DuVoice	# Integration

Title	Avaya ACM/SES SIP
Document	avaya-acm-ses-sip
Last update	2012/05/02
DuVoice Versions	5.10, 5.00
Switch Versions	

Table of Contents

Features Supported

Message Lights	Yes
Forward to personal greeting RNA	Yes

Forward to personal greeting Busy	Yes
Forward to personal greeting DND	
Different busy greeting	Yes
Auto login to a mailbox	Yes
Trunk ID multiple tenant	
Voicemail transfer	
Record a call	
DID Fax	No
Caller ID with message	Yes
Trunk to trunk transfer (unsupervised)	
Loop start disconnect	No
Tone disconnect	No
SIP disconnect	Yes
Supervised transfers	No
Blind transfers	Yes
Call screening	No
Caller queuing	No

Hospitality Features Supported

Property management integration (PMS)	
Room phone control	
Wakeup calls	Yes
Failed wakeup alerting	
Guest name changes	
Room Clean - Room Dirty Status	

PBX Requirements and Programming

Requirements

- Avaya ACM CM v5 or above.
- Avaya SIP Enablement Services
- Avaya Communications Manager

Known Issues

None.

Programming

This section describes the procedure for setting up a SIP trunk between Avaya Communication manager and Avaya SES.

Capacity Verification

Enter the **display system-parameters customer-option** command. Verify the number of OPS stations for the voicemail system. You will need one for each line on the DuVoice system.

```
                            OPTIONAL FEATURES

G3 Version: V15                                  Software Package: Standard
  Location: 1                                 RFA System ID (SID): 1
  Platform: 18                                RFA Module ID (MID): 1

                                                                USED
                             Platform Maximum Ports:2850       46
                                  Maximum Stations: 14         10
                          Maximum XMOBILE Stations: 0          0
                Maximum Off-PBX Telephones - EC500: 0          0
                Maximum Off-PBX Telephones -   OPS: 100        2
                Maximum Off-PBX Telephones - PBFMC: 0          0
                Maximum Off-PBX Telephones - PVFMC: 0          0
                Maximum Off-PBX Telephones - SCCAN: 0          0
```

On tab 2 verify the number of SIP trunks available.

```
                              OPTIONAL FEATURES

IP PORT CAPACITIES                                                             USED
                     Maximum Administered H.323 Trunks: 10     3
           Maximum Concurrently Registered IP Stations: 12     0
             Maximum Administered Remote Office Trunks: 0      0
Maximum Concurrently Registered Remote Office Stations: 0      0
              Maximum Concurrently Registered IP eCons: 0      0
  Max Concur Registered Unauthenticated H.323 Stations: 0      0
                  Maximum Video Capable H.323 Stations: 0      0
                   Maximum Video Capable IP Softphones: 0      0
                       Maximum Administered SIP Trunks: 10     10
  Maximum Administered Ad-hoc Video Conferencing Ports: 0      0
   Maximum Number of DS1 Boards with Echo Cancellation: 0      0
                             Maximum TN2501 VAL Boards: 10     0
                     Maximum Media Gateway VAL Sources: 1      1
           Maximum TN2602 Boards with 80 VoIP Channels: 0      0
          Maximum TN2602 Boards with 320 VoIP Channels: 0      0
   Maximum Number of Expanded Meet-me Conference Ports: 0      0
```

IP Codec

Enter **change ip-codec-set <c>** command, where c is a number between 1 and 7, inclusive.

```
                          IP Codec Set

    Codec Set: 7

    Audio        Silence       Frames   Packet
    Codec        Suppression   Per Pkt  Size(ms)
 1: G.711MU           n          2        20
 2:
 3:
 4:
 5:
 6:
 7:


     Media Encryption
 1: none
 2:
 3:
```

Configure IP Network Region

Enter **change ip-network-region <n>** command, where **n** is a number between 1 and 250 inclusive, and configure the following.

- Authoritative Domain
  This domain must match the SIP Domain value configured in Avaya SES.
- Intra-region IP-IP Direct Audio
  Default value for this field is yes.
- Inter-region IP-IP Direct Audio
  Default value for this field is yes.
- Codec Set
  Set this to the value of the codec set selected under **IP Codec**.

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

Configure IP Node Name

Enter **change node-names ip**. Add a node name for Avaya SES along with its IP address.

IP NODE NAMES

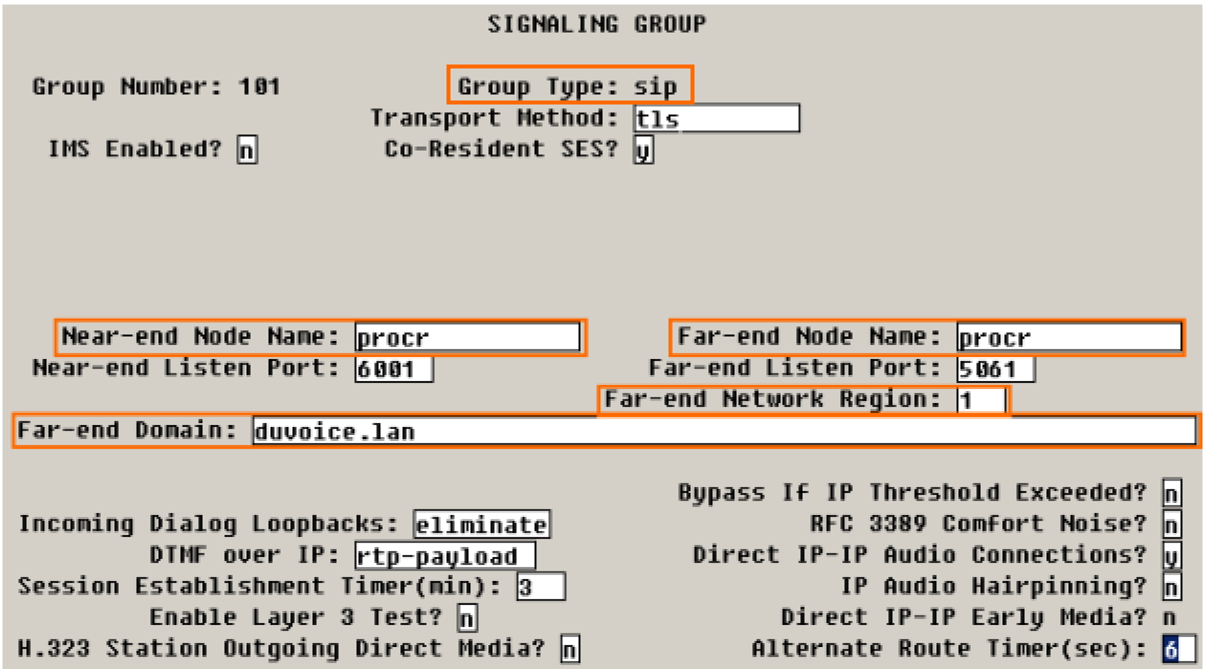
Name	IP Address
SES	192.168.11.70
default	0.0.0.0
duvoice	192.168.11.26
msqserver	192.168.11.62
procr	192.168.11.70

Configure SIP Signaling

Enter **add signaling-group <s>** command, where **s** is an available signaling group and configure the following.

- Group Type
  Set to **sip**.
- Near-end Node Name
  Set to procr.

- Far-end Node Name
  Set to the Avaya SES name or procr if hosted on the same board.
- Far-end Network Region
  Set to the region configured under **IP Network Region**.
- Far-end Domain
  Set to the value entered in **Authoritative Domain**.



SIGNALING GROUP

Group Number: 101 Group Type: sip
Transport Method: tls
IMS Enabled? n Co-Resident SES? y

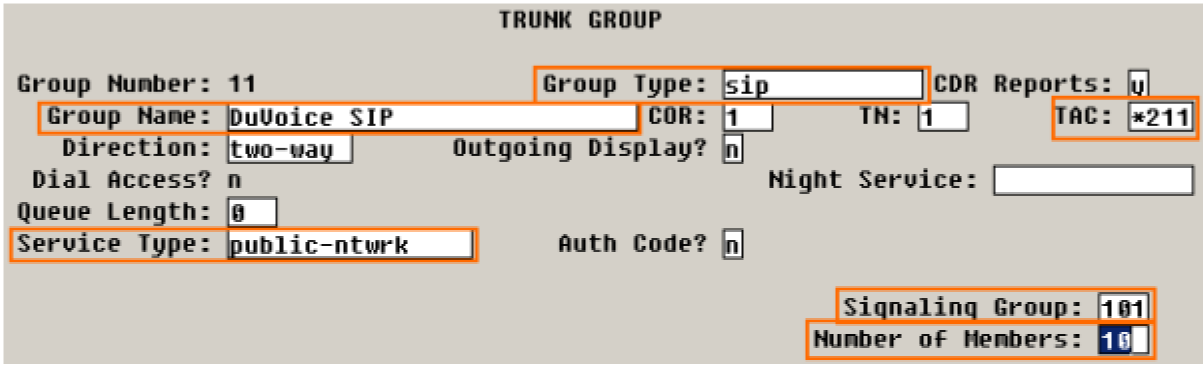
Near-end Node Name: procr Far-end Node Name: procr
Near-end Listen Port: 6001 Far-end Listen Port: 5061
Far-end Network Region: 1
Far-end Domain: duvoice.lan

Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3 IP Audio Hairpinning? n
Enable Layer 3 Test? n Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n Alternate Route Timer(sec): 6

Configure SIP Trunk

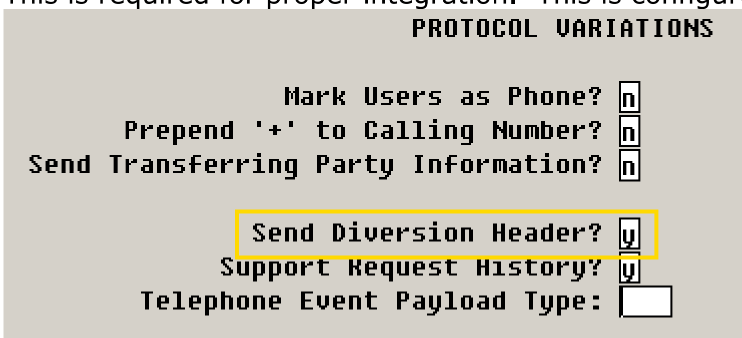
Enter
Enter **add trunk-group <t>** command, where **t** is an unallocated trunk group and configure the following.

- Group Type
  Set the Group Type field to **sip**.
- Group Name
  Enter a descriptive name.
- TAC (Trunk Access Code)
  Set to any available trunk access code.
- Signaling Group
  Set to the Group Number field value to the signaling group.
- Number of Members
  Allowed value is between 0 and 255. Set this value to twice the number of voice ports on the DuVoice.
  Note: *Each SIP call between two SIP endpoints requires two SIP trunks for the duration of the call.*
- Service Type
  This must be set to public-ntwrk to support proper integration.


TRUNK GROUP

Group Number: 11 Group Type: sip CDR Reports: y
Group Name: DuVoice SIP COR: 1 TN: 1 TAC: *211
Direction: two-way Outgoing Display? n
Dial Access? n Night Service:
Queue Length: 0
Service Type: public-ntwrk Auth Code? n
Signaling Group: 101
Number of Members: 10

Enable Divesion Header
This is required for proper integration. This is configured on tab 4.

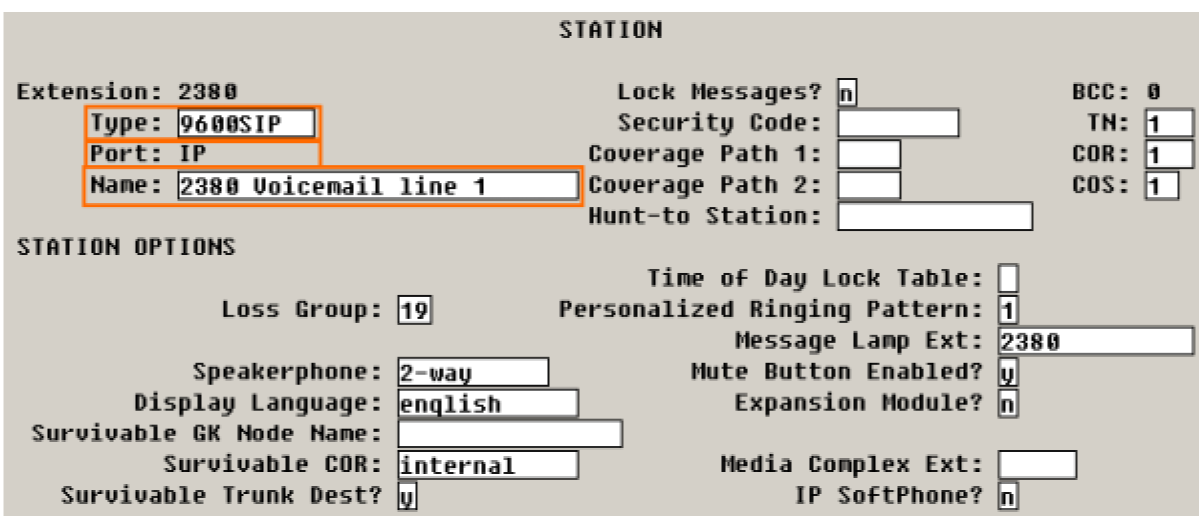

PROTOCOL VARIATIONS

Mark Users as Phone? n
Prepend '+' to Calling Number? n
Send Transferring Party Information? n

Send Diversion Header? y
Support Request History? y
Telephone Event Payload Type:

Configure SIP Endpoint
Enter **add station s**, where **s** is an extension valid in the provisioned dial plan. Create a station for each line of the voicemail.

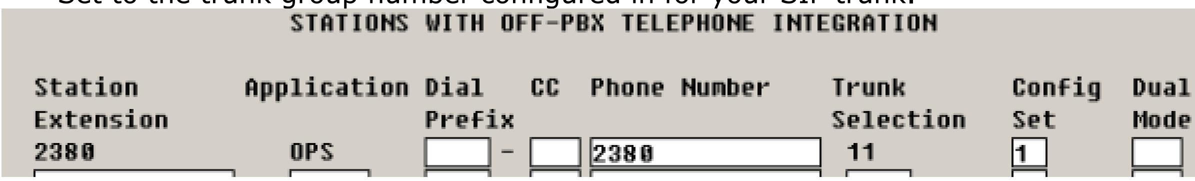
- Type
  Set to 9600SIP.
- Name
  Enter a descriptive name for this station.
- On tab 6 set the SIP Trunk to the trunk number created above.


STATION

Extension: 2380 Lock Messages? n BCC: 0
Type: 9600SIP Security Code: TN: 1
Port: IP Coverage Path 1: COR: 1
Name: 2380 Voicemail line 1 Coverage Path 2: COS: 1
Hunt-to Station:
STATION OPTIONS
Time of Day Lock Table:
Loss Group: 19 Personalized Ringing Pattern: 1
Message Lamp Ext: 2380
Speakerphone: 2-way Mute Button Enabled? y
Display Language: english Expansion Module? n
Survivable GK Node Name:
Survivable COR: internal Media Complex Ext:
Survivable Trunk Dest? y IP SoftPhone? n

Enter **add off-pbx-telephone station-mapping** command and configure the following. Perform this for each voicemail port configured above.

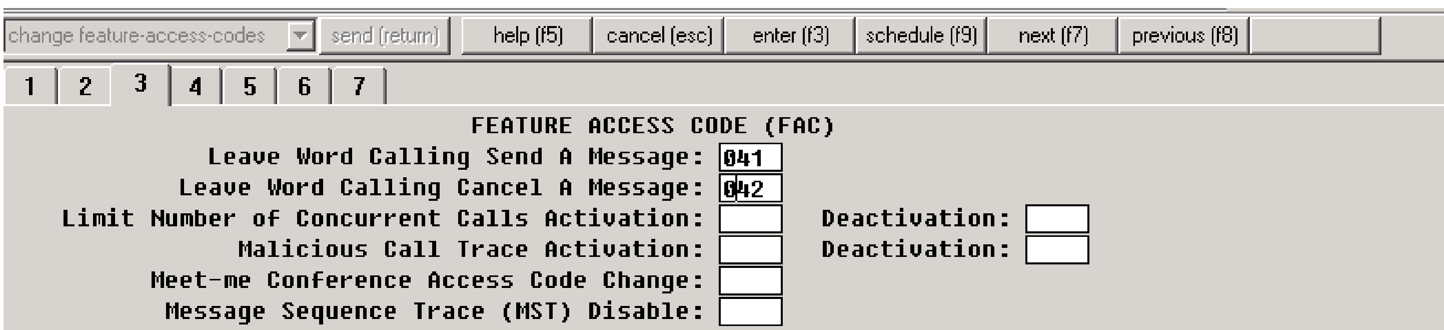
- Station Extension
  Set the extension of the OPS station as configured above.
- Application
  Set to **OPS**.
- Phone Number
  Enter the number the line will use for registration and call termination.
- Trunk Selection
  Set to the trunk group number configured in for your SIP trunk.



Configure LWC MWI

Enter change feature-access-codes command and enter a feature access code for LWC on/ off. Check your dialplan and other feature codes for an available code.

Note: You cannot use a pound sign for any feature code dialed by DuVoice system.

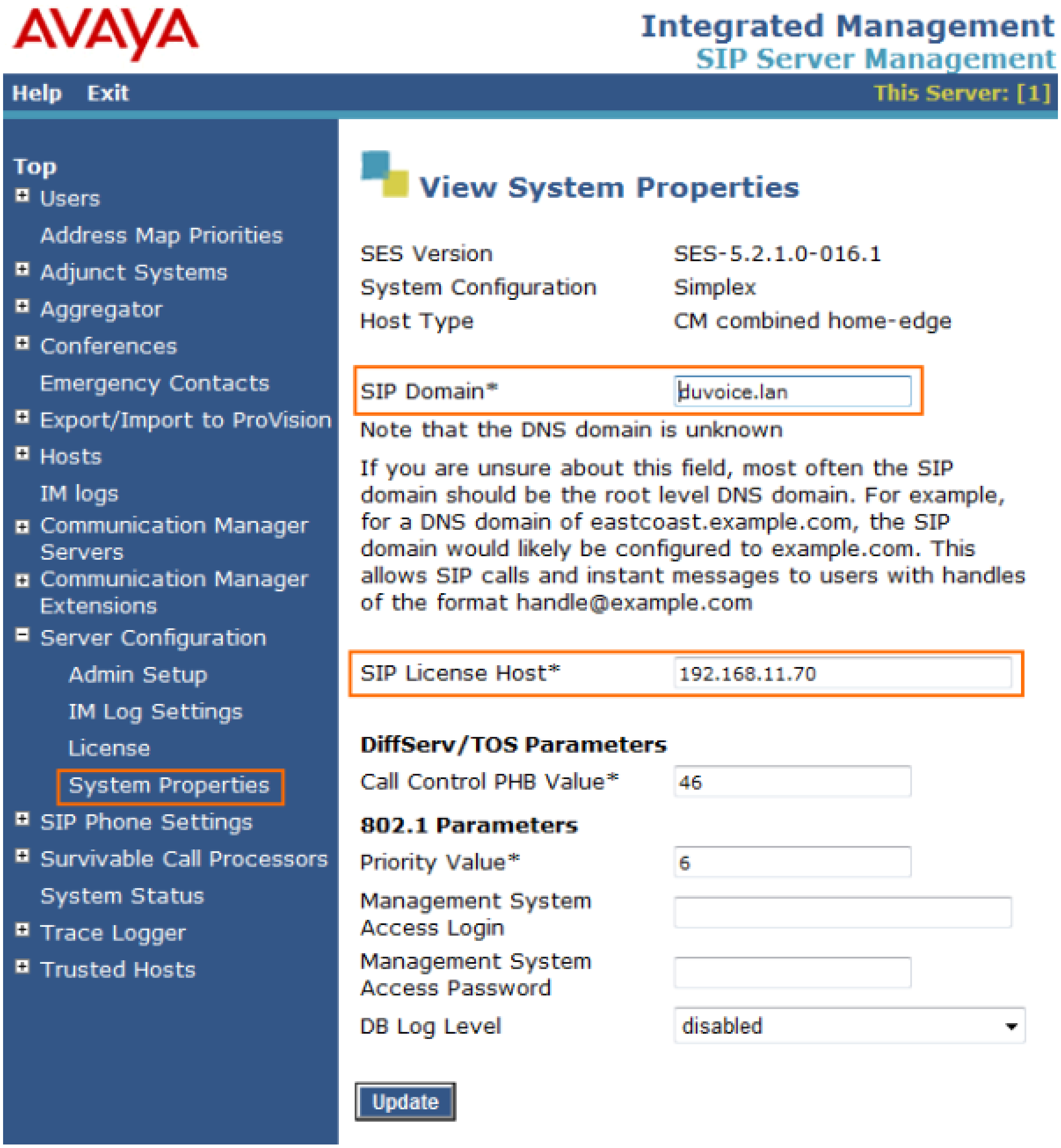


Configure Avaya SIP Enabledment Services

Configuration of the Avaya SES must be performed via Internet Explorer only and will not operate correctly on any other browsers.

Login to your SES via Internet Explorer and set the System Properties

- SIP Domain
  Enter the domain configured for your IP Network Region.
- Enter the SIP License Host.

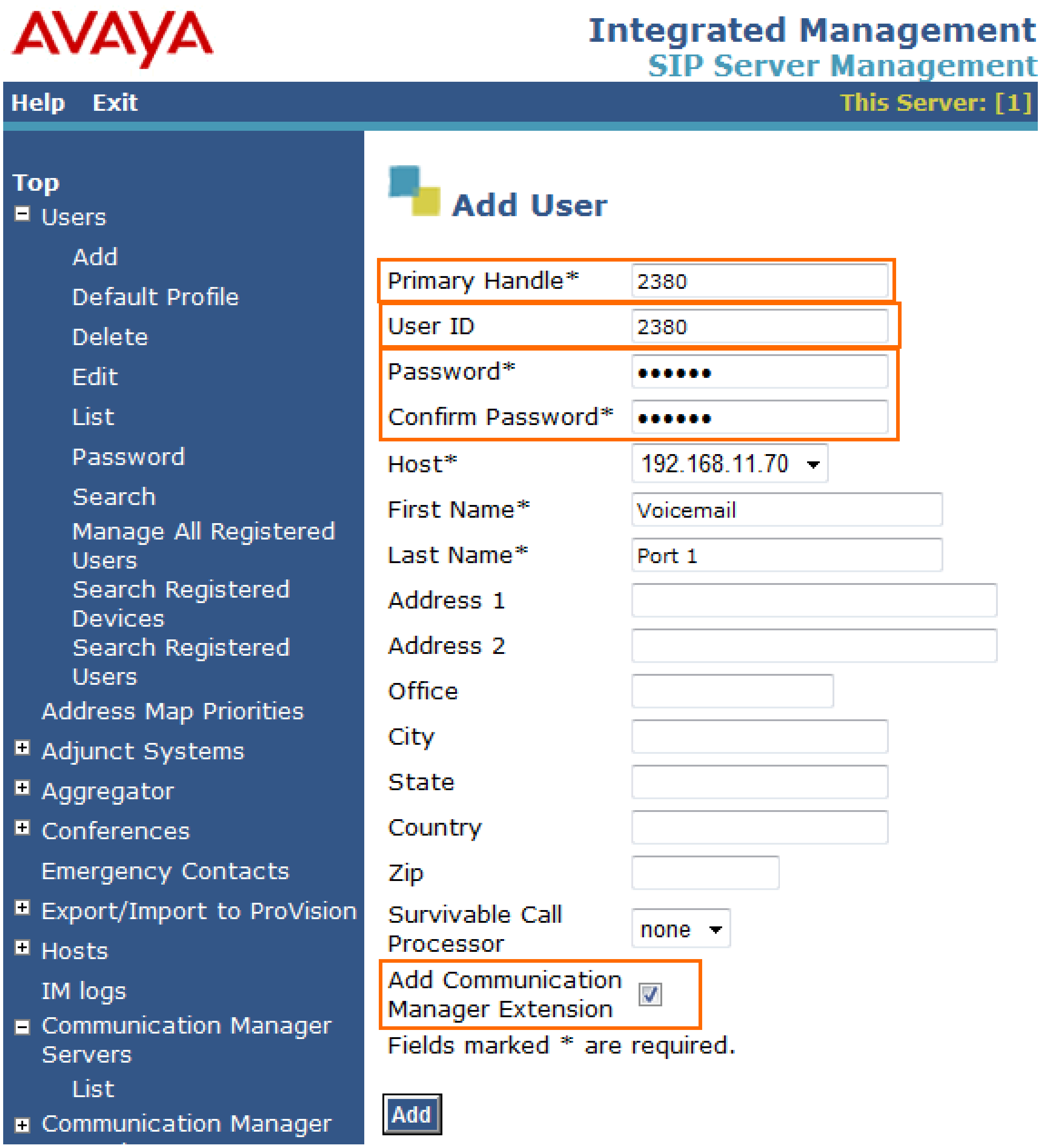


Add matching user to the SES for the extensions created on the ACM.

- Primary Handle
  Set this to the extension number.
- User ID
  Set this to the extension number. This field will be used as the Username for the voicemail port in System Configuration.
- Password
  Enter a password. This field will be used as the Password for the voicemail port in System Configuration.
- First Name/Second Name
  Enter a descriptive first and last name.
- Add Communication Manager Extension

Be sure to check this box. This will create a mapping for this extension between the SES and ACM.

When the Add button is pressed enter the extension number and confirm the Communication Manager Server field is correct and click Add.



Configure Voicemail

This document assumes the voicemail software has already been installed.

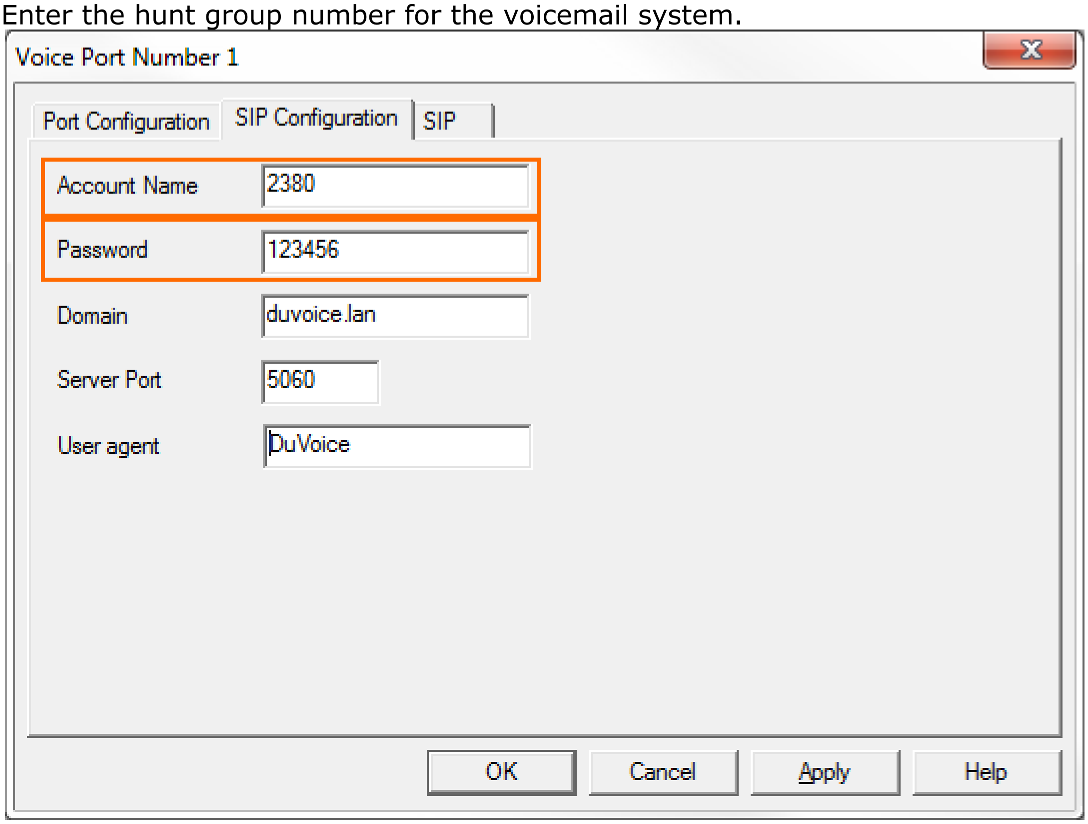
Configure Ports

Each voicemail port must be configured with it's corresponding User in the SES.

Start System Configuration

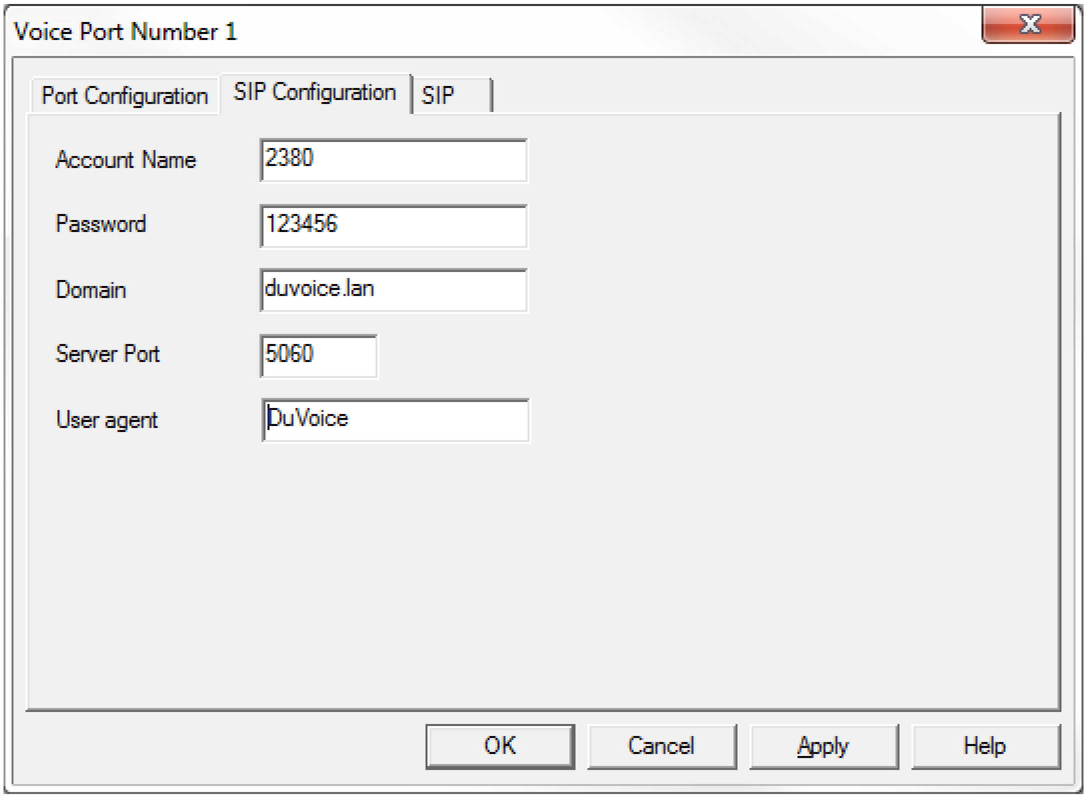
Double click the first configured port.

- Extension number
  Enter the extension number for this port.
- Hunt group extension is a member of
  Enter the hunt group number for the voicemail system.



Click the **SIP Configuration** tab.

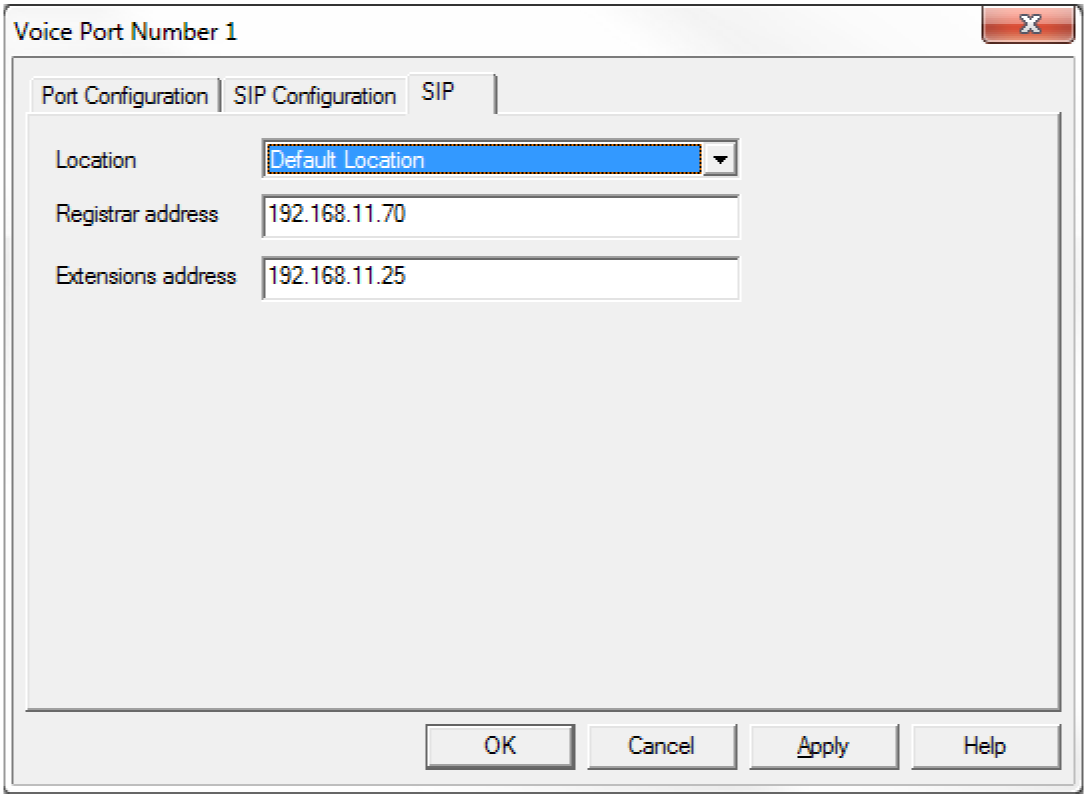
- Account Name
  Enter the user name for this port entered in the SES.
- Password
  Enter the password entered for this user in the SES.
- Domain
  Enter the domain name the SES is configured for.
- Server Port
  The default is 5060.
- User agent
  The default value is DuVoice.



Click the **SIP** tab.

This tab is a duplicate of the SIP tab located under Features|Connectors. The settings on this tab are not port specific, but rather Location specific.

- Location
  If this port is located on a different location than the default choose the location.
- Registrar address
  Enter the address of the SES server.
- Extensions address
  Confirm this is the address the system will use to communicate with the switch on.



Click Ok and perform these actions on all the remaining ports.