SAMSUNG

Enterprise IP Solutions

Officeserv SIP Station Manual

Edition 3

Software version 4.7

New features available in 4.7 software

- MPS feature provides support for NAT.
    - SIP phones can be used through NAT boundaries.
- Sending DTMF from Digital and SMT phones to SIP phones is supported.
    - Sends DTMF to door phones to operate relays
- H.264 Video on SIP phones is supported.
- Call forwarding unreachable can be set on SIP phones.
    - Enables incoming call redirection when the SIP phone is not registered.
- Additional security features
    - IP White list to identify trusted IP addresses
    - There are no default user ID and passwords for SIP stations
    - The User ID and password cannot be the same.
    - SIP station passwords must be longer than 6 digits and may contain alpha and other type characters.
    - The SIP station password is not shown in the DM display and is encrypted in the database.
    - The “Comm Exclusive” feature provides additional protection against attacks. It can be set to ignore messages from IP addresses that repeatedly send failed registration attempts.
- Unconditional ringing groups can now include SIP stations
    - In the OS7030 up to 8 SIP station members can be in a group (16 total members)
    - In the other systems up to 10 SIP station members can be in a group (32 total members)
    - Only 4 SIP phones in the group ring at the same time. There is a 50mS delay between groups of ringing SIP Phones.
- Automatic addition of the Trunk Group Access code for SIP phones
    - Allows SIP phones to use their call log to make outgoing trunk calls.
- SIP Phone Auth code dialling
    - Allows a SIP phone to provide an authorisation code when making outgoing calls.
    - The SIP station dials the "auth feature code" + "auth code" + "trunk access code" + "outgoing digits” as a single number (not overlap sending)
- SIP Station diversion header
    - The OfficeServ sends the identity of the forwarding phone to the SIP station that receives the forwarded call.
    - Example: A Party (STN or TRK) calls Phone B (FWD set to SIP Phone) → SIP phone receives the call and shows the diversion information on its display.
    - Diversion information is only displayed on SIP phones that support the feature.
- TLS and sRTP can be used with SIP Stations.
    - TLS encrypts the SIP signalling
    - sRTP encrypts the voice packets
    - Must be supported by the SIP phone

What are SIP stations?

The IP voice functionality of SIP stations in the OfficeServ is similar to standard IP phones, but instead of using Samsung's proprietary protocol they use SIP.

The significant difference between SIP stations and Samsung IP Phones is the number of features provided and how they are used. Samsung IP Phones integrate with the OfficeServ to perform many features using soft keys and feature buttons, while OfficeServ SIP stations can only provide functionality for very basic features and call states.

SIP stations can send voice traffic directly between each other, and to IP phones, when they are in the same private network.

SIP stations use MGI ports when communicating with devices that are non-IP based (ISDN trunks, Digital phones, etc.). The MPS feature can be used if the other endpoint is also IP based.

In this document the term 'endpoint' is used in many descriptions; in SIP terminology the word 'endpoint' refers to a device that can send and receive SIP transactions. An endpoint can be an Officeserv system, a SIP Phone, a SIP carrier's proxy server, or any other equipment that communicates using SIP.

How SIP phones communicate with the OfficeServ

The OfficeServ requires that SIP Phones use registration and authentication before it will allow them to make and receive calls.

Registration is used by SIP devices to tell a registration server (also known as a *Registrar*) where to find the user of a particular account. A *Registrar* is built into the OfficeServ so that SIP phones can register their location. For example: A SIP phone will tell the OfficeServ that it wants to use a specific station number by sending a REGISTER message to the OfficeServ. After a SIP phone successfully registers its station number with the OfficeServ, the OfficeServ will know which device to send incoming calls for that station number. The station number, password details and IP address of the device sending the message are important details included in the registration messages.

The OfficeServ will reject the registration request if the station number or password is incorrect, and as result, calls will not be sent to the SIP phone.

Authentication is used by SIP endpoints to verify that SIP INVITE messages received from another device are from a valid SIP account, and not from a malicious third party pretending to be that account.

The *SIP Station* feature in the OfficeServ does not use authentication to verify that the SIP phone making a call is the valid holder of the account. However, the OfficeServ does check the source IP address of SIP stations attempting to make outgoing trunk calls; and the call from the SIP phone is rejected if their source IP address does not match the IP address of a registered station.

Compatibility of 3rd party SIP devices

The SIP Station feature allows SIP capable devices to make calls via the Officeserv. However, many of the enhanced station features that are available on SIP phones are product specific. This usually happens because the developer has taken a unique approach to the design of the SIP message flow for a feature, or that the SIP standard has not covered the feature that they wish to provide. In addition, there is no organisation that tests, approves, or regulates the implementation of features using SIP. So errors and bugs in the SIP functionality of SIP devices can be a common problem. The consequence of these incompatibilities and bugs is that although a device may be labelled as being SIP compliant it does not necessarily mean that it will work properly.

Before offering SIP devices to a customer always check that the functionality that they require of that device will operate correctly on the Officeserv.

Use the ***Testing Process*** and ***Interoperability Checklist*** sections of this document to help you determine which features are compatible between the OfficeServ and the IP phone; and to help you document the operation of those features.

Testing process

The recommended test process is as follows:

1. Determine which of the features sought by the customer are provided by the SIP device.

2. Verify that those SIP features are supported by the Officeserv.

3. Read the information in this guide that describes the features provided by the OfficeServ.

4. Configure the Officeserv and SIP device and test all of the required features.

5. Document the settings in the Officeserv and the SIP device.

6. Document the actions the user must perform to use the features.

Interoperability checklist

The ***SIP Station features*** section of this document describes all of the features supported by the Officeserv's SIP station feature. Use the information in that section and the SIP device's user guide to perform tests and fill in the details in the following chart. This chart will create a checklist of the useable features and keep a record of any special considerations of which the user must be aware when using that product. The chart provided on the next page is provided to help you document the test process and record the results.

For each test check that:

1. The appropriate tones are heard during the relevant call states.
   For example, Hold tone during the hold state and ring tone during the ring state.

2. That the name, number and status displays are correct.
   Note that SIP stations have an inherent problem with updating display information when the call state changes. If the SIP device does not show the correct called or calling party number after a transfer or recall it is not necessarily a fault. The OfficeServ does not send information purely related to phone displays to the SIP device because this functionality is not specified in the SIP standards. This issue is not unique to the OfficeServ system, although some SIP based PBXs may offer support for additional display information using a proprietary method.

3. That there is two-way speech during conversation states.
   SIP calls require the set up and tear down of voice paths between the devices participating in the call. When features like Transfer and Conferencing are performed the voice paths and their destinations will change several times, so it is important to check that those paths have been correctly re-assigned after those features have been used.

4. That at the end of a call all of the participating devices can release the call properly.
   It is very important that all devices involved in a call are returned to the idle state after the call is complete.

SIP Station feature checklist		
SIP Phone brand, model and software version:		
OfficeServ system type and software version:		
Feature	Test Result	Notes
Registration		
Basic Call – SIP Phone to SIP Phone		
Basic Call – SIP Phone to Digital Phone		
Calling Name & Number Display		
Hold/Resume – Station to Station.		
Hold recall		
Music on Hold		
Consultation Call – Return to original caller		
Consultation Call – Toggle between members		
Consultation Transfer – Complete transfer		
Consultation Transfer – Cancel transfer		
Blind Transfer – Call Answered		
Blind Transfer – Unanswered Recall		
Call Pickup		
Call Forward All		
Call Forward Busy		
Call Forward No Answer		
Call Forward clear		
Conference		
Call Park (System Hold)		
DND		
Voice Mail Indication		
Voice mail retrieve		

Configuring SIP stations in the OfficeServ

There are a couple of steps that must be completed before SIP stations can be configured in the Officeserv. The first step is to obtain a SIP license key that grants permission to use the required number of SIP stations. The second step is to change at least one of the cards in the virtual cabinet to 'SIP STN' to allow you to configure SIP phones. Both steps are explained in greater detail in the following sections.

Licensing SIP ports

There must be SIP station ports licensed in the Officeserv for the SIP stations to function. The license screen displays two types of SIP stations; *SIP Phone* and *3rd SIP Phone*. This document describes the use of third party SIP devices, so the license required is for *3rd SIP Phones*.

The number of licensed SIP stations is the maximum number of SIP phones that can be registered in the system. New registrations will fail if the amount of SIP phones trying to register exceeds the number of phones licensed.

The maximum number of SIP stations that can be programmed in an Officeserv depends on the type of Officeserv being configured. The limits to the maximum number of programmable SIP stations are described in the section of this document titled "**Assigning Virtual Cards**". SIP station license keys can be obtained that allow the number of SIP stations to exceed the number of programmable ports, but the maximum number of *usable* SIP stations is fixed at the number of programmed ports.

The Officeserv license key numbers are entered in **License Key (DM 2.1.4)**

MGI and MPS port requirements

The OfficeServ may need a MGI port license in addition to a SIP Station license. An MGI license is only required for the OS7030, OS7100 or OS7220S and then only when the inbuilt MGI ports are used. A license for the inbuilt MGI ports is not required when an MGI or OAS card is installed in the OS7100 or OS7200S.

SIP stations may use the MPS feature when they exchange voice traffic with other VoIP devices connected to the OfficeServ. For example: SIP trunks use MPS channels while in the conversation state with a SIP station.

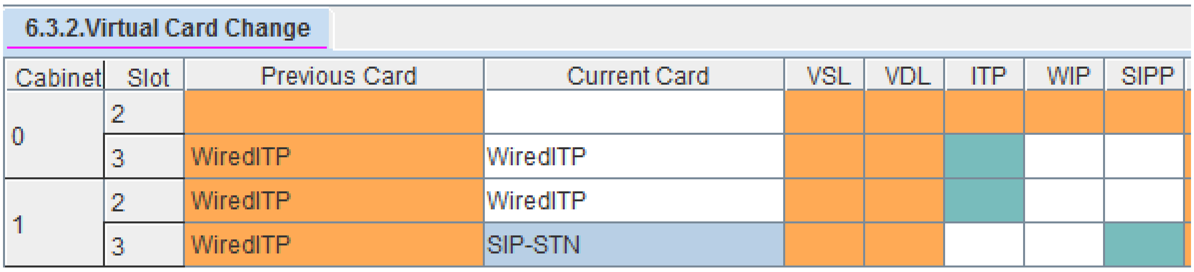
The MPS feature is also used when the network link between the SIP phone and the OfficeServ includes NAT boundaries. This occurs even if the other device is an IP phone or another SIP phone.

Features that use a speech path from a TDM source, such as music on hold and internal zone paging, always use MGI ports.

The total number of MGI ports needed in the system depends on the number of VoIP stations that need to communicate with non-VoIP stations and trunks. There must be enough MGI ports available to service the SIP stations as well as any other VoIP devices, such as IP Phones, SPnet trunks and SIP trunks.

Assigning Virtual cards

The virtual cards for SIP stations are not enabled in the default database, so they must be assigned for the virtual ports to become available. **Virtual Card Change (DM 6.3.2)** is used to change the virtual card configuration.



6.3.2.Virtual Card Change

Cabinet	Slot	Previous Card	Current Card	VSL	VDL	ITP	WIP	SIPP
0	2							
	3	WiredITP	WiredITP					
1	2	WiredITP	WiredITP					
	3	WiredITP	SIP-STN					

The number of virtual ports that can be assigned to SIP stations depends on the type of Officeserv system that is being configured.

SIP Phone Capacity

OS7030	16
OS7100	56
OS7200S	64
OS7200 (MP20)	128
OS7400	480

SIP Phone Capacity with TLS enabled

OS7200 (MP20)	64
OS7400	160

Configuring SIP station ports

The SIP station ports are configured in **SIP Phone Information (DM 2.7.2)**

2.7.2.SIP Phone Information

Tel Number	User ID	Password
217		
218		
219		
220		

The User ID and Password for the SIP phones are blank after the virtual card is assigned.

Settings in SIP Phone Information (DM2.7.2) page

User ID
Password
Tone Source
Call Waiting
Call forwarding Unreachable
DTMF Type
Insert Trunk Port
Insert Trunk Type
NAT Check
Use IP White List

Tel number and User ID

When a virtual card is assigned the *Tel number* and the *User ID* are blank. The *Tel Number* matches the *User ID sent by the SIP phone*. The *User ID* is used for authentication and cannot be blank.

Password

This is the password that the SIP Phone must send to the OfficeServ when it is asked to provide authentication. Passwords must be at least 6 digits long (max 8) and can use alpha-numeric characters along with some special characters, such as: !@#$%^&* _

NOTE: If the **Use IP White List** setting (see below) is enabled and the IP address of the SIP Phone is not in the **Phone IP White List** then registration is *not* possible.

Tone Source

The *Tone Source* parameter is set to either ‘Use System Tone’ or ‘Use Phone Tone’. When the *Tone Source* parameter is set to ‘Use System Tone’ the Officeserv sends hold tone from the MGI card. When the *Tone Source* parameter is set to ‘Use Phone Tone’ the hold tone is sent directly from the phone that put the call on hold.

Note that when the *Tone Source* parameter is set to ‘Use Phone Tone’, and the SIP phone is put on hold, the OfficeServ tells the SIP phone that the phone that put them on hold is sending the hold tone. This means that the ‘holding’ phone must generate hold tone and send it directly to the SIP phone.

In most cases this parameter should be set to 'Use System Tone' so that the SIP station will receive hold tone from the OfficeServ*. Unexpected results can occur if it is set to 'Use Phone Tone', as the phone that is supposed to provide the hold tone may not be able to provide it.*

Call Waiting

Call Waiting is no longer supported by SIP stations. Even though this option is provided and can be changed from "Disable" to "Enable", it has been deactivated in system software and does not function. The DID call waiting feature is still available for incoming trunk calls, but using that feature on SIP stations may result in those calls becoming lost.

Call forwarding Unreachable

Incoming calls are forwarded to this destination if SIP station is not registered to this SIP account. The destination must be a station number.

DTMF Type

The OfficeServ can send DTMF information to SIP stations. This is provided so that SMT and Digital phones can send door unlock codes to the SIP based door stations. "RFC2833" sends tones in band and "INFO" sends the tones as SIP messages. "RFC2833" is used by most door phones.

Insert Trunk Port

An LCR access code, Trunk number or Trunk group is programmed in this field. The **Insert Trunk Type** option determines how this number is used.

Insert Trunk Type

Determines how the value in the **Insert Trunk Port** field is used.

INCOMING: The value in the **Insert Trunk Port** field is added to the front of the CID on incoming trunk calls. This number can be the LCR access code, Trunk number or Trunk group. Therefore, when the SIP phone uses its local call log to make a call the required trunk access code is appended to the front of the dialled number.

For example: When the **Insert Trunk Port** is "802" and the **Insert Trunk Type** is set to "Incoming" and a trunk call is received with the CID of 0398722900 the call log in the SIP phone contains *802*0398722900.

Note: This uses the `From` and `Contact` header in the SIP INVITE message. The SIP phone should use this information in its call log.

OUTGOING: The OfficeServ checks the length of the dialled number and if it is longer than 4 digits adds the **Insert Trunk Port** code to the start of the number.

If the dialled number is 4 or less digits long the OfficeServ does not add the **Insert Trunk Port** code.

NONE: The OfficeServ does not use the **Insert Trunk Port** code at all.

NAT Check

When NAT is in the path between the OfficeServ and a SIP station this parameter helps the system identify the IP address and Port of the SIP station. It is used when the OfficeServ receives SIP messages from a SIP station via NAT in which the source port number in the IP packet header and the SIP `Contact` header are different.

1) **NAT Check** is set to “IP” (Default)

Compare only the IP address in the IP packet header with the value in the `Contact` header. If it is different the OfficeServ sends the SIP INVITE messages to the IP address and Port written in the `Contact` header.

2) **NAT Check** is set to “IP & Port”

Compare the IP address and Port in IP packet header with those values in the `Contact` header. If they are different, the OfficeServ sends the SIP INVITE message to the IP address and Port in the IP packet header.

Use IP White List

When the **IP White List setting** is “Enabled” (Default is “Enabled”) the OfficeServ checks the **Phone IP White List** for the IP address of the SIP phone. If the IP address is not in the table the SIP phone will not be able to register or make calls.

Set the **IP White List setting** to “Disable” to stop the SIP Stations from using this feature.

The **IP White List** contains IP addresses that are trusted by the system. The IP addresses are programmed in the page: **Phone IP White List (DM5.13.10)**. If a range of IP addresses need to be programmed, e.g. 192.168.1.1 – 192.168.1.254, then the broadcast address of the class-full IP range needs to be configured. For example: to allow all SIP phones in the range 192.168.1.1 to 192.168.254 to register the entry in the **Phone IP White List** is programmed as 192.168.1.255.

Note that the White List feature provides additional security to the OfficeServ and should be enabled whenever possible.

TLS and sRTP

- **TLS** is Transport Layer Security and it encrypts the SIP signalling.
- **sRTP** is Secure Real Time Protocol and encrypts the voice traffic.

The SIP Phone must support TLS and/or sRTP to use these features. The configuration of TLS and sRTP in SIP phones is beyond the scope of this document – please consult the administration and programming guides provided by the supplier of the SIP phones for more information.

WARNING: These features are not supported by all OfficeServ models. And they have an effect on the number of endpoints and concurrent calls in the OfficeServ systems that support them. Please read the OfficeServ 4.7 feature guide for details.

Configuring TLS in the OfficeServ

SIP station TLS related parameters are configured in **DM 5.2.12 SIP Stack/Ext/Trunk Options**.

TLS Port

This is the UDP port that the OfficeServ system listens for TLS packets. Its default value is 5061.

SIP Connection Reuse

This is only valid if the transport protocol is TCP or TLS. If this parameter is set to ENABLE the OfficeServ will reuse the TCP or TLS connection which has been established during SIP REGISTER process. If this is set to DISABLE the OfficeServ will make new TCP or TLS connections whenever a call is made. You can reduce the load by setting this parameter to ENABLE, but the SIP station must also support SIP Connection Reuse.

SIP Mutual TLS Enable

If this is set to ENABLE the OfficeServ will request certificate from SIP Peering or SIP Station.

SIP Validate Any TLS Certificate

If this is set to DISABLE the OfficeServ will abort a TLS connection when the certificate sent from the SIP Station is unknown. When this parameter is set to DISABLE the certificate used by the SIP Station must already be loaded in the SD card in the OfficeServ. If this is set to ENABLE the OfficeServ will omit the verification process for the certificate.

Configuring sRTP in the OfficeServ

Enabling sRTP can affect other stations and trunks in the OfficeServ, please be careful when configuring this feature.

- The supported encryption method is AES-CM mode and the supported authentication method is HMAC-SHA1.
- The Master Key size is 128 bit

Set the **sRTP Algorithm** option in **DM 2.1.5 System Options** to use AES_CM_128_HMAC_SHA1_80.

The SIP phone will have to request sRTP operation when it wants to use this feature.

SIP stations and the OfficeServ *Private with Public* feature

The *Private with Public* functionality of the Officeserv is designed to enable IP phones and IP trunks to operate through a NAT (Network Address Translation) boundary at the local (OfficeServ system) end of a link. This feature requires that the main processor card and MGI card are configured with both private and public IP addresses. The configured public addresses are typically the same as the public address of the NAT device that links the customer site to the Internet.

The OfficeServ assumes that a SIP phone is in the public network when its IP address is not in the same subnet range as the Officeserv. This means that when a SIP station and the OfficeServ are in the same private network, but the IP address of the SIP station is in a different subnet, the OfficeServ will incorrectly use its own public address in the SIP messages. Fortunately, the OfficeServ can be programmed to treat the addresses of SIP phones in other subnets as private addresses.

Private IP addresses that are outside of the OfficeServ's subnet range are programmed in **Private IP Address (DM 5.2.8)**. When the IP address of a SIP station is added to this table the Officeserv assumes that it can be reached *without* going through a NAT boundary, and that the SIP station is able to send voice *directly* to other VoIP devices in the private network.

5.2.8.Private IP Address

Entry No	Private IP Address
1	192.168.80.255
2	0.0.0.0
3	0.0.0.0
4	0.0.0.0

Note: Whenever a SIP station calls another SIP station, or IP phone, on the same private network the voice packets will be sent directly between them and will not use the OAS/MGI card.

SIP Extension Options

The **SIP Stack/Ext/Trunk Options screen (DM 5.2.12) screen** is used to set the parameters that are common to all SIP stations in the Officeserv.

SIP Extension Configuration	Signal Port	5060
	IPUMS/IVR Signal Port	5070
	SIP Expire Time (sec)	600
	NAT Reg Expire Time	60

Signal Port

The S*ignal Port* is the IP port number on which the Officeserv will expect to receive SIP messages from SIP phones. In addition, outgoing SIP messages from the Officeserv to the SIP phones will use the *Signal Port* value as the source port in the IP packets. The default value for the signal port is 5060.

SIP Expire Time

The *SIP Expire Time* is used by the Officeserv to ascertain if a SIP phone is still active. The OfficeServ starts this timer when a SIP phone successfully registers. If this timer expires before the SIP phone makes another registration attempt it will be de-registered by the OfficeServ.

The default value is 600 seconds (10 minutes). The registration timer configured in the SIP phone should have the same value.

5.2.12.SIP Stack/Ext/Trunk Options

Item		Value
	Comm Exclusive	Response
	Common MSG Block Timer (Sec)	600
	Register MSG Block Timer (Sec)	60
	Register Retry Limit	2

Comm Exclusive

There are two settings for this feature: “Response” and “No Response”. “No Response” gives greater protection against attacks, because the OfficeServ will not respond to registration requests from IP addresses that fail to send the correct User ID information. This helps protect the system from flooding attacks from attackers. “Response” will always send a rejection message on failed registration attempts.

Register MSG Block Timer

This is the period of time that the OfficeServ will block messages from a suspect IP address.

Register Retry Limit

This is the number of times that a SIP Phone can attempt, and fail, to register before being blocked by the OfficeServ

Displaying SIP station status

The Registration status of the SIP phones is displayed in the **SIP Phone Status screen (DM 6.2.3)**.

6.2.3.SIP Phone Status

Tel Number	Current Status	Phone Type	Public IP Address	Public Port	Signal Port	Protocol
217	Registered	Non Samsung SIPP	192.168.1.2	47268	47268	UDP
218	Not Registered	Non Samsung SIPP	192.168.1.103	50707	50707	UDP
219	Not Registered	Disconnected	0.0.0.0	0	0	UDP
220	Not Registered	Disconnected	0.0.0.0	0	0	UDP

Tel Number

The *Tel Number* is the contact number of the SIP phone. It is used for SIP station registration and when calls are made to and from the SIP phone.

Current Status

The value shown in the *Current Status* field is either 'Registered' or 'Not Registered'. 'Registered' means that a SIP phone using that station number was able to register with the OfficeServ. It has done this by sending REGISTER messages containing the correct password details to the OfficeServ. 'Not Registered' means that a SIP phone has not registered with the OfficeServ to use that station number, or that a SIP phone has attempted, but failed, to register. It can also mean that the registration has expired for the SIP phone using that number.

Phone Type

SIP stations that are not made by Samsung will show as 'Non Samsung SIPP'.

Public IP address

This field shows the IP address of the SIP phone registered to use this station number. When the SIP phone is unregistered the IP address is 0.0.0.0.

Access Information

Attempts to register with the OfficeServ are recorded in the **Access Information (DM 5.2.21)** table.

5.2.21.Access Information

Index	Date	IP Address	Caller ID	Reason
0	13.4.10 1:4:54	192.168.1.2	217	SIP REGISTER IP/TELNO Invalid
1	13.4.10 1:4:54	192.168.1.2	217	SIP REGISTER IP/TELNO Invalid
2	13.4.10 1:4:55	192.168.1.2	217	SIP REGISTER IP/TELNO Invalid
3	13.4.10 1:4:57	192.168.1.2	217	SIP REGISTER IP/TELNO Invalid
4	13.4.10 1:5:1	192.168.1.2	217	SIP REGISTER IP/TELNO Invalid
5	13.4.10 1:5:5	192.168.1.2	217	SIP REGISTER IP/TELNO Invalid
6	13.4.10 1:5:9	192.168.1.2	217	SIP REGISTER IP/TELNO Invalid

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HOW TO: Quick Guide to configuring SIP Stations

The following process explains how to set up a basic SIP station in the OfficeServ. Please read the detailed command information provided earlier in this document for more information about the settings.

The X-Lite software SIP phone can be used to test the SIP stations. It can be downloaded for free from: http://www.counterpath.com/x-lite.html

Step 1. The OfficeServ must be configured to operate with VoIP phones.

- The IP addresses of the OfficeServ's main processor and OAS/MGI card have already been configured.
- The SIP phone has been installed on a PC and is working properly.

Step 2. The license is configured in **License Key (DM 2.1.4)**
A license key is required to permit Samsung SIP and Non-Samsung SIP phones to work with the Officeserv.

Step 3. Assign a virtual card for SIP Stations in **Virtual Card Change (DM 6.3.2).**
The card type is "**SIP-STN**".

Step 4. Configure a SIP station in **SIP Phone Information (DM 2.7.2).**
The *User ID* and *Password* are the critical values and the password must be at least 6 digits long.

If the white list feature is being used then the network address of the SIP phone needs to be programmed in **Phone IP White List (DM5.13.10)**. If the white list feature is not being used the "Use White List" option should be "Disabled".

Step 5. Use the **SIP Phone Status (DM 6.2.3)** to find out if the SIP phone has successfully registered with the system.

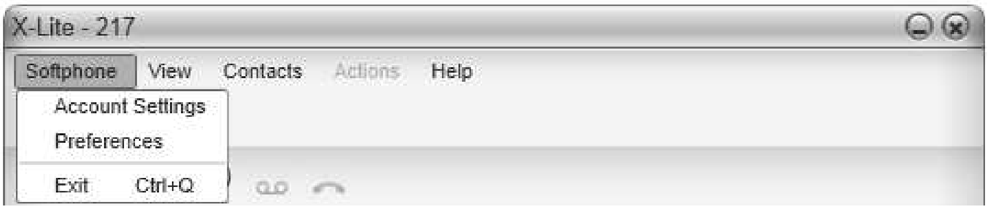
Step 6. Check the *SIP expire time* in **SIP Station Options (DM 5.2.12)**. The SIP phone re-registration time must be set to match this value (600 seconds, by default).

SIP phone programming

Most SIP phones have a large number of settings that may, or may not, be required for that phone to work. However, when configuring SIP phones to work with the Officeserv it is usually sufficient to just program certain key parameters.

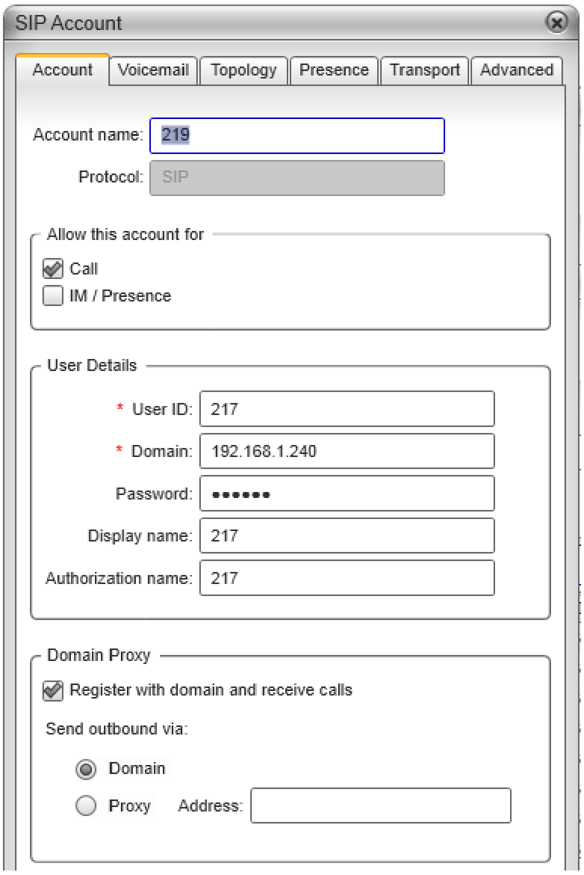
Note: Wireshark can be run on the same PC that is running X-Lite; and this makes it simple to capture, view, and troubleshoot the SIP traffic being sent between the SIP Phone and the OfficeServ.

Go to the account settings

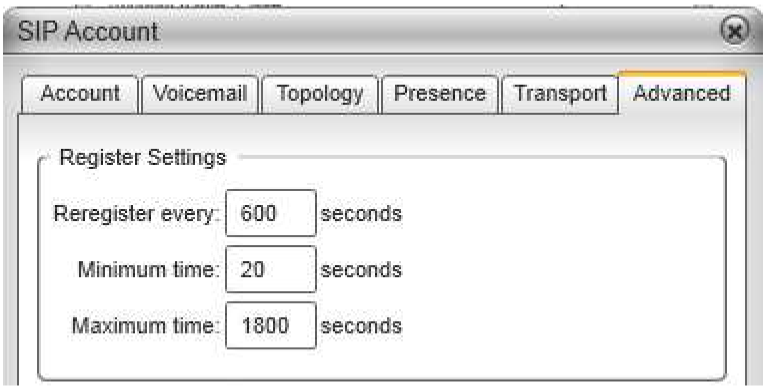


Key parameters in the account screen

X-Lite *User ID* = OfficeServ *Tel Number*
X-Lite *Authorization name* = OfficeServ *User ID*
X-Lite *Password* = OfficeServ *Password*
X-Lite *Domain* = The IP address of the OfficeServ main processor
X-Lite: Enable "Register with domain and receive calls"



In the *Advanced* tab check that the *Reregister* time matches the setting in the OfficeServ.



SIP Station features

The SIP station features supported by the Officeserv are described in detail in the following section. Be aware that although these features are supported by the OfficeServ, they may not be supported by the SIP phone. Use the feature descriptions to determine what features are supported by the SIP phone you are using.

Officeserv supported features – Tables

This chart is a list of the SIP station features supported by the Officeserv. Use this chart to make notes about the feature compatibility of SIP phones that you are testing.

Feature	SIP Phone comments
Registration	
Basic Call	
Hold/Resume	
Music on Hold	
Consultation Call	
Transfer (Consultation)	
Transfer (Blind)	
Call Pickup	
Call Forward (All)	
Call Forward (Busy)	
Call Forward (No Answer)	
Conference	
Call Park (System Hold)	
DND	
Voice Mail Indication	

Use this chart to record the dial access codes for the SIP station features.

Feature Name		Feature Code
Call Forward	Call Forward Clear	600
	All Call Forward	601
	Busy Forward	602
	No Answer Forward	603
	Busy / No Answer Forward	604
	Call Waiting Off	
	Call Back Set	
DND	DND Off	
	DND On	
Call Park	Call Park	
Conference	Conference	
Call Pickup	Call Pickup	

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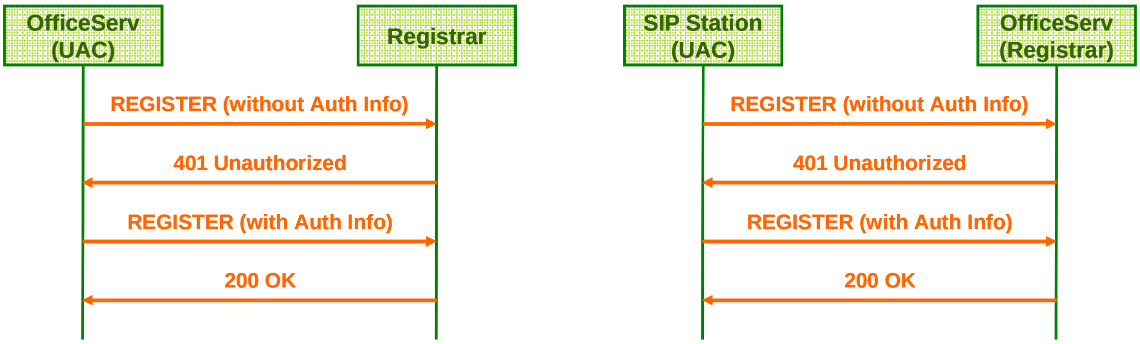
Registration

Registration is the process of one SIP endpoint telling another endpoint that a SIP account resides on the device sending the registration messages. The SIP Phone must register with the Officeserv for the SIP station features to function properly.

The number of SIP stations *licensed* in the Officeserv must be equal, or greater, that the number of SIP phones registering with the Officeserv. In the license screen the 3rd party SIP phones are called 'Non Samsung SIP Phone'.

Registration traffic flow

The diagram below shows the flow of SIP registration messages. This diagram shows that the process of registration of SIP stations is the same as the process used by SIP trunks. *Note: UAC means 'User Agent Client' and is the term used in the SIP RFC documents to describe the SIP endpoint that is initiating a SIP transaction dialog, which in this case is registration.*



The SIP registration process goes through four steps.

1. The UAC sends a REGISTER message without authentication information.
2. The Registrar sends back a '401 Unauthorised' response, which includes information needed by the UAC to perform authentication.
3. Upon receiving the '401 Unauthorised' response the UAC replies with a REGISTER message that contains the authentication information.
4. After confirming the validity of the authentication data contained in the REGISTER message from the UAC the Registrar confirms the UAC's registration.

Basic calls

Basic calls are calls directly to and from SIP stations. SIP phones will only send a call to the Officeserv after all of the digits of the destination number have been dialled. This means that dial tone is provided by the SIP phone, not by the Officeserv.

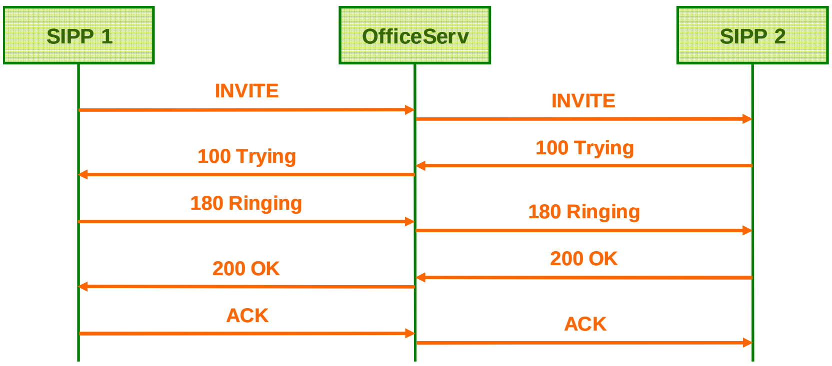
SIP stations use the Officeserv to setup and control calls to other phones and external destinations. Calls made to the SIP phones are also controlled by the Officeserv. *A SIP phone must be registered with the Officeserv before calls will be sent to that phone.*

The path that the voice packets take during a call depends on the type and location of the source and destination devices. Calls between SIP phones and TDM destinations, such as ISDN trunks and digital handsets, will always use MGI ports. Calls between IP devices, such as IP Phones and SIP phones, will try to send voice packets directly between each other and not use MGI ports. If the SIP phones do not support the same codecs, or the OfficeServ believes that they are in separate unconnected networks, it will use the MGI card to relay traffic between them. Note that Multi-party calls, such as conferences, will use the MGI card.

The SIP Station feature in the OfficeServ does not use authentication to verify that the SIP phone making a call is the valid holder of an account. It does check that the CLI of the SIP phone making the call is a registered SIP station, but it does not check that the SIP phone knows the correct password for the SIP account. When the CLI of a SIP phone is not registered in the OfficeServ the OfficeServ will assume that it is receiving an incoming trunk call and it will not allow the phone to use station features.

Basic Call traffic flow

The diagram below shows the SIP message transaction used for a call between two SIP phones, SIPP1 and SIPP2. The SIP phones send all SIP messages to the Officeserv to set up the call, they do not send any SIP messages to each other.



Normally the RTP streams (not shown in the diagram) will be sent directly between SIPP1 and SIPP2, which saves MGI resources. If one of the phones is a Samsung IP phone, and it is on the same network as the SIP phone, the RTP streams will be sent directly between the SIP phone and the IP phone and not use MGI resources.

Note: When a SIP phone calls a destination that is busy the Officeserv will send a '486 Busy Here' SIP message to the SIP phone.

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Hold/Resume

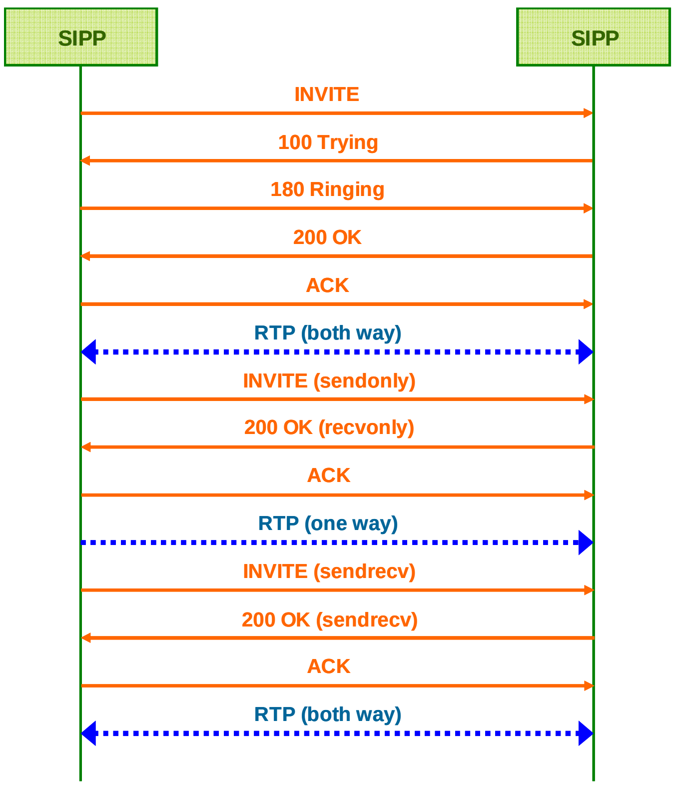
The *Hold/Resume* feature allows a SIP phone to tell the Officeserv to put an active call on hold. It is also used by the Officeserv to tell the SIP phone that the currently active call has been put on hold.

SIP uses a technique called re-INVITE to put a station on hold. A re-INVITE is normal INVITE message that is sent during an active session, but the re-INVITE message contains a field in its Session Description Protocol (SDP) section that specifies that the RTP stream is either: *sendonly*, *sendrecv*, or *recvonly*. SDP is defined in RFC 2327 and describes media attributes for the call, such as: codec type, IP address and port for the RTP stream.

When *sendonly* is specified in a re-INVITE message the endpoint sending the message is telling the other endpoint that it is not listening to the RTP stream, it is 'only sending' it – which effectively puts the call on hold. The endpoint that receives this re-INVITE message will typically respond with a '200 OK' message, which indicates that it is in *recvonly* mode. When the controlling endpoint wants to take the call off hold it sends a re-INVITE message specifying *sendrecv* – this tells the receiving endpoint that it is ready to *resume* the two-way transmission of RTP packets. The endpoint receiving this message will reply with a '200 OK' message indicating that it has returned to *sendrecv* mode and a voice path will be re-established.

Hold/Resume Call traffic flow

The *Hold/Resume* message flow is shown in the diagram below.



Most of the features described in this document need to interrupt the voice stream, and therefore the Hold/Resume technique is used extensively in the SIP message flow.

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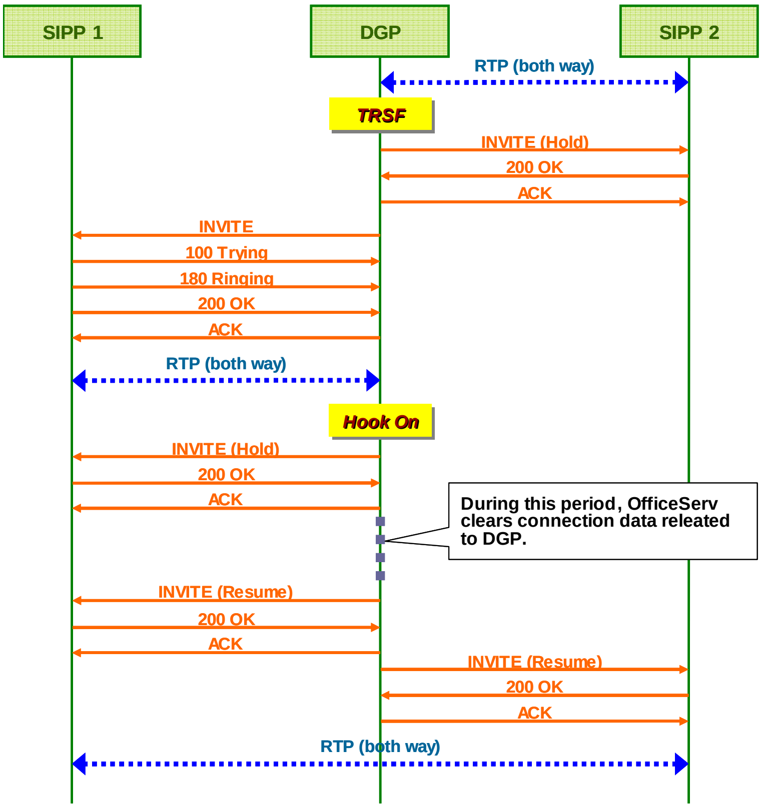
Call Transfer

Call transfer is a complex feature in SIP, and SIP message flows vary considerably depending on which device is performing the transfer. For example, the SIP message flow for a transferred call comprising of two SIP stations and a Digital phone – with the Digital phone performing the transfer, is considerably different to a transferred call comprising of three SIP phones – with a SIP phone performing the transfer. So this document will not try to describe all possibilities, but it will show the SIP message flow for a call transfer by a digital phone between two SIP phones. For information about SIP phones performing transfers see the **Transfer Using Eyebeam Phones** section of this document.

Consultation transfer

A consultation transfer is performed when a station completes the transfer of a call after conversing with the party that the call is being transferred to.

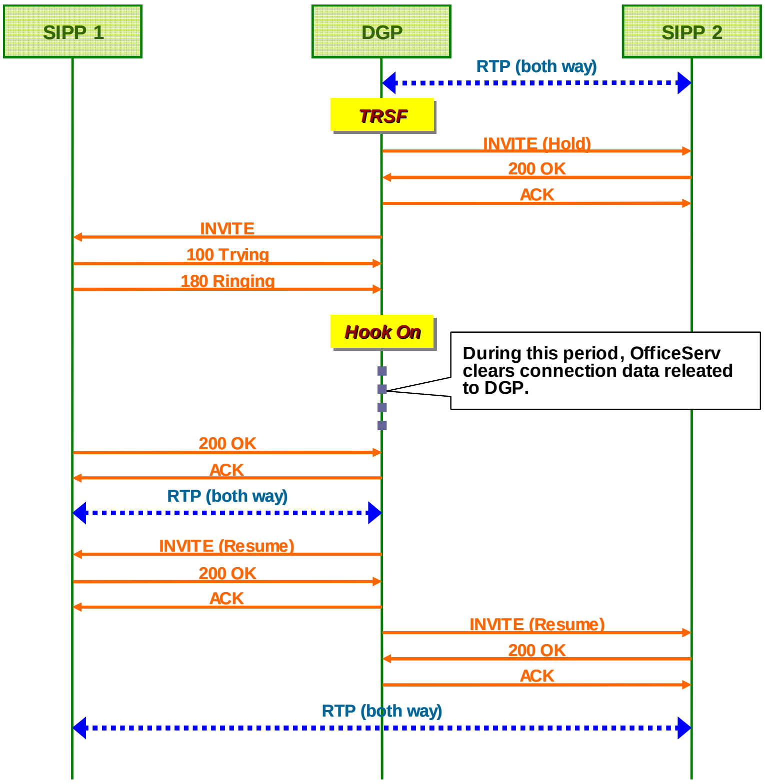


This scenario starts with a Digital Phone (DGP) and a SIP Phone (SIPP2) already in an active conversation, and the following steps show the SIP messages sent when the DGP performs a consultation transfer.

1. The DGP initiates transfer process by pushing TRSF button: this puts SIPP 2 in the hold state.
2. The DGP makes a second call which sets up a SIP session with SIPP1.
3. After the DGP hangs up the Officeserv clears all connection data related to the DGP because it is no longer part of the call.
4. After the DGP hangs up the Officeserv resumes each held SIP session, changing the RTP destinations so that SIPP 1 and SIPP2 can communicate directly with each other without using the MGI.

Blind Transfer

A blind transfer occurs when the phone transferring the call hangs up before the phone receiving the transfer answers the call. In the flow diagram below the Digital Phone (DGP) is performing the transfer.



The following steps are performed during a blind transfer.

1. The Digital Phone (DGP) initiates the transfer process by pushing the TRSF button: this puts SIP Phone 2 (SIPP 2) in the hold state.
2. The DGP makes the second call.
3. After the DGP hangs up the Officeserv clears all the connection data related to the DGP because it is no longer part of the call.
4. The Officeserv resumes each held SIP session, changing the SIP terminal's RTP destination so that SIPP 1 and SIPP 2 can communicate directly each other without using the MGI.

The main difference between a blind transfer and a consultation transfer is that the transferor hangs up in the ringback state when performing a blind transfer. From the SIP phone's perspective the SIP message flow and its mechanisms are almost the same as a consultation transfer.

There is one important point for blind transfers: whoever becomes the transferor (i.e., standing in the middle in the call flow) 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 the Officeserv cannot process the subsequent blind transfer properly unless this condition is met.

Lab test results have found that some SIP terminals do not send a hold message and the result is that the Officeserv fails to execute the Blind Transfer.

Consultation Call

A consultation call is basically the first part of a consultation transfer; the difference is that the station transferring the call does not hang up after the call is answered by the phone receiving the transfer. Instead of hanging up, the transferring phone toggles between the other two calls in the conversation by repetitively pressing the TRANSFER key.

Dialling feature codes from SIP phones

SIP is designed to set up sessions between endpoints. For example, calls between SIP phones are SIP sessions. Dialled feature codes are not valid destination numbers; therefore, when the Officeserv receives an INVITE message containing a feature code instead of a phone number, it cannot process the INVITE in the same way as a normal session.

When the Officeserv receives a feature activation code it will respond with a '*480 Temporarily Unavailable'* SIP message. It does not matter if the feature activation succeeds or fails, this message is always sent.

Call Forward

There are four types of Call Forward that can be set on SIP stations, All Call Forward, Busy Forward, No Answer Forward and Busy/No Answer Forward. A Call Forward cancellation can also be performed.

Call forwards are set by the SIP phone dialling the feature activation code and the destination number. The default feature activation codes are:

600 Call Forward Clear
601 Call Forward All
602 Call Forward Busy
603 Call Forward No Answer
604 Call Forward Busy/No Answer

Note: The response from the Officeserv to the activation/deactivation of this feature will be '*480 Temporarily Unavailable*', please read the ***Dialling feature codes from SIP phones*** description for details.

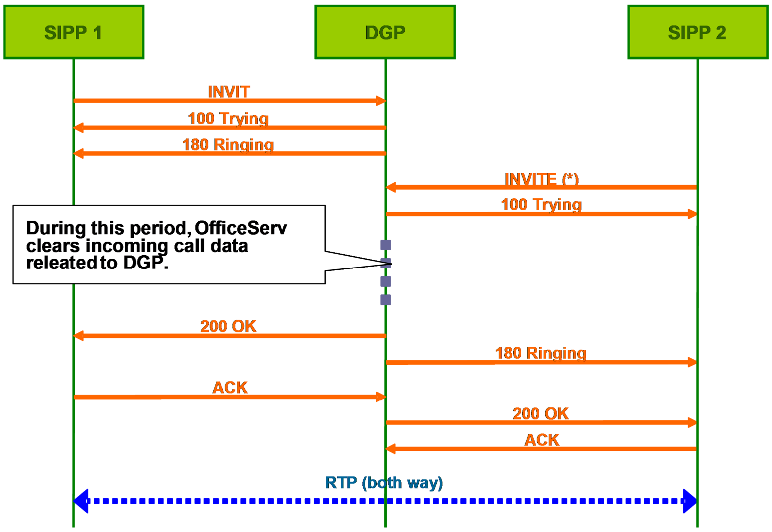
Call Pickup

Call Pickup allows a SIP station to answer a call on behalf of a ringing station. The three types of pickup supported by the SIP stations are *My Group Pickup*, *Group Pickup* and *Directed Pickup*.

My Group Pickup allows a member of a pickup group to answer a call on behalf of another member of the same pickup group. All SIP phones are in the same pickup group by default. There is no default feature access code for MYGPICK in the number plan, so it must be assigned before the feature can be used.

Group Pickup allows a station in any group to answer a call ringing on a station in any other group.

Directed Pickup allows a station to pick up a call ringing on another station by dialling the direct pickup code and the number of the ringing station.



The diagram above shows the flow of SIP messages when a SIP station calls a DGP, and then another SIP phone picks up that call using the *My Group Pickup* feature. SIPP1, DGP and SIPP2 are all in the same pickup group.

1. SIP station (SIPP1) calls the Digital Phone (DGP).
2. The DGP rings and SIP station SIPP1 is in the ring back state,
3. Another SIP station (SIPP2) dials the *Call Pickup* feature code.
4. The original call is picked up by SIPP2 and a connection is made between the DGP and SIPP2.

Conference

Of all the SIP supplementary services implemented in the Officeserv the *Conference* feature may be the most complicated feature in both usage and in its internal mechanisms.

When using the conferencing feature, all stations, including SIP stations, should use the MGI for handling RTP packets instead of their own conferencing DSP. This is because the Officeserv gathers all the RTP packets being transmitted during the conference onto a single conference chip in the MGI by repeatedly using SIP Hold/Resumes. A maximum of five members in a conference are allowed.

The *conference* feature access code is '46' by default and, like any other supplementary feature access code, this can be changed. When a SIP phone dials the conference feature code the Officeserv responds with a '*480 Temporarily Unavailable*' message to indicate that the feature has been accepted.

The SIP phone setting up the conference must be able to make calls on multiple line buttons. This is essential because the phone dials the conference feature code and a station number without taking the initial call off hold. The conference setup will fail if the SIP phone has only a single line. Multiple line buttons do not require multiple SIP station accounts in the Officeserv, as they all make calls using the same account number.

Using the conference feature

The operation of the conference feature follows the steps described below. SIP station SIPP1 is the initiator and controller of the conference, SIP station SIPP2 is the first member station and SIP station SIPP3 is the second member station. The X-lite Eyebeam SIP phone was used as the conference initiator - SIPP 1.

1. SIPP1 has called SIPP2 and the call is in progress.
2. SIPP1 puts the call on hold by pressing HOLD.
3. SIPP1 dials the conference feature access code '46' and presses the DIAL button.
4. SIPP1 waits for the "*480 Temporarily Unavailable"* response from the Officeserv and then presses the HANG UP button.
   SIPP 2 is now in the conference state waiting for more members to arrive
5. SIPP1 dials the number of SIPP3 and presses the DIAL button.
6. SIPP3 answers the call.
7. SIPP1 puts the call to SIPP3 on hold by pressing HOLD.
8. SIPP1 dials the conference feature access code '46' and presses the DIAL button.
9. SIPP1 waits for the "*480 Temporarily Unavailable"* response from the Officeserv and then presses the HANG UP button.
   SIPP 2 and SIPP3 are now in the conference state waiting for more members to arrive
10. SIPP1 dials the conference feature access code '46' and presses the DIAL button.
11. SIPP1 waits for the "*480 Temporarily Unavailable"* response from the Officeserv and then presses the HANG UP button.
12. SIPP1 takes the call off hold.
    SIPP1, SIPP2 and SIPP3 are now in a conference

The key point in understanding how the SIP station conference feature works is to know what actions are taken by the Officeserv when the conference feature code is dialled. The usage of the conference feature code is similar to the usage of the ‘CONF’ soft key on Samsung Digital phones.

The following description explains more about the operation of the conference feature.

1. Conferences are be made by a station in an active session putting the call on hold and then dialling the conference feature code. Whenever a phone wants to make a new conference, or add a new member to the conference group, it must first send a Re-Invite message to put the current session in the *Hold* state. It then sends the conference feature code in the form of a normal INVITE message. The Officeserv responds to the feature code with a ‘*480 Temporarily Unavailable*’ message and the conferencing phone goes back in to the IDLE state. It is then ready to take the next action.

2. As for the other members of the conference: when the conference feature code is received from the station initiating the conference the Officeserv puts any stations already involved in the call into a single conference session through the MGI. This is done to ensure that they send/receive RTP packets to each other via the MGI, instead of by direct transmission to each other.

3. At this point the station setting up the conference has two options: A) Make another call to invite another member into the conference. B) Stop adding members. When the conferencing station chooses option A it invites another member station to the conference by sending out an INVITE message to the Officeserv, and then it goes back to step 1 again. However, when it chooses option B it simply needs to send the conference feature code again without taking any other action.

4. When the Officeserv receives consecutive conference feature codes from the station that is setting up the conference it knows that the owner station does not want to invite additional conference members. The Officeserv sends session resume messages to all stations in the conference so that they can communicate with each other via the MGI.

5. Finally, the station setting up the conference MUST send a session resume message to actually join in the conference and communicate with other stations. This is because the owner station has set its session into hold mode while setting up the conference, which disabled its RTP transmission. To resume the session it needs to send a final Re-Invite message to the Officeserv, which will then connect the RTP stream into the conference by sending a 200 OK response to tell the SIP phone that the IP address of the MGI card is the destination for RTP packets. Without resuming the hold session the owner station cannot listen nor speak to the conference, even though it is in the conference room.

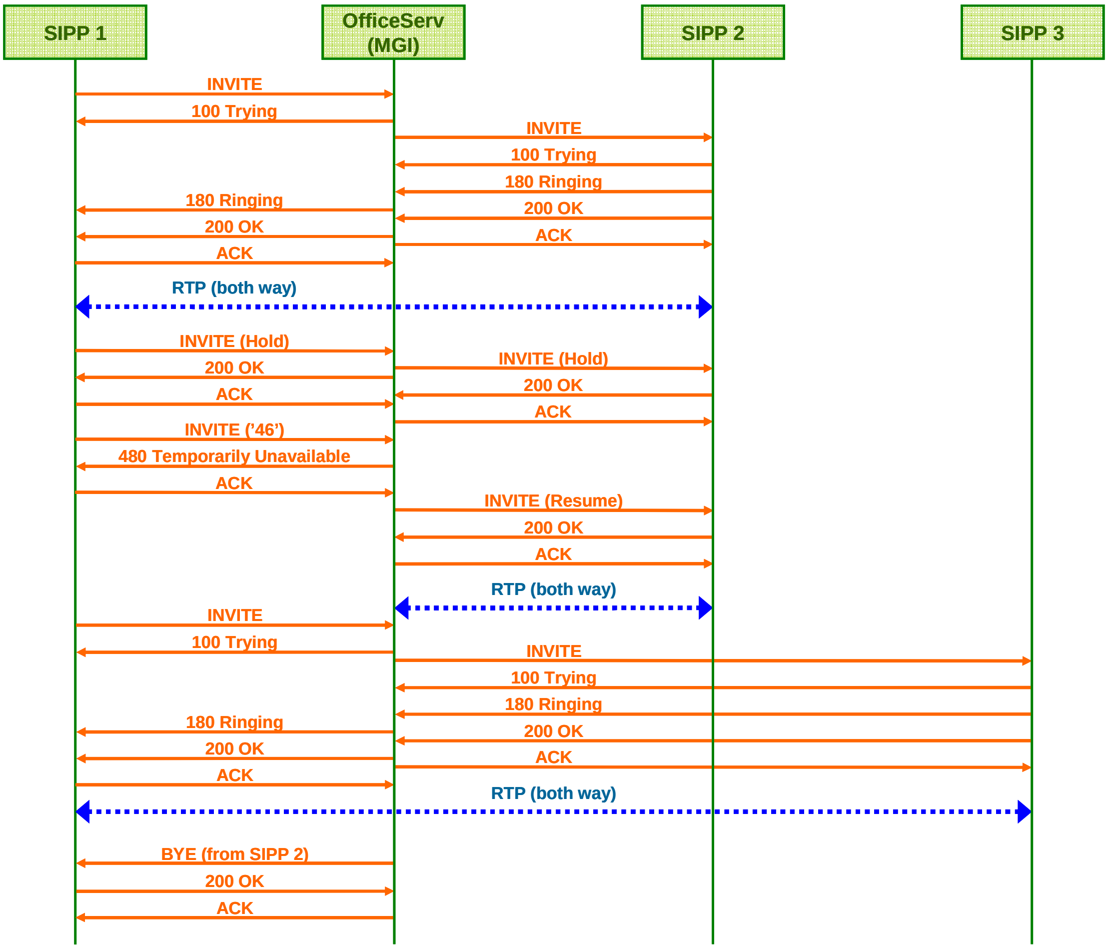
The next section shows the messages sent and received by the SIP phone and the Officeserv during the setup and management of the conference. Reading this section will help make the above description easier to understand.

Conferencing feature - SIP Message Flow

The notes and flow charts below show the SIP messages exchanged when a conference call is set up. There are two parts to the diagram: *Conference feature SIP messages – figure 1,* and *Conference feature SIP messages – figure 2*.

As shown in *Conference feature SIP messages – figure 1* below, SIPP 1 calls SIPP 2 through the Officeserv and establishes a two-way conversation. The RTP stream is sent directly between SIPP 1 and SIPP 2.

Conference feature SIP messages – figure 1



SIPP 1 starts the process of setting up the conference by sending an INVITE (hold) message, which is sometimes referred to as a Re-Invite. The INVITE (hold) message is relayed through the Officeserv to SIPP 2 and the user of SIPP2 will hear hold tone. An INVITE (hold) message is usually generated by the SIP Phone user pressing the *Transfer* button.

The owner of the conference, which is SIPP 1, then sends the conference feature access code, '46', to the Officeserv.

Note that there must be an INVITE (hold) before the INVITE '46' message, as the Officeserv always has to put a conference member into the hold state before actually adding it into the conference. Some SIP terminals will automatically put the current session in hold mode before they send out the INVITE message '46', but most do not. In all cases, the call needs to be put on hold before the conference feature code is sent.

The conference feature code '46' gives a conference signal to the Officeserv so that the Officeserv can take the necessary actions needed to set up a conference, such as modifying RTP transmissions using Re-INVITE messages.

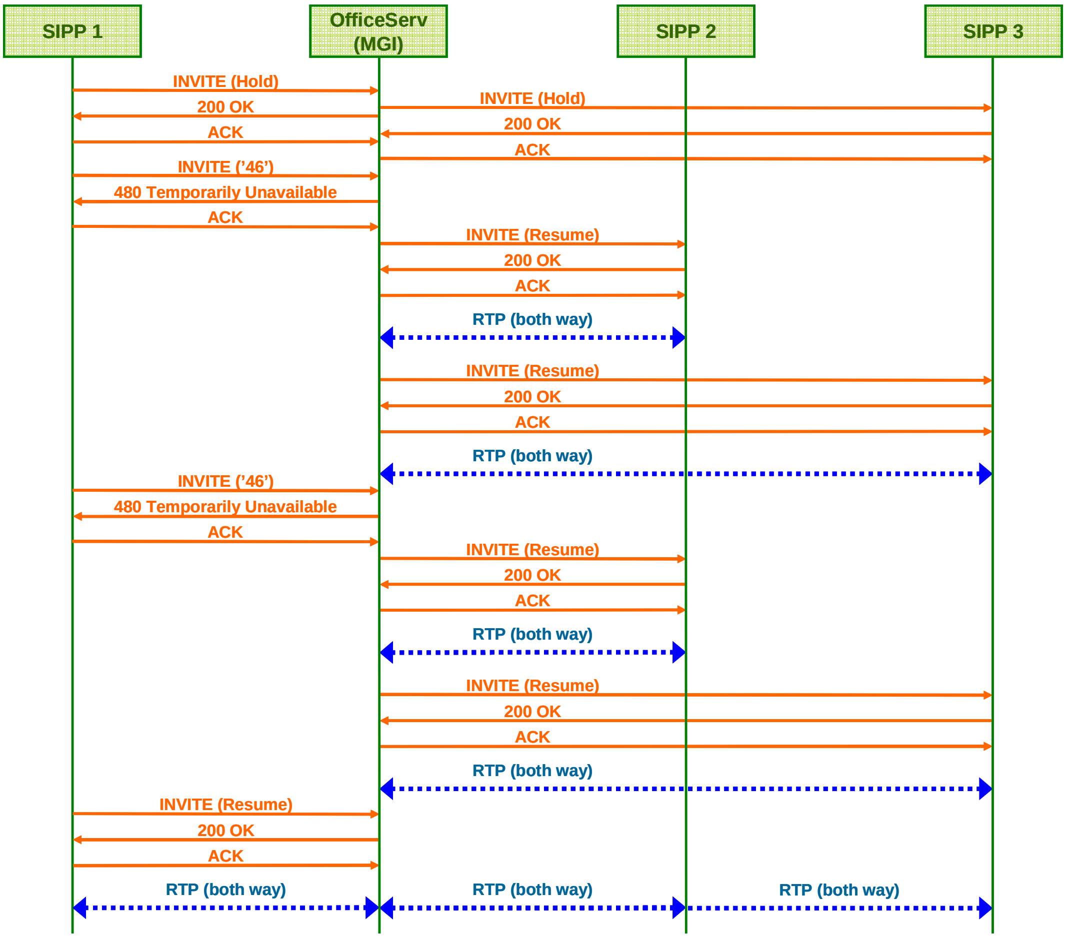
Immediately after the Officeserv receives the conference feature access code the session with SIPP 2 is resumed and the RTP packets are sent between SIPP 2 and the MGI card; instead of between SIPP 1 and SIPP 2. The conference owner, SIPP 1, is now in an idle state and can make a call to invite another member to the conference, which in this example is SIPP 3.

When the session between SIPP 1and SIPP 3 is established the Officeserv clears the session between SIPP 1 and SIPP 2 because this session is no longer useful. And another important reason for clearing this session is that many SIP terminals support a maximum of two concurrent connections, so leaving the original connection active will prevent SIPP 1's sending out the next '46' conference feature code, which may lead to a failure in further conference processing.

Now SIPP 1 is in an active session with SIPP 3, and the session with SIPP 2 is gone. Meanwhile, SIPP 2 is put into a session with the Officeserv MGI card.

The exchange of SIP messages for the remainder of this description is shown in the *Conference feature SIP messages – figure 2 diagram* below.

Conference feature SIP messages – figure 2



To put SIPP 3 into the conference session, which SIPP 2 is already in, SIPP 1 puts the call to SIPP 3 on hold and then sends the conference feature code. Note that before sending out the conference feature code SIPP 1 puts its session with SIPP 3 into hold mode. After SIPP 1 puts the call to SIPP 3 on hold it goes back to the IDLE state so that it can choose to either end adding conference members or to add an additional member.

At this point the Officeserv modifies the RTP session for SIPP 3 to change the destination of the RTP stream to the MGI card so SIPP 3 can be joined into the conference.

As SIPP 1 chooses to stop adding additional members to the conference it sends the conference feature code again. When the Officeserv receives consecutive conference feature codes from the conferencing station it sends the Re-INVITE (resume) message to all terminals, including the conferencing station, to change the destination of RTP packets to the MGI card. For the stations that are already in the conference and directing their RTP packets to the MGI card this Re-INVITE message may be redundant, but should not cause any problems.

Finally, in order to actually communicate in the conference the conferencing station MUST resume the session which was held just before sending the conference feature code. In some cases conferencing SIP will not send the last INVITE (resume) message, therefore it is important to not forget it.

Call Park (System Hold)

The Call Park (System Hold) feature is used by a station to put a call into a hold state where it can be retrieved by itself or another station. The Call Park feature is activated by the SIP station using the feature code and can only be applied to a single call.

Note operation of this feature depends on hold/transfer functionality of the SIP phone. Some SIP phones do not support the SIP signalling methods used by this feature.

The feature names in the Officeserv number plan are HOLD and HOLDPK. HOLD is used to put the call on hold and HOLDPK is used to retrieve the call from hold. The hold park function is enabled for station to station calls in *DM Menu 5.14.1 Transfer/Recall/Pickup Options → Pickup Held Station* and in KMMC: MMC 210 – STNHOLD PICK

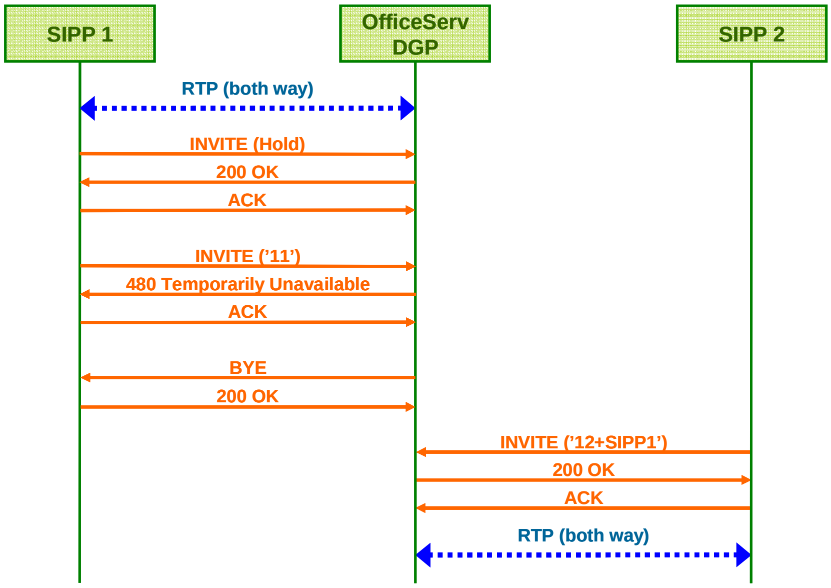
The default feature access codes are:

HOLD	11
HOLDPK	12

To park a call the user puts the call on hold (or transfer) and then dials '11' and the station number. The station retrieving the call dials '12' and then the number of the station that parked the call, for example: '12220'.

The following example and diagram shows the SIP messages exchanged when a SIP Phone (SIPP1) puts a Digital Phone (DGP) in Call Park, and then another SIP station (SIPP2) retrieves the call.

1. SIPP1 is in a conversation with DGP.
2. SIPP1 presses Hold
3. SIPP1 dials '11' and waits for the response from the system (*480 Temp unavailable*)
   The DGP is now in the Call Park state and SIPP1 is IDLE
4. SIPP2 goes off-hook and dials '12' and the station number of SIPP1
   The call has been retrieved, so DGP and SIPP2 are in conversation.



If the call is not retrieved it recalls to the parking station. The wait time is set in DM 5.14.1 *Transfer/Recall/Pickup Options* → *E-Hold Recall time* expires.

DND (Do Not Disturb)

When a SIP station registers the *Do Not Disturb* (DND) feature the Officeserv will block any calls going to that station.

When a SIP station is in Do Not Disturb mode it does not receive any messages and therefore does not have to send any responses.

Calls to SIP stations in DND mode get a ‘Busy’ response and the display on IP Phones and Digital phones will show ‘DND’.

There is no indication on the SIP phone that it is in DND mode, so the user must remember to de-activate it when it is no longer required. Otherwise they will not receive any calls and will have no visible reason why they are not getting them.

The DND feature is manually set and unset by the station user dialling the DND feature access code followed by either a ‘1’ to enable it, or a ‘0’ to disable it.

The feature name for *Do Not Disturb* in the system’s number plan is DND and the default access code is ‘40’. The user dials ‘401’ to activate DND on their phone and ‘400’ to de-activate it. If necessary, the access code can be changed to another number.

The steps to set and unset the DND feature are described below.

1. To activate the DND feature the SIP station user goes off-hook and dials ‘400’.
2. The OfficeServ responds with the message “480 Temporarily Unavailable”.
3. To de-activate the DND feature the SIP station user goes off-hook and dials ‘401’.
4. The OfficeServ responds with the message “480 Temporarily Unavailable”.

MWI (Message Waiting Indication)

The Message Waiting Indication (MWI) feature is implemented so that the Voice Mail System (VMS) in the Officeserv can notify a SIP Phone that a voice message is waiting for them. After the VMS records a new message in a mail box it notifies the SIP Station linked to that mailbox. The MWI feature is an inbuilt function of the OfficeServ and therefore no feature access code is required.

The MWI message sent to the SIP station uses the standard SIP NOTIFY method and is compatible with the MWI format used by many standard SIP phones (i.e. Cisco 7960).

Most SIP stations will give a visual notification when a voice message is waiting for them. SIP Stations that do not support MWI notification may appear not to react to the MWI message.

When the SIP user has listened to the voice message the VMS will clear the MWI indication by sending a SIP NOTIFY message to the SIP phone.