



**IP Office 500
SIP Trunking Configuration Guide
using AccessLine SIP Trunking**



Version 1.1

August 30, 2012

Introduction

These Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider AccessLine and an Avaya IP Office 500 solution, Release 8.

The AccessLine SIP Trunking service referenced within these Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks directly as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

Prerequisite for Configuring Avaya Office 500 with AccessLine SIP Trunks

AccessLine Configuration Information

You should have the configuration information from Accessline before you start the install. This information will provide settings for the SIP connection(s).

The information you will receive from Accessline will be composed of the following information. Your order will be different from this information:

```
SIP Trunk ID: dgw5713954
Password: Akkbid24
SBC DNA Name: usbc.accessline.com
SBC IP Address: 64.28.113.10
SBX Port: 5060
```

Call processing has been setup to allow up to X simultaneous calls.

The following new numbers and routing have been provisioned in our systems to route via the SIP trunks to the PBX. Please ensure they are programmed in the customer's phone equipment to route appropriately and advise the customer of these new numbers:

2062645134

The following 911 callback number has been configured in our network as the default for this location: **2062645107**. Make sure it routes appropriately at the customer's site.

Avaya IP Office 500 Requirements

Avaya IP Office 500 must be using R8, a sip trunking license, and a VCM channel or module is required.

Firewall/Router Setup

Make sure that Port 6060 is forwarded from your router to your internal IP address of your IP office system.

SIP-based VOIP enabled PBX or SIP phones connected to AccessLine's Service via our SIP trunking service MUST be installed in a secure **trusted** zone behind a Firewall and not exposed to the public internet. This means the PBX or SIP phones should never be put into a router's DMZ (allows untrusted access).

The firewall must block all inbound internet (untrusted) traffic to the PBX or SIP phones. The firewall can be configured to allow inbound traffic from trusted devices from remote (satellite) locations.

You do not have to block outbound traffic from your private network to the internet, but Network-Address-Translation, or "NAT" must be enabled. NAT allows the AccessLine Service to send calls to the PBX or SIP Phones. If the firewall has multiple NAT settings, you must select the NAT setting that "Address Restricted" and not "Endpoint Independent". If you do want to limit outbound internet traffic on the firewall, then you need to open SIP related ports on the firewall to allow AccessLine's Service to function properly.

Protocol	Port(s)
UDP	6060
TCP	6061
UDP	30000 – 50000

Support

For technical support on the Avaya products described in these Application Notes visit <http://support.avaya.com> .

PBX Configuration and Setup

SIP Trunk CREATION:

Under "Line", right click and select New, SIP line. Enter the information as detailed below using your supplied Proxy Address and username and password from Accessline.

The screenshot shows the configuration window for "SIP Line - Line 18". The "ITSP Proxy Address" field is highlighted with a red box and contains the text "usbc.accessline.com". In the "Network Configuration" section, the "Layer 4 Protocol" is set to "UDP", "Use Network Topology Info" is set to "None", "Send Port" is set to "6060", and "Listen Port" is set to "6060". The "Send Port" and "Listen Port" fields are also highlighted with red boxes. Other fields include "Explicit DNS Server(s)" (0.0.0.0), "Calls Route via Registrar" (checked), and "Separate Registrar" (empty).

Select your codecs:

Default codec for AccessLine SIP Trunking is G729 Primary with G711 Secondary.

NOTE: If you are using channels in this Sip Trunk for Faxing, Select Fax Transport Support:G711 And Check Re-invite Supported.

The screenshot shows the "Codec Selection" section of the "SIP Line - Line 18" configuration window. The "Codec Selection" dropdown is set to "System Default". Below it, there are two lists: "Unused" (empty) and "Selected" (containing G.729(a) 8K CS-ACELP, G.711 ALAW 64K, G.711 ULAW 64K, and G.723.1 6K3 MP-MLQ). To the right of these lists are navigation buttons: >>>, ↑, <<<, ↓, and >>>. Below the lists, there are checkboxes for "VoIP Silence Suppression", "Re-invite Supported" (checked), "Use Offerer's Preferred Codec", "Codec Lockdown", and "PRACK/100rel Supported". At the bottom, there are dropdowns for "Fax Transport Support" (set to G.711), "Call Initiation Timeout (s)" (set to 4), and "DTMF Support" (set to RFC2833).

Create Sip Credentials:

Use the information provided by AccessLine to configure the credentials.

The screenshot displays the 'SIP Line - Line 18' configuration window. At the top, there are tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'SIP Credentials' tab is active, showing a table with the following data:

Index	UserName	Authentication Name	Contact	Expiry (mins)	Register
1	dgw5713954	dgw5713954	dgw5713954	60	True
2	dgw5752209	dgw5752209	dgw5752209	60	True

Buttons for 'Add...', 'Remove', and 'Edit...' are located to the right of the table. Below the table is the 'Edit SIP Credentials' dialog box, which is highlighted with a red border. The dialog box contains the following fields:

- User name: dgw5713954
- Authentication Name: dgw5713954
- Contact: dgw5713954
- Password: *****
- Expiry (mins): 60
- Registration required:

'OK' and 'Cancel' buttons are located to the right of the dialog box.

Create Sip URI:

This is where you create the incoming and outgoing Groups necessary for the next step.

Line			SIP Line - Line 18								
Line Number	Line Type	Line SubType	SIP Line	Transport	SIP URI	VoIP	T38 Fax	SIP Credentials			
Analogue Trunk			Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	Analogue Trunk		1	100 100	<None>	2062645134	2062645134	2062645134	None	1: dgw57139...	10
2	Analogue Trunk		2	101 101	<None>	2062645107	2062645107	2062645107	None	1: dgw57139...	10
3	Analogue Trunk		3	102 102	<None>	4252927166	4252927166	4252927166	None	2: dgw57522...	10
SIP Line											
18	SIP Line										

Line			SIP Line - Line 18								
Line Number	Line Type	Line SubType	SIP Line	Transport	SIP URI	VoIP	T38 Fax	SIP Credentials			
Analogue Trunk			Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	Analogue Trunk		1	100 100	<None>	2062645134	2062645134	2062645134	None	1: dgw57139...	10
2	Analogue Trunk		2	101 101	<None>	2062645107	2062645107	2062645107	None	1: dgw57139...	10
3	Analogue Trunk		3	102 102	<None>	4252927166	4252927166	4252927166	None	2: dgw57522...	10
SIP Line											
18	SIP Line										

Edit Channel

Via: <None>

Local URI: 2062645134

Contact: 2062645134

Display Name: 2062645134

PAI: None

Registration: 1: dgw5713954

Incoming Group: 100

Outgoing Group: 100

Max Calls per Channel: 10

Local URI, Contact and Display Name options:

- As Shown above, entering the DID Number. Use this option to create multiple channels to be routed based on incoming calls.
- “Use Internal Data” Use this if you have users or hunt groups that need to be directed based on incoming CLID. There are “SIP” Tabs for each user and hunt group where you would put the DID

that would correspond to that user or hunt group. Note that if you do not have any of these settings set to use internal data, there will be no SIP tab in your hunt groups or User sections. LOOK AT ALTERNATE SCENARIO AT THE END OF THIS DOCUMENT FOR FURTHER INSTRUCTIONS.

The above example screenshot shows the following scenario:

Sip Trunk 18:

- Channel 1 is our main incoming/outgoing company phone number. When an incoming call is received on that “line” (206-264-5134) the call will be routed to Incoming call group 100
- Channel 2 is our 911 number given to us by accessline. This number is not known to anyone on the outside.
- Channel 3 is our fax line

Incoming Call Route

Create your incoming call groups based on the Group ID’s you specified in the SIP URI

Using our current scenario, callers sent to call group 100 will be routed to the auto attendant, whereas a dispatcher calling in to the 911 caller ID will be routed to the EMERGENCY hunt group set up to ring at all the desks, bypassing the AA.

The image shows two screenshots of the Avaya IP Office configuration interface. The top screenshot shows the configuration for Incoming Call Route 100. The table below shows the data for this route:

Line Group ID	Incoming Number	Destination
100		AA:Auto Attendant
101		250 Emergency
102		207 Fax

The right-hand pane shows the Destinations tab for group 100, with the Default Value set to AA:Auto Attendant.

The bottom screenshot shows the configuration for Incoming Call Route 101. The table below shows the data for this route:

Line Group ID	Incoming Number	Destination
100		AA:Auto Attendant
101		250 Emergency
102		207 Fax

The right-hand pane shows the Destinations tab for group 101, with the Default Value set to 250 Emergency.

Calls coming in on group 102 will be routed to the fax at extension 207.

Emergency (911) Hunt Group

AccessLine configures a unique TN (telephone number) per customer location. The TN is registered with our E911 provider. When AccessLine receive a 911 call from the PBX, we forward to our E911 provider with the registered 911 TN as the CLI (regardless of what was passed from the PBX). The 911 TN is used by the Emergency responders to call back to the location.

In this example the inbound routing is setup to ring all phones.

HuntGroup			Collective Group Emergency: 250							
System Name	Name	Extension	Hunt Group	Queuing	Overflow	Fallback	Voicemail	Voice Recording	Announcem	
Collective Group										
	Emergency	250	Name		Emergency					<input type="checkbox"/> CCF
	Main	200	Extension		250					
			Ring Mode		Collective				No Ans	
			Hold Music Source		No Change					
			Agent's Status on No-Answer Applies To		None					
			User List							
			Extension	Name						
			<input checked="" type="checkbox"/>	203	Conference Room					
			<input checked="" type="checkbox"/>	206	Extn206					
			<input checked="" type="checkbox"/>	207	Fax					
			<input checked="" type="checkbox"/>	205	Home worker					
			<input checked="" type="checkbox"/>	202	Jackie Someone					
			<input checked="" type="checkbox"/>	299	Melanie HOME IP					
			<input checked="" type="checkbox"/>	201	Melanie Person					
			<input checked="" type="checkbox"/>	204	Rob					
			<input checked="" type="checkbox"/>	210	Robert					
			<input checked="" type="checkbox"/>	215	Sip User					

Outgoing Call Route

Automatic Route Selection (outgoing call route). All of our outgoing calls will be routed using line group ID 100, except emergency, which is set to 101.

ARS		Main*	
Name	Time Profile	ARS	
Main		Dial Delay Time	System Default (4) <input type="checkbox"/> Check User Call Barring
		In Service	<input checked="" type="checkbox"/> Out of Service Route
		Time Profile	<None> Out of Hours Route
Code	Telephone Number	Feature	Line Group ID
11	911	Dial Emergency	101
911	911	Dial Emergency	101
0N;	0N	Dial 3K1	100
1N;	1N	Dial 3K1	100
XN;	N	Dial 3K1	100
XXXXXXXXXN	N	Dial 3K1	100

Emergency (911) Setup

The system default has the outgoing call group (ARS) setup for 911 dialing.

Code	Telephone Number	Feature	Line Group ID
11	911	Dial Emergency	101
911	911	Dial Emergency	101

What is recommended is to create an Emergency hunt group that every user is a member of, and use the 911 number provided by AccessLine in the SIP tab of the hunt group.

Faxing:

Create a fax user, and create shortcodes that reference the outgoing group ID that you want to fax out on.

The screenshot shows two configuration panels. The left panel is titled 'User' and lists several users, with 'Fax' (extension 207) selected. The right panel is titled 'Fax: 207' and shows configuration options for 'ShortCodes'. Below this, a table lists the configured shortcodes.

User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording
				Code	Telephone Number	Feature	Line Group ID	
			9N;	N	Dial 3K1		102	
			91N	N	Dial 3K1		102	
			XXXXXXXXXXN	N	Dial 3K1		102	

Click on the physical extension port that the fax is connected to and select Fax Machine:

The screenshot shows two configuration panels. The left panel is titled 'Extension' and lists various extension types, with 'Analogue Extension' (extension 207) selected. The right panel is titled 'Analogue Extension: 7 207' and shows configuration options for 'Equipment Classification'. The 'FAX Machine' option is selected under this category.

Flash Hook Pulse Width configuration:

- Use System Defaults
- Minimum Width: 20 ms
- Maximum Width: 500 ms

Message Waiting Lamp Indication Type:

- None

Hook Persistency: 100 ms

ALTERNATE SCENARIO BASED ON USING INTERNAL DATA IN THE SIP URI FIELDS:

The screenshot shows the 'SIP Line - Line 18' configuration window. At the top, there are tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. Below the tabs is a table with the following columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. The table contains one row with the following values: Channel 1, Groups 19 19, Via <None>, Local URI Use Internal Data, Contact Use Internal Data, Display Name Use Internal Data, PAI None, Credential 1: dgw5713954, and Max Calls 10. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table is an 'Edit Channel' form with the following fields: Via (set to <None>), Local URI (set to Use Internal Data), Contact (set to Use Internal Data), Display Name (set to Use Internal Data), PAI (set to None), Registration (set to 1: dgw5713954), Incoming Group (set to 19), Outgoing Group (set to 19), and Max Calls per Channel (set to 10). At the bottom right of the form are 'OK' and 'Cancel' buttons.

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	19 19	<None>	Use Internal Data	Use Internal Data	Use Internal Data	None	1: dgw5713954	10

If you opted to “Use Internal Data” in the SIP URI, here is a screen shots detailing the “SIP” tabs for each user:

See below in the User\Sip Settings, you will enter the CLID that you want the outgoing line to route:

The screenshot shows the 'Jackie Someone: 202' SIP settings window. At the top, there are tabs for 'Menu Programming', 'Mobility', 'Phone Manager Options', 'Hunt Group Membership', 'Announcements', 'SIP', and 'Personal Directory'. The 'SIP' tab is selected. Below the tabs are the following fields: SIP Name (2062645134), SIP Display Name (Alias) (Jackie Someone), Contact (2062645134), and an 'Anonymous' checkbox which is unchecked.

INCOMING CALL ROUTING:

Note, if you have specified the DID number inside the users SIP Tab. You can use a . (dot) as the destination and the call will route to that user.

Incoming Call Route		
Line Group ID	Incoming Number	Destination
100	.	
101		250 Emergency
102		207 Fax

100	
Standard	Voice Recording
	Destinations
	TimeProfile
	Destination
	Default Value
*	

If not, program the incoming call route accordingly.

Mobile Twinnig Configuration

To support mobile twinning calls with the AccessLine SIP Trunk you must adjust the settings for RTP keep alive.

Under: system-LAN1-VOIP go to the bottom and change the scope of RTP keepalives from disabled to RTP and set initial keepalives to Enabled. Save config, merge and 2 way audio will be present on mobile twinning calls.

The screenshot shows the configuration page for system-LAN1-VOIP. The 'RTP keepalives' section at the bottom is highlighted with a red box. The 'Scope' dropdown is set to 'RTP' and the 'Initial keepalives' dropdown is set to 'Enabled'. Other settings include 'H.323 Gatekeeper Enable', 'SIP Trunks Enable', 'SIP Registrar Enable', 'RTP Port Number Range' (49152-53246), 'DiffServ Settings', and 'DHCP Settings'.