Application Notes for Configuring SIP Trunking between the Skype SIP Service and an Avaya IP Office 6.1 Telephony Solution
Issue 1.0

Abstract

These Application Notes describe the steps to configure trunking using the Session Initiation Protocol (SIP) between the Skype SIP Service and Avaya IP Office. The Avaya solution consists of Avaya IP Office and various Telephones.

Table of Contents

1.0 Introduction

These Application Notes describe the procedure for configuring Session Initiation Protocol (SIP) trunking between the Skype SIP trunking network and an Avaya SIP telephony solution consisting of Avaya IP Office and Avaya telephones. Avaya IP, digital and analog telephones can be used. The information provided is specific for the SIP trunk inter-working, unrelated configuration is not considered within this guide. This document assumes that the installer has undergone Avaya approved training and has a working knowledge of IP Office installations.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 is the primary specification governing this protocol. In the configuration described in these Application Notes, SIP is used as the signaling protocol between the Avaya components and the network service offered by Skype. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

Note that FAX transmission is not supported by Skype.

1.1 Document Change History

Date	Version	Summary of Changes
December 8, 2010	0.1	Original publication
January 10, 2011	1.0	Document issue.

2.0 System Software / Loadware

To achieve successful interoperability between the IP Office and Skype SIP Trunking, the various network elements must be running the version of software as shown below:

System	Platform	Firmware
IP Office 500	All platforms	GA Release 6.1.5
IP Office Phones	All supported phones	As provided by Release 6.1.5

Table 1 Validated Equipment and Software

3.0 Features

3.1 Features Supported

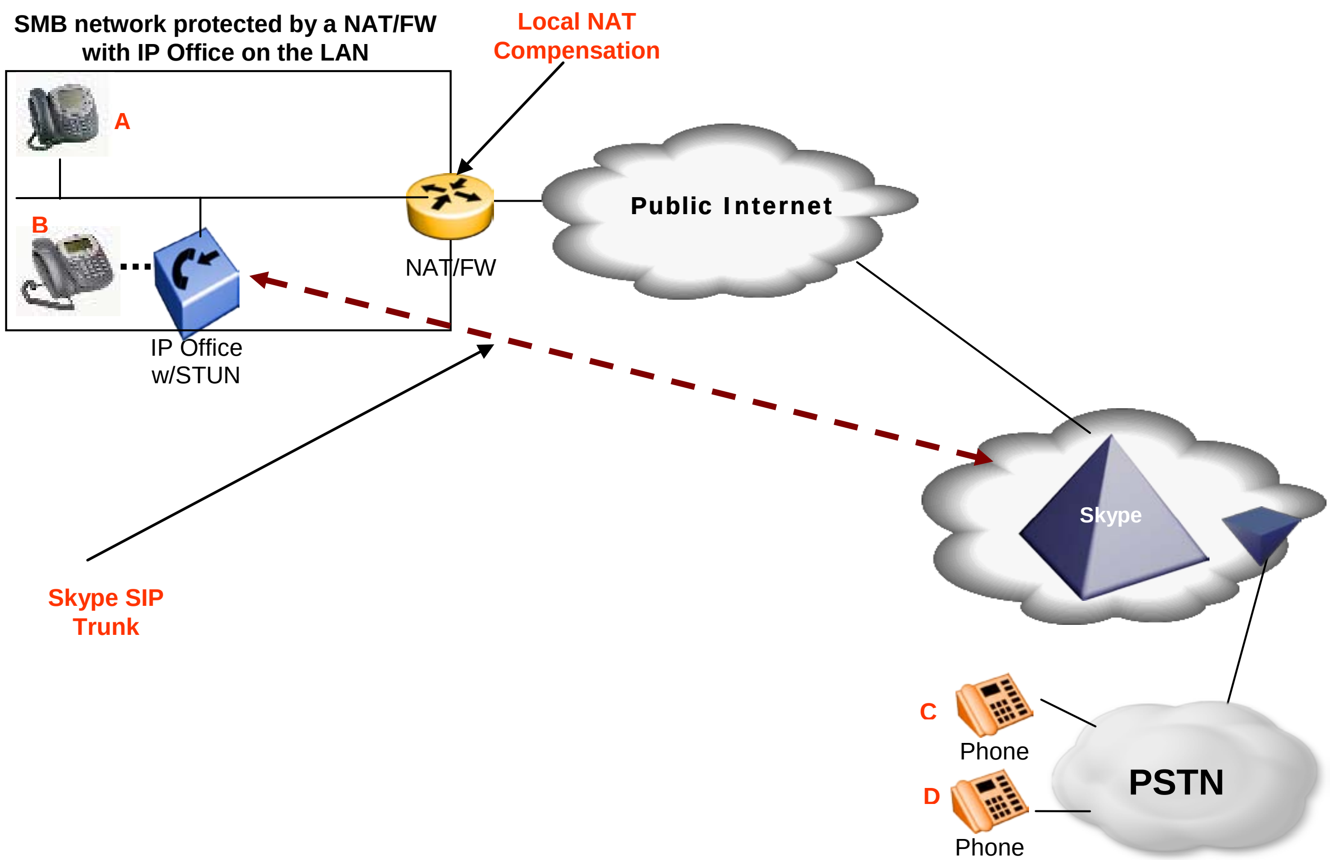
- Domain Name Resolution
- Provisioning and Skype Manager
- Basic Skype Calls
- Basic PSTN Calls
- DTMF (RFC 2833)
- Dial Plan
- Toll Free Calls
- Voice Mail and MWI
- Early Media and Codec Negotiation
- Call Forward (All, Busy, No Answer)
- Call Park and Pickup
- Blind Transfer
- Assisted Transfer
- Conference (Three Way Adhoc Conference)
- Caller ID Presentation
- Extension Mapping

3.2 Technical Caveats

1. The IP Office 6.1.55304 Critical Patch is required if the IP Office is configured with a DNS service in Manager > System Settings > DNS > DNS Service IP Address. The patch is required to resolve the sip.skype.com ITSP Domain Name. The critical patch is not required if the 1Q11 Service Pack is installed.
2. FAX is not supported by Skype.
3. Calls from Skype users appear on IPOffice with no CLID and SkypeID as a Display Name in the cases below. Avaya tracking: MRDB00041005.
    a. Skype user does not have an Online Number
    b. Skype User does not have a configured Caller ID
4. The Windows Skype client version 4.2.0.158 displays 'Failed Call' when calling an IP Office set that is busy.
5. Skype does not support Early Media in SIP.
6. IPOffice CLIR (Caller ID Restriction) feature of IP Office is not compatible with Skype SIPConnect service – all outgoing calls with CLIR fail. Avaya tracking: MRDB00041191.

3.3 Reference Configuration

The following diagram illustrates the configuration:



The following endpoints were used for testing:

Endpoint	Type	Extension	Number
A	4621SW IP	250	250
B	5410	201	201
C	PSTN	--	6137123456
D	PSTN	---	6137123457
Skype User ID	---	--	99051000111234
Skype User incoming phone number	---	--	19785281234
Skype User business account extension	---	--	777

3.4 IP Office Configuration

All configuration steps for Avaya IP Office were performed using the IP Office Manager application. This application presents the administrator with a hierarchy of icons for the various components which can be configured.

In order to configure a SIP trunk between IP Office and Skype, do the following:

3.4.1 Licenses

A license is required for SIP Trunk Channels, which can be confirmed by selecting the “License” icon in Manager.

3.4.2 System

Select the “System” icon in Manager and enter the parameters shown in the following table:

Tab	Parameter	Usage
System	Name	Enter a name to be assigned to IP Office for identification purposes.
	Locale	Select the locale for the installation from the drop-down menu.
LAN1 – LAN Settings	IP Address	Enter the IP address assigned to IP Office.
	IP Mask	Enter the network mask assigned to IP Office.
LAN1 – Network Topology	Binding Refresh Time	Enter 30 seconds for the IP Office to send recurring ‘SIP Options requests’ to the remote proxy terminating the trunk. Those requests will keep the port open through the firewall.
	Run STUN on startup	Select if the IP Office is behind a NAT/FW and IP Office is going to be doing Local NAT compensation. STUN (Simple Traversal of UDP through NAT) is a mechanism used with UDP SIP to overcome the effect of NAT firewalls.
	Run STUN	Click the Run STUN button to test STUN operation if the IP Office is behind a NAT/FW and IP Office is going to be doing Local NAT compensation. STUN (Simple Traversal of UDP through NAT) is a

		mechanism used with UDP SIP to overcome the effect of NAT firewalls.
DNS	DNS Service IP Address	This is the IP address of a DNS Server.

3.4.3 IP Route

Select the “IP Route” icon in Manager and create a default route with the parameters shown in the following table:

Parameter	Usage
IP Address	Enter “0.0.0.0”.
IP Mask	Enter “0.0.0.0”.
Gateway IP Address	Enter the IP address of the default router.
Destination	Select “LAN1” from the drop-down list.

3.4.4 SIP Lines

Select the “Line” icon shown in Manager and right click to add a new SIP Line for the SIP trunk with the parameters shown in the following table.

A SIP URI must bee added for each:

1. Skype SIP User.
2. Each incoming call phone number of the Skype SIP User.
3. Each Skype business account with an extension number for incoming calls that is associated with the Skype SIP User.

Tab	Parameter	Usage
SIP Line	Line Number	This value is assigned automatically when the line is created.
	ITSP Domain Name	Set this field to: sip.skype.com
	International Prefix	This prefix is added to calls identified as not being national.
Transport	ITSP Proxy Address	Leave blank if DNS is configured in System Settings > DNS to resolve the ITSP Domain Name. Otherwise enter IP address: 63.209.144.201
SIP Credentials	User name	Add a New SIP Credential where User name for Skype registration authentication is the Skype SIP User like 99051000111234

	Authentication name	The Authentication name should match the User name above.
	Contact	The Authentication name should match the User name above.
	Password	Enter the Skype SIP User password provided for the registration authentication.
	Registration Required	Leave selected for registration before making calls.
SIP URI- Add a SIP URI for the Skype SIP User.	Local URI	Select “Use Credentials User Name”.
	Contact	Select “Use Credentials User Name”.
	Display Name	Select “Use Credentials User Name”.
	Registration	Select the account credentials configured on the line's SIP Credentials tab.
	Incoming Group	Enter the number used in SIP Line - Line Number which was assigned automatically when the line is created.
	Outgoing Group	Enter the same number selected for “Incoming Group” for this SIP URI.
SIP URI- Add a SIP URI for each incoming call phone number of the Skype SIP User.	Local URI	Enter the incoming call phone number of the Skype SIP User like: 19785281234
	Contact	Enter the same number entered for Local URI.
	Display Name	Enter the same number entered for Local URI.
	Registration	Select “0: <None>”.
	Incoming Group	Enter a number different to that used in by other SIP URI – Incoming Groups. The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls.
	Outgoing Group	Enter the same number selected for “Incoming Group” for this SIP URI.
SIP URI- Add a SIP URI for each Skype business account with an extension number for incoming calls that is associated with the Skype SIP	Local URI	Enter the extension number for incoming calls of the Skype business account associated with the Skype SIP User.

User.	Contact	Enter the same number entered for Local URI.
	Display Name	Enter the same number entered for Local URI.
	Registration	Select “0: <None>”.
	Incoming Group	Enter a number different to that used in by other SIP URI – Incoming Groups. The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls.
	Outgoing Group	Enter the same number selected for “Incoming Group” for this SIP URI.
VOIP	Compression Mode	This field can be changed to define the compression mode (codec) or modes offered during call setup. The Skype supported codecs are G.711 ALAW 64K, G.711 ULAW 64K, G.729(a) 8K CS-ACELP.
	Re-invite Supported	Select so Re-Invite can be used during a session to change the characteristics of the session.
	Fax Transport Support	Ensure Fax Transport Support is not selected.

3.4.1 Call Routing

3.4.1.1 Outgoing Call Routing

Select the “Short Code” icon shown in Manager to create a shortcode to route outgoing calls from Avaya IP Office to the Skype Network. The IP Office user will dial a digit followed by the phone number N to be called. Create a new shortcode with the values shown in the following table:

Parameter	Usage
Code	Enter: <digit>N; For example: 8N;
Feature	Select “Dial” from the drop-down menu.
Telephone Number	Enter N
Line Group Id	Select the Line Group Id from the drop-down list that matches the SIP URI – Outgoing Group of the Skype SIP User.

3.4.1.1 Incoming Call Routing

Select the “Incoming Call Route” icon shown in Manager; create a new incoming call route with the values shown in the following table for each SIP URI:

1. Skype SIP User.
2. Each incoming call phone number of the Skype SIP User.
3. Each Skype business account with an extension number for incoming calls that is associated with the Skype SIP User.

Tab	Parameter	Usage
Standard	Line Group Id	Select the Line Group Id from the drop-down list that matches the SIP URI – Incoming Group of the Skype SIP User.
	Incoming Number	Leave the blank default.
Destinations	Destination	Either enter the destination manually or select the destination for the call from the drop-down list.

4.0 Verification Steps

The correct configuration of the system can be verified by performing the following steps:

1. Verify that the local extensions on Avaya IP Office can call and talk to each other.
2. Verify that the local extensions on Avaya IP Office and the telephones attached to the PSTN can call each other.
3. Verify that if the Skype SIP User has an incoming call phone number that the PSTN can reach the expected Avaya IP Office extension.
4. Verify that if the Skype SIP User has a Skype business account with an extension number for incoming calls that Skype users can call the Skype business account to reach the expected Avaya IP Office extension.
5. Verify that if the Skype SIP User has a Skype business account with no extension number for incoming calls that Skype users can call the Skype business account to reach the expected Avaya IP Office extension.

5.0 Conclusion

These Application Notes contain instructions for configuring Avaya IP Office to connect to the Skype SIP network via a SIP trunk. All test cases passed with exception noted in section: Technical Caveats.

5.1 Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com.