

Spectrum Enterprise SIP Trunking Service Avaya IPO10 IP PBX Configuration Guide

About Spectrum Enterprise:

Spectrum Enterprise is a division of Charter Communications following a merger with Time Warner Cable and acquisition of Bright House Networks. Spectrum Enterprise is a national provider of scalable, fiber technology solutions. The Spectrum Enterprise portfolio includes networking and managed services solutions, including Internet access, Ethernet and Managed Network Services, Voice, TV and Cloud solutions. Our industry-leading team of experts works closely with clients to achieve greater business success.

About this document:

Spectrum Enterprise assures IP PBX compatibility by conducting interoperability testing to ensure any potential compatibility issues have been resolved prior to installation. Please review the IP PBX configuration instructions in this guide prior to your installation date.

Be advised that this document may contain references to Charter or Charter Business. All references to Charter should be read as Spectrum Enterprise.

Thank you,

Spectrum Enterprise

Avaya IPO10 IP PBX Configuration Document



1 Introduction

These Application Notes describe the steps necessary for configuring Session Initiation Protocol (SIP) Trunking service between the legacy Charter Communications Platform and an Avaya SIP-enabled enterprise solution. It does not provide any information how to provision, configure or use the features of the switch. Please refer to the documentation provided with the IP PBX or contact the vendor.

1.1 Service Limitation

Charter Business has conducted thorough testing of the Avaya IP PBX and has determined that the combination of Charter Business SIP Trunks and the Avaya IP-PBX **DOES NOT** support consistent fax receipt or transmission. The customer should make alternative service arrangements in order to support their faxing needs.

2 Configure IP Office

This section describes the IP Office configuration required to interwork with Charter SIP Trunking service. IP Office is configured through Avaya IP Office Manager (IP Office Manager) which is a PC application.

Step	Action	Result
1	Navigate to Start > Programs > IP Office > Manager	Launch IP Office Manager
2	Navigate to File > Open Configuration	Pop-up Window Appears
3	Select The Proper IP Office	
4	Log In With The Appropriate Credentials	Management Window Appears
5	Go To Next Table	

2.1 Licensing

The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative

Step	Action	Result
6	Click: License in the Navigation Pane and SIP Trunk Channels in the Group Pane	
7	Confirm There Is A Valid License With Sufficient Instances (Trunk Channels) in the Details Pane	
8	Procedure Complete	

IP Offices						<u> - 10</u>	
BOOTP (5)	License Remote Server						
Operator (3) IP500V2 Main	License Mode License Normal						
	Licensed Version 10.0						
Control Unit (4)	PLDS Host ID						
Extension (48)	PLDS File Status Valid						
Group (1)							
Short Code (69) Seprice (0)	Feature	Key	Instances	Status	Expiration Date	Source	•
	Avaya Softphone Licence	N/A	100	Valid	Never	PLDS Nod	la
Incoming Call Route (3)	VMPro TTS (Scansoft)	N/A	40	Valid	Never	PLDS Nod	la
	VMPro TTS Professional	N/A	40	Valid	Never	PLDS Nod	ia 👘
- M Directory (0)	IPSec Tunnelling	N/A	1	Valid	Never	PLDS Nod	la
Erewall Profile (1)	Power User	N/A	384	Valid	Never	PLDS Nod	ia 👘
E-1 IP Route (5)	Avaya IP endpoints	N/A	384	Valid	Never	PLDS Nod	la
Account Code (0)	IP500 Voice Networking Channels	N/A	32	Valid	Never	PLDS Nod	fa
License (90)	SIP Trunk Channels	N/A	128	Valid	Never	PLDS Noc	Ja
	IP500 Universal PRI (Additional cha	N/A	100	Valid	Never	PLDS Not	fa
User Rights (8)	CTI Link Pro	N/A	1	Valid	Never	PLDS Nod	la
Location (0)	Wave User	N/A	16	Valid	Never	PLDS Nod	fa
Authorization Code (0)	3rd Party IP Endpoints	N/A	384	Valid	Never	PLDS Nod	la

2.2 System

Configure the necessary system settings. In an Avaya IP Office, the LAN2 tab settings correspond to the Avaya IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side). For the compliance test, the LAN1 interface was used to connect Avaya IP Office to the enterprise private network (LAN), LAN2 was not used.

2.2.1 System - LAN1 Tab

In the sample configuration, IP500V2 Main was used as the system name. The LAN port connects to Charter's Modular Access Router, across the enterprise LAN (private) network. The LAN1 settings correspond to the LAN port in IP Office.

Step	Action	Result
1	To Access LAN1 Settings, Navigate to System(1) > IP5000V2 Main in the Navigation Pane	
2	In the Details Pane Navigate to the LAN1 > LAN Settings Tab	
3	Set IP Address: LAN IP address, e.g. 10.64.70.60.	
4	Set IP Mask: LAN subnet mask, e.g. 255.255.255.0.	
5	All other parameeters should be set according to customer requirements	
6	Click: OK to commit	
7	Go to Next Table	

IP Offices	IF IF	2500V2 Main
BOOTP (5) Operator (3) IP500V2 Main IP500V2 Main	System LAN1 LAN2 DNS Voicemail Telephony Directory LAN Settings VoIP Network Topology Volp	y Services System Events
System (1) IP500V2 Main IP500V2 Main IP5	IP Address 10 64 70 60 IP Mask 255 255 255 0	
	Primary Trans. IP Address 0 0 0 0 RIP Mode None Image: Image	•
AAS (1) Incoming Call Route (3) WAN Port (0) Directory (0)	Number Of DHCP IP Addresses 200 💭 DHCP Mode	Advand
Firewall Profile (0) Firewall Profile (1) Firewall Profile (1) Firewall Profile (0) Firewall Profile (0) Firewall Profile (0) Firewall Profile (0)	Server & Chent & Diatin & Disabled	Advanced
 Iser Rights (8) ✓ ARS (2) ✓ Location (0) ✓ Authorization Code (0) 		

Step	Action	Result
8	Click On: VoIP Tab	
9	Check: H323 Gatekeeper Enable to allow Avaya IP Telephones/Softphone using the H.323 protocol to register.	
10	Check: SIP Trunks Enable to enable the configuration of SIP Trunk connecting to Charter.	
11	Check: SIP Registrar Enable to allow Avaya IP Telephones/Softphone to register using the SIP protocol.	
12	Enter: the Domain Name of the enterprise under Doma in Name.	
13	Verify: the UDP Port and TCP Port numbers under La yer 4 Protocol are set to 5060.	
14	Verify: the RTP Port Number Range settings for a specific range for the RTP traffic.	
	The Port Range (Minimum) and Port Range (Maximum) values were kept as default.	
15	In the Keepalives Section, Set Scope: RTP-RTCP	This will cause the IP Office to send RTP keepalive packets at the beginning of the calls and every 30 seconds thereafter if no other RTP traffic is present.
16	Set Periodic Timeout: 30	
17	Set Initial Keepalives: Enabled	

18	All other parameters should be set according to customer requirements.	
19	Click: OK to commit	
20	Go to Next Table	

IP Offices	E IP500V2 Main
 ⊕ & BOOTP (5) ⊕ Ø Operator (3) ⊕ IP500V2 Main ⊕ Surteen (1) 	System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR VCM VoIP VoIP Security LAN Settings VoIP Network Topology VolP VolP VolP Security
	Image: With a state in the
	Image: SIP Trunks Enable Image: SIP Registrar Enable Image: Auto-create Extension/User Image: SIP Remote Extension Enable
- WAN Port (0) Directory (0) Time Profile (0) B B Firewall Profile (1)	SIP Domain Name avaya.lab.com SIP Registrar FQDN avaya.lab.com
 IP Route (5) Account Code (0) License (90) 	UDP UDP Port 5060 Remote UDP Port 5060
- ₩ Tunnel (0) ⊕ ₩ User Rights (8) ⊕ ★ ARS (2)	Layer 4 Protocol V TCP TCP Port 5060 Remote TCP Port 5060
Authorization Code (0)	Challenge Expiration Time (sec) 10
	Port Number Range Minimum 49152 Maximum 53246
	Port Number Range (NAT) Minimum 49152 - Maximum 53246 -
	Enable RTCP Monitoring on Port 5005 RTCP collector IP address for phones 0 . 0 . 0 . 0
	Keepalives Scope Initial keepalives Enabled

Step	Action	Result
21	Click On: Network Topology Tab	
22	Select: Firewall/NAT Type from the pull-down menu to the option that matches the network configuration. In the compliance testing, it was set to Op en Internet .	With this configuration, even though the default STUN settings are populated, they will not be used.
23	Set Binding Refresh Time (seconds): Desired Value , the value of 300 (or every 5 minutes) was used during the compliance testing.	This value is used to determine the frequency that IP Office will send OPTIONS heartbeat to the service provider.
23	Set Public IP address: The IP Address assigned under the LAN Settings tab, e.g., 10.64.70.60.	
24	Set Public Port: 5060 for UDP	

25	All other parameters should be set according to customer requirements.	
26	Click: OK to commit	
27	Go to Next Table	

IP Offices	E IP500V2 Main
 ⊕- & BOOTP (5) ⊕ Ø Operator (3) ⊕ IP500V2 Main ⊕ System (1) ⊕ IP500V2 Main 	System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR VCM VoIP VoIP Security LAN Settings VoIP Network Topology Network Topology
⊕ (*) Line (25) ⊕ -∞ Control Unit (4) ⊕ -∞ Extension (48) ⊕ -∰ Group (1) ⊕ -∰ Short Code (69) -∰ Service (0)	STUN Server Address 69/30/168.13 STUN Port 34/8 Firewall/NAT Type Open Internet • Binding Refresh Time (sec) 300 • Public IP Address 10 64 70 60
	Public Port UDP 5060 TCP 5060 TLS 5061
License (90) -₩i Tunnel (0) ⊕-₩i User Rights (8) ⊕-₩i ARS (2) -₩i Location (0) -₩i Authorization Code (0)	Run STUN on startup

2.2.2 - Telephony Tab

Step	Action	Result
28	Navigate to: Telephony > Telephony Tab in the Details Pane	
29	Choose: the Companding Law typical for the enterprise location, U-Law was used.	
30	Uncheck: Inhibit Off-Switch Forward/Transfer box	Allow call forwarding and call transfers to the PSTN via the SIP trunk to the service provider.
31	All other parameters should be set according to customer requirements.	
32	Click: OK to commit	
33	Go to Next Table	

2.2.3 System - VoIP TAB

Step	Action	Result
34	Navigate To: System (1) > IP500V2 Main in the Navigation Pane	
35	Select: VoIP tab	
36	The RFC2833 Default Payload field is new in IP Office release 10. The default value 101 was used.	It allows the manual configuration of the payload type used on SIP calls that are initiated by the IP Office.
37	For Codec Selection : select the codecs and codec order of preference on the right, under the Select ed column. The Default Codec Selection area enables the codec preference order to be configured on a system-wide basis. The buttons between the two lists can be used to move codecs between the Unused and Selected lists, and to change the order of the codecs in the Selected codecs list. The example below shows the codecs used for IP phones (SIP and H.323), the system's default codecs and order was used.	By default, all IP lines and phones (SIP and H.323) will use the system default codec selection shown here, unless configured otherwise for a specific line or extension.
38	Click: OK to commit	
39	Procedure Completed	



<note>

Note: The codec selections defined under this section (System – VoIP Tab) are the codecs selected for the IP phones/extensions. The codec selections defined under **Section 2.4.6** (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk).

2.3 IP Route

In the reference configuration, the IP Office LAN1 interface and the private interface of the Charter's Modular Access Router resided on the same IP subnet, so an IP route was not necessary. In an actual customer configuration, these two interfaces may be in different IP subnets, and in that case an IP route would have to be created to specify the IP address of the gateway or router where the IP Office needs to send the packets, in order to reach the IP subnet where the Charter Modular Access Router resides.

To create an IP route to specify the IP address of the gateway or router where the IP Office needs to send the packets in order to reach the IP subnet where the Charter's Modular Access Router resides (if located in different subnets).

Step	Action	Result
1	In The Left Navigation Pane, Right click on IP Route > Select New	
2	Set the IP Address and IP Mask to 0.0.0.0 to make this the default route.	
3	Set Gateway IP Address to the IP Address of the gateway/route used to route calls to the public network (to the network where Charter's Modular Access Router resides, if located in different subnet), e.g., 10.64.70.1 .	
4	Set Destination to LAN1 from the pull-down menu	
5	Click: OK to commit	
6	Procedure Completed	

IP Offices		0.0.0.0
BOOTP (5)	IP Route	
E IP500V2 Main	IP Address	0 . 0 . 0 . 0
System (1)	IP Mask	0 . 0 . 0 . 0
 ●-「子 Line (25) ●- ●- ○ Control Unit (4) 	Gateway IP Address	10 . 64 . 70 . 1
Extension (48)	Destination	LAN1
Group (1)	Metric	0
Service (0)		Proxy ARP
RAS (1) Incoming Call Route (3)		
WAN Port (0)		
Time Profile (0)		
P Route (5)		
172.31.21.0 192.168.128.0		
192.168.99.0 201.163.64.14		
Account Code (0)		
- ilki Tunnel (0)		
er Rights (8) er ¥ ARS (2)		
Location (0)		

2.4 SIP Line

A SIP Line is needed to establish the SIP connection between IP Office and the Charter SIP Trunking Service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by Avaya IP Office Manager to create a SIP Line. Follow the steps in Sections 2.4.1 and 2.4.2 to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses.
- SIP trunk Registration Credentials.
- SIP URI entries.
- Setting of the Use Network Topology Info field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Section 2.4.3** to **2.4.7**.

Alternatively, a SIP Line can be created manually. To do so, right-click on Line in the Navigation pane and select New à SIP Line. Then, follow the steps outlined in Sections 2.4.3 to 2.4.7

2.4.1 Importing a SIP Line Template

<note>

Note: DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not

include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500v2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

-		
Step	Action	Results
1	Copy a previously created template file to a location (e.g., C:\ <i>Temp</i>) on the same computer where IP Office Manager is installed. By default, the template file name will have the format <user b="" supplied<=""> text>.xml, where the <user supplied="" text=""></user> portion is entered during template file creation.</user>	
2	<note> Note: If necessary, the <user supplied="" text=""> portion of the template file name may be modified, however the <user supplied="" text="">.xml format of the file name must be maintained. For example, an original template file Test.xml could be changed to Test1.xml. The template file name is selected in Section 5.4.2, step 1, to create a new SIP Line.</user></user></note>	
3	Import the template into IP Office Manager. From IP Office Manager, select Tools > Import Templates in Manager. File Edit View Tools Help Extension Renumber IP Offic IP Offic IP Offic Operator (3) IP SolV2 Main IP SolV2 Main IP SolV2 Main IP SolV2 Main IP Offic IP Offic	A folder browser will open. Select the directory used to stor C:Tremp). Browse For Folder Select a folder to import templates from - ProgramData Pierror SUPPORT SupPORT Swwork Temp Swwork Temp Support Market Store Market Store Ma

4	In the reference configuration, template files CharterIPO10.xml was imported. The template files are automatically copied into the IP Office default template location, C:\Program Files\Avaya\IP Office\Manager\Templates .	
5	After the import is complete, a final import status pop-up window will open stating success or failure. Click OK .	
6	Go To Next Table	

2.4.2 Creating a SIP Trunk From An XML Template

Step	Action		
7	To create the SIP Trunk from a template,	right-click on Line in the Navigation pane, and select New from Temp	plate > Open from file.
	IP Offices	E	
	BOOTP (5) Gerator (3) Development (3)	SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials : Line Number 17 - ITSP Domain Name Local Domain Name SIP Ctrl+X Ctul C	
	Paste	Ctrl+C Ctrl+V Ctrl+Del	
		Ctrl+T 0	
	Export as Templa	ate Open from file	

M Open								
Comput	ter 🕨 Avaya eSOE (C:) 🕨 Program	Files (x86) ► Avaya ► IP Office ► N	Manager 🕨 Template	5	4	5 Search Tem	plates	
Organize 🔻 New fol	der						80 v	
🔶 Favorites	Name	Date modified	Туре	Size				
📰 Desktop 👔 Downloads	CharterIP010.xml	1/17/2017 7:41 AM	XML Document		4 KB			
Secent Places								
Avava eSOE (C:)								
TOSHIBA (E:)								
.								
File	name					Template File	es (*.xml)	
						Open		Cance
fter the import is compl	lete, a final import status po	p-up window will open stating	success or failu	ure. Clic	k OK			_
ter the import is complete Template Provision	lete, a final import status po oning SIP Trunk created suc	p-up window will open stating	g success or failu	ure. Clic	k OK			
fter the import is complete Template Provision	lete, a final import status po oning SIP Trunk created suc	p-up window will open stating	g success or failu	ure. Clic	k OK			

<note>

Note: It is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 2.4.3** to **2.4.7**.

2.4.3 SIP Line - SIP Line Tab

Step	Action	Results
11	Click on the SIP Line tab in the Details pane	
12	Leave the ITSP Domain Name blank	If this field is left blank, then IP Office inserts the ITSP Proxy Address from the Transport tab as the ITSP Domain in the SIP messaging.
13	Verify that URI Type is set to SIP	
14	Verify that In Service box is checked, which is the default value	This makes the trunk available to incoming and outgoing calls.
15	Verify that Check OOS box is checked, the default value.	IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the Binding Refresh Time for LAN1, as shown in Section 2.2.1 .
16	Verify that Refresh Method is set to Auto .	
17	Verify that Timer (seconds) is set to On Demand	
18	Under Redirect and Transfer, set Incoming Supervised REFER and Outgoing Supervised REFER to Never (see Sections 2.1).	
19	All other parameters should be set to default or according to customer requirements.	
20	Click OK to commit	
21	Go to Next Table	

IP Offices	E	SIP Line - Line	17	
BOOTP (5)	SIP Line Transport SIP URI VoIP T	38 Fax SIP Credentials SIP Advanced Engineering	1	
□ System (1)	Line Number	17 🔺	In Service	
IP500V2 Main	ITSP Domain Name		Check OOS	
-(* Line (25) - ~ 1	Local Domain Name			
2	URI Type	SIP	Session Timers	
18	Location	Cloud	Refresh Method	Auto
			Timer (sec)	On Demand
- % 23 - f 7 201	Prefix			
-17 202	National Prefix	0		
-17 203	International Prefix	00		
-17 205	Country Code		Redirect and Transfer	_
-13 207	Name Priority	System Default 👻	Incoming Supervised REFER	Never
-11 208	Description	Service Provider	Outgoing Supervised REFER	Never
-17 210			Send 302 Moved Temporarily	
-17 212			Outgoing Blind REFER	

2.4.4 SIP Line - Transport Tab

Step	Action	Results
22	Click on the Transport tab in the Details pane	

23	Set the ITSP Proxy Address to the IP address of the inside interface (or private side) assigned to Charter's Modular Access Router.	
24	Set the Layer 4 Protocol to UDP.	
25	Set Use Network Topology Info to LAN1 as configured in Section 2.2.	
26	Set the Send Port to 5060.	
27	All other parameters should be set to default or according to customer requirements.	
28	Click OK to commit	
29	Go to Next Table	

IP Offices	SIP Line - Line 17	
B- ■ BOOTP (5) Grave Operator (3) Grave IP500V2 Main Grave System (1) Grave IP500V2 Main Dr 13 Line (25)	SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials SIP Advanced Engineering ITSP Proxy Address 10.64.70.54 Interview Interview	
	Layer 4 Protocol UDP Send Port 5060 Use Network Topology Info LAN 1 Listen Port 5060	* *
	Explicit DNS Server(s) 0	
	Separate Registrar	

2.4.5 SIP Line - SIP URI Tab

A SIP URI entry needs to be created to match each incoming number that IP Office will accept on this line.

Step	Action	Results
30	In the SIP URI Tab click Add	The New Channel area will appear at the bottom of the pane.
31	To edit an existing entry, click an entry in the list at the top, and click the Edit button.	
32	Set Local URI, Contact, Display Name to Use Internal Data.	
33	Set Identity under Identity to Auto.	
34	Set Header under Identity to P Asserted ID	
35	Set Send Caller ID under Forwarding and Twinning to Diversi on Header.	
36	Set Diversion Header to Auto.	
37	Associate this line with an incoming line group by entering a line group number in the Incoming Group field.	This line group number will be used in defining incoming call routes for this line.

38	Associate the line to an outgoing line group using the Outgoing Group field.	The outgoing line group number is used in defining short codes for routing outbound traffic to this line.For the compliance test, a new incoming and outgoing group 17 was defined that only contains this line (line 17).
39	Set Max Sessions to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.	
40	Click OK to commit	
41	Click OK to commit again	
42	Go to Next Table	

2.4.6 SIP Line -VoIP Tab

Step	Action	Results
43	Select the VoIP tab to set the Voice over Internet Protocol Parameters of the SIP Line	The New Channel area will appear at the bottom of the pane.
44	In the sample configuration, the Codec Selection was configured using the Custom option , allowing an explicit order of codecs to be specified for the SIP Line. The buttons allow setting the specific order of preference for the codecs to be used on the SIP Line, as shown. Charter only supports codec G.711ULAW for audio.	
45	Select G.711 for Fax Transport Support (Refer to Section 2.1).	
46	Set the DTMF Support field to RFC2833.	This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
47	Check the Re-invite Supported box	Allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
48	Check the PRACK/100rel Supported box	Advertise the support for reliable provisional responses and Early Media to Charter.
49	Set Diversion Header to Auto.	
50	Default values may be used for all other parameters.	
51	Click OK to commit	
52	Go to Next Table	

IP Offices	E		SIP Line - Line 17	
■ ♣ BOOTP (5) ■ ⊕ Operator (3) ■ ■ IP500V2 Main ● ■ IP500V2 Main ● ● IP500V2 Main ● 1 1 ●<	SIP Line Transport SIP L	Unused G.711 ALAW 64K G.722 64K G.729(a) 8K CS-ACELP G.723.1 6K3 MP-MLQ	Selected G.711 ULAW 64K C<<<	 VoIP Silence Suppression Local Hold Music Re-invite Supported Codec Lockdown Allow Direct Media Path Force direct media with phones PRACK/100rel Supported G.711 Fax ECAN
-13 207 -13 207 -13 208 -13 209	Fax Transport Support DTMF Support	G.711 RFC2833		•
-13 210	Media Security	Disabled	•	

<note>

Note: The codec selections defined under this section (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk). The codec selections defined under Section 2.2.3 (System – VoIP tab) are the codecs selected for the IP phones/extension (H.323 and SIP).

2.4.7 SIP Line - SIP Advanced Tab

Step	Action	Results
53	Select the SIP Advanced tab to configure IP Office to use the PAI header for privacy calls	For outbound calls with privacy enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with "anonymous". IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing purposes. By default, IP Office will use the PPI header for privacy.
54	Verify the Call Routing Method is set to Request URI .	
55	Check the box for Use PAI for Privacy .	
56	Default values may be used for all other parameters.	
57	Click OK to commit	
58	Procedure Completed	

IP Offices	E	SIP Line - Line 17		
BOOTP (5)	SIP Line Transport SIP URI VoIP	T38 Fax SIP Credentials SIP Advanced Engineering		
⊕ ⊕ Operator (3) ⊕ IP500V2 Main ⊕ System (1) ↓ IP500V2 Main ⊕ 19500V2 Main ⊕ 19500V2 Main ⊕ 11 ↓ 1 ↓ 20 ↓ 1 ↓ 21 ↓ 19 ↓ 20 ↓ 21 ↓ 22 ↓ 23 ↓ 17 ↓ 20 ↓ 21 ↓ 22 ↓ 23 ↓ 17 ↓ 20 ↓ 19 ↓ 20 ↓ 19 ↓ 20 ↓ 19 ↓ 20 ↓ 19 ↓ 20 ↓ 17 ↓ 21 ↓ 17 ↓ 21 ↓ 21 ↓ 17 ↓ 21	Addressing Association Method Call Routing Method Suppress DNS SRV Lookups Identity Use "phone-context" Add user=phone Use + for International	By Source IP address	Media Allow Empty INVITE Send Empty re-INVITE Allow To Tag Change P-Early-Media Support Send SilenceSupp=Off Force Early Direct Media Media Connection Preservation Indicate HOLD	None Disabled
	Use PAI for Privacy Use Domain for PAI Swap From and PAI/Diversion Caller ID from From header Send From In Clear Cache Auth Credentials User-Agent and Server Headers Send Location Info	V V Never	Call Control Call Initiation Timeout (s) Call Queuing Timeout (mins) Service Busy Response on No User Responding Send Action on CAC Location Limit Suppress Q.850 Reason Header Emulate NOTIFY for REFER No REFER if using Diversion	4 - 5 - 486 - Busy Here 408-Request Timeout Allow Voicemail

2.5 Extension

In this section, an example of an Avaya IP Office Extension will be illustrated. In the interest of brevity, not all users and extensions will be presented, since the configuration can be easily extrapolated to other users and extensions.

Step	Action	Results
1	Right click on Extension	
2	Select New > Select H323 or SIP	
3	Select the Extension Tab	
4	Go to Next Table	

Following is an example of extension 1502; this extension corresponds to an H.323 extension:

IP Offices	E	H.323 Extension: 8012 1502
IP500V2 Main	Extension VoIP	
⊕-~~ Control Unit (4)	Extension ID	8012
	Base Extension	1502
40 102 1102 40 103 1103	Phone Password	
40 104 1104 40 105 1105	Confirm Phone Password	
- 40 106 1106 - 40 107 1107	Caller Display Type	On 👻
- 40 108 1108	Reset Volume After Calls	
- 40 110 1110 - 40 110 1110	Device Type	Avaya 9641
-\$ 112 1112	Location	Automatic
	Fallback As Remote Worker	Auto 🗸
- 116 1116	Module	0
	Port	0
- 40 26 1504 - 40 27 1505	Disable Speakerphone	

Step	Action	Results
5	Select the V OIP tab	
6	Use default values on V oIP tab	By default, all IP phones (SIP and H.323) will use the system default codec selection configured under the System VoIP tab (Section 2.2.3), unless configured otherwise for a specific extension by selecting Custom under Codec Selection o n the screenshot shown below.
7	Procedure Completed	

The example below shows the codecs used for IP phones (SIP and H.323).

IP Offices	E	H.323 E	xtension: 8012 1502	
IP500V2 Main	Extension VolP			
Control Unit (4)	IP Address	0 . 0 . 0 . 0		VoIP Silence Suppression
Extension (48)				Enable Faststart for
- 102 1102	MAC Address	00 00 00 00 00 00		non-Avaya IP phones
- 4 103 1103				Out Of Band DTME
- 40 104 1104	Codec Selection	System Default	•	
- 105 1105		Unused	Selected	Local Tones
		G.722 64K	G.711 ULAW 64K	III Allow Direct Made Dath
			G.711 ALAW 64K	Allow Direct Media Path
- 40 100 1100			G.729(a) 8K CS-ACELP	
- 110 1110		T	G.723.1 6K3 MP-MLQ	
@ 111 1111				
- 40 112 1112		<<<		
40 113 1113				
- 40 114 1114				
- 4 115 1115				
- 4 116 1116				
8008 1 501				
8012 1502				
-40 23 1303	Reserve License	None	-	
-40 27 1505	TOMANDA	D.C. H		
- 40 28 1506	TDM->IP Gain	Default	•	
- 4 29 1507	IP->TDM Gain	Default	•	
- 40 30 1508				
- 40 31 1509	Supplementary Services	None	•	
- 4 32 1510				
- 8009 1540	Media Security	Same as System (Disabled)	•	
- 💊 8011 1541				

2.6 Users

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in Section 2.4.

Step	Action	Results
1	Navigate to User in the left Navigation pane	
2	Select the name of the user to be modified in the center Group pane	
3	Go to Next Table	

In the example below, the name of the user is "H323 Ext 1502":

IP Offices	E	H323 ext 1502: 1502
1115 Extn1115	User Voicemail DND Shor	rt Codes Source Numbers Telephony Forwarding Dial In Voice Recording Button Programming
1110 Extn1110		
1504 Exten1504	Name	H323 ext 1502
-1505 Extn1505	Password	
-1506 Extn1506	Passifold	
1507 Extn1507	Confirm Password	••••
1508 Extn1508		
1509 Extn1509	Unique Identity	
1510 Extn1510	Conference PIN	
1596 Extn1596	Conference Part	
1597 Extn1597	Confirm Audio Conference PIN	
1598 Extn1598		
1599 Extn1599	Account Status	Enabled
1600 Extn1600	Eull Name	
1601 Extn1601	Full Name	
1603 Extn1603	Extension	1502
	Email Address	
- 1541 H323 ext1541	Locale	▼
-1501 IP H323 9640	Drivity	¢ _
- 1552 Rem W SIP 1552	Phoney	· ·
-1570 sip1570	System Phone Rights	None
-1571 sip1571		
-1572 sip1572	Profile	Rasic User
- 1575 sip1575 - 1576 sip1576	- Come	Recentionist
1580 sip1580		Enable Softshope
1595 Soft H323 1595		Enable one-X Portal Services
1557 WebRTC1557		Enable one X TeleCommuter
B- Group (1)		Enable Remote Worker
Short Code (69) Service (0)		
⊕-4, RAS (1)		
Incoming Call Route (3) WAN Port (0)		Enable Mobile VoIP Client
Directory (0)		Send Mobility Email
Time Profile (0) Frewall Profile (1)		Web Collaboration
IP Route (5) Account Code (0)		Exclude From Directory
License (90)		
User Rights (8)	Device Type	Avaya 9641
⊕- ¥ ARS (2)		
Authorization Code (0)		

In the example below, the name of the user is "Soft SIP 1550".

Step	Action	Results
4	This is an Avaya IP Office Softphone user, set the Profile to Power User	
5	Ceck Enable Softphone	
6	Go to Next Table	

IP Offices		Ξ						Sof	t SIP 1550	: 1550			
1115 Extn1115	^	User Vo	icemail	DND	Short	Codes	Source Numbe	rs Telephony	Forwarding	Dial In	Voice Recording	Butto	on Programming
1503 Extn1503analog		Name				Soft SIP 1	550						
1504 Extn1504													
1505 Extn1505		Password				••••							8
1507 Extn1507		Confirm Pa	assword			••••							
1508 Extn1508		Unique Ide	untitu										
1509 Extn1509		Onique tue	anny										
1596 Extn1596		Conference	e PIN										
1597 Extn1597		Confirm A	udio Con	ference	PIN								
1598 Extn1598													1
1599 Extn1599		Account St	tatus			Enabled						•	J
-1601 Extn1601		Full Name											
1602 Extn1602		Extension				1550							
1502 H323 ext 1502		Exection											
1542 H323 ext 1542		Email Addr	ress										
- 1541 H323 ext1541		Locale				[-]
1540 IP H323 1540		D. J. J. J.								_			,
- 1552 Rem W SIP 1552		Priority				2						•	J
1570 sip1570		System Pho	one Right	ts		None						•	
1571 sip1571												_	
1572 sip1572		Profile				Power U	ser					•	
						Recep	tionist						
1580 sip1580					Г	Enabl	e Softnhone						
1595 Soft H323 1595					L	E Crabi	C Sonphone						
-1557 WebRTC1557						Enabl	e one-X Portal	Services					
1558 WebRTC1558						Enabl	e one-X TeleCo	mmuter					
B Short Code (69)						Enabl	e Remote Work	er					
Service (0)	ш					Enabl	e Communicat	or					
Incoming Call Route (3)	ш					Enabl	e Mobile VoIP (Client					
WAN Port (0)						Send	Mobility Email						
Time Profile (0)						Web (Collaboration						
Firewall Profile (1)													
IP Route (5) Account Code (0)						Exclus	de From Directo	ory					
License (90)								-					
- Will Tunnel (0)		Device Two	e		-	Unknow	n SIP device						
User Rights (8)		Cence typ		_ ⊲		- Introw	and active						
	11												
Authorization Code (0)													

Step	Action	Results
7	Select the Voic email Tab	The screen shows the Voicemail tab for the user with extension 1502. Voicemail password can be configured using the Voicemail Code and Confirm Voicemail Code parameters.
		In the verification of these Application Notes, incoming calls from Charter to this user were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved
		successfully. Voice mail navigation and retrieval were performed locally and from PSTN telephones to test DTMF using RFC 2833.
8	Go to Next Table	

IP Offices	H		H323 e	ext 1502: '	1502		
1115 Extn1115	User Voicemail DND Short	Codes Source Numbers	Telephony F	Forwarding D	Dial In Voi	ice Recording	Button Programming
	Voicemail Code				Vo	oicemail On	
1505 Extn1505	Confirm Voicemail Code				E Ve	oicemail Help	
1506 Extn1506	Voicemail Email				📃 Vo	oicemail Ringb	ack
1508 Extn1508					🗌 Vo	oicemail Email	Reading
1510 Extn1510					🗐 UI	MS Web Servic	es
-1596 Extn1596							
1597 Extn1597							
1599 Extn1599	- Voicemail Email						
1600 Extn1600	Contraction of Contraction	Alert					
1601 Extn1601	Copy Porward	Alert					
1603 Extn1603	DTMF Breakout						
-1502 H323 ext 1502 1542 H323 ext 1542	Reception/Breakout (DTMF 0)	System Default ()			•		
- 1541 H323 ext1541	0						
- 1540 IP H323 1540	Breakout (DTMF 2)	System Default ()			-		
-1552 Rem_W SIP 1552	1						
	Breakout (DTMF 3)	System Default ()			-		
-1572 sip1572		-,					
-1575 sip1575							

Step	Action	Results
9	Select the Mobility tab	
10	Check Mobility Features box	
11	Check Mobile Twinning box	
12	Configure Twinned Mobile Number field with the number to dial to reach the twinned telephone,	
	including the dial access code "9", in this case 917864571234.	
13	Other options can be set according to customer requirements	
14	Go to next Table	

IP Offices	Z			H32	23 ext 1	502: 1502*		
1115 Extn1115 1116 Extn1116	Dial In	Voice Recording	Button Programming	Menu Programming	Mobility	Group Membership	Announcements	SIP
1503 Extn1503analog 1504 Extn1504 1505 Extn1505 1506 Extn1506 1507 Extn1507 1508 Extn1508 1509 Extn1508	Twinr Maxin	ernal Twinning ned Handset num Number of Ca vin Bridge Appearar	<none> 1 nces</none>					¥
1510 Extn1510 1596 Extn1596 1597 Extn1597 1598 Extn1598 1599 Extn1599	Tw Tw	vin Coverage Appea vin Line Appearance bility Features	srances IS]				
1600 Extn1600 1601 Extn1601 1602 Extn1602 1603 Extn1603	₩ Me Tv (ii Tv	obile Twinning winned Mobile Nun ncluding dial access winning Time Profil	nber s code) 91786457123	4				-
1502 H323 ext 1502 1542 H323 ext 1542 1541 H323 ext 1542 1541 H323 ext 1541 1540 IP H323 1540	M	lobile Dial Delay (se lobile Answer Guard	c) 2 d (sec) 0 *					
- 1501 IP H323 9640 - 1552 Rem_W SIP 1552 - 1570 sip1570 - 1571 sip1571		Hunt group calls of Forwarded calls el	ligible for mobile twir igible for mobile twin	ning				
- 1572 sip1572 - 1575 sip1575 - 1576 sip1576 - 1580 sip1580	in on	e-X Mobile Client						
1550 Soft SIP 1550	M 1	obile Callback						

Step	Action	Results
15	Select the Button Programming tab on the user	
16	select the button to program to turn Mobil Twinning on and off	
17	Click on Edit > Emulation > Twinning	In the sample below, button 4 was programmed to turn Mobil Twinning on and off on user 1502.
18	Go to next Table	

IP Offices		i.	2						H32	23 ext 1	502: 1502*
1115 Extn1115 1116 Extn1116	^	ſ	Dial In	Voi	ce Recording	Button Programming	Men	u Progr	amming	Mobility	Group Membership
1503 Extn1503analog			Button		Label	Action		Action	n Data		
1504 Extn1504			1			Appearance		a=			
			2			Appearance		b=			
			3			Appearance		c=			
1508 Extn1508			4			Twinning					
1509 Extn1509			5								
1510 Extn1510			6								
1597 Extn1597			7								
			8								
			9								
1600 Extn1600			10								
1601 Extn1601			11								
1602 Extn1002			12								
1502 H323 ext 1502			13								
1542 H323 ext 1542			14								
			15								

Step	Action	Results
19	Select SIP tab	the values entered for the SIP Name and Contact fields are used as the user part of the SIP URI in the "From" and "Contact" headers for outgoing SIP trunk calls. In addition, these settings are used to match against the SIP URI of incoming calls without having to enter this number as an explicit SIP URI for the SIP line (Section 2.4)
20	The SIP Name and Contact are set to one of the DID numbers assigned to the enterprise by Charter.	In the example, DID number 3031231273 was used.
21	The SIP Display Name (Alias) parameter can optionally be configured with a descriptive name.	
22	If all calls involving this user should be considered private, then the Anonymous box may be checked to withhold the Caller ID information from the network.	
23	Procedure Completed	

IP Offices	H323 ext 1502: 1502*
1115 Extn1115 1116 Extn1116	Dial In Voice Recording Button Programming Menu Programming Mobility Group Membership Announcements SIP Personal Directory
	SIP Name 3031231273
1505 Extn1505	SIP Display Name (Alias) H323 ext 1502
-1507 Extn1507	Contact 3031231273
1510 Extn1510 1596 Extn1596	Anonymous
1597 Extn1597	
1598 Extn1598	
1600 Extn1600 1601 Extn1601	
1602 Extn1602	
1502 H323 ext 1502	
-1542 H323 ext 1542	

2.7 Incoming Call Route

An incoming call route maps inbound DID numbers on a specific line to internal extensions, hunt groups, short codes, etc., within the IP Office system.

In a scenario like the one used for the compliance test, only one incoming route is needed, which allows any incoming number arriving on the SIP trunk to reach any predefined extension in IP Office. The routing decision for the call is based on the parameters previously configured for **Call Routing Method** and **SIP URI (Section 2.4.5)** and the users **SIP Name** and **Contact**, already populated with the assigned Charter DID numbers **(Section 2.6)**.

Step	Action	Results
1	From the left Navigation pane, right-click on Incoming Call Route and select New.	
2	On the Details pane (not shown), Select the Standard tab	
3	Set Bearer Capacity to Any Voice.	
4	Set the Line Group ID to the incoming line group of the SIP line defined in Section 5.4.	
5	Default values may be used for all other parameters.	
6	Go to Next Table	

IP Offices				17
IP Offices BOOTP (5) Coperator (3) IP500V2 Main System (1) Control Unit (4) Extension (48) User (50) Group (1) Short Code (69) Service (0) RAS (1) Incoming Call Route (3)	Standard Bearer Ca Line Grou Incoming Incoming Locale	Voice Recording pability p ID Number Sub Address CLI	Destinations Any Voice 17	17 • •
 Inconning Call Rodre (3) I7 0 0 WAN Port (0) Directory (0) Time Profile (0) Firewall Profile (1) Firewall Profile (0) Firewall Profile (1) Firewall Profile (0) Firewall Profile (1) Firewall Profile (0) Firewall Profile (1) Firewall Profile (1) Firewall Profile (0) Firewall	Locale Priority Tag Hold Mus Ring Tone	ic Source e Override	1 - Low System Source None	•

Step	Action	Results
7	Under the Destinations tab, enter "." for the Default Value .	This setting will allow the call to be routed to any destination with a value on its SIP Name f ield, entered on the SIP tab of that User , which matches the number present on the user part of the incoming Request URI.
8	Click OK to commit	
9	Procedure Completed	

IP Offices	Ш		1	17	
BOOTP (5)	Stand	ard Voice Recording Destinations			
E		TimeProfile	Destination		Fallback Extension
E System (1)	+	Default Value		-	
E-C Line (25)					
E-4 Extension (48)					
🕑 📲 User (50)					
🕀 🎆 Group (1)					
Service (0)					
B- RAS (1)					
E-C Incoming Call Route (3)					
42 0					

2.8 Outbound Call Routing

For outbound call routing, a combination of system short codes and Automatic Route Selection (ARS) entries are used. With ARS, features like time-based routing criteria and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. While detailed coverage of ARS is beyond the scope of these Application Notes, and alternate routing was not used in the reference configuration, this section includes some basic screen illustrations of the ARS settings used during the compliance testing.

2.8.1 Short Codes and Automatic Route Selection

Step	Action	Results
1	To create the short code used for ARS right-click on $\ensuremath{\textbf{Short}}$ Code in the $\ensuremath{\textbf{Navigation}}$ pane	
2	Select New	The screen below shows the creation of the short code 9N use d in the reference configuration. When the Avaya IP Office users dialed 9 plus any number N, calls were directed to Line Group 17 which configurable via ARS.
3	In the Code field, enter the dial string which will trigger this short code. In this case, 9N was used (note that the semi-colon is not used here).	
4	Set Feature to Dial.	This is the action that the short code will perform.
5	Set Telephone Number to N .	The value N represents the number dialed by the user after removing the 9 prefix. This value is passed to ARS.
6	Set the Line Group ID to 50: Main to be directed to Line Group 17, which is configurable via ARS.	
7	Set the Locale to United States (US English).	
8	Click the OK to commit.	
9	Go to Next Table	

IP Offices	Ш		9N: Dial
9X *41	Short Code		
9x *43	Code	9N	
9x *44 9x *45*N#	Feature	Dial	
9× *46 9× *47	Telephone Number	Ν	
9× *48 9× *49	Line Group ID	50: Main	
9× *50	Locale	United States (US English)	
•••• 9× *52	Force Account Code		
9× *53*N# 9× *55	Force Authorization Code		
9× 8N: 9× 9N 9× FNE00			
- 🍪 Service (0)			

The following screen shows a sample ARS configuration for the route **50: Main**. Note the sequence of **X**'s used in the **Code** field of the entries to specify the exact number of digits to be expected, following the access code and the first set of digits on the string. This type of setting results in a much quicker response in the delivery of the call by IP Office.

Step	Action	Results
10	To create a short code to be used for ARS, select ARS > 50 : Main on the Navigation pane	
11	Select New	
12	In the Code field, enter the dial string which will trigger this short code. In this case, 1 followed by 10 X 's to represent the exact number of digits.	
13	Set Feature to Dial.	This is the action that the short code will perform.
14	Set Telephone Number to 1N .	The value N represents the additional number of digits dialed by the user after dialing 1 (The 9 will be stripped off).
15	Set the Line Group ID to the Line Group number being used for the SIP Line, in this case Line Group ID 17 was used.	
16	Set the Locale to United States (US English).	
17	Click the OK to commit.	
18	Procedure Completed	

Edit Short Code				
Code	1XXXXXXXXXX] [ОК
Feature	Dial	•		
Telephone Number	1N			Cancel
Line Group ID	17	•		
Locale	United States (US English)	•		
Force Account Code			•	
Force Authorization Code				

The following screenshot shows the ARS dial pattern entry after it was added.

IP Offices	E			Main		C
B-R BOOTP (5)	ARS					
e-≂ IP500V2 Main e-≂ System (1)	ARS Route ID	50		Secondary Dial tone		
⊕-17 Line (25) ⊕-∞ Control Unit (4)	Route Name	Main		SystemTone	•	
	Dial Delay Time	System Default (4)	*	Check User Call Barring		
	Description					
Incoming Call Route (3) WAN Port (0) Directory (0)	In Service	V		Out of Service Route	<none></none>	•
Time Profile (0) Green Profile (1) Firewall Profile (1) Free Profile (5)	Time Profile	<none></none>	Ŧ	Out of Hours Route	<none></none>	•
Account Code (0)		Ţ				
- United (0)	Code	Telephone Number	Feature	Line Group ID	^	Add
B-XARS (2)	16X	16N	Dial	17		
S0: Main	1X000000000	1N	Dial	17		Remove
51: Outbound Fax	210000000	2N	Dial	17	8 r	5.04
- Location (0)	411	411	Dial	17		Edit
Authorization Code (0)	611	611	Dial	17		
	60000000	6N	Dial	17		
	81XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX	81N	Dial	22	*	
	Alternate Route Priorit	y Level 3 J ime 30		Alternate Route	<none></none>]

2.9 Save Configuration

When desired, send the configuration changes made in Avaya IP Office Manager to the Avaya IP Office server in order for the changes to take effect.

Step	Action	Result
1	Navigate to File > Save Configuration in the menu bar at the top left of the screen	Save the configuration performed in the preceding sections. Once the configuration is validated, a screen similar to the following will appear, with either the Merge or the Immediate radio button chosen based on the nature of the configuration changes made since the last save. Note that clicking OK may cause a service disruption due to system reboot.
2	Click OK if desired	
3	Procedure Completed	

Save Configuration	
IP Office Settings	
IP500V2 Main	
Configuration Reboot Mode	
Merge	
Immediate	
When Free	
Timed	
Reboot Time	
09:56	
Call Barring	
Incoming Calls	
Outgoing Calls	
OK Cancel	Help

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