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

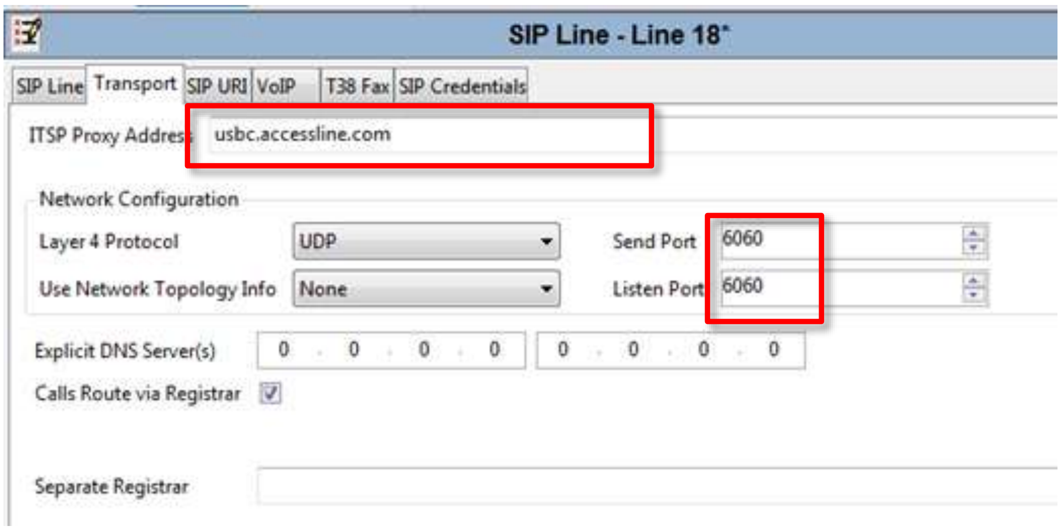
Support

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

PBX Configuration and Setup

SIP Trunk CREATION:

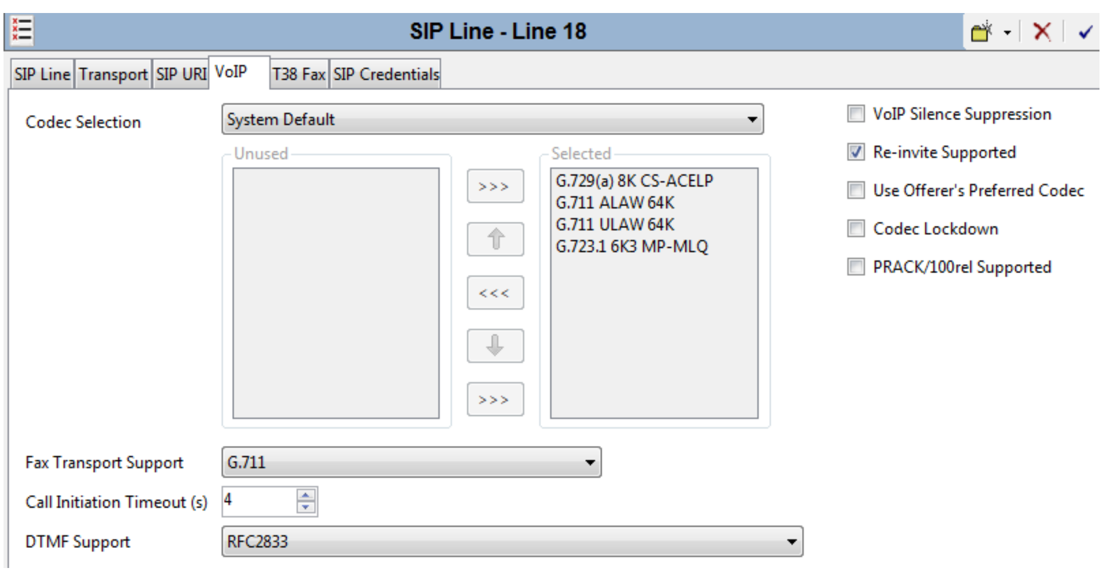
Under "Line", right click and select New, SIP line. Enter the information as detailed below using your supplied Proxy Address and username and password from Accessline.



Select your codecs:

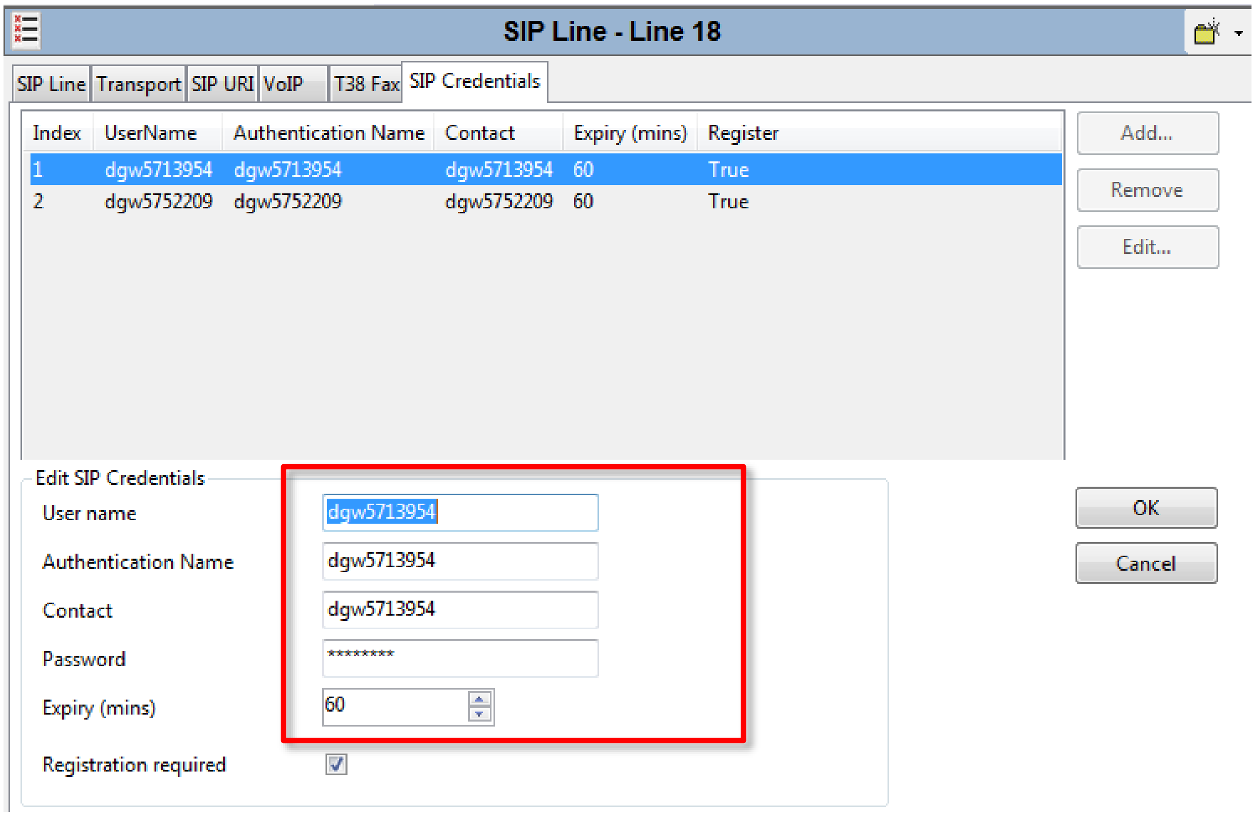
Default codec for AccessLine SIP Trunking is G729 Primary with G711 Secondary.

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



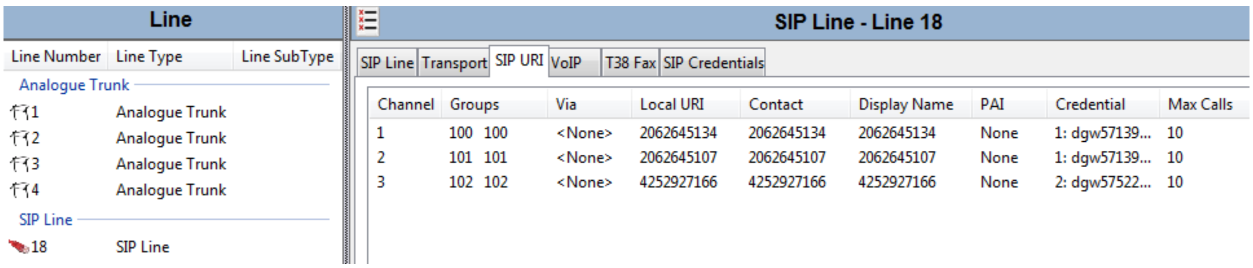
Create Sip Credentials:

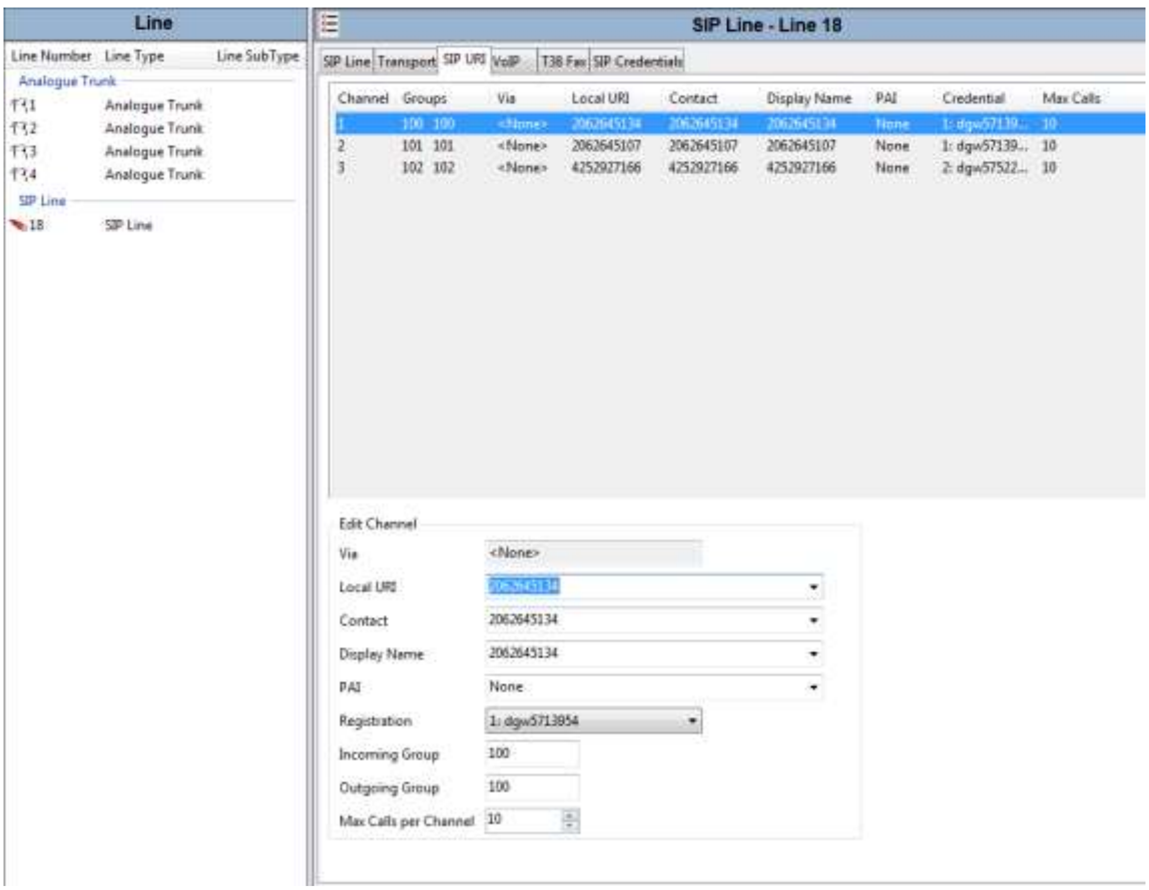
Use the information provided by AccessLine to configure the credentials.



Create Sip URI:

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





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:

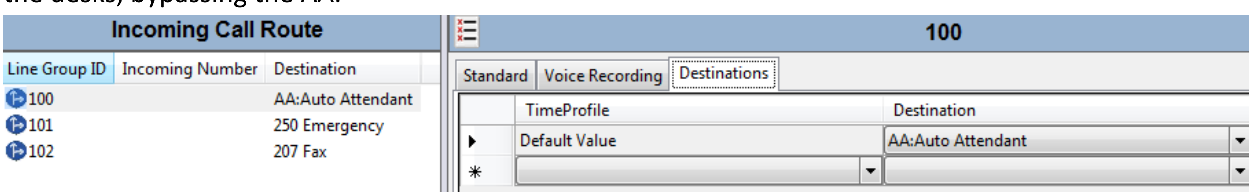
Sip Trunk 18:

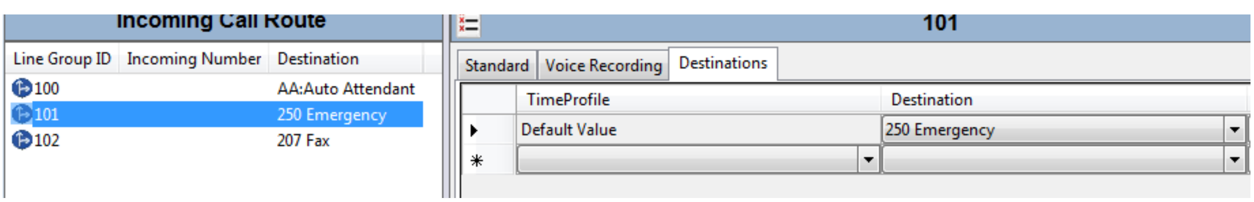
- Channel 1 is our main incoming/outgoing company phone number. When an incoming call is received on that "line" (206-264-5134) the call will be routed to Incoming call group 100
- Channel 2 is our 911 number given to us by accessline. This number is not known to anyone on the outside.
- Channel 3 is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



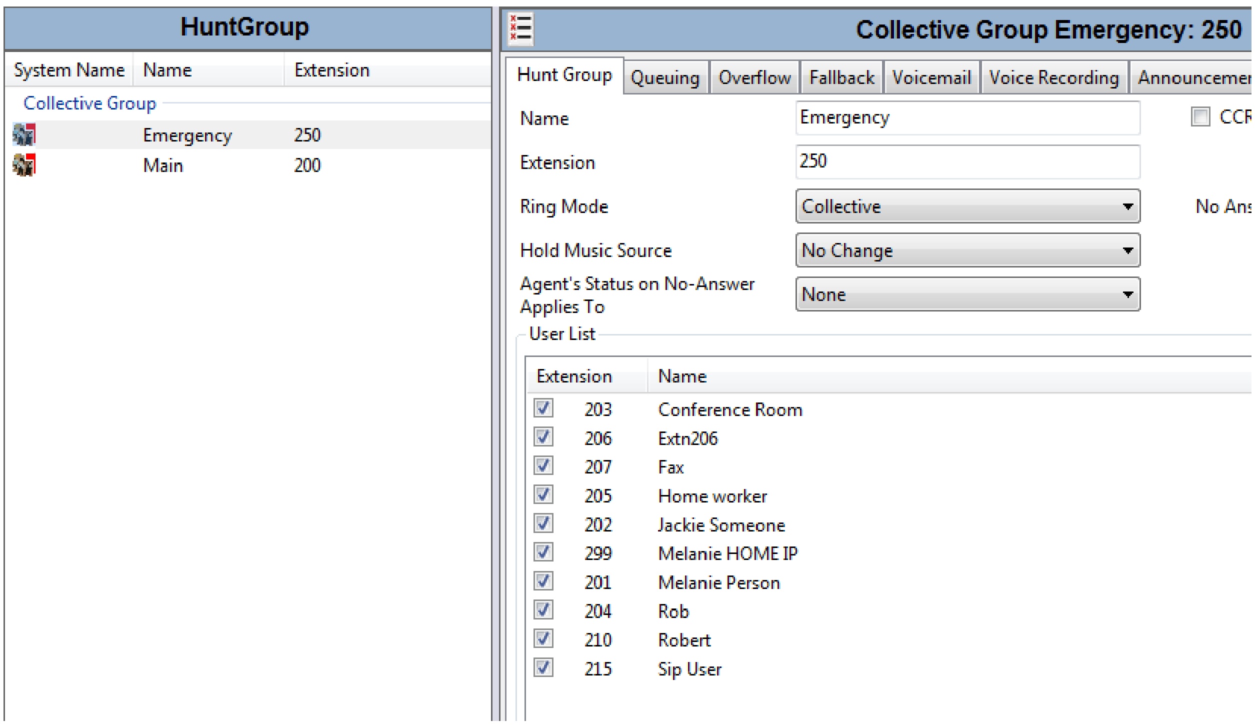


Calls coming in on group 102 will be routed to the fax at extension 207.

Emergency (911) Hunt Group

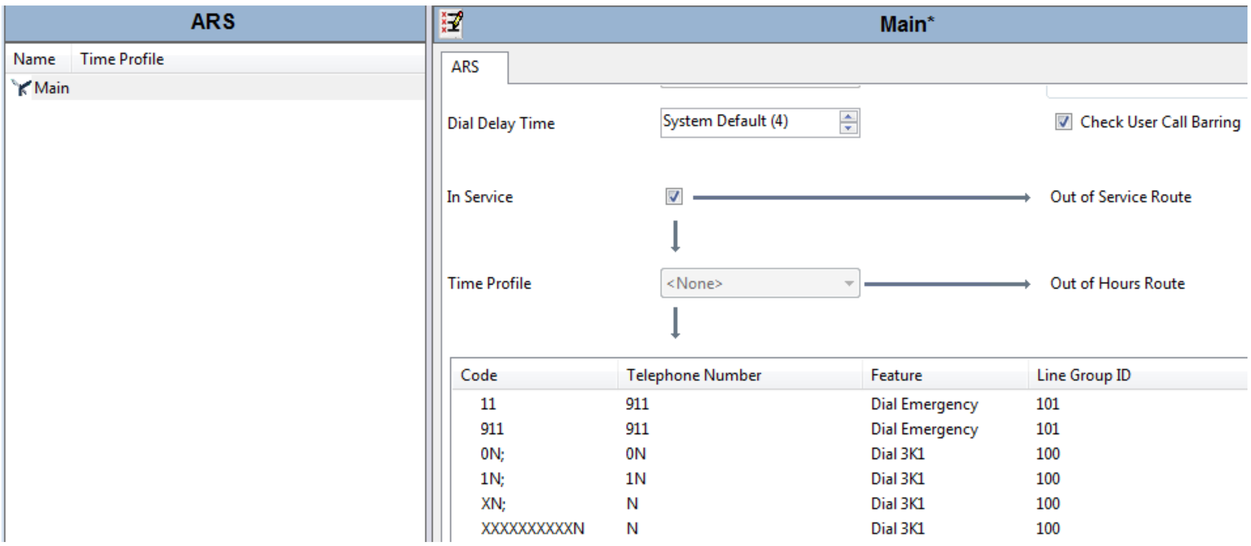
AccessLine configures a unique TN (telephone number) per customer location. The TN is registered with our E911 provider. When AccessLine receive a 911 call from the PBX, we forward to our E911 provider with the registered 911 TN as the CLI (regardless of what was passed from the PBX). The 911 TN is used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

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



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.



Emergency (911) Setup

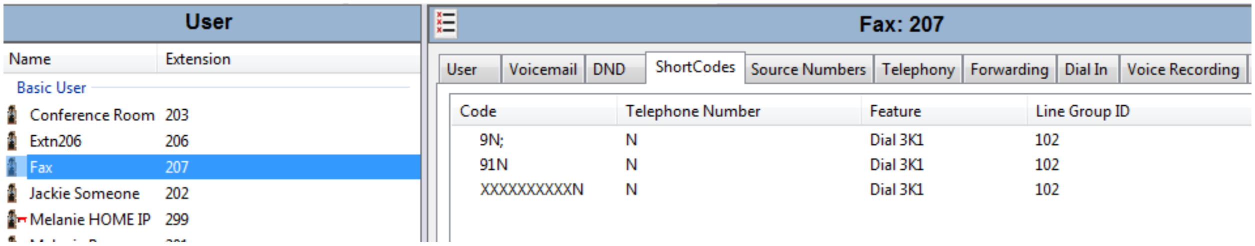
The system default has the outgoing call group (ARS) setup for 911 dialing.

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

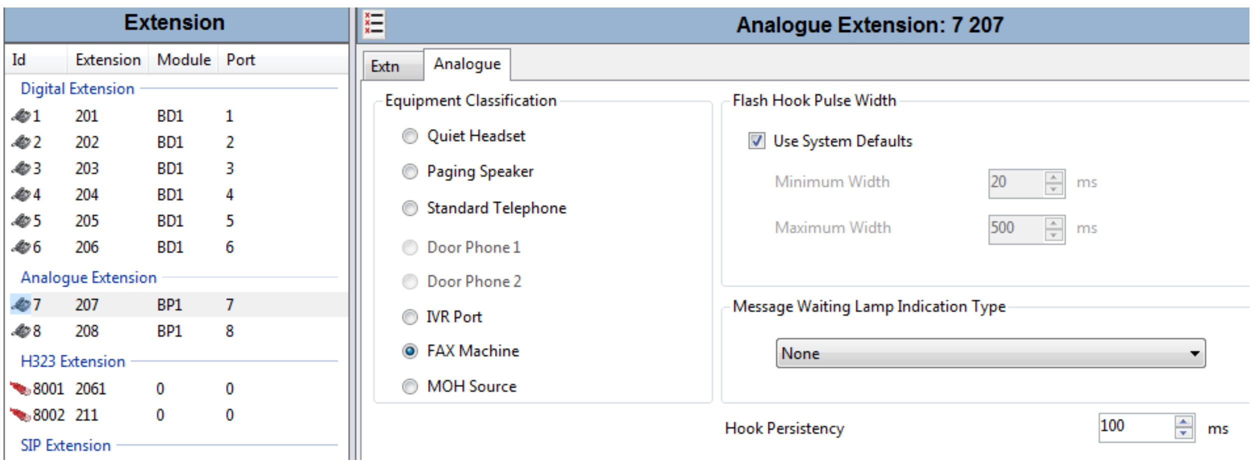
What is recommended is to create an Emergency hunt group that every user is a member of, and use the 911 number provided by AccessLine in the SIP tab of the hunt group.

Faxing:

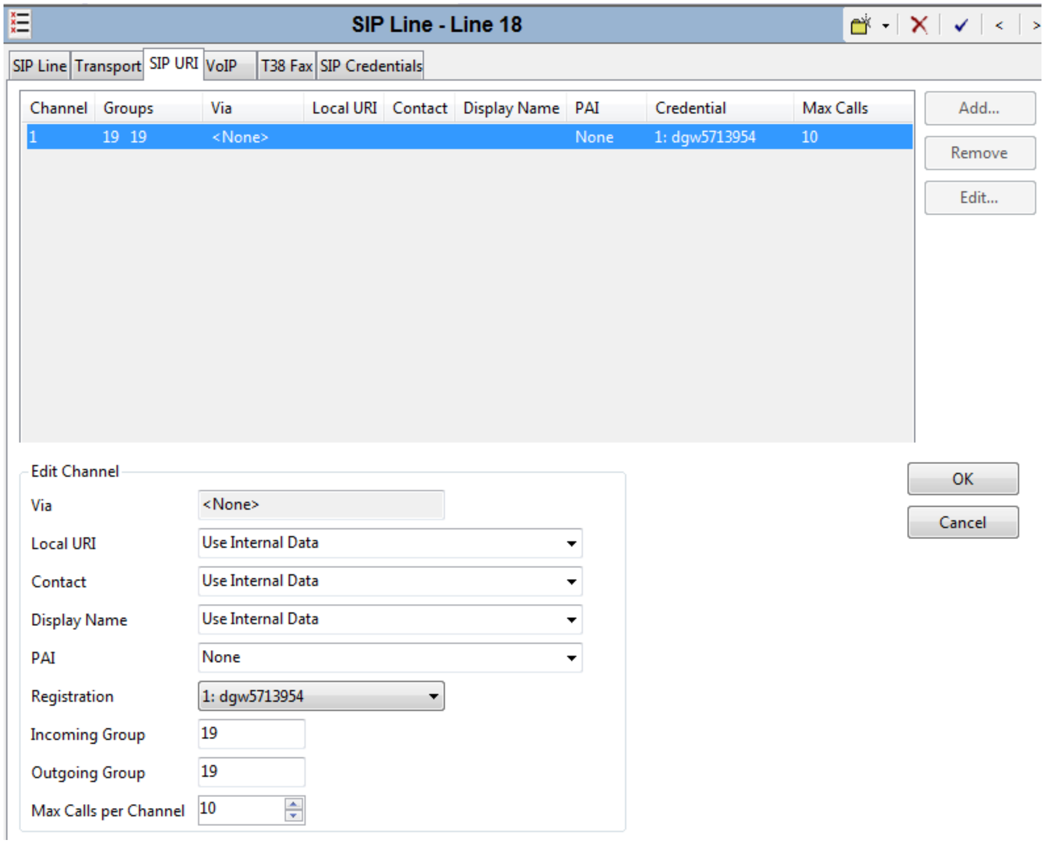
Create a fax user, and create shortcodes that reference the outgoing group ID that you want to fax out on.



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

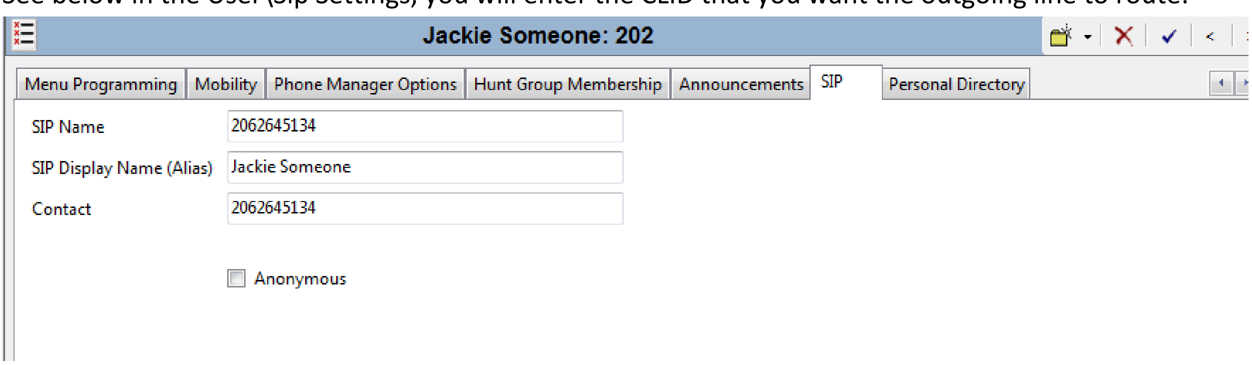


ALTERNATE SCENARIO BASED ON USING INTERNAL DATA IN THE SIP URI FIELDS:



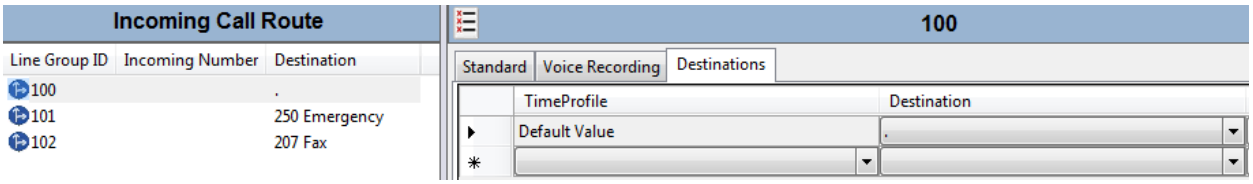
If you opted to “Use Internal Data” in the SIP URI, here is a screen shots detailing the “SIP” tabs for each user:

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



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.



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.