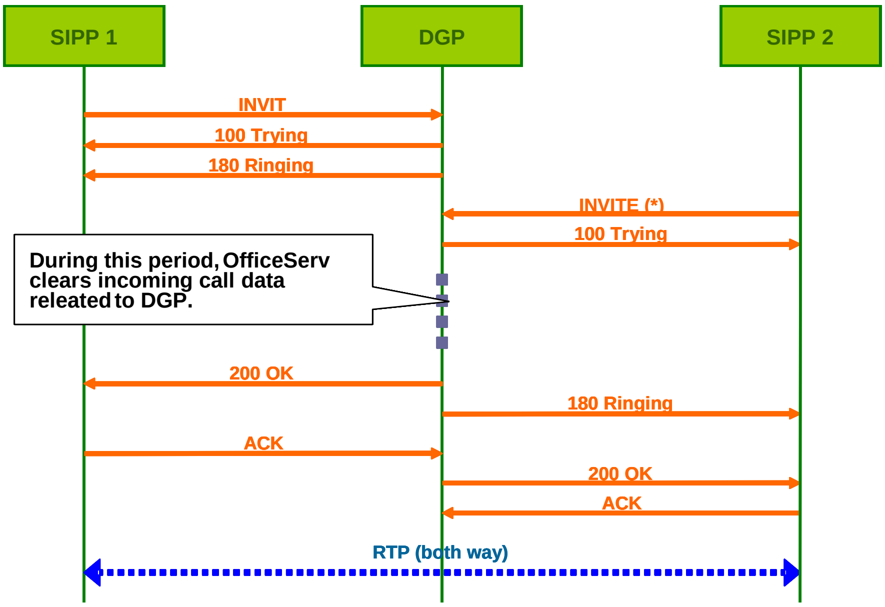

Figure 10. Call Pickup from DGP by SIP Station

When picking up a call from a SIP station, only the feature code is needed as the scenario described below. (2001:DGP, 3332:SMT-i3015, 3334:WIP6000)

i. From a SIP station (3332), dial a DGP number (2001).

ii. DGP (2001) rings and the SIP station (3332) is in a Ringback state,

iii. From another SIP station (3334), simply dial Call Pickup feature code of '*'.

iv. The original call is picked up by 3334, and a connection is made between 3332 and 3334.

4.7. Conference

Conference feature may be the most complicated feature both in usage and in its internal mechanism among SIP supplementary services implemented in OfficeServ. 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 in a conference are allowed.

4.7.1. Using MMC724 Conference Feature Codes

By default, the feature code for Conference specified in MMC724 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 made is to know what actions are taken 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 about more internal story.

i. Conference can be made by a conference owner station using the conference feature code while it is in an active session. 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.

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

iii. 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 i again. However, when choose option #2, it simply needs to send the conference feature code once more to OfficeServ system without taking any other action.

iv. 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 the all station in the conference so that they can communicate each other via MGI.

v. Finally, the conference owner station MUST resume its hold session to 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 in the conference room.

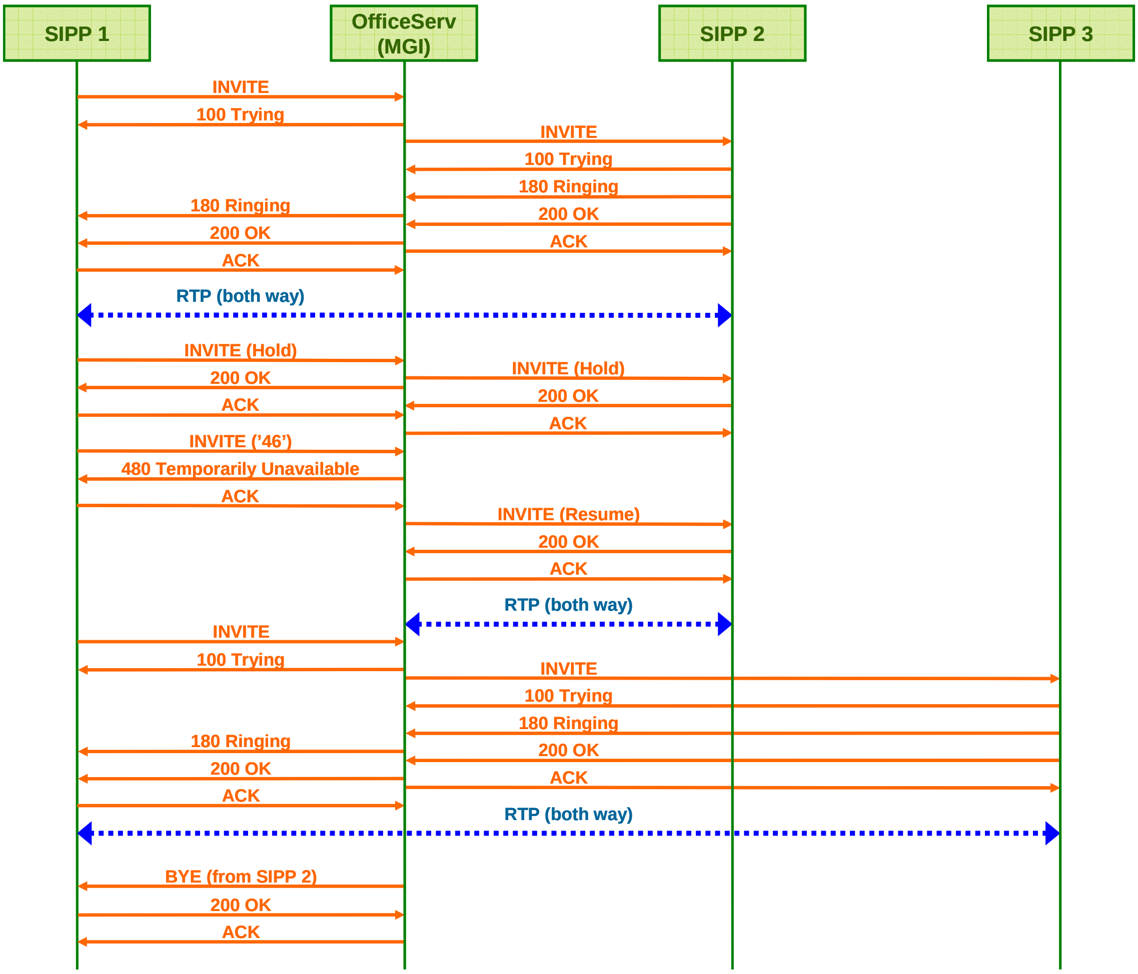

Figure 11. Conference made by a SIP station (#1)

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 for 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 resuming it. 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, manual hold mode switch 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 more important for disconnecting is that as many SIP terminals support maximum 2 concurrent connections only, leaving the original connection will block SIPP 1's next sending out of '46' conference feature code, which may lead 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).

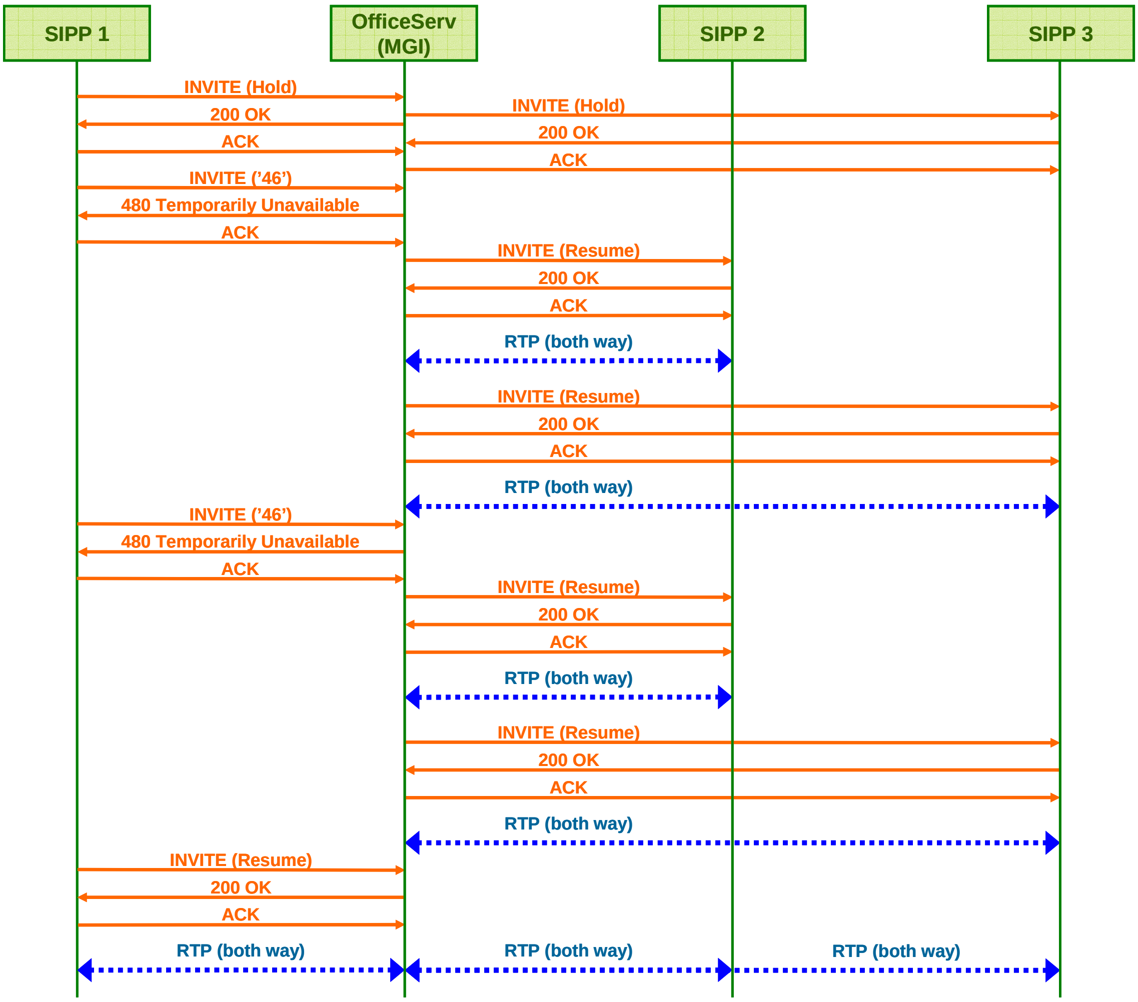

Figure 12. Conference made by a SIP station (#2)

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 to hold mode again. After this, SIPP 1 goes back to virtually 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 gives 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 there is no harm obviously.

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 in the conference. In many cases, users are apt to miss this last process of resuming the conference owner station, therefore it is important not to forget it.

4.8. Call Park (System Hold)

Using Call Park feature, a SIP station can park the call in an active session, and later 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. And as in the case of call pick up feature, a Call Park feature code is applied only for a single call.

4.8.1. Using MMC724 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 MMC724 FEAT DIAL NUMBER option. By default, the feature codes are set to 11 and 12 respectively, and of course operator can change these values if he/she wants. To enable retrieving a parked call using MMC724 feature code, operator should set MMC option of STNHOLD PICK in MMC210.

Table 8. MMC724 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 original call parker number.

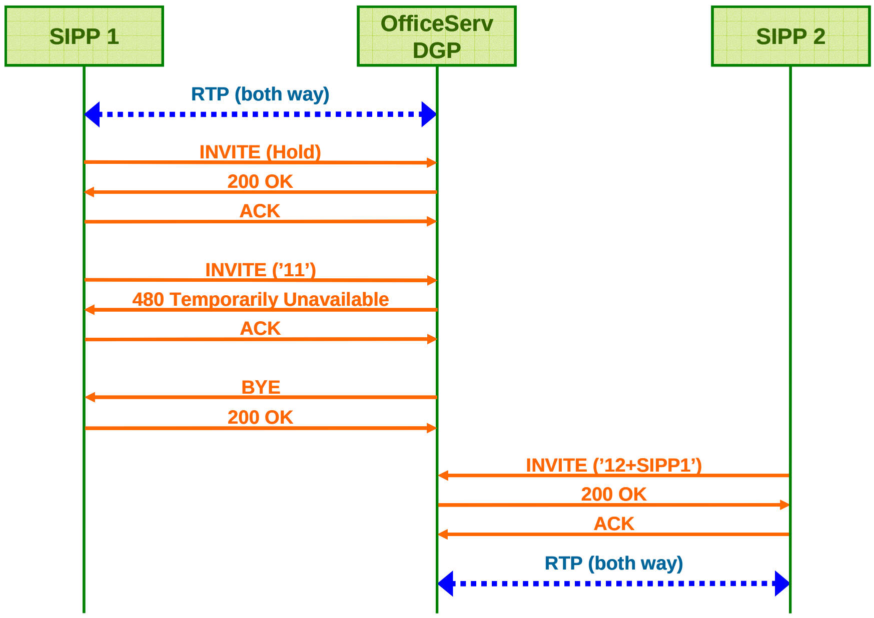

Figure 13. Call Park from SIP Station

Following is the sample scenario for Call Park. (2001/2002:DGP, 3332:SMT-i3015)

i. From a SIP station (3332), dial a DGP number (2001).

ii. DGP (2001) answers and an active connection is made between 3332 and 2001.

iii. To park the current call, the SIP station (3332) first puts the call in Hold mode, selecting Hold button. (When using WIP6000, this process is unnecessary because it is automatically done right before sending the call park feature code)

iv. Then, the SIP station (3332), parks the call by dialing the MMC724 Call Park feature code of '11'. (The feature code name of Call Park is a bit tricky because the feature code name is HOLD (System Hold), not CALL PARK. However, it does call park from a perspective of SIP supplementary service)

v. The call is parked onto OfficeServ system, and the SIP station (3332) is now in an IDLE state.

vi. From another DGP (2002), retrieve the parked call by dialing '123332'. (It's not 122001)

vii. 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 call the original call parker to make the connection again, which seems to be a Call Back function.

4.9. Call Back

Call Back feature enables a caller to receive a call from its original callee when the callee becomes IDLE, which was in BUSY state at the time of caller's initial inviting.

The biggest difference between case of using SIP station and case of using legacy K/P 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 callee, mean while legacy K/P station can ask the Call Back service directly through LCD interface while it is in ringback state or in busy-received state. Of course a SIP station would not ask for Call Back service if it had not received any busy response from the callee.

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

4.9.1. Using MMC724 Call Back Feature Codes

When ask Call Back service from DGP, it simply select CBK soft key from LCD after receiving busy response from the target terminal. Therefore, here it is necessary to mention how to use MMC724 Call Back feature code when using other standard SIP terminal such as Samsung's SMT-i3015.

Default Call Back feature code specified in CBK in MMC724 FEAT DIAL NUMBER option is '44'. So, use this code + target station number when asking Call Back from a SIP terminal.

Table 9. Call Back Feature Code List

Call Back	Feature Codes
Call Back	44

Following test scenario is for Call Back feature. (3332, 3334, 3340: SMT-i3015)

i. From a SIP station (3332), dial another SIP station (3340).

ii. 3340 answers and an active connection is made between 3332 and 3340

iii. 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.

iv. 3334 asks Call Back service to OfficeServ dialing Call Back feature code + target station number.(i.e., '443332')

v. 3332 hooks on to end the connection with 3340.

vi. Knowing the SIP station (3332) becomes IDLE state, OfficeServ calls 3334 who asked 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.

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

viii. Now the call is made between 3334 and 3332.

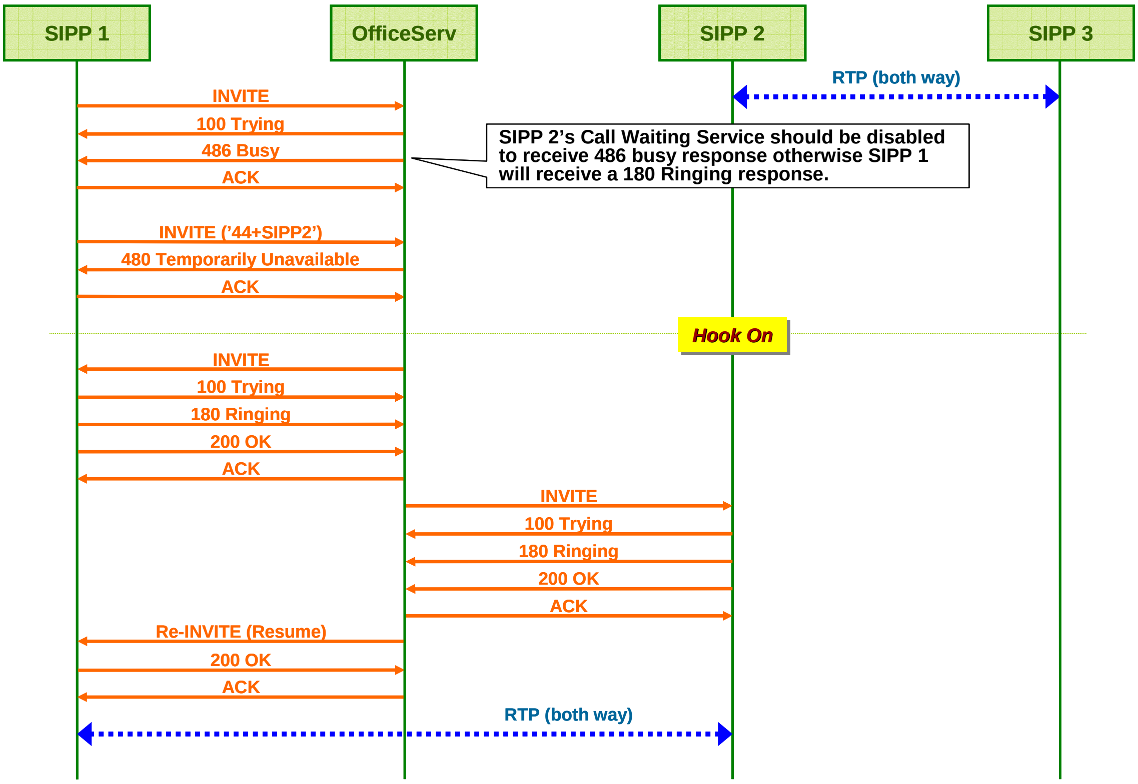

Figure 14. Call Back from SIP Station

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.

4.10. 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 hence does not need to give any response. DND setting lasts valid until it is manually unset, so be careful not to forget to unset DND when the service is not necessary anymore otherwise the terminal will not receive any incoming call ever.

4.10.1. Using MMC724 DND Feature Codes

To practice DND feature, operator should use DND feature code specified in DND in MMC724 FEAT DIAL NUMBER option. By default, OfficeServ has DND MMC724 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 unset.

Table 10. Call Back Feature Code List11. Call Back Feature Code List

DND	Feature Codes
DND	40
DND Set	401
DND Unset	400

Following is the scenario for DND setup. (3332:SMT-i3015, 3340:WIP6000)

ix. From a SIP station (3340), ask DND service dialing DND feature code of '401'.

x. From another SIP station (3332), try to make a call to 3340 who has set DND service on, then OfficeServ gives 486 busy response back to 3332 on behalf of 3340.

xi. From a SIP station (3340), ask unsetting DND service dialing feature code of '400'.

4.11. 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).

Though 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 seem to react on the MWI message.