

How to Configure the Avaya IP Office 6.1 for use with Integra Telecom SIP Solutions

Overview

This document provides a reference for configuration of the Avaya IP Office to connect to Integra Telecom SIP Trunks. The document covers a basic setup with required steps for interoperability with Integra Telecom only.

Test Network Setup

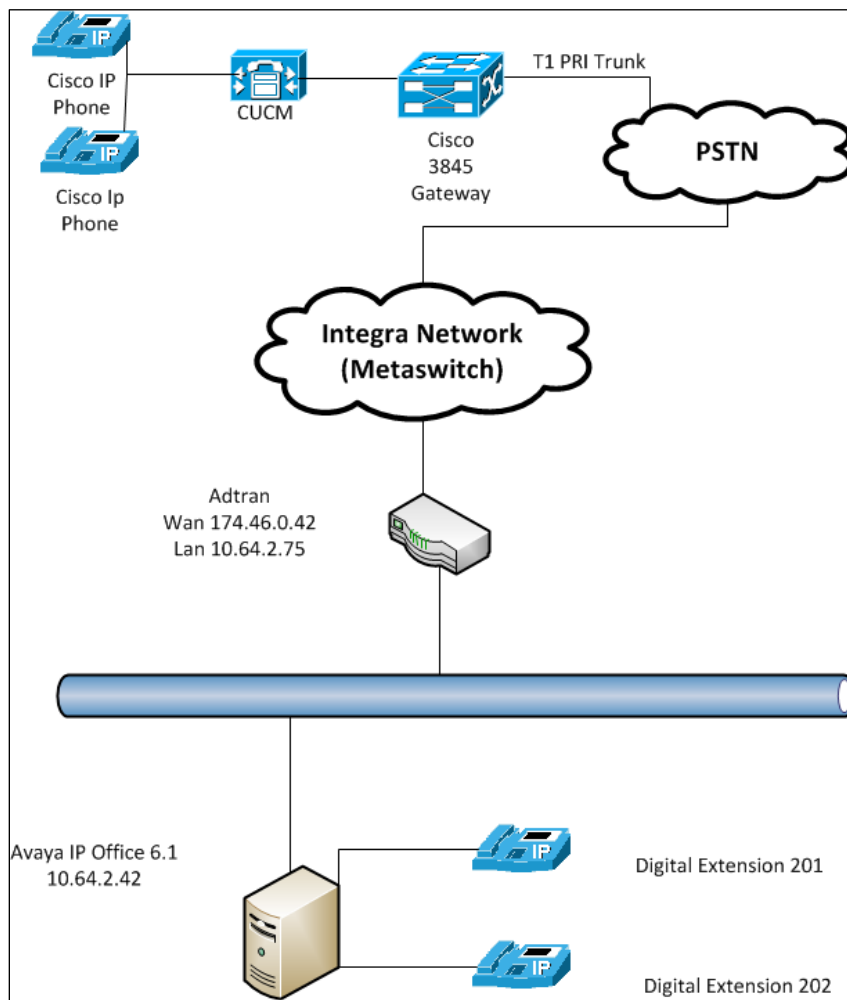


Figure 1 Avaya IP Office Network Layout

Devices Under Test

Devices Under Test	Version
Avaya IP Office (IP-PBX)	6.1
Adtran NetVanta 3305	17.09.02.00

3rd Party Components and their Versions

3 rd Party Product/Components	Version
Cisco 6509 switch	12.2(33)SXH1

Limitations

These are the Avaya IP Office Limitations:

- Avaya IP Office does not support Blind Transfer.

Configuration in the Avaya IP Office (IP-PBX)

This section shows the configuration for Avaya IP Office *to Integra Network passing thru Adtran*. All values provided are for example purposes only and actual values will be provided by Integra Implementation Team.

The following steps show a quick sequence on how to configure Avaya IP Office with Adtran.

1. **System settings** (Detailed Information in section 2.1)
2. **Sip Trunk**(Detailed Information in section 2.2)
3. **Incoming Call Routing** (Detailed Information in section 2.3)
4. **Outgoing Call Routing** (Detailed Information in section 2.4)
5. **Configure T38/G711 Fax** (Detailed information in section 2.5)

System Settings

Configure the system settings by navigating to System → LAN1 → LANSettings. This is the IP address and subnet mask for the Avaya IP Office. The values below are for example purposes only and the actual values will be provided by Integra Implementation Team

1. Set the IP Address; 10.64.2.42
2. Set the IP Mask; 255.255.0.0

The figure below shows "LAN Settings" screen

The screenshot displays the 'LAN Settings' configuration page for LAN1. The 'IP Address' field is set to 10.64.2.42 and the 'IP Mask' field is set to 255.255.0.0. The 'Primary Trans. IP Address' is 0.0.0.0. The 'RIP Mode' is set to 'None'. The 'Enable NAT' checkbox is unchecked. The 'Number Of DHCP IP Addresses' is set to 200. The 'DHCP Mode' is set to 'Disabled'. There is an 'Advanced' button at the bottom right.

Figure 2 LAN Settings

The next screen is System → LAN1 → VoIP. Enable Sip Trunks and Sip Registrar for the Avaya IP Office.

1. Set the H323 Gatekeeper Enable with a check mark in the box.
2. Set the SIP Trunks Enable with a check mark in the box.
3. Set the SIP Registrar Enable with a check mark in the box.
4. Set the DSCP mask to 24

The figure below shows “Voip” screen.

The screenshot displays the configuration interface for VoIP settings. The top navigation bar includes tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, and SMTP. Under the LAN1 tab, there are sub-tabs for LAN Settings, VoIP, Network Topology, and SIP Registrar. The VoIP sub-tab is active, and the SIP Registrar sub-tab is selected. A red box highlights the following checked options: H323 Gatekeeper Enable, SIP Trunks Enable, and SIP Registrar Enable. Below these, there are sections for H323 Auto-create Extn, H323 Auto-create User, and Enable RTCP Monitoring On Port 5005. The RTP Port Number Range is set to Port Range (Minimum) 49152 and Port Range (Maximum) 53246. The DiffServ Settings section includes DSCP (Hex) 88, FC, DSCP Mask (Hex) 60, and SIG DSCP (Hex) 46. A red box highlights the DSCP Mask field, which is set to 24. The DHCP Settings section includes Primary Site Specific Option Number (SSON) 176, Secondary Site Specific Option Number (SSON) 242, VLAN Not Present, 1100 Voice VLAN Site Specific Option Number (SSON) 232, and 1100 Voice VLAN IDs. The RTP keepalives section includes Scope Disabled, Periodic timeout 0, and Initial keepalives Disabled.

Figure 3 Voip

The next screen is System → LAN1 → Network Topology.

1. Confirm that Firewall/NAT Type is set to Unknown.
2. Confirm that Run STUN on startup is not checked.

The figure below shows “Network Topology” screen.

The screenshot shows the 'Network Topology' configuration window. It has tabs for 'LAN Settings', 'VoIP', 'Network Topology', and 'SIP Registrar'. The 'Network Topology' tab is active. The 'Network Topology Discovery' section contains the following fields:

- STUN Server IP Address: 75 . 101 . 138 . 128
- STUN Port: 3478
- Firewall/NAT Type: Unknown (highlighted with a red box)
- Binding Refresh Time (seconds): 30
- Public IP Address: 0 . 0 . 0 . 0
- Public Port: 1095

At the bottom right, there are two buttons: 'Run STUN' and 'Cancel'. Below the 'Run STUN' button, there is a checkbox labeled 'Run STUN on startup' which is unchecked (highlighted with a red box).

Figure 4 Network Topology

The next screen is System → LAN1 → SIP Registrar. The SIP Registrar tab is used to define the Domain Name, port numbers for signaling on the Avaya IP Office. The values below are for example purposes only and the actual values will be provided by Integra Implementation Team.

1. Set the Domain Name; vancwasn.integravoip.net
2. Set the Layer 4 Protocol; Both TCP & UDP
3. Set the TCP port; 5060
4. Set the UDP port; 5060
5. Confirm that Auto-create Extn/User is unchecked

The figure below shows “SIP Registrar” screen.

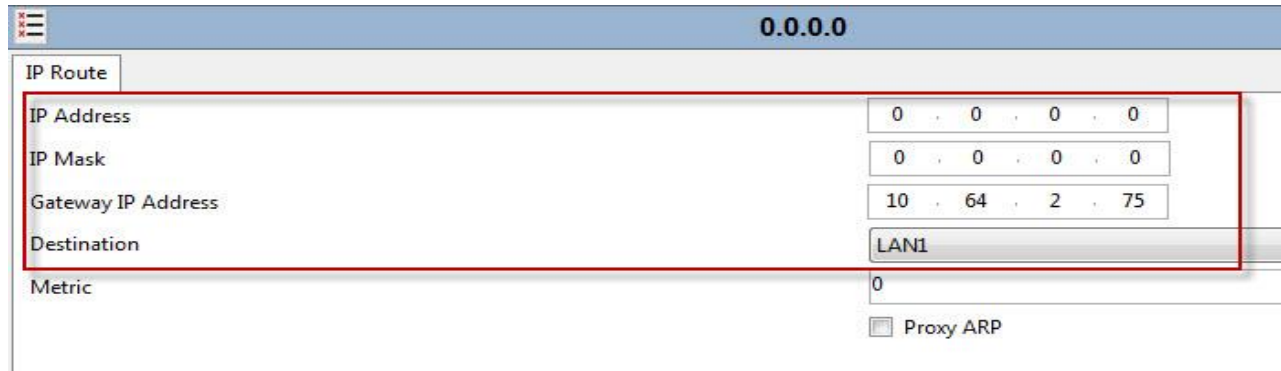
LAN Settings	VoIP	Network Topology	SIP Registrar
Domain Name	vancwasn.integravoip.net		
Layer 4 Protocol	Both TCP & UDP ▼		
TCP Port	5060 ▲▼		
UDP Port	5060 ▲▼		
Challenge Expiry Time (secs)	10 ▲▼		
Auto-create Extn/User	<input type="checkbox"/>		

Figure 5 SIP Registrar

The next section completes the System setup. The next screen is System → IP Route → 0.0.0.0. The 0.0.0.0 tab is used to define the default gateway on the Avaya IP Office. The Gateway IP Address must be the LAN side IP Address of Adtran. The values below are for example purposes only and the actual values will be provided by Integra Implementation Team.

1. Set IP Address; 0.0.0.0
2. Set IP Mask; 0.0.0.0
3. Set Gateway IP Address; 10.64.2.75.
4. Set the Destination to LAN1

The figure below shows “0.0.0.0” screen.



0.0.0.0	
IP Address	0 . 0 . 0 . 0
IP Mask	0 . 0 . 0 . 0
Gateway IP Address	10 . 64 . 2 . 75
Destination	LAN1
Metric	0
<input type="checkbox"/> Proxy ARP	

Figure 6 IP Route 0.0.0.0

Sip Trunk

Configure the sip trunk in Avaya IP Office by navigating to Line→17 →SIP Line. The ITSP Domain Name must match the Domain Name entered in [Figure 5](#). The values below are for example purposes only and the actual values will be provided by Integra Implementation Team.

1. Set the ITSP Domain Name; vancwasn.integravoip.net
2. Set the In Service check box; If not checked then check
3. Set the Call Routing Method: Request URI
4. Set the REFER supported check box; If not checked then check

The figure below shows “Sip Line” screen.

The screenshot shows the configuration for SIP Line 17. The 'SIP URI' tab is active. Key fields include:

- Line Number: 17
- ITSP Domain Name: vancwasn.integravoip.net
- In Service:
- Use Tel URI:
- Check OOS:
- Call Routing Method: Request URI
- Originator number for forwarded and twinning calls: (empty)
- Send Caller ID: None
- REFER Support:
- Incoming: Auto
- Outgoing: Auto

Figure 7 Sip Line

The next screen is Line→17 →Transport. The ITSP Proxy Address must be the same as the Gateway IP Address entered in [Figure 6](#). The values below are for example purposes only and the actual values will be provided by Integra Implementation Team.

1. Set the ITSP Proxy Address to 10.64.2.75
2. Set the Layer 4 Protocol; UDP
3. Set the Use Network Topology Info;None
4. Set the Send Port; 5060
5. Set the Listen Port; 5060
6. Set the Calls Route via Registrar check box; if not checked then check

The figure below shows “Transport” screen.

The screenshot shows the configuration for the Transport tab of SIP Line 17. Key fields include:

- ITSP Proxy Address: 10.64.2.75
- Network Configuration:
 - Layer 4 Protocol: UDP
 - Send Port: 5060
 - Use Network Topology Info: None
 - Listen Port: 5060
- Explicit DNS Server(s): 0 . 0 . 0 . 0 | 0 . 0 . 0 . 0
- Calls Route via Registrar:
- Separate Registrar: (empty)

Figure 8 Transport

The next screen is Line→17 →SIP URI. The SIP URI tab is has multiple uses. For the SIP trunk configuration this implementation is using Channel 1. This SIP URI tab only has two entries; Channel's 1 and 2 there could be more entries if the deployment has more did numbers. There will be more explanation when the SIP URI is addressed in sections 2.3 and 2.4. The values below are for example purposes only and the actual values will be provided by Integra Implementation Team.

The figure below shows "SIP URI " screen.

The screenshot shows a web interface for configuring SIP Line 17. The 'SIP URI' tab is selected, displaying a table with the following data:

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	19 17	<None>	3608529756	3608529756	3608529756	None	1: 3608529756	10
2	20 16	<None>	3608529762	3608529762	3608529762	None	1: 3608529756	10

Figure 9 SIP URI

The next screen is Line→17 →VoIP.

1. Set the Compression Mode; Check G711 ULAW 64K and G729(a) 8K CS-ACELP
2. Set the Re-invite Supported check box; if not checked then check
3. Set the DTMF Support; RFC2833

The figure below shows "VoIP" screen.

Figure 10 VoIP

The next screen is Line→17 →SIP Credentials. The SIP Credentials are supplied by Integra. An Username/Authentication Name and Password will be issued as the login credentials. This information will be entered in the SIP Credentials screen. Each time a new sip credentials is created it is referenced by an index number. For purpose of this example Index 1 was created. The values below are for example purposes only and the actual values will be provided by Integra Implementation Team

1. Set the User name; XXXXXXXXXXXX
2. Set the Authentication Name; XXXXXXXXXXXX
3. Set the Password; yyyyyyyyyyyyyyy
4. Set the Expiry; 61
5. Set the Registration required check box; if not checked then check

The figure below shows "SIP Credentials" screen.

SIP Line	Transport	SIP URI	VoIP	T38 Fax	SIP Credentials														
					<table border="1"> <thead> <tr> <th>Index</th> <th>UserName</th> <th>Authentication Name</th> <th>Contact</th> <th>Password</th> <th>Expiry</th> <th>Register</th> </tr> </thead> <tbody> <tr> <td>1</td> <td>XXXXXXXXXX</td> <td>XXXXXXXXXX</td> <td></td> <td>yyyyyyyyyy</td> <td>61</td> <td>True</td> </tr> </tbody> </table>	Index	UserName	Authentication Name	Contact	Password	Expiry	Register	1	XXXXXXXXXX	XXXXXXXXXX		yyyyyyyyyy	61	True
Index	UserName	Authentication Name	Contact	Password	Expiry	Register													
1	XXXXXXXXXX	XXXXXXXXXX		yyyyyyyyyy	61	True													

Figure 11 SIP Credentials

Incoming Call Routing

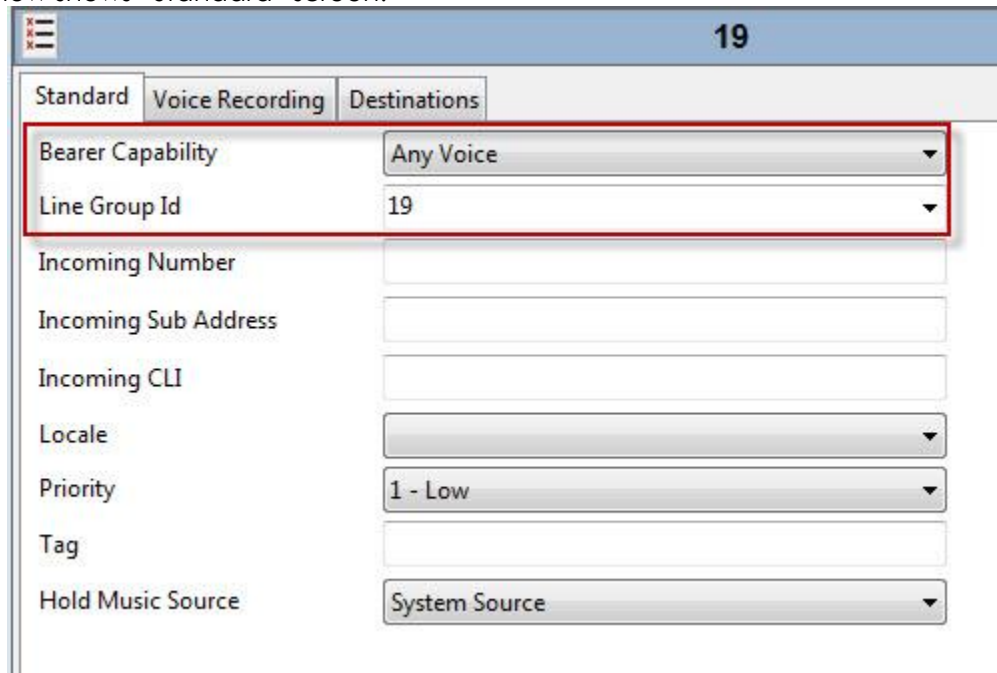
When Avaya IP Office is implemented using Integra Telecom as the Service Provider Integra provides a 10 digit User name which is a PSTN number. This PSTN number is registered with a SIP trunk, from Avaya IP office to Integra Telecom. For this example 360-852-9756 is used. When 360-852-9756 is dialed an Invite is sent to Avaya IP office. The information below is used in routing the call to the destination.

- Avaya IP office uses the Request URI as provisioned in [Figure 7](#) to route incoming calls.
- Avaya IP office uses Index number of "1" from SIP Credentials as provisioned in [Figure 11](#). The Index number of 1 is used as well as the number in the Request URI to find a match in the SIP URI [Figure 9](#)
- Avaya IP office uses SIP URI [Figure 9](#) to find the Incoming Group. In this example the Incoming Group is 19
- Avaya IP office uses Incoming Call route 19 [Figure 13](#) to find the extension to route the call to.

Configure the Incoming Call Route in Avaya IP Office by navigating to Incoming Call Route → 19 → Standard.

1. Set the Bearer Capability; Any Voice
2. Set the Line Group ID; 19

The figure below shows "Standard" screen.



19		
Standard	Voice Recording	Destinations
Bearer Capability	Any Voice	
Line Group Id	19	
Incoming Number		
Incoming Sub Address		
Incoming CLI		
Locale		
Priority	1 - Low	
Tag		
Hold Music Source	System Source	

Figure 12 Standard

The next screen is Incoming Call Route → 19 → Destinations. The Destination has a pull down for the extensions, ring group, auto attendant... that have been configured. In this example the call terminates to extension 202.

1. Set the Destination; 202

The figure below shows "Destinations" screen.

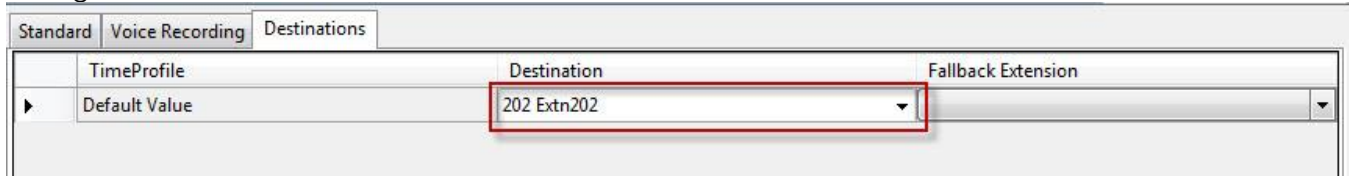


Figure 13 Destinations

Outgoing Call Routing

When using Avaya IP Office with Integra Telecom there are a couple of different options to configuring a user to make out bound calls.

The first method is for the user to show the caller id of the registration number. The second method is for the user to show a caller id other than that of the registration number.

1.1.1 Method 1 Outgoing Call Routing

For this example we will use extension 202, extension 202 is associated with user 202. When extension 202 dials 9 to access the trunk and either 1+10 or 10 digits. The terminating caller will see the Registration number as the caller id.

- The User → 202 → ShortCodes [Figure 14](#) is checked for a pattern match. If there is no pattern match in this Short Codes section then the system Short Code is checked.
- The Short Code → 9N; [Figure 15](#) is checked to see which Line Group the call should be sent to. For this example it should be sent to Line Group 17
- The SIP URI [Figure 9](#) is now searched for Outgoing Group 17
- The Outgoing Group 17 is associated with the Registration XXXXXXXXXXXX and uses the values shown in [Figure 16](#)
- Any user that makes an out bound call by dialing 9 + will show XXXXXXXXXXXX as the Calling party information

Configure/confirm the User 202 ShortCodes by navigating to User → 202 → ShortCodes. You can see that there is no match for the 9 that was dialed as the first digit by extension 202.

The figure below shows "ShortCodes" screen.

Extn202: 202			
User	Voicemail	DND	ShortCodes
Code	Telephone Number	Feature	Line Group Id
*FWD0	201=201	Forward Number	0
*FWD1	912142425995=912142425995	Forward Number	0

Figure 14 Extension ShortCodes

The next screen is Short Code →9N;. The Line Group ID is chosen based on the OutGoing Group that was entered in the SIP URI screen [Figure 9](#)

1. Set the Code; 9N;
2. Set the Feature;Dial
3. Set the Telephone Number; N
4. Set the Line Group ID; 17

The figure below shows "9N;" screen.

9N;: Dial	
Short Code	
Code	9N;
Feature	Dial
Telephone Number	N
Line Group Id	17
Locale	
Force Account Code	<input type="checkbox"/>

Figure 15 Short Code 9N;

The next screen is SIP URI *Figure 9*. If you highlight Channel 1 and click Edit you can see that:

1. Local URI is set to Use Credentials User Name
2. Contact is set to Use Credentials User Name
3. Display Name is set to Use Credentials User Name
4. Registration is set to Index 1 XXXXXXXXXX
5. Outgoing Group is set to 17

The figure below shows Channel 1 information

Edit Channel	
Via	<None>
Local URI	Use Credentials User Name
Contact	Use Credentials User Name
Display Name	Use Credentials User Name
PAI	None
Registration	1: XXXXXXXXXX
Incoming Group	19
Outgoing Group	17
Max Calls per Channel	10

Figure 16 Channel 1 Information

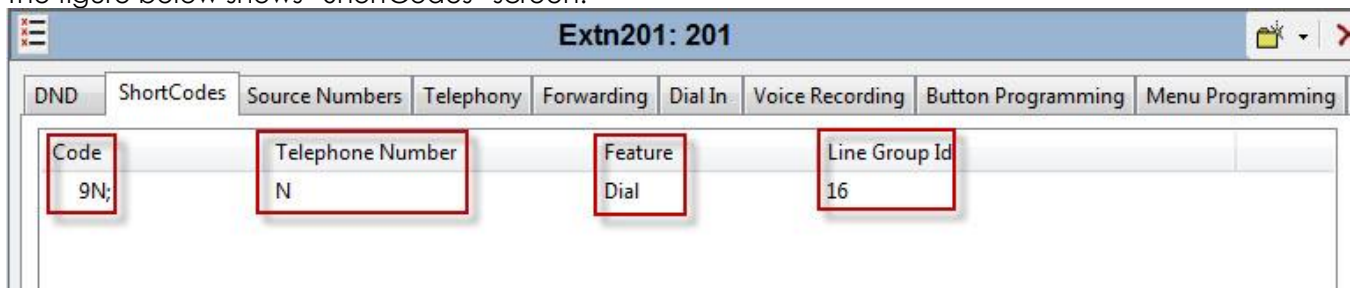
1.1.2 Method 2 Outgoing Call Call Routing

When the Avaya IP Office Administrator received the registration information from Integra Telecom, Integra Telecom supplied an additional DID number. This number is 360 852 9762. The Avaya IP Office extension 201 would like this number to be its caller id. To accomplish this the following actions were performed:

- The User → 201 → ShortCodes *Figure 17* is checked for a pattern match. If there is no pattern match in this Short Codes section then the system Short Code is checked. There is a pattern match in this users ShortCodes table
- The Short Code → 9N; *Figure 15* is skipped. For this example it should be sent to Line Group Id 16 as seen in *Figure 17*
- The SIP URI *Figure 9* is now searched for Outgoing Group 16
- The Outgoing Group 16 is associated with the Registration XXXXXXXXXX and uses the values shown in *Figure 19*
- Any user that makes an out bound call by dialing 9 + will show 3608529752 as the Calling party information

Configure/confirm the User 201 ShortCodes by navigating to User → 201 → ShortCodes. You can see that there is a match for the 9 that was dialed as the first digit by extension 201.

The figure below shows “ShortCodes” screen.



Code	Telephone Number	Feature	Line Group Id
9N;	N	Dial	16

Figure 17 ShortCodes 201

The next screen is Short Code →9N;. The Line Group ID is chosen based on the OutGoing Group that was entered in the SIP URI screen [Figure 9](#)

1. Set the Code; 9N;
2. Set the Feature;Dial
3. Set the Telephone Number; N
4. Set the Line Group ID; 17

The figure below shows “9N;” screen.

The screenshot shows a configuration window titled "9N; Dial". The window has a header bar with a menu icon and the title. Below the header, there is a "Short Code" section. The fields are as follows:

Code	9N;
Feature	Dial
Telephone Number	N
Line Group Id	17
Locale	
Force Account Code	<input type="checkbox"/>

Figure 18 System Short Code

The next screen is SIP URI [Figure 9](#) . If you highlight Channel 2 and click Edit you can see that:

1. Local URI is set to 3608529762
2. Contact is set to 3608529762
3. Display Name is set to 3608529762
4. Registration is set to Index 1 XXXXXXXXXX
5. Outgoing Group is set to 16

The figure below shows Channel 2 information

Edit Channel

Via	<None>
Local URI	3608529762
Contact	3608529762
Display Name	3608529762
PAI	None
Registration	1.
Incoming Group	20
Outgoing Group	16
Max Calls per Channel	10

Figure 19 Channel 2 information

Configure T38/G711 Fax

After following sections 2.1 – 2.3 the Avaya IP Office PBX will send and receive faxes via G711. To use T38 the following change must be made in Line → 17 → VoIP. Assign the directory number as you would with any other extension with the information in section 2.3.

1. Set the Fax Transport Support check box; If not checked then check

The figure below shows “VoIP” screen.

SIP Line - Line 17*

SIP Line	Transport	SIP URI	VoIP	T38 Fax	SIP Credentials
Compression Mode	Advanced	<input checked="" type="checkbox"/> G.711 ULAW 64K <input checked="" type="checkbox"/> G.729(a) 8K CS-ACELP <input type="checkbox"/> G.711 ALAW 64K <input type="checkbox"/> G.723.1 6K3 MP-MLQ	<input type="checkbox"/> VoIP Silence Suppression <input checked="" type="checkbox"/> Fax Transport Support <input checked="" type="checkbox"/> Re-invite Supported <input type="checkbox"/> Use Offerer's Preferred Codec	Call Initiation Timeout (s)	9
DTMF Support		RFC2833			

Figure 20 Sip Line Voip with Fax

The next screen is Line → 17 → T38 Fax. There are no changes required in this screen with the exception of item 1 below. Please confirm the settings in this screen are identical to the ones in the Avaya IP Office being configured.

1. Set the Use Default Values check box; If not checked then check

The figure below shows “T38 Fax” screen.

SIP Line - Line 17*					
SIP Line	Transport	SIP URI	VoIP	T38 Fax	SIP Credentials
T38 Fax Version	3				
Transport	UDPTL				
Redundancy					
Low Speed	0				
High Speed	0				
TCF Method	Trans TCF				
Max Bit Rate (bps)	14400				
EFlag Start Timer (msecs)	2600				
EFlag Stop Timer (msecs)	2300				
Tx Network Timeout (secs)	150				
<input checked="" type="checkbox"/> Use Default Values					
<input checked="" type="checkbox"/> Scan Line Fix-up <input checked="" type="checkbox"/> TFOP Enhancement <input type="checkbox"/> Disable T30 ECM <input type="checkbox"/> Disable EFlags For First DIS <input type="checkbox"/> Disable T30 MR Compression <input type="checkbox"/> NSF Override					
Country Code					0
Vendor Code					0

Figure 21 T38 Screen