



Avaya Solution & Interoperability Test Lab

Application Notes for Konftel 300IP with Avaya Communication Manager - Issue 1.0

Abstract

These Application Notes document compliance testing the Konftel 300IP with Avaya IP and digital telephones controlled by Avaya Communication Manager. The Konftel 300IP communicates with Avaya Communication Manager via its connection to the Avaya SIP Enablement Services platform and enables meeting or conference participants to simultaneously participate in a telephone conversation.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

The purpose of these Application Notes is to illustrate how the Konftel 300IP can be used within a telephone system consisting of Avaya IP and digital telephones controlled by Avaya Communication Manager. The Konftel 300IP functions as a SIP phone with two accounts which serve as separate line appearances, and contains a microphone and loudspeaker, which effectively extends the range from which the unit can be used to include an area of 30 square meters. Placed within a conference room, the Konftel 300IP enables all of the participants in the room to take part in a telephone conversation. The unit also performs echo cancellation to avoid feedback problems that might otherwise occur.

The Konftel 300IP has a keypad/display, shown in the figure below, which serves as a telephone keypad, as well as providing additional functions.



Figure 1: Konftel 300IP Keypad /Display

This document details the configuration used for compliance testing with Konftel 300IP with Avaya Communication Manager. The diagram below depicts the configuration used for compliance testing.

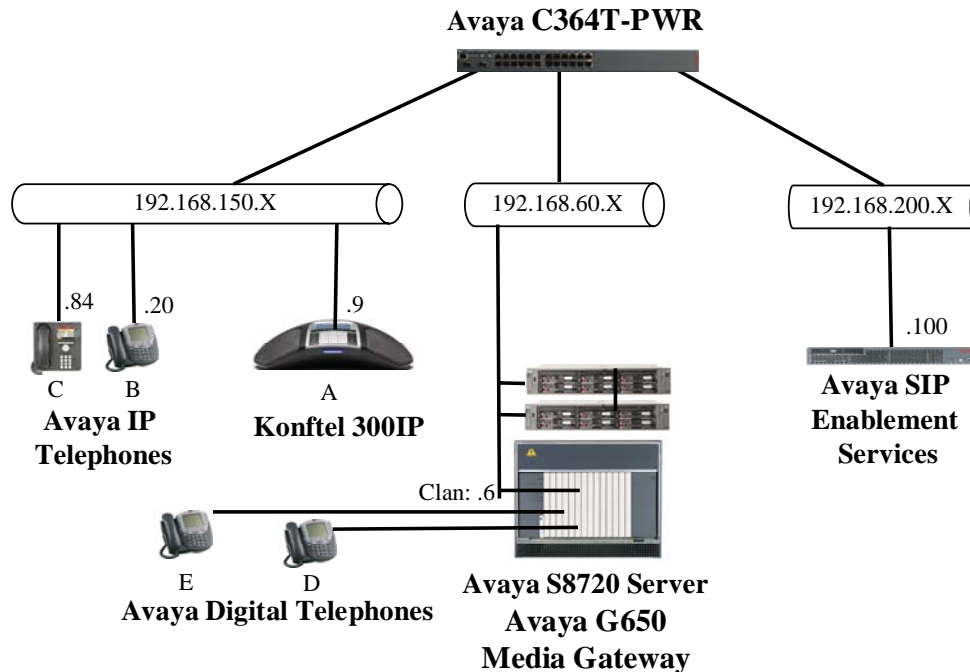


Figure 2: Test Configuration

The configuration that was used for testing consists of an Avaya G650 Media Gateway, and an Avaya S8720 Server. The Avaya telephones and the Konftel 300IP were located at physically separate locations to ensure that sound from the test location could not be heard other than via the telephone connection. Note that the Konftel 300IP was able to operate solely from the power that it received from the Avaya C364T-PWR Ethernet switch to which it was attached. The unit is also shipped with a power supply which can be used if Power over Ethernet is unavailable.

The following table contains additional information about each of the telephone endpoints depicted in the System Configuration diagram:

Diagram	Ext	Endpoint
A1	68001	Konftel 300 IP / Line 1
A2	68002	Konftel 300 IP/ Line 2
B	60121	Avaya 4620 SW
C	60093	IP Avaya 9640
D	60007	Avaya 2420
E	60008	Avaya 2420
X		ISDN telephone

Table 1: Extensions Used for Testing

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration:

Equipment	Software Version
Avaya S8720 Server / Avaya Communication Manager	R015x.01.1.415.1
Avaya SIP Enablement Services	SES-5.0.0.0-825.31
Avaya 2410 Digital Telephone	5.0
Avaya 4620SW IP Telephone	2.887 (H.323)
Avaya 9640 Telephone	1.5 (H.323)
Avaya C364T-PWR Ethernet Switch	4.5.14
Konftel 300 IP	1.0.2

Table 2: Version Numbers of Equipment and Software

3. Configuration

3.1. Configuration of the Avaya S8720 Server

The configuration and verification operations illustrated in this section were performed using the Avaya Communication Manager System Administration Terminal (SAT).

3.1.1. Verify system-parameters customer-options

Use the **display system-parameters customer-options** command to verify that Avaya Communication Manager is configured to meet the minimum requirements to support the configuration used for these tests. Those items shown in **bold** indicate required values or minimum capacity requirements. If these are not met in the configuration, please contact an Avaya representative for further assistance.

Verify that the number of SIP trunks supported by the system is sufficient for the combination of trunks to the two Konftel 300IP SIP accounts and addition SIP endpoints that might be included in the configuration.

display system-parameters customer-options		Page	2 of 11
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
	Maximum Administered H.323 Trunks:	10	0
	Maximum Concurrently Registered IP Stations:	50	10
	Maximum Administered Remote Office Trunks:	0	0
Maximum Concurrently Registered Remote Office Stations:	0	0	0
	Maximum Concurrently Registered IP eCons:	0	0
Max Concur Registered Unauthenticated H.323 Stations:	0	0	0
	Maximum Video Capable H.323 Stations:	0	0
	Maximum Video Capable IP Softphones:	0	0
	Maximum Administered SIP Trunks:	200	20
	Maximum Number of DS1 Boards with Echo Cancellation:	0	0
	Maximum TN2501 VAL Boards:	1	0
	Maximum G250/G350/G700 VAL Sources:	0	0
	Maximum TN2602 Boards with 80 VoIP Channels:	0	0
	Maximum TN2602 Boards with 320 VoIP Channels:	0	0
	Maximum Number of Expanded Meet-me Conference Ports:	0	0

Figure 3: System-Parameters Customer-Options Form, Page. 2

Verify that IP Stations are supported by the system.

display system-parameters customer-options		Page	4 of 11
OPTIONAL FEATURES			
Emergency Access to Attendant? y	IP Stations? y		
Enable 'dadmin' Login? y			
Enhanced Conferencing? y	ISDN Feature Plus? n		
Enhanced EC500? y	ISDN/SIP Network Call Redirection? y		
Enterprise Survivable Server? n	ISDN-BRI Trunks? y		
Enterprise Wide Licensing? n	ISDN-PRI? y		
ESS Administration? n	Local Survivable Processor? n		
Extended Cvg/Fwd Admin? y	Malicious Call Trace? n		
External Device Alarm Admin? n	Media Encryption Over IP? n		
Five Port Networks Max Per MCC? n	Mode Code for Centralized Voice Mail? n		
Flexible Billing? n			
Forced Entry of Account Codes? n	Multifrequency Signaling? y		
Global Call Classification? n	Multimedia Call Handling (Basic)? n		
Hospitality (Basic)? y	Multimedia Call Handling (Enhanced)? n		
Hospitality (G3V3 Enhancements)? n	Multimedia IP SIP Trunking? n		
IP Trunks? y			
IP Attendant Consoles? y			

Figure 4: System-Parameters Customer-Options Form, Page 4

Verify that sufficient IP Phones are allocated to support the system configuration.

display system-parameters customer-options			Page 10 of 11
MAXIMUM IP REGISTRATIONS BY PRODUCT ID			
Product ID	Rel. Limit	Used	
IP_API_A	: 1000	0	
IP_API_B	: 1000	0	
IP_API_C	: 1000	0	
IP_Agent	: 1000	0	
IP_IR_A	: 1000	0	
IP_Phone	: 12000	4	
IP_ROMax	: 12000	0	
IP_Soft	: 1000	0	
IP_eCons	: 128	0	
oneX_Comm	: 12000	0	

Figure 5: System-Parameters Customer-Options Form Page 10

3.1.2. Configure Dial Plan

Use the **change dialplan analysis** command to specify that dialed strings which begin with “6” are extensions. Include the number used for a Trunk Access Code in **Figure 15**.

change dialplan analysis						Page 1 of 12			
DIAL PLAN ANALYSIS TABLE									
						Percent Full:		1	
Dialed	Total	Call		Dialed	Total	Call	Dialed	Total	Call
String	Length	Type		String	Length	Type	String	Length	Type
6	5	ext							
*83	3	dac							

Figure 6: Dialplan Analysis Form

3.1.3. Configure IP Network Interface

Use the **change node-names ip** command to configure IP addresses as shown in the following table.

Parameter	Usage
clan	Enter the IP address of the Avaya Control LAN (CLAN) interface.
ses	Enter the IP address of the SES server.

Table 3: Node-Names IP Parameters

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
clan	192.168.60.6	
default	0.0.0.0	
ses	192.168.200.100	

Figure 7: Node-Names IP Form, Page 1

Use the **change ip-network-region <x>** command to designate a network region to be used for the IP telephone communications using the parameters shown in the following table, where <x> is the network region assigned to the SIP signaling group in **Figure 14**.

Parameter	Usage
Location	Enter “1”.
Authoritative Domain	Enter the domain name to be used for SIP communications. This must be the same as is specified in Figure 23 .
Name	Enter a name to identify the region.
Codec Set	Enter the number of the codec set defined in Figure 9 .

Table 4: IP-Network-Region Parameters

```

change ip-network-region 1                                     Page 1 of 19
                                IP NETWORK REGION
Region: 1
Location: 1      Authoritative Domain: ffm.com
Name: FFM
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: yes
Codec Set: 1          Inter-region IP-IP Direct Audio: yes
                      IP Audio Hairpinning? n
  UDP Port Min: 2048
  UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS      RTCP Reporting Enabled? y
Call Control PHB Value: 46    RTCP MONITOR SERVER PARAMETERS
Audio PHB Value: 46          Use Default Server Parameters? y
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS          RSVP Enabled? n
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5

```

Figure 8: IP-Network-Region Form, Page 1

Use the **change ip-codec-set** command to designate a codec set to be used. The codec selected by Konftel 300IP users is dependent on fidelity requirements and bandwidth availability. Most of the testing was done with the G.711A codec, although other codec combinations were tested to insured proper codec interoperation. The codec selection configured here must be compatible with the codecs configured for the Konftel 300IP in **Figure 30**.

Parameter	Usage
Audio Codec	Enter “G.711A”.

Table 5: IP-Codec-Set Parameters

change change ip-codec-set 1				Page 1 of 2
IP Codec Set				
Codec Set: 1				
Audio	Silence	Frames	Packet	
Codec	Suppression	Per Pkt	Size(ms)	
1: G.711A	n	2	20	

Figure 9: IP-Codec-Set Form, Page 1

3.1.4. Configure Station Interface for the Konftel 300IP

Use the **add station <x>** command to allocate a station for Konftel 300IP where <x> is the extension for the Konftel 300IPs as shown in **Table 1**. Repeat this to add a second station to be used by the second Konftel 300IP SIP account.

Parameter	Usage
Type (p. 1)	Enter the station type for an Avaya IP phone which can be configured as a SIP phone which supports least four call appearances.
Name (p. 1)	Enter the name of the user which is to be associated with the phone. Use distinct names for stations provisioned for the separate Konftel 300 IP SIP accounts.
BUTTON ASSIGNMENTS (p. 4)	Create an additional call appearance, so that there are a total of four call appearances to enable the Konftel 300IP to create group conferences with a total of five parties (including itself).

Table 6: Configuration Konftel 300IP Station

add station 68001			Page 1 of 5
STATION			
Extension: 68001	Lock Messages? n	BCC: 0	
Type: 4610	Security Code:	TN: 1	
Port: S00140	Coverage Path 1:	COR: 1	
Name: Konftel Line 1	Coverage Path 2:	COS: 1	
Hunt-to Station:			
STATION OPTIONS			
Loss Group: 19		Time of Day Lock Table:	
Speakerphone: 2-way		Personalized Ringing Pattern: 1	
Display Language: english		Message Lamp Ext: 68001	
Survivable GK Node Name:		Mute Button Enabled? y	
Survivable COR: internal		Media Complex Ext:	
Survivable Trunk Dest? y		IP SoftPhone? n	
IP Video? n			

Figure 10: Konftel 300IP Station Form, Page 1

add station 68001		Page 4 of 5	
STATION			
SITE DATA			
Room:		Headset?	n
Jack:		Speaker?	n
Cable:		Mounting:	d
Floor:		Cord Length:	0
Building:		Set Color:	
ABBREVIATED DIALING			
List1:	List2:	List3:	
BUTTON ASSIGNMENTS			
1: call-appr	5:		
2: call-appr	6:		
3: call-appr	7:		
4: call-appr	8:		

Figure 11: Konftel 300IP Station Form, Page 4

3.1.5. Configure Avaya IP Stations

Use the **add station <x>** command to allocate stations for Avaya IP Telephones where <x> is the extension for the each of the Avaya IP Stations as shown in **Table 1**.

Parameter	Usage
Type	Enter the type of station to be configured. Any Avaya IP station type can be used.
Name	Enter the name of the user which is to be associated with the phone.

Table 7: Configuration Avaya IP Station

add station 60121		Page 1 of 5
STATION		
Extension: 60121	Lock Messages? n	BCC: 0
Type: 4620	Security Code:	TN: 1
Port: S00009	Coverage Path 1:	COR: 1
Name: extn 60121	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
	Time of Day Lock Table:	
Loss Group: 19	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 60121	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english		
Survivable GK Node Name:	Media Complex Ext:	
Survivable COR: internal	IP SoftPhone? n	
Survivable Trunk Dest? y		
	Customizable Labels? y	

Figure 12: Avaya IP Station Form

3.1.6. Configure Avaya Digital Stations

Use the **add station <x>** command to allocate a stations for Avaya Telephones where <x> is the extension for the each of the Avaya digital stations shown in **Table 1**.

Parameter	Usage
Type	Enter the type of station to be configured.
Port	Enter the board and port address of the interface to which the station is attached.
Name	Enter the name of the user which is to be associated with the phone.

Table 8: Configuration Avaya Digital Station

add station 60007		Page 1 of 5
STATION		
Extension: 60007	Lock Messages? n	BCC: 0
Type: 2410	Security Code:	TN: 1
Port: 01A0507	Coverage Path 1:	COR: 1
Name: extn 60007	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
	Time of Day Lock Table:	
Loss Group: 2	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 60007	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	
	IP Video? n	
	Customizable Labels? y	

Figure 13: Avaya Digital Station Form

3.1.7. Configure Interface to SIP Enablement Services

Use the **add signaling-group** command to allocate a signaling group for the interface to SES using the following parameters:

Parameter	Usage
Group Type	Enter “sip”.
Near-end Node Name	Enter “clan” (defined in Figure 7) to designate the Control LAN as the near end node name.
Far-end Node Name	Enter “ses” to assign the SES server as the far end node name.
DTMF over IP	Enter “rtp-payload”. This value is used to have Avaya Communication Manager send DTMF transmissions using RFC 2833, which corresponds to the default DTMF method employed by the Konftel 300IP, as shown in Figure 30 .
Direct IP-IP Audio Connections	Enter “y” to allow direct IP-IP endpoint connections (shuffling).

Table 9: Signaling-Group Parameters

add signaling-group 83		Page 1 of 1
SIGNALING GROUP		
Group Number: 1	Group Type: sip	
	Transport Method: tls	
Near-end Node Name: clan	Far-end Node Name: ses	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
	Far-end Network Region: 1	
Far-end Domain:		
	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	Direct IP-IP Audio Connections? y	
	IP Audio Hairpinning? n	
Enable Layer 3 Test? n		
Session Establishment Timer(min): 3		

Figure 14: Signaling-Group Form

Use the **add trunk-group** <*n*> command, where <*n*> is an unused trunk number, to allocate a trunk group to be used as an interface to the SIP Enablement Services server. Use the parameters shown in the following table.

Parameter	Usage
Group Type	Enter “sip”.
Group Name	Assign a name for identification purposes.
TAC	Enter the Trunk Access Code allocated in Figure 6 .
Service Type	Enter “tie”.
Signaling Group	Enter the number of the signaling group allocated in Figure 14 .
Number of Members	Enter a number large enough to support the maximum number of anticipated simultaneous calls to be made via the SIP trunk.

Table 10: Trunk-Group Parameters

add trunk-group 83		Page 1 of 21	
TRUNK GROUP			
Group Number: 83	Group Type: sip	CDR Reports: y	
Group Name: SIP	COR: 1	TN: 1	TAC: *83
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Signaling Group: 83	
		Number of Members: 255	

Figure 15: Trunk-Group Form, Page 1

Use the **add off-pbx-telephone station-mapping <x>** command for each of the Konftel 300IP SIP extensions shown in **Table 1**.

Parameter	Usage
Station Extension (p.1)	The extension of the SIP telephone. This is the station provisioned in Figure 10 .
Application (p.1)	Enter “OPS”.
Phone Number (p.1)	Enter the extension.
Trunk Selection (p.1)	Enter the number of the SIP trunk which is allocated in Figure 15 .
Call Limit (p.2)	Enter “4” to allow group conference operations.

Table 11: off-pbx-telephone station-mapping Parameters

add off-pbx-telephone station-mapping 68001						Page 1 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION						
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set
68001	OPS	-		68001	83	1

Figure 16: off-pbx-telephone station-mapping Form, Page 1

change off-pbx-telephone station-mapping 68001						Page 2 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION						
Station Extension	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location	
68001	4	both	all	none		

Figure 17: off-pbx-telephone station-mapping Form, Page 2

At the completion of all Avaya Communication Manager configuration operations, use the **save translation all** command to save the changes.

3.2. Avaya SIP Enablement Services

Configure SES by entering “<SES IP Address>/admin/” in a web browser. After entering the administrator name and password, the following screen content is displayed. Select “Launch Administration Web Interface”.

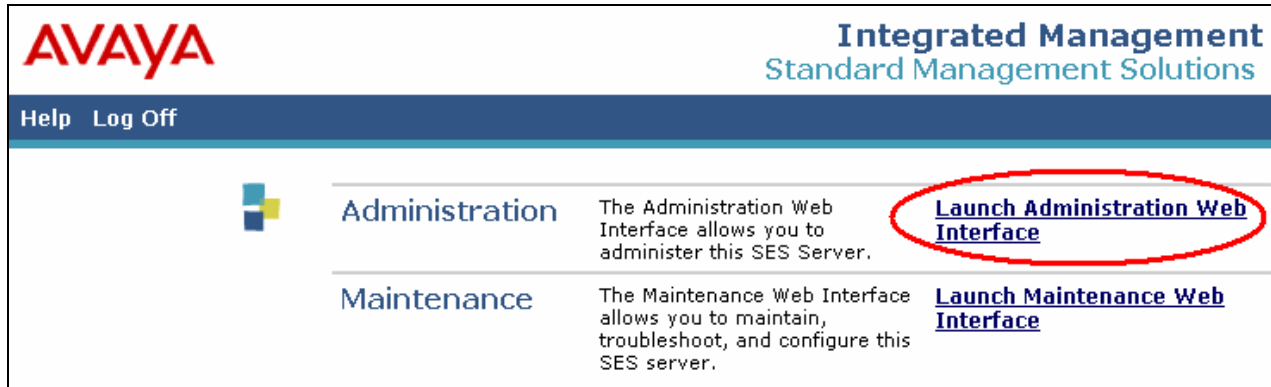



Figure 18: Launch Maintenance Web Interface Screen

Click “Setup” on the Top SES menu.



AVAYA Integrated Management
SIP Server Management
Help Exit Server: 192.168.200.100

Top
Setup
+ Users
Address Map Priorities
+ Adjunct Systems
+ Conferences
Emergency Contacts
+ Export/Import to ProVision
+ Hosts
IM logs
+ Media Servers
+ Media Server Extensions
+ Server Configuration
+ SIP Phone Settings
+ Survivable Call Processors
System Status
+ Trace Logger
+ Trusted Hosts

Top

Manage Users	Add and delete Users.
Manage Address Map Priorities	Adjust Address Map Priorities.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Conferencing	Add and delete Conference Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Export Import to ProVision	Export and import data using ProVision on this host.
Manage Hosts	Add and delete Hosts.
IM logs	Download IM Logs.
Manage Media Servers	Add and delete Media Servers.
Manage Media Server Extensions	Add and delete Media Server Extensions.
Server Configuration	View Properties of the system.
Manage SIP Phone Settings	Add/Delete Phone Settings
Manage Survivable Call Processors	Add and delete Survivable Call Processors.
System Status	View System Status.
Trace Logger	Manage SIP Trace Logs.
Manage Trusted Hosts	Add and delete Trusted Hosts.

Figure 19: Initiate “Setup” from Top SES Configuration Screen

3.2.1. Setup Dataservice

Click “Setup Dataservice”.

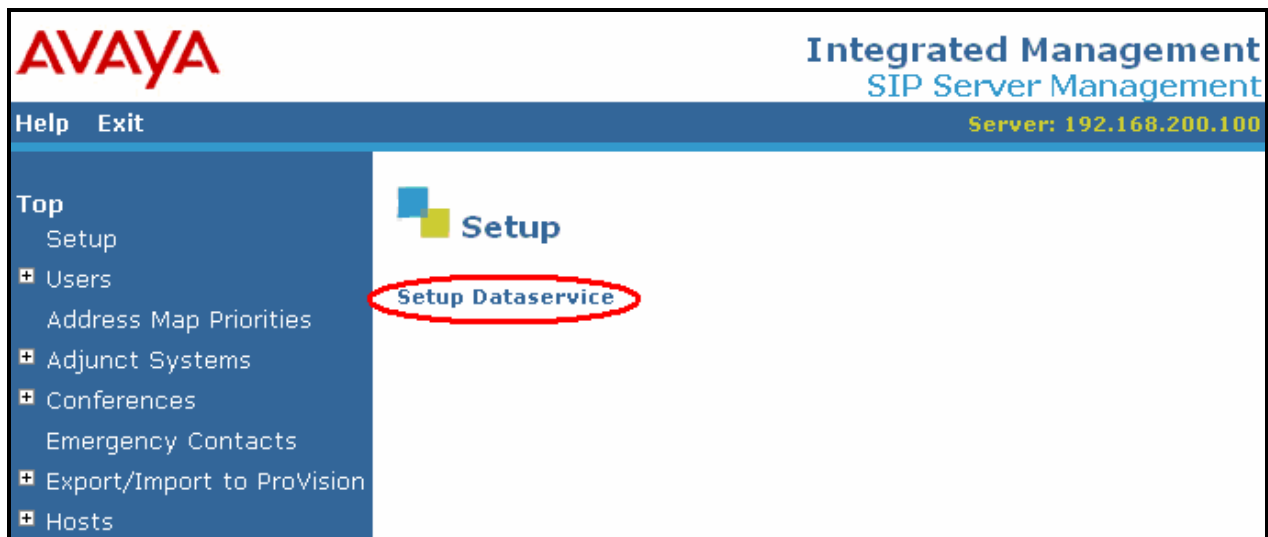


Figure 20: Initiate “Setup Dataservice” from Top Setup Screen

Select “This server is the SES Master Administration System for the SES Network”, and click “Setup”, and “Continue” for the screen that follows.

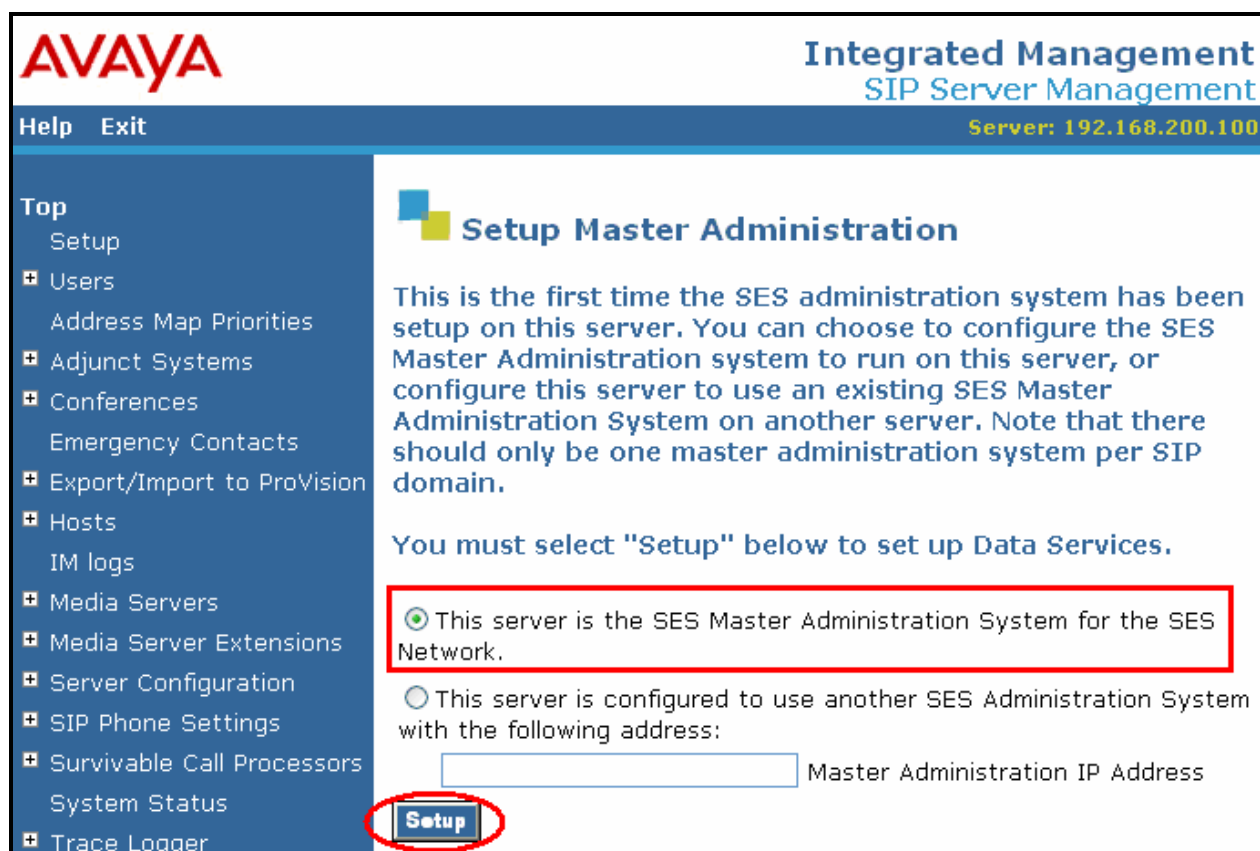


Figure 21: “Setup Master Administration” Screen

3.2.2. Setup SIP Domain

Click “Setup SIP domain”.



Figure 22: “Setup SIP Domain” Screen

Enter values in this screen as shown in the following table, and click “Update”, followed by “Ok” for the following screen.

Parameter	Usage
SIP Domain	Enter the same value as was used for “Authoritative Domain” in Figure 8 .
License Host	Enter the IP address of the license host, in this case the IP address of this SES server.

Table 12: Parameters for System Properties

AVAYA Integrated Management SIP Server Management
 Help Exit Server: 192.168.200.100

Top

- Users
 - Address Map Priorities
- Adjunct Systems
- Certificate Management
- Conferences
 - Emergency Contacts
- Export/Import to ProVision
- Hosts
 - IM logs
- Media Servers
- Media Server Extensions
- Server Configuration
 - Admin Setup
 - IM Log Settings
 - License

View System Properties

SES Version SES-5.0.0.0-825.31
 System Configuration simplex
 Host Type SES combined home-edge

SIP Domain* ffm.com

Note that the DNS domain is ffm.com

If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

SIP License Host* 192.168.200.100

Figure 23: View System Properties Screen

3.2.3. Add Media Server Interface

Navigate to **Media Servers** → **Add** from the “Top” level menu shown in **Figure 19**, and specify the interface parameters as shown in the following table.

Parameter	Usage
Media Server Interface Name	Enter a descriptive name for this interface.
Host	Select the IP address of the SES server from the drop-down menu.
SIP Trunk IP Address	Enter the IP address of the CLAN interface specified in Figure 7 and Figure 14 .
Media Server Admin Address	Enter the IP address of the Avaya Server administration interface.
Media Server Admin Login	Enter an administrator user ID for the media server, such as the User ID used in Figure 18
Media Server Admin Password	Enter the password for the administrator user ID, such as the Password used with the User ID in Figure 18 , or contact Avaya for more information on creating a valid user ID.

Table 13: “Add Media Server Interface” Parameters

AVAYA Integrated Management
SIP Server Management
Server: 192.168.200.100

Help Exit

Top

- Users
 - Address Map Priorities
- Adjunct Systems
- Certificate Management
- Conferences
 - Emergency Contacts
- Export/Import to ProVision
- Hosts
 - IM logs
- Media Servers
 - Add
 - List
- Media Server Extensions
- Server Configuration
- SIP Phone Settings
- Survivable Call Processors
- System Status
- Trace Logger

Add Media Server Interface

Media Server Interface Name* S8720

Host 192.168.200.100

SIP Trunk

SIP Trunk Link Type ☐ TCP ☒ TLS

SIP Trunk IP Address* 192.168.60.6

Media Server

Media Server Admin Address (see Help) 192.168.60.100

Media Server Admin Port 5022

Media Server Admin Login [REDACTED]

Media Server Admin Password [REDACTED]

Media Server Admin Password Confirm [REDACTED]

SMS Connection Type ☒ SSH ☐ Telnet ☐ Not Available

Figure 24: SES Add Media Server Interface Screen

3.2.4. Add Hosts

Navigate to **Hosts** → **Add Host** from the top level screen shown in **Figure 19**. Enter values in this screen as shown in the following table, accepting the default values for those parameters which are not listed. Click the “Add” button upon completion and the “Continue” button when the following screen is displayed.

Parameter	Usage
Host IP Address	Enter the IP address of the SES server.
Profile Service Password	Enter the password which was entered from the initial setup script when SES was installed.

Table 14: “Add Host” Parameters

AVAYA Integrated Management SIP Server Management
Server: 192.168.200.100

Help Exit

Top

- Users
- Address Map Priorities
- Adjunct Systems
- Certificate Management
- Conferences
- Emergency Contacts
- Export/Import to ProVision
- Hosts
 - List
 - Migrate Home/Edge
 - IM logs
- Media Servers
- Media Server Extensions
- Server Configuration
- SIP Phone Settings
- Survivable Call Processors
- System Status
- Trace Logger
- Trusted Hosts

Add Host

Host IP Address* 192.168.200.100

Profile Service Password*

Host Type SES combined home-edge

Parent none

Listen Protocols ☒ UDP ☒ TCP ☒ TLS

Link Protocols ☐ UDP ☐ TCP ☒ TLS

Access Control Policy (Default) ☒ Allow All ☐ Deny All

Emergency Contacts Policy ☒ Allow ☐ Deny

Minimum Registration (seconds) 300 Registration Expiration Timer (seconds)* 86400

Line Reservation Timer (seconds) 30

* Outbound Routing Allowed ☒ Internal ☒ External From

Figure 25: SES Add Host Screen

3.2.5. Add Users

From the “Users” menu in the left frame, click “Add”, and enter the parameters shown in the following screen, for each of the SIP extensions used for Konftel 300IP SIP accounts, as shown in **Table 1**. Stations for each of these users are allocated in for Avaya Communication Manager in **Figure 10** and **Figure 16**. Accounts are provisioned on the Konftel 300IP for these users in **Figure 29**.

Parameter	Usage
Primary Handle	Enter the extension to be assigned to the user.
User ID	Enter the extension to be assigned to the user.
Password / Confirm	Enter the password to be assigned to the user.
First / Last Name	Enter a name for identification purposes.
Add Media Server Extension	Check this box, to add an extension for this user.

Table 15: User Configuration Parameters

Figure 26: SES Add User Screen

The “Add Media Server Extension” screen will appear after the user has been added in the previous step. Enter the “Extension” for the Konftel 300IP SIP account shown in **Table 1** for the user which was created in the previous step, select the corresponding “Media Server” from the drop-down list (see **Figure 24**), and click “Add”. Note that each extension which is added must also be configured in **Figure 29**.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo, the title "Integrated Management SIP Server Management", and the server IP address "Server: 192.168.200.100". A navigation sidebar on the left lists options: "Top", "Users", "Add", "Default Profile", "Delete", "Edit", "List", "Password", and "Search". The main content area is titled "Add Media Server Extension". It contains two input fields: "Extension" with the value "68001" and "Media Server" with a dropdown menu showing "S8720". Below these fields is a note: "Fields marked * are required." and an "Add" button. The "Extension" and "Media Server" fields, along with the "Add" button, are highlighted with red boxes.

Figure 27: SES Add Media Server Extension Screen


3.3. Configuration of the Konftel 300IP

The Konftel 300IP can either be assigned an IP address manually, or via DHCP via the keypad interface (see **Figure 1**) on the Konftel 300IP, using the following sequence.

- Press the “Menu” key.
- Select “Settings”.
- Select “Advanced”, and enter the administrator PIN when prompted.
- Select “Network”.
- Select either “DHCP” or “STATIC” followed by the IP address to be assigned to the unit.

Once the Konftel 300IP has been assigned an IP address, the remainder of the configuration procedure can be preformed either with the keyboard/display of the Konftel 300IP, or via Web browser, as illustrated by the remainder of this section of the document.

To use a web browser, enter the IP address of the Konftel 300IP into the URI “Address” field of the browser, which causes the screen shown below to appear. Select “Admin” from the “Profile” drop-down menu, enter the administrator PIN, and click “Login”.



The screenshot shows the Konftel 300IP web interface. At the top left is the Konftel logo, and at the top right, it says "You are logged out" with a blue "Login" link. Below this is a dark grey navigation bar with the following links: "Status", "Phone book", "Call list", and "Settings". The main content area is titled "Login" and contains a "Profile" dropdown menu set to "Admin", a "PIN" input field with four black dots, and a "Login" button.

Figure 28: Konftel 300IP Login Screen

3.3.1. Configure SIP Accounts

Select the “Settings” tab from the top of the screen, and then “SIP” from row of the underlying set of tabs. Enter the parameters shown in the following table for each of the accounts (S8720-1 and S8720-2 below).

Parameter	Usage
Enable account	Select the “Yes” radio button.
Account name	Enter a descriptive name for the account.
User	Enter the “User ID” which was allocated in Figure 26 .
Registrar	Enter the IP address of SES.
Proxy	Enter the IP address of SES.
Authentication name	Enter the “User ID” which was allocated in Figure 26 .
Password	Enter the “Password” which was allocated in Figure 26 .

Table 16: User Configuration Parameters

The screenshot displays the Konftel 300IP SIP Settings screen. The top navigation bar includes 'Status', 'Phone book', 'Call list', and 'Settings'. The 'Settings' tab is selected, and the 'SIP' sub-tab is active. The screen is divided into sections for 'Account 1', 'Account 2', 'NAT Traversal', and 'Transport'. For each account, there are fields for 'Enable account' (radio buttons for Yes/No), 'Account name', 'User', 'Registrar', 'Proxy', 'Realm', 'Authentication name', 'Password', and 'Registration interval'. Red boxes highlight the 'Enable account' section and the 'Authentication name' and 'Password' fields for both accounts. Below the accounts, there is a 'NAT Traversal' section with options for STUN, Offer ICE, and TURN, and a 'Transport' section with radio buttons for UDP, TCP, TLS, and SIPS. A note at the bottom right says 'Please check corresponding media signalling setting'.

Figure 29: Konftel 300IP SIP Settings Screen

3.3.2. Configure Media Settings

Select the embedded “Media” tab from within the “Setting” tab. The codec selected by Konftel 300IP users is dependent on fidelity requirements and bandwidth availability. Most of the testing was done with the G.711A codec, although other codec combinations were tested to insured proper codec interoperation. The codec selection configured here must be compatible with the codecs configured for Avaya Communication Manager in **Figure 9**. Click “Save” to complete the configuration sequence.

Codec	Priority
G722	0 - Disabled
G711 Alaw	4 - High
G711 Ulaw	0 - Disabled
G729	0 - Disabled

Security

SRTP ☒ Disabled ☐ Optional ☐ Mandatory

Secure signalling ☐ No ☒ TLS ☐ SIPs *Please check corresponding SIP transport setting*

VAD

Enable VAD ☐ Yes ☒ No

DTMF

DTMF Signalling ☒ RFC 2833 ☐ SIP Info ☐ Inband

Figure 30: Konftel 300IP Media Settings Screen

4. Interoperability Compliance Testing

The objective of the compliance testing done on the Konftel 300IP product was to verify that it is compatible with Avaya Communication Manager and Avaya SIP Enablement Services. This includes verifying that the essential Konftel 300IP features function properly when used with Avaya Communication Manager, and that Avaya Communication Manager features are not hindered by the interaction with the Konftel 300IP. Furthermore, Konftel 300IP's robustness was verified.

4.1. General Test Approach

The test method employed can be described as follows:

- Avaya Communication Manager was configured to support various local H.323 IP telephones, as well as both the SIP accounts of the Konftel 300IP.
- The Konftel 300IP was configured to use its two SIP accounts to act as separate SIP telephone endpoints.
- Various telephony operations involving the Konftel 300IP and Avaya Communication Manager were performed manually.
- The SIP protocol exchanges were monitored with a protocol trace program to verify the correct protocol exchanges.

4.2. Test Results

The only issue encountered during testing was that if Avaya Communication Manager and the Konftel 300IP are configured with compatible codec sets, with differing priorities, the actual codec selected differs depending on whether Direct IP-IP Audio Connections are enabled. This is a minor problem which can be avoided by changing the codec priority configuration such that Avaya Communication Manager and the Konftel 300IP each use the same codec priority order.

5. Verification Steps

- Verify that the Konftel 300IP can register and re-register with the Avaya SIP Enablement Services server.
- Verify that the Konftel 300IP can make and receive calls from both of its SIP account lines.
- Verify that the Konftel 300IP can make multiple simultaneous calls and toggle between those calls.
- Verify that the Konftel 300IP hold/retrieve feature is compatible with Avaya Communication Manager.
- Verify that the Konftel 300IP can create conferences manually.
- Verify that the Konftel 300IP can create multi-party conferences with up to five participants (including itself) using the Konftel 300IP “group conference” feature.
- Verify that the Konftel 300IP codec support is compatible with Avaya Communication Manager.
- Verify that the Konftel 300IP DTMF facility is compatible with Avaya Communication Manager.
- Verify that the Konftel 300IP is compatible with the direct IP-IP media stream capability of Avaya Communication Manager.
- Verify that the Konftel 300IP can operate from both its external power supply or for power that it received via its Power Over Ethernet connection.
- Verify that the Konftel 300IP can recover from interruptions to its Ethernet interface (when operating from its external power supply).

6. Support

Support for Konftel products is available at

- Web-based support: <http://www.konftel.com/>
- Email: info@konftel.com
- International help desk: +46 90706489
- North American help: +1 866-606-4728

7. Conclusion

The Konftel 300IP can be used with Avaya Communication Manager and Avaya SIP Enablement Services to enable those present in a room to participate in a telephone conversation. The configuration described in these Application Notes has been successfully compliance tested.

8. Additional References

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

- [1] *Administrator Guide for Avaya Communication Manager*, January 2008, Issue 4.0, Document Number 03-300509.
- [2] *Feature Description and Implementation for Avaya Communication Manager*, January 2008, Issue 6, Document Number 555-245-205.
- [3] *Installing and Administering SIP Enablement Services*, January 2008, Issue 5.0, Document Number 03-600768.
- [4] *SIP Enablement Services (SES) Implementation Guide*, January 2008, Issue 5, Document Number 16-300140.
- [5] *User Guide Konftel 300IP (English)*, available at www.konftel.se

Several Internet Engineering Task Force (IETF) standards track RFC documents were referenced within these Application Notes. The RFC documents may be obtained at: <http://www.rfc-editor.org/rfcsearch.html>.

- [6] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
- [7] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard

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