SIP Trunking Configuration Guide

for

Avaya™ IP Office version 6.0 with IP Office Manager 8.0 SIP Trunking Configuration Guide

Document Revision 1.4

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1 Audience

This document is intended for the SIP Trunk customer’s technical staff and Avaya Value Added Retailer (VAR) having installation and operational responsibilities.

2 Introduction

This Configuration Guide describes configuration steps for Cox SIP Trunking to an Avaya IP Office v6.0 with IP Office Manager v8.0. Cox SIP Trunking is a scalable and efficient IP Trunking telecommunication solution for your business that provides all the traditional services such as Direct Inward Dialing, Hunting, Calling Name, Calling Number, Local/Long Distance and Cox network-based Business Continuity options, including:

- Burstable Trunk Capacity – Dynamically increases call capacity during peak busy periods so your customers never receive a busy signal.
- Call Forward Always – On the trunk group pilot number for all calls in case of an outage (flood, fire, power outage, etc.).
- Call Forward Not Reachable – On the trunk group pilot number that operates on a per-call contingency basis to forward the call to any PSTN number (e.g. call center or alternate office location) during temporary call completion impairments.
- Route Exhaustion – automatic reroute of trunk group calls to any PSTN phone number (i.e., a call center) if calls can’t be completed to the PBX.
- Support for geo-redundant PBX deployments and automatic reroute of SIP Trunks to the backup customer data center.

All calls are routed over Cox’s national fiber network with guaranteed Quality of Service (QoS); calls never traverse the Internet.

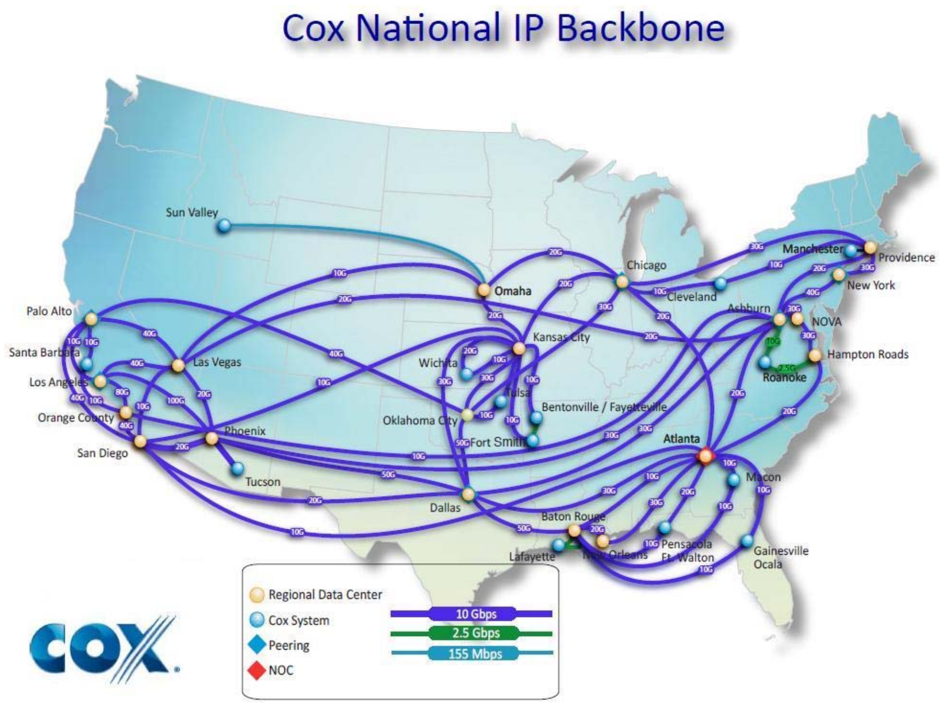

Figure 1 – Cox Fiber Network

3 Network Topology

The high level Cox SIP Trunk network architecture is depicted below. The key network elements are:

- IP PBX – Customer PBX for terminating SIP Trunks.
- Cox Enterprise Session Border Controller (E-SBC) – The E-SBC is a smart service demarcation device and SIP Application Layer Gateway (ALG) installed and managed by Cox.
- High Availability & Geo-Redundant core Session Border Controllers (SBC) and Broadsoft SIP Call Servers for maximum survivability and reliability.
- PSTN Gateway for connections to the Public Switched Telephone Network (PSTN).

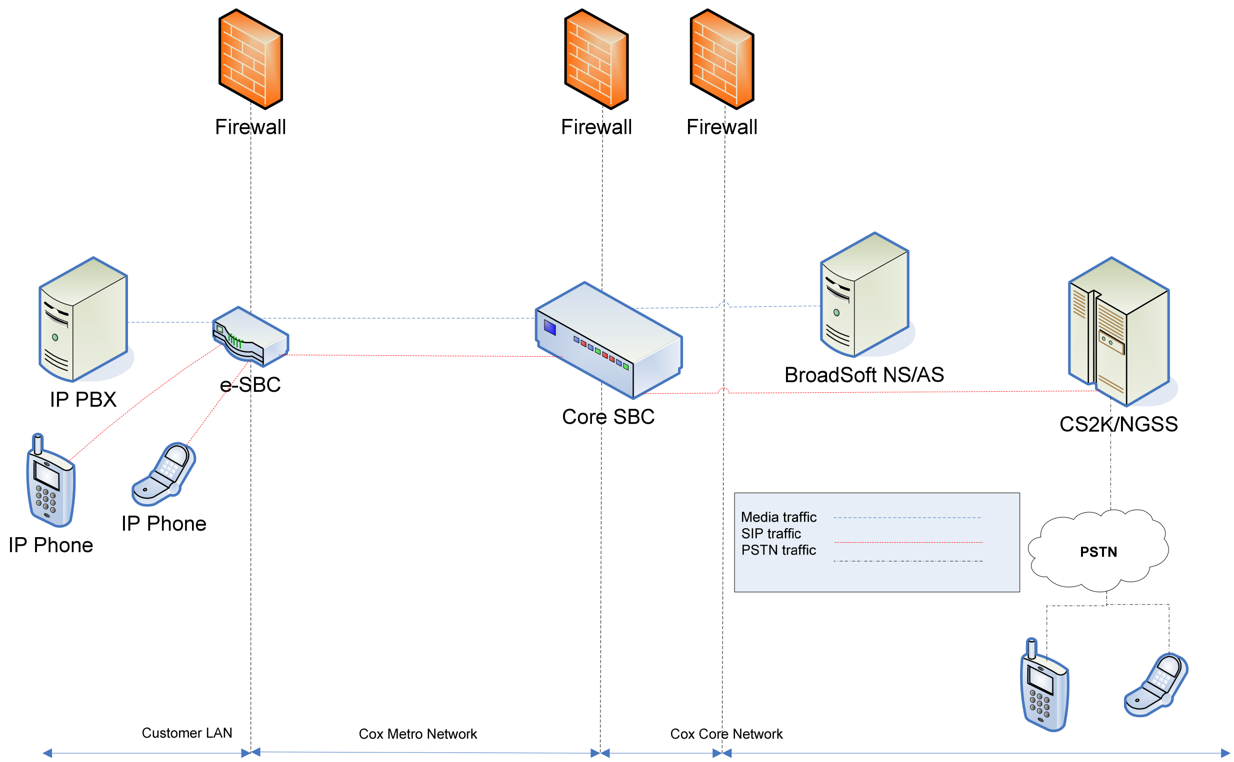

Figure 2 – Network Reference Architecture

This SIP Trunk network architecture is replicated across the Cox operating regions for scalability and operational autonomy.

Cox will deploy one or more Enterprise Session Border Controllers (E-SBCs) to meet call capacity, customer data center geo-redundancy and trunk group requirements. The E-SBC is owned and managed by Cox and is the service demarcation point. The E-SBC performs SIP ALG, SIP normalization, NAT, security, traffic shaping/prioritization, performance reporting and remote diagnostic functions.

4 System Components

The SIP Trunk reference lab configuration is illustrated in **Figure 3** and is representative of an Avaya IP Office PBX deployment.

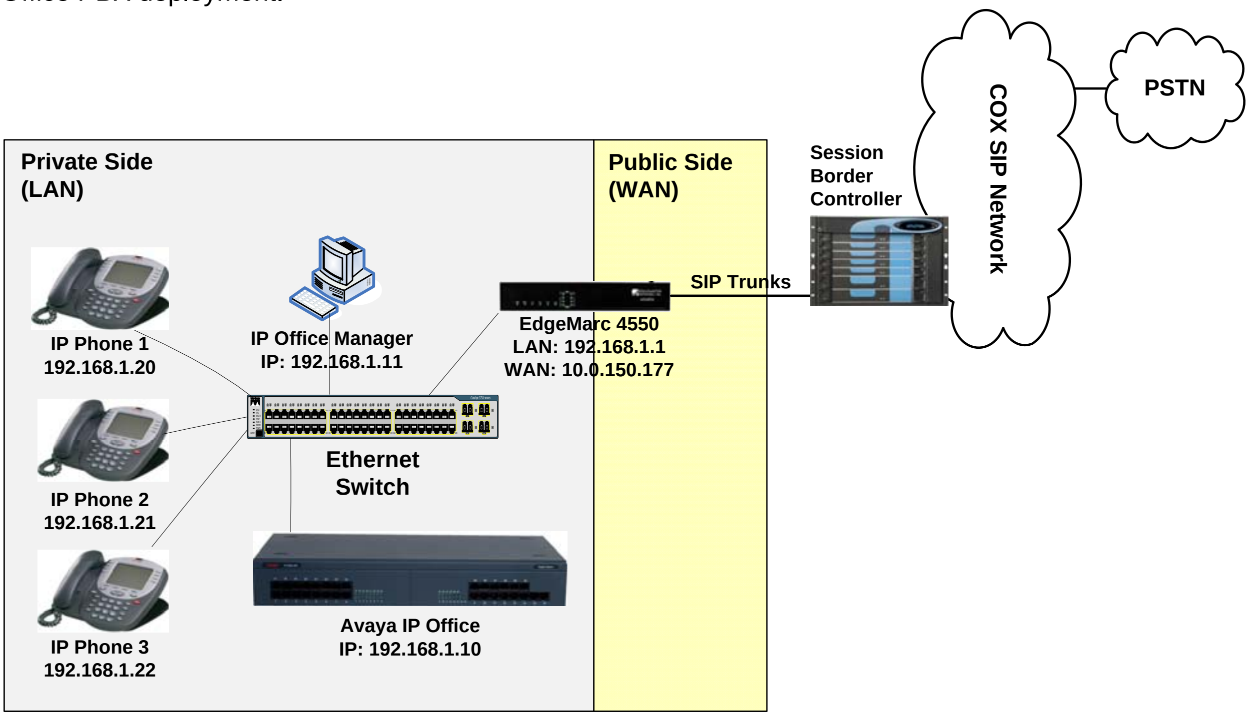

Figure 3 – Avaya IP Office SIP Trunking Lab Network

4.1 Hardware Components

- Avaya IP Office 6.0
- Avaya IP Office Manager 8.0
- Avaya IP Office Voice Mail Pro
- Avaya IP Phone
- Avaya Analog Phone
- EdgeMarc 4550 Enterprise Session Border Controller (E-SBC)

4.2 Software Requirements

- Avaya IP Office 6.0 release
- Avaya IP Office Manager 8.0 release
- EdgeMarc 4550 9.12.5 release

4.3 Example Configuration Information

The specific values listed in the table below and in subsequent sections are used in this lab configuration described in this document, and are for **illustrative purposes only**. Customers must obtain and use the specific values for their own specific configurations.

Table 1 – IP Addresses

Component	Cox Lab Value	Your Value
EdgeMarc E-SBC		
• LAN IP Address	192.168.1.1	
Avaya IP Office		
• IP Address	192.168.1.10	
Avaya IP Office Manager (also TFTP server)		
• IP Address	192.168.1.11	

5 Features

5.1 Supported Features

- SIP Registration Mode (REGISTER request) with the Cox network
- Basic Calls using G.711ulaw
- Calling Party Number Presentation and Restriction
- Call Transfer
- Call Forwarding
- RFC2833 transcoding

5.2 Unsupported Features

- Codec negotiation of G.729, G.726, and others
- T.38 Fax relay

6 Caveats

- Must select 'To Header' in Call Routing Method in Avaya IP Office SIP Line configuration to insure proper indexing of the User's DID number coming from Cox's network.
- Avaya IP Office does not support SIP DIVERSION header when call forwarding is enabled. Avaya IP Office sends a new SIP INVITE for call forwarding. See call forward example below.
- Avaya IP Office bridges (i.e. hairpins) both call legs during call transfer, and does not release the SIP Trunk.
- Avaya IP Office only supports "Anonymous" Caller ID using Feature Access Codes or Star Codes, see detail description below.

7 Avaya IP Office System Configuration

7.1.1 Setup System Information

You will need to connect to the Avaya IP Office using a Microsoft© Windows® server with network access configured with the IP Office Manager, VoiceProClient, and IP Office Status clients.

Enter the Name for the IP Office Server and select its locale.

You may provide the information for the remaining areas in the System tab as indicated below in Figure 4. Select Manager as Phone File Server Type.

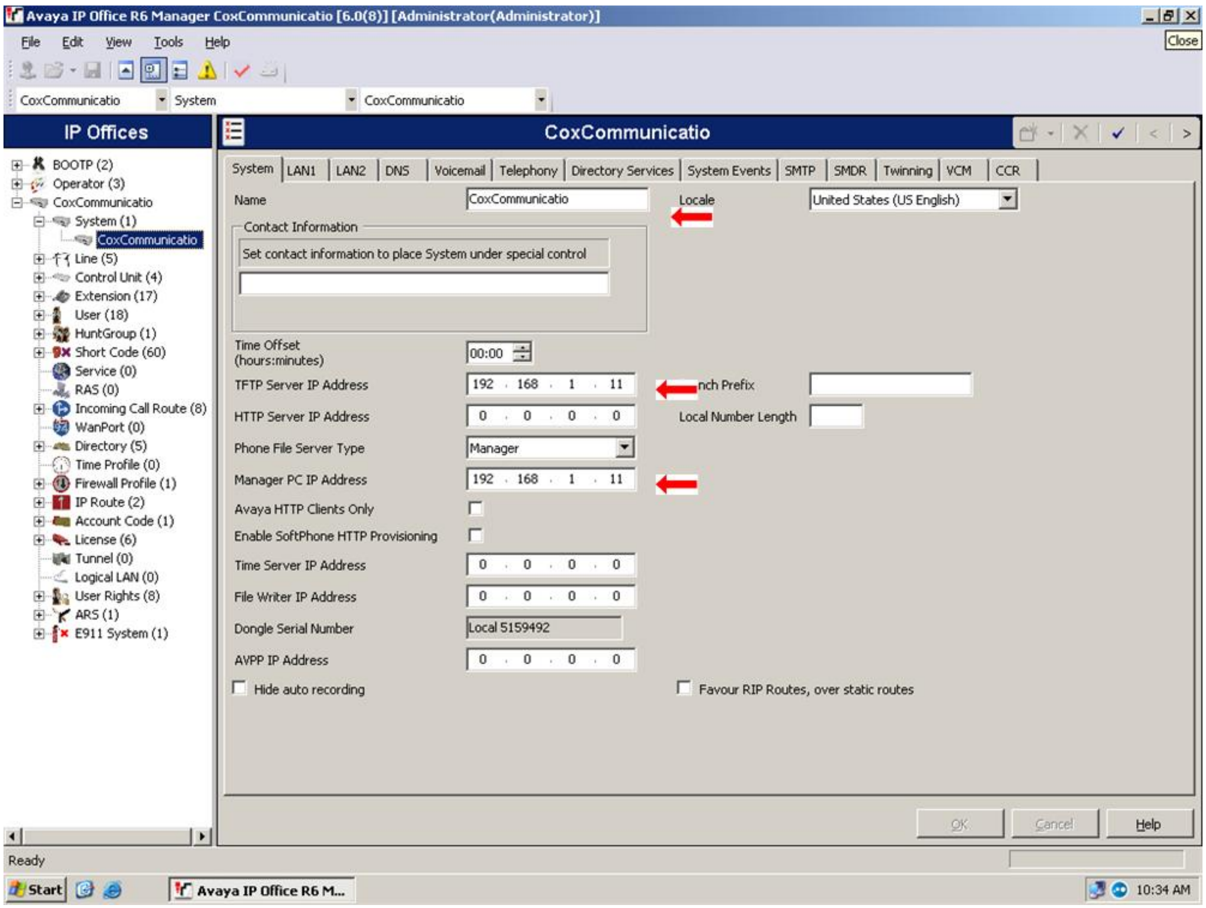

Figure 4 – System Setup

7.1.2 LAN1 → LAN Settings Information

Enter the IP address and IP net mask of the IP Office server. For RIP mode select 'None' from the drop down unless otherwise required.

The IP Office server is also capable of being activated as a DHCP server and can provide IP addresses to clients attempting to register on the LAN side of the network. If the network already has a DHCP server(s) pre-existing and the network is to be responsible for the registrations, then disable the DHCP server on LAN1 for the IP Office server.

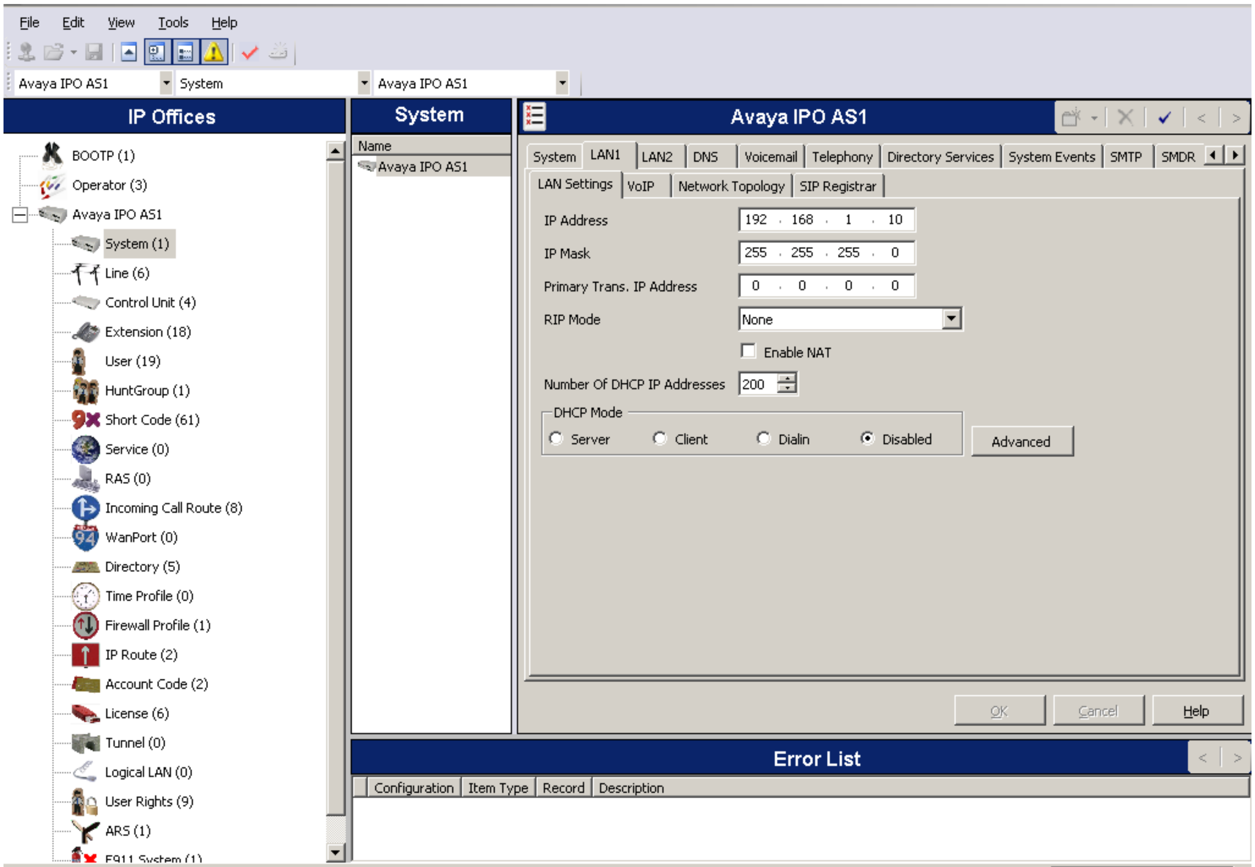

Figure 5 – LAN Settings

7.1.3 LAN1 → VoIP Information

For the IP Office server to use SIP Trunks, it must be enabled. A license is required for this and is activated after installation. See section 7.2 for license information.

Make sure the SIP Trunk Enable, SIP Registrar Enable, and H323 Auto-create Extn, are selected. See Figure 6 below.

If a VLAN is configured, select the proper option from the drop down list.

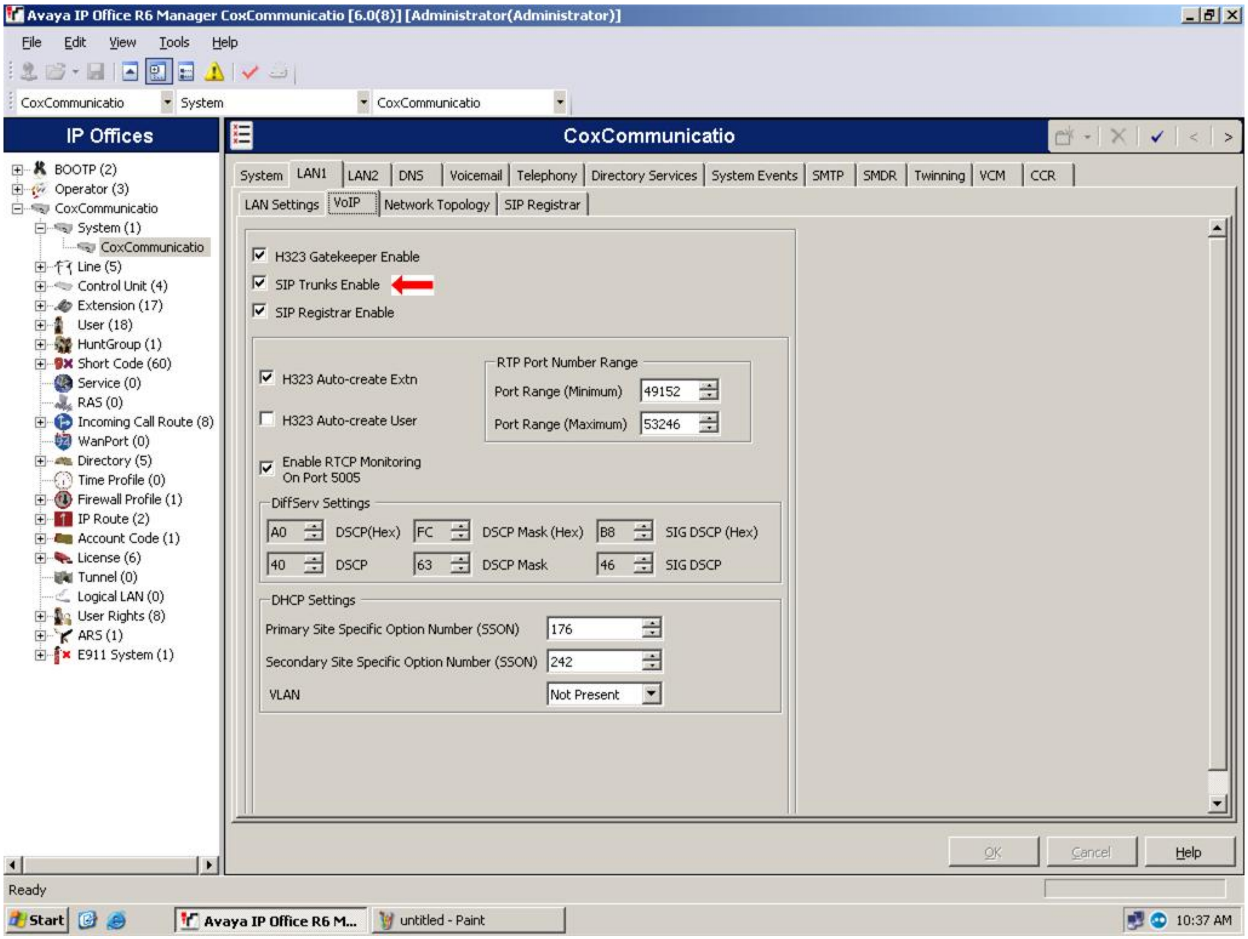

Figure 6 – LAN1: VoIP Information

7.1.4 LAN1 → Network Topology

For the IP Office server to use SIP Trunks, you **must not** input a STUN Server IP address. A Public IP Address is optional.

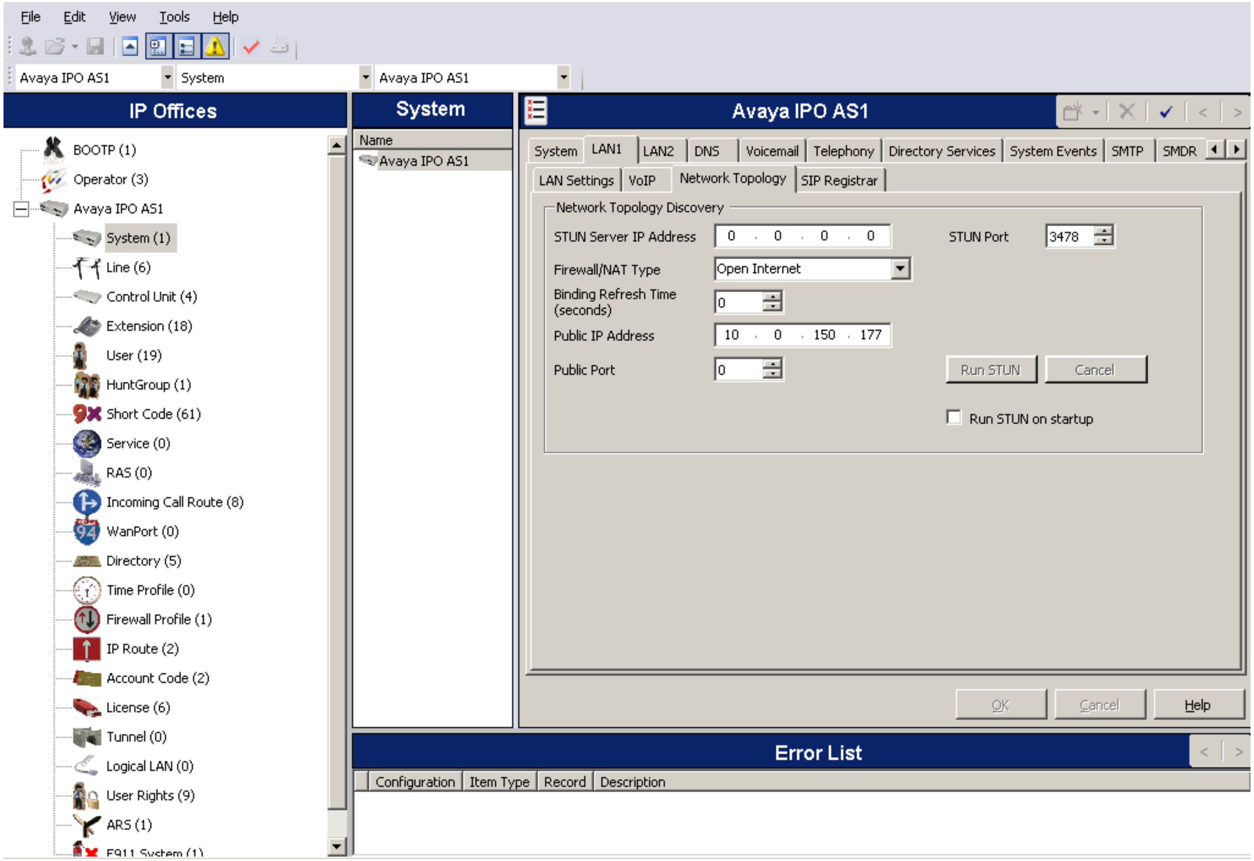

Figure 7 – LAN1: Network Information

7.1.5 Setup LAN1 → SIP Registrar

For the IP Office server to use SIP Trunks, the Domain Name is **not used**.

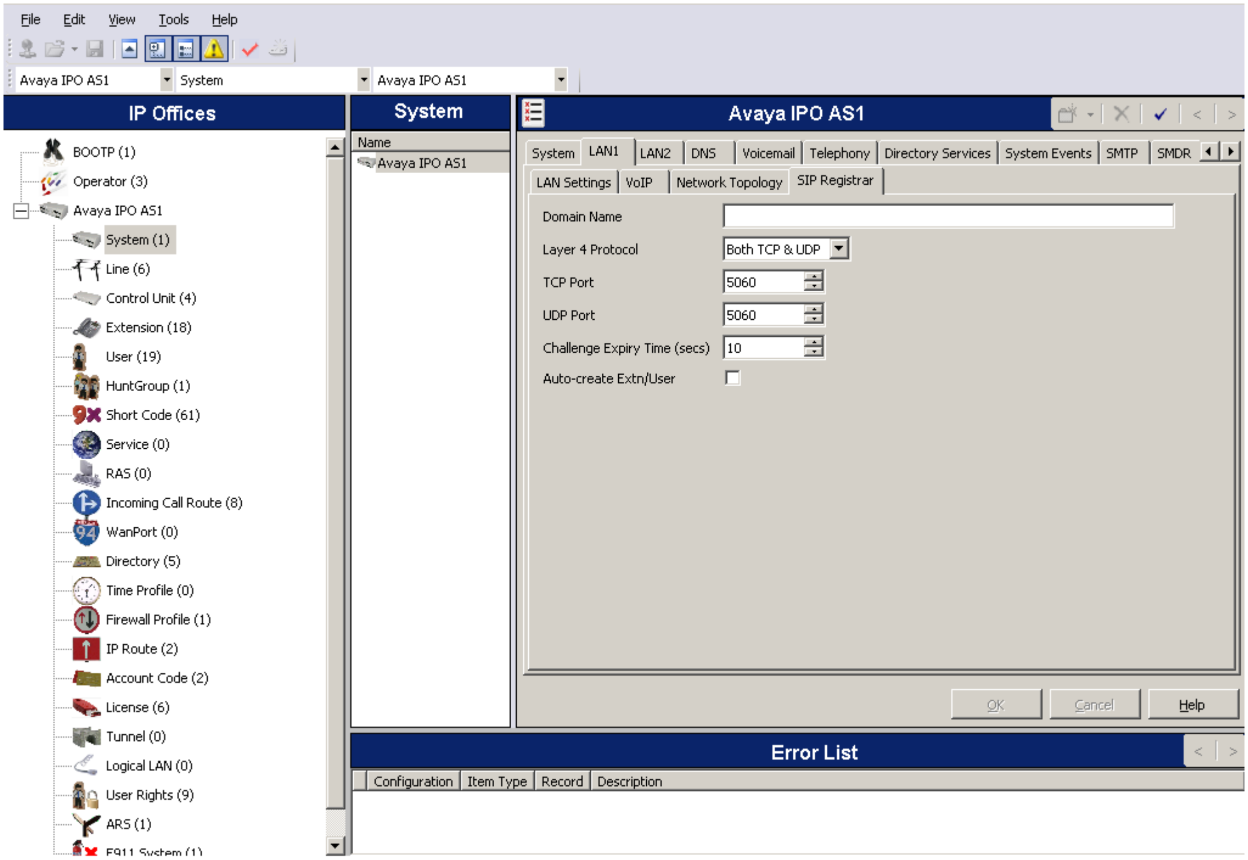

Figure 8 – LAN1: SIP Registrar

7.1.6 Setup LAN2 (WAN) → LAN Settings Information

The LAN 2 tab is intended to be used if the network administrator requires it but it is normally used as another way to access the IP Office server in the event the LAN port is un-accessible.

The WAN information is not required to be provided in this tab in order for the IP Office server to work. This information is needed if you wish to connect to the IP Office server from another PC or server. Matching of the IP addresses for the same network is needed for this direct connection to work.

Depending on how the system came up during the boot process will dictate the DHCP mode and settings that the server will use. For more information on this scenario please see the Appendix.

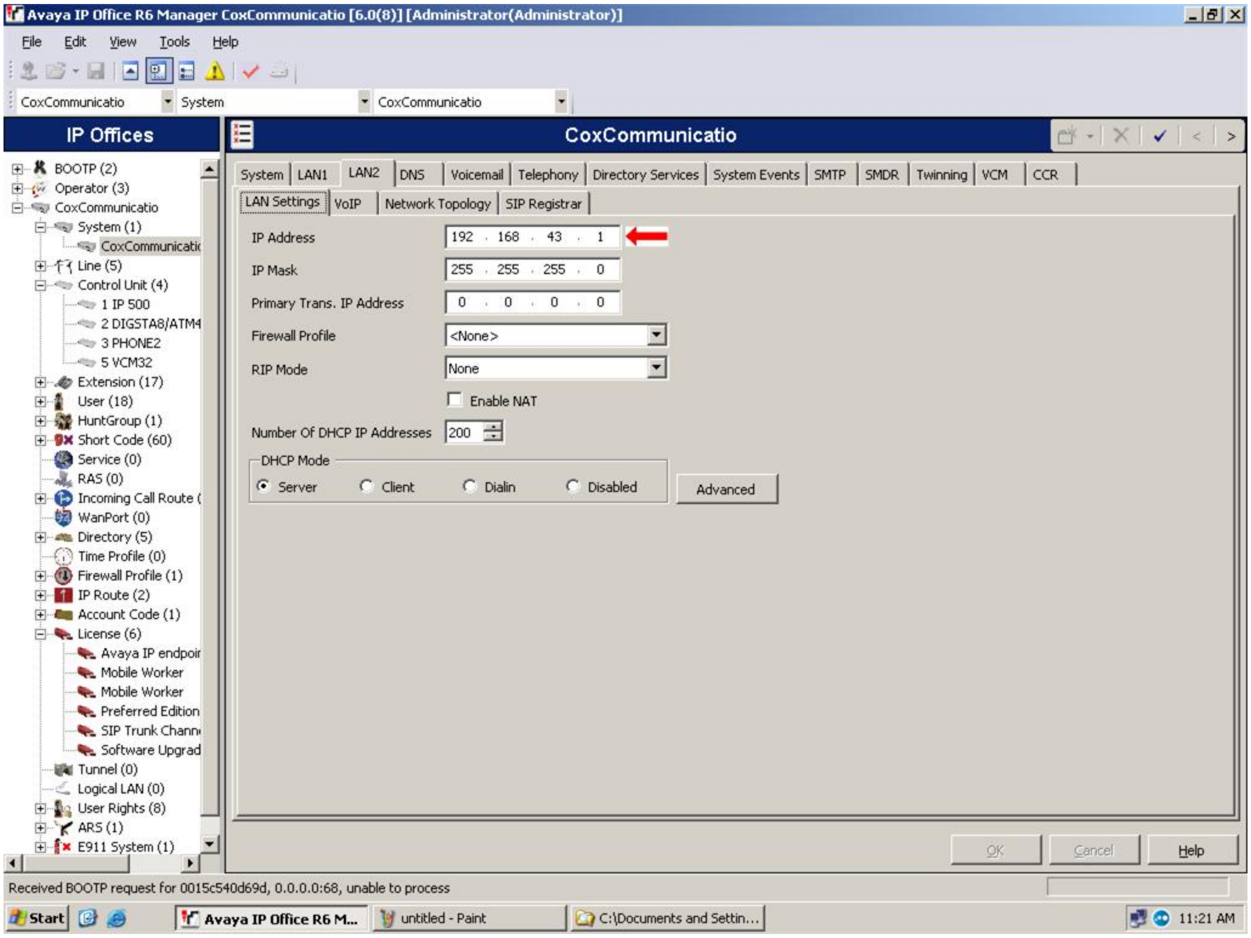

Figure 9 – LAN2 (WAN) Settings

7.1.7 Setup LAN2 (WAN) → VoIP Information

For the IP Office server to use SIP Trunks, it **must** be enabled. A license is required for this and is activated after installation. See section 2 for license information.

Check the following as shown in to **Figure 10** to enable these features:

- **SIP Trunk Enable**
- **SIP Registrar Enable**
- **H323 Auto-create Extn**

If a VLAN is configured, select the proper option from the drop down list.

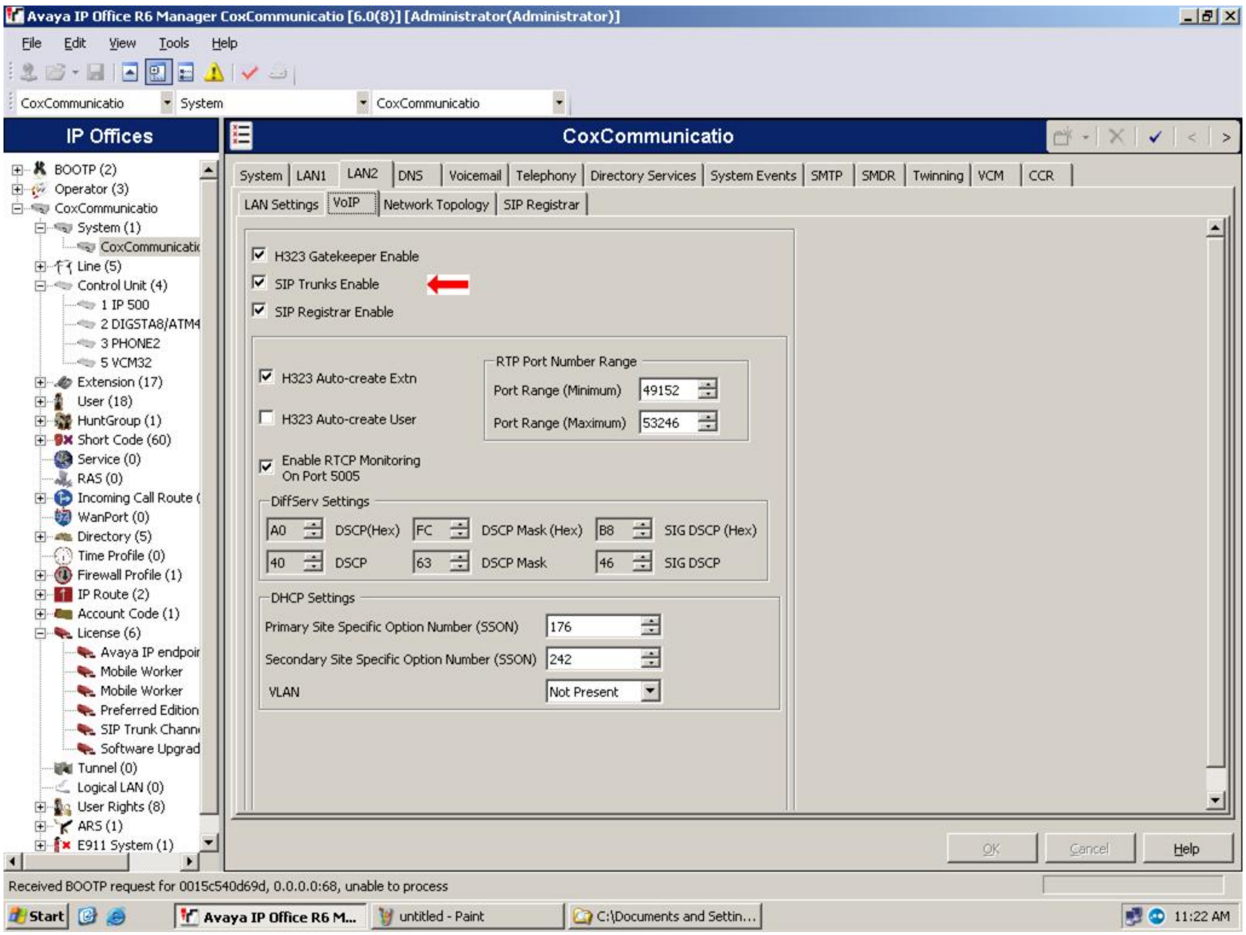

Figure 10 – LAN2 (WAN): Enable SIP Trunks

7.1.8 LAN2 (WAN) → Network Topology

Optional STUN Server IP Address support.

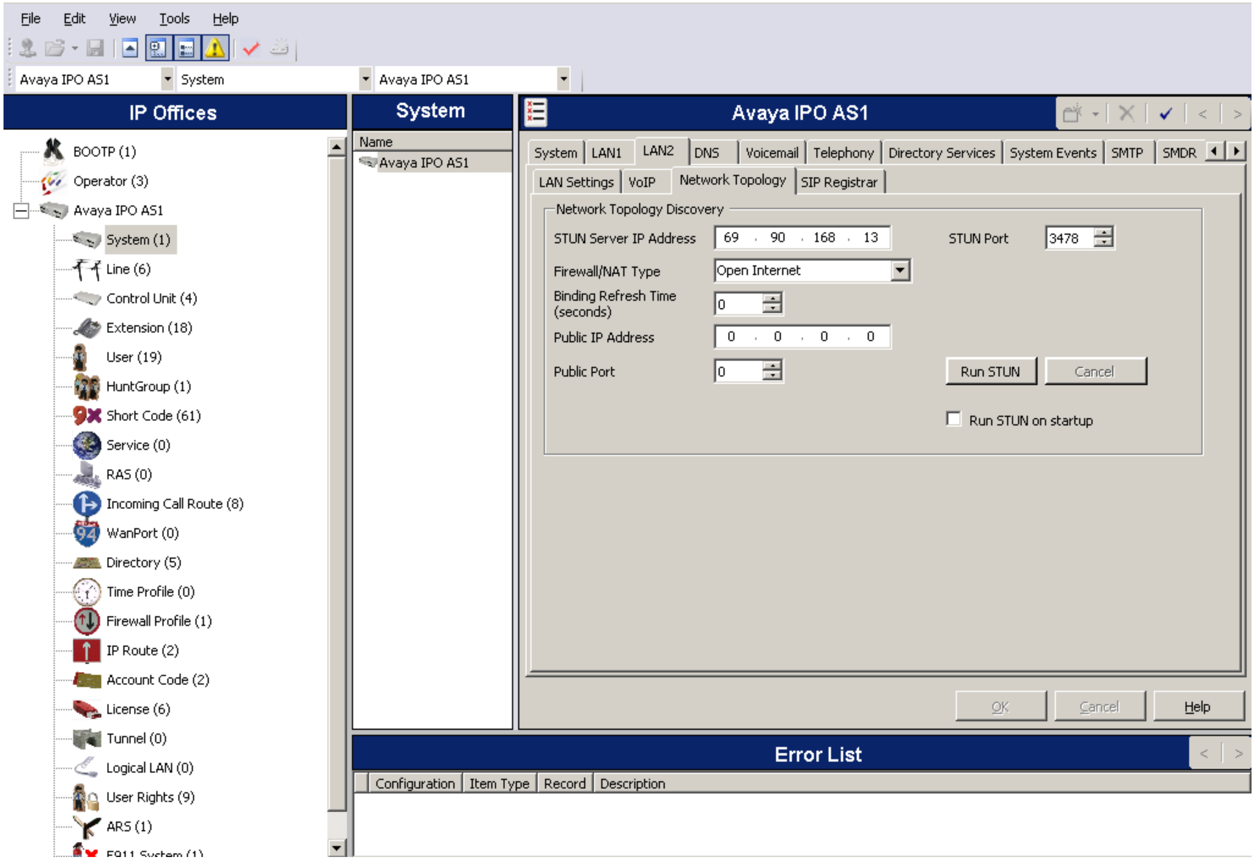

Figure 11 – LAN2 (WAN): Network Topology

7.1.9 Setup LAN2 (WAN) → SIP Registrar Information

The SIP Registrar Domain Name is not required and ignored by Cox if a value is input. Please leave it blank as shown:

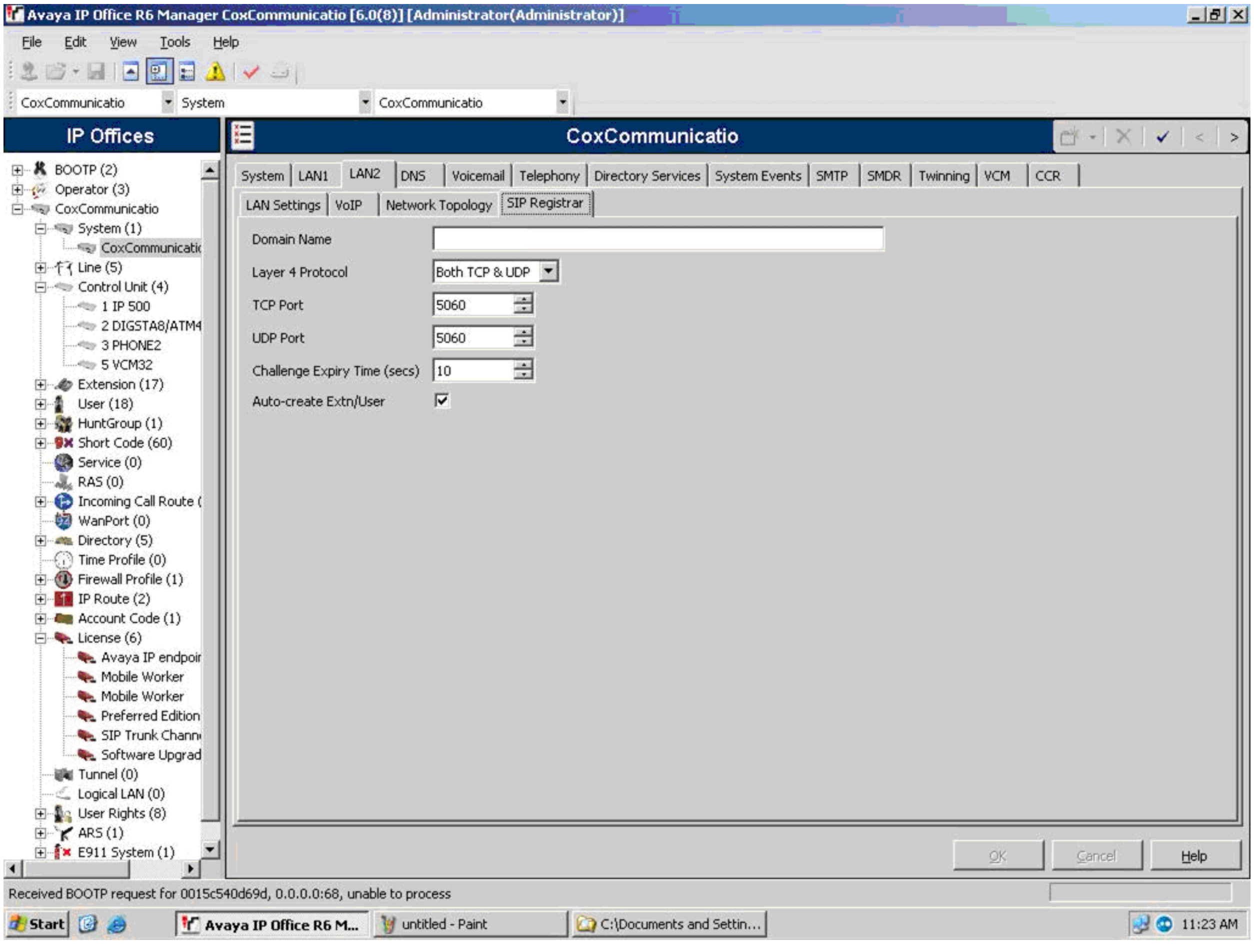

Figure 12 – LAN2 (WAN): SIP Registrar

7.2 Avaya IP Office SIP Configuration Guide

7.2.1 How to Check For SIP Trunking Licenses

To make calls using SIP you must have a valid license that can be purchased through Avaya business partners, in the number of 1, 5, 10, 20 or a combination of all the above up to 128 instances of the same license. Avaya will provide a license that will need to be inserted into license form. An example is provided in **Figure 13**. License can be shared among different SIP trunks; the number of instances represents the maximum number of calls that can be dialed or received at the same time by IP Office using any of its SIP trunks.

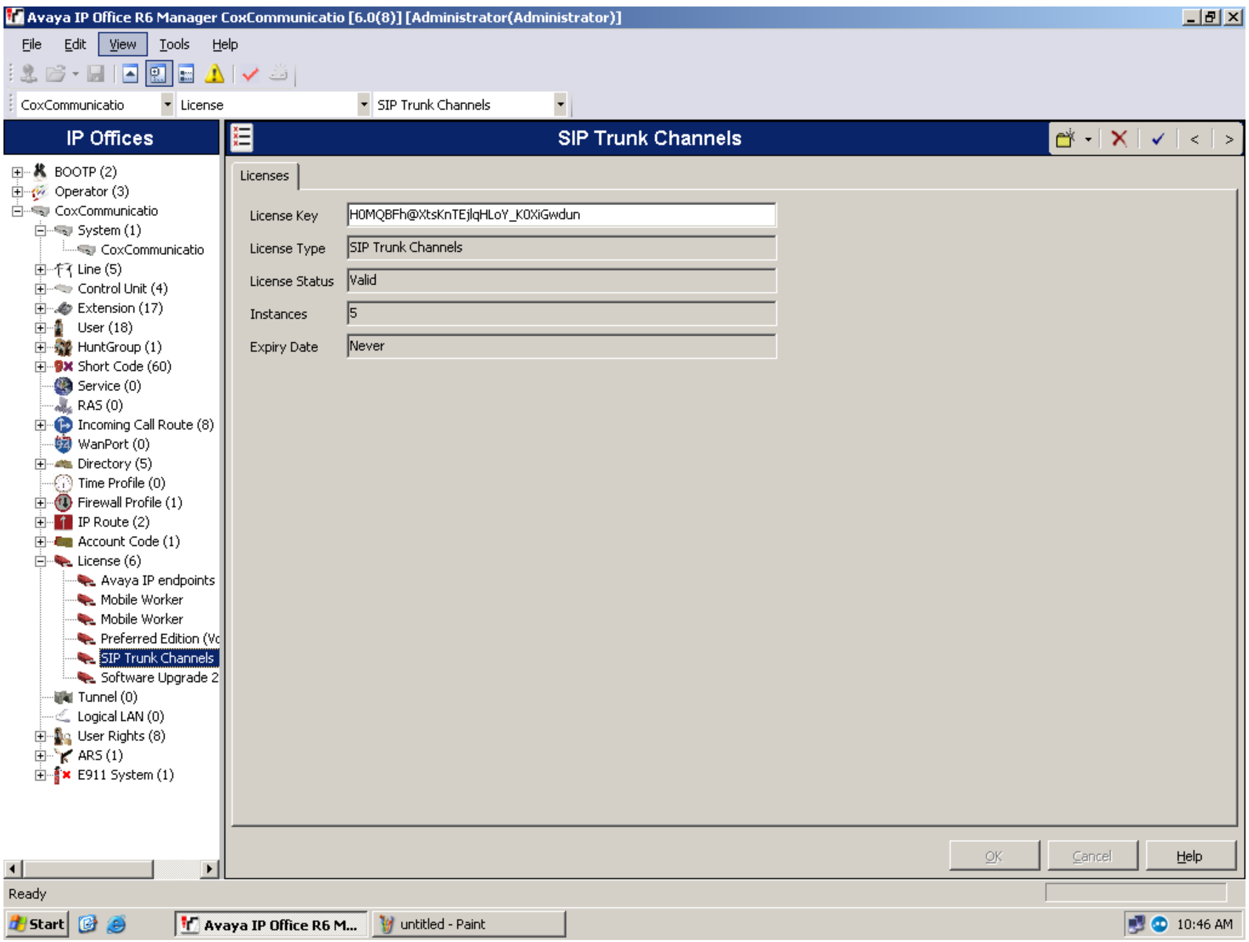

Figure 13 – SIP Trunk Licenses

- *License Key* is the license identifier that will be provided by Avaya business partners.
- *License Type* must be set to *SIP Trunk Channel*.
- *License Status* should be set to *Valid*, if the acquired license is a valid one.
- *Instances*, will display the number of license instance that have been purchased.
- *Expiry Date* will indicate the expiration of the license

Business®

7.2.2 Configure IP Routes to Cox IP Network

The Cox E-SBC IP Address must be input for the Gateway IP Address. This is used for setting up the SIP Trunk static route table. Please refer to **Figure 3** for the example IP addresses used in the lab reference network. Your actual IP addresses may be different.

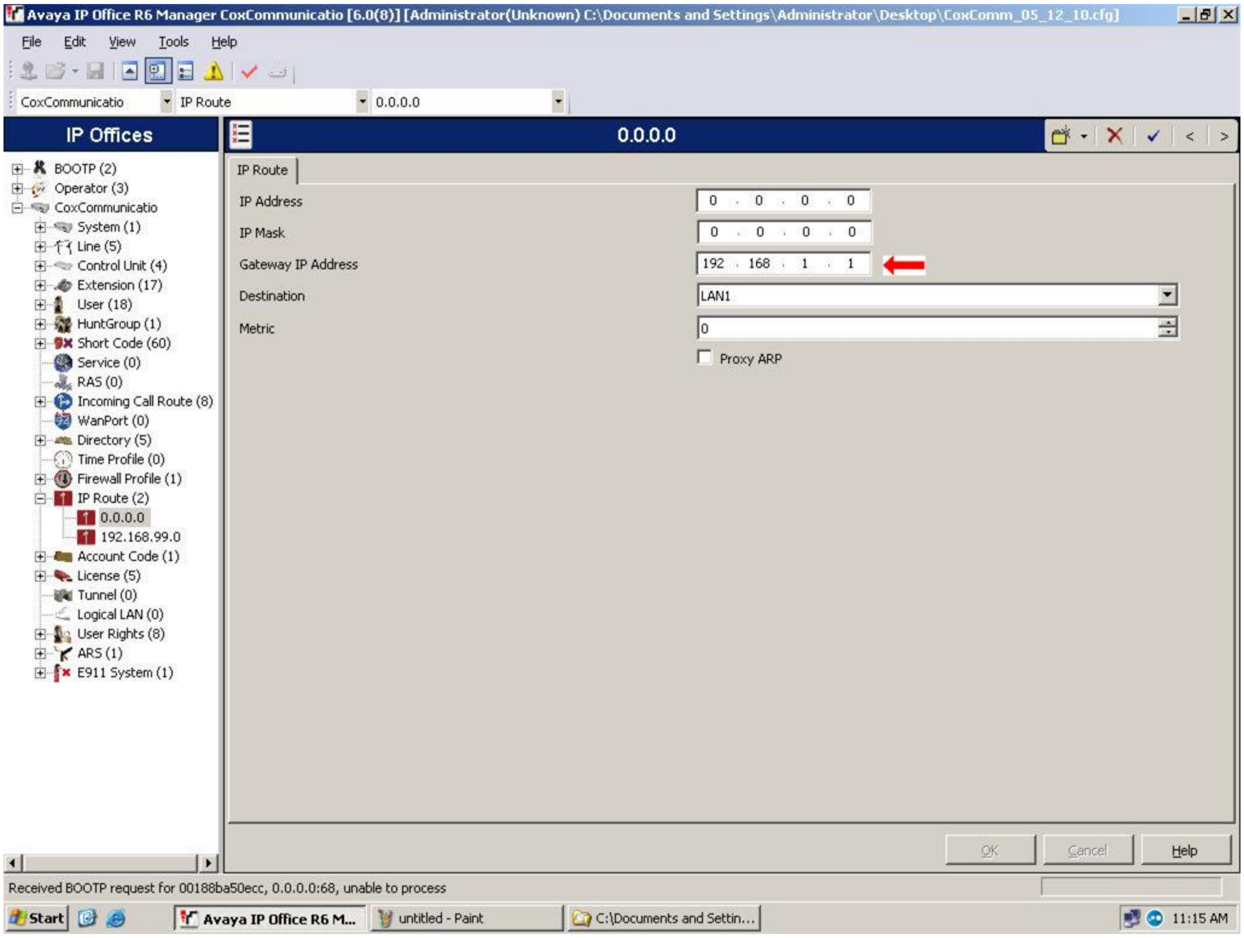

Figure 14 – IP Routes to the Cox Network

7.2.3 Setting Up Primary Line → SIP Line

This section deals with the SIP line tab on the SIP Line configuration. Enter the required information as listed below. See **Figure 3** for a cross reference of the lab IP addresses shown here. Your IP addresses may be different.

- **Line number** is automatically assigned by the IP Office Manager application when the SIP Line is created.
- **ITSP Domain Name** must be the static LAN IP address of the Cox E-SBC. Cox does not use DNS and the E-SBC LAN IP Address must be input instead of a domain name. The Cox E-SBC will internally translate the LAN IP Address to a Domain Name within the Cox network.
- **ITSP IP Address** must be the static LAN IP address of the Cox E-SBC.
- **In Service** shall be checked.
- **Use TEL URI** shall not be checked.
- **Use Network Topology Info** is set it to LAN 1.

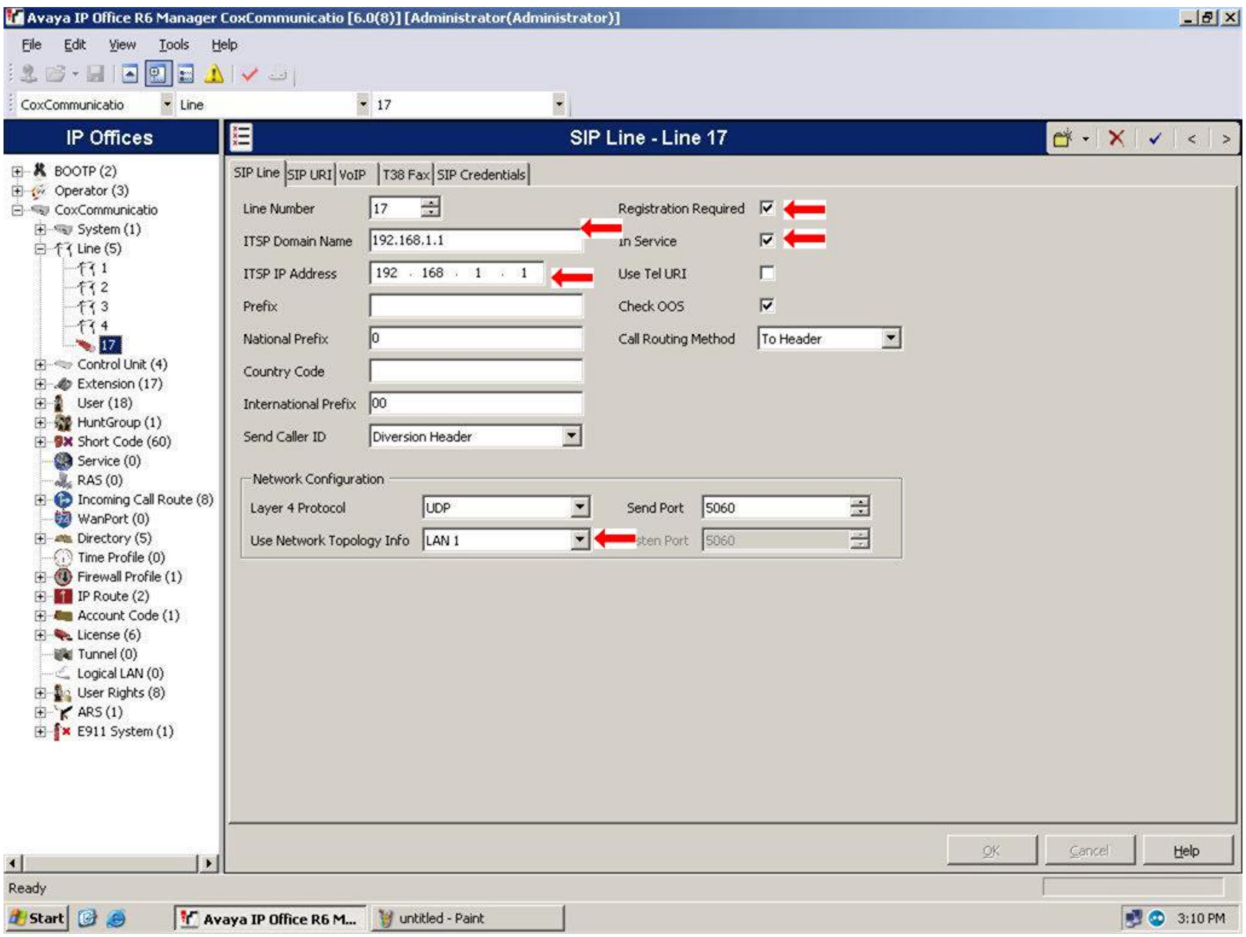

Figure 15 – SIP Primary Line

Setting Up Phone Lines and ICR → Standard

This section deals with how the Direct Inward Dial (DID) telephone numbers (TN's) are assigned to users in the IP Office and how they are routing via the "Incoming Call Route". Enter the required information as listed below for every DID number in the PBX inventory.

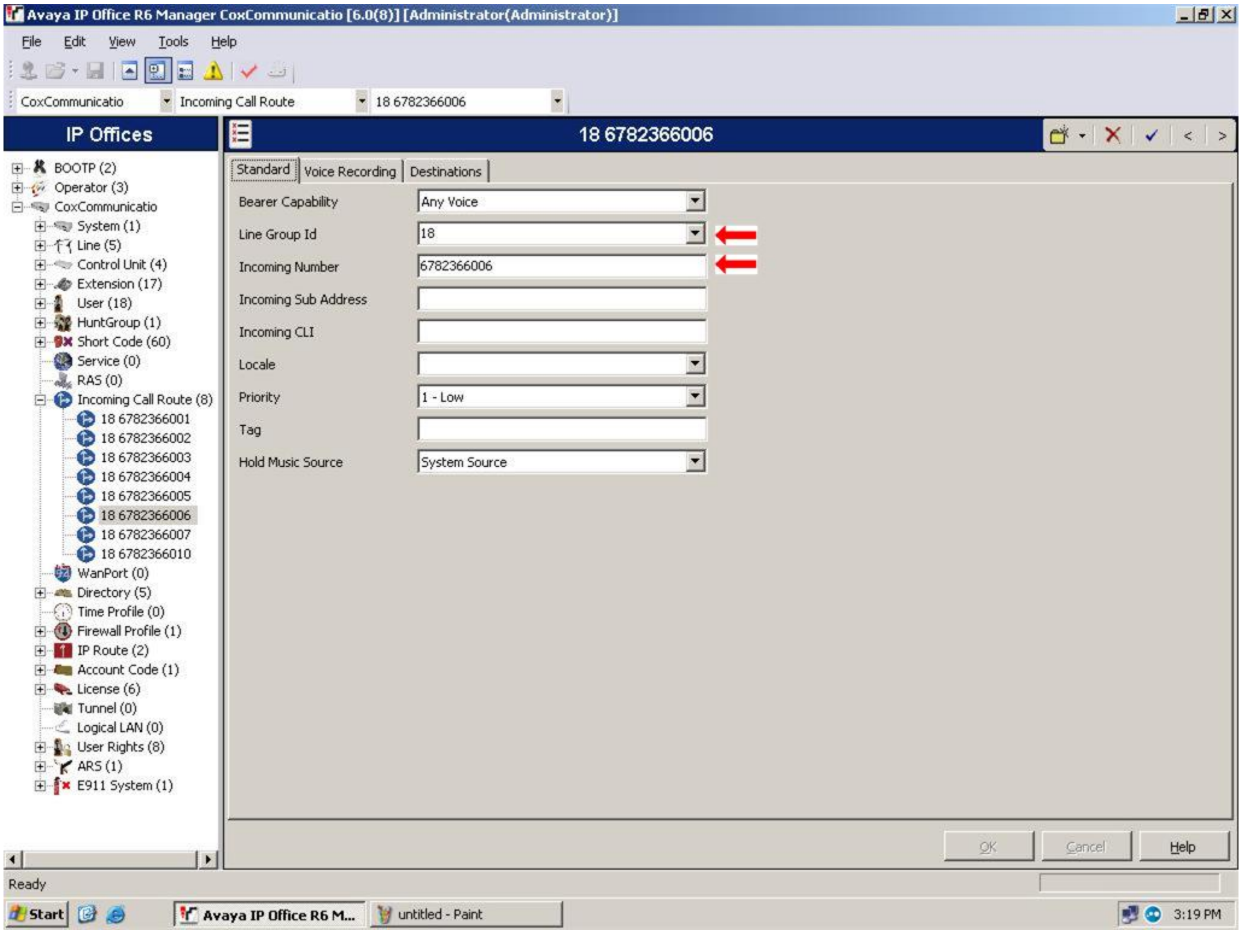

Figure 16 – Incoming Call Route

Once the incoming number has been entered, select the “Destinations” tab and set to the proper destination using the drop down menu as shown in **Figure 17**.

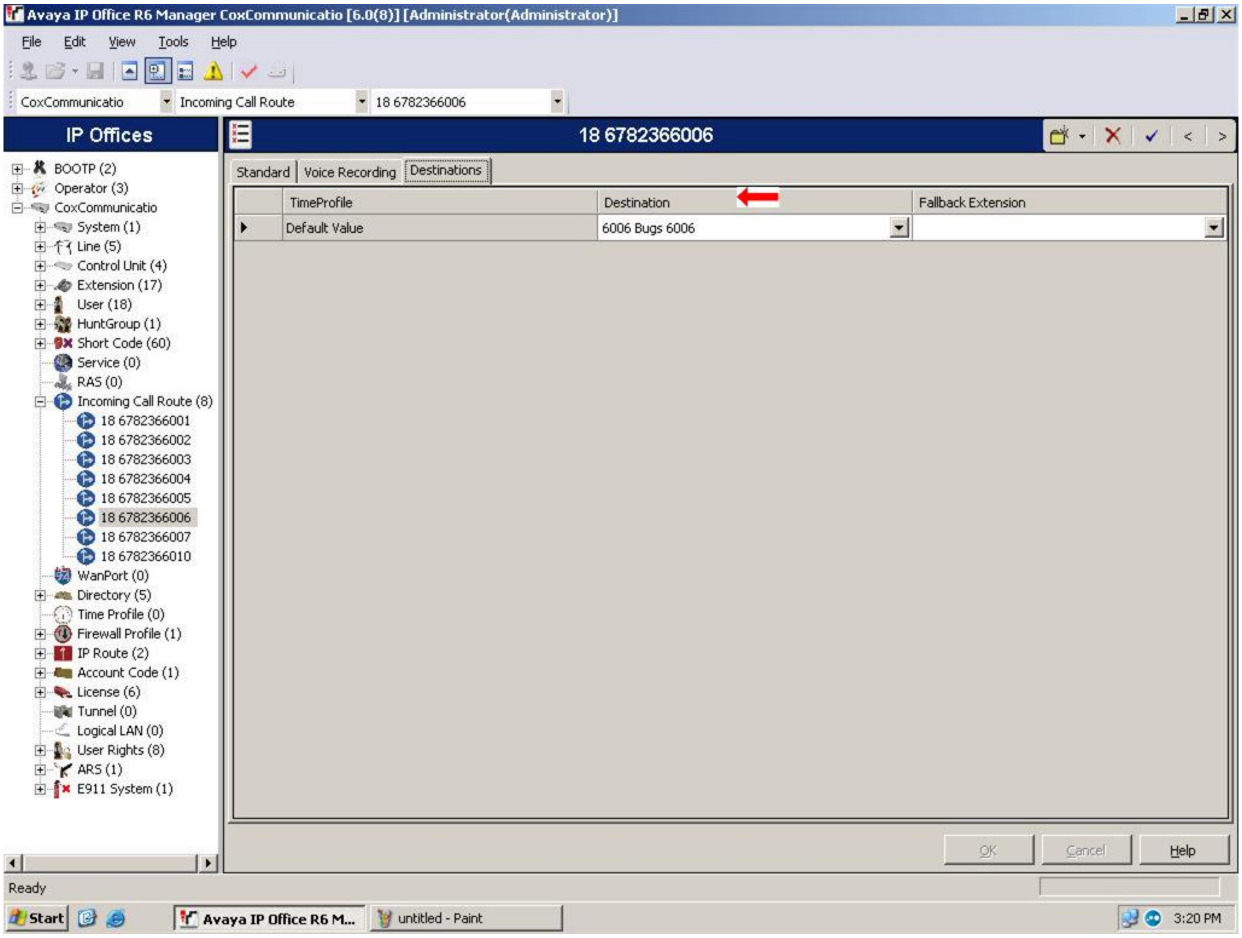

Figure 17 – Incoming Call Route Destinations

7.2.4 Configure the SIP URI

To facilitate the use of proper translation of identities across the network, it is necessary to format the URI for SIP messages. Enter the required information as illustrated in the follow screen:

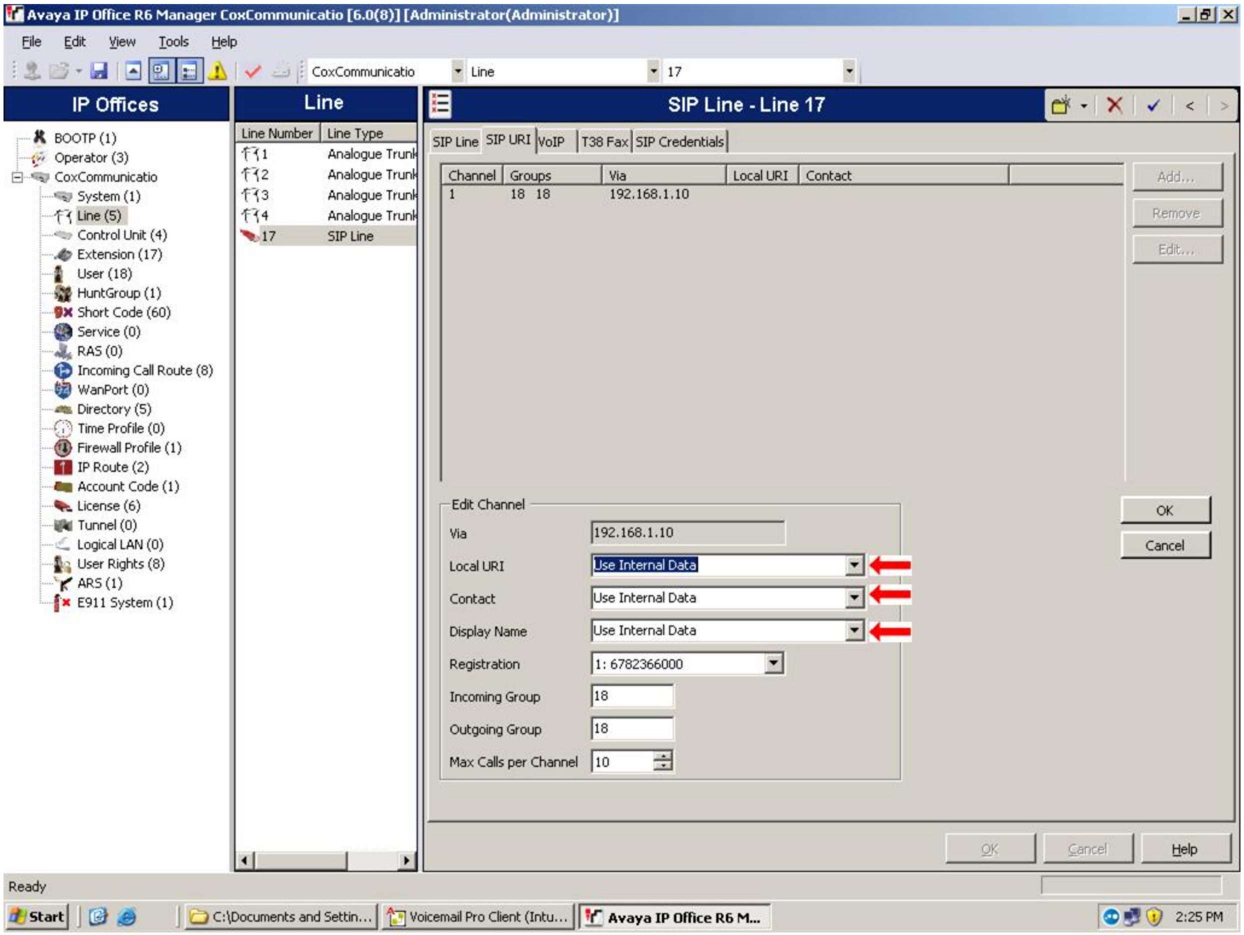

Figure 18 – Incoming Call Route: SIP URI

To provide the use of proper aliasing of names and caller ID, the SIP URI must be set to "Use Internal Data". This will allow the "SIP" tab in the user section to be available. Please refer to the next section for information on configuring this particular tab.

IMPORTANT: The Trunk Group Pilot Number, a.k.a. the Cox Billing Telephone Number (BTN) must be entered in the "Registration" field in **Figure 18**.

7.2.5 SIP Lines → VOIP

Select the SIP Lines → VoIP tab and enter the following values in the Compression Mode → Advanced fields:

- Select **G.711 ULAW 64K**
- **Call Initiation Timeout** to 5 seconds
- **DTMF Support** RFC2833
- Check **FAX Transport Support**
- Check **Re-Invite Supported**

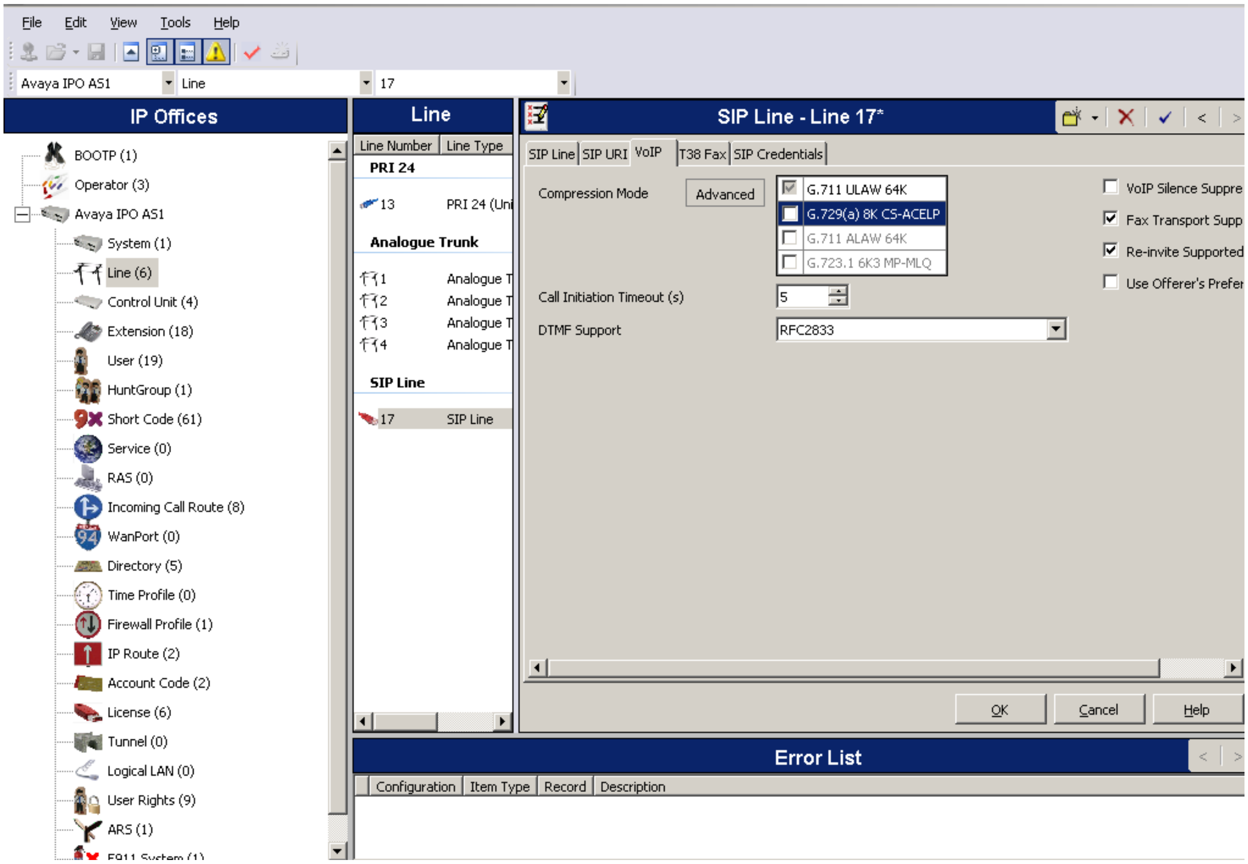

Figure 19 – SIP Lines: VoIP Compression Mode

7.2.6 SIP Lines → T.38 Fax

Note: Please check with your Cox Account Representative to verify T.38 support is available in your area.

Select the SIP Lines → T38 Fax tab and enter the following values:

- **T.38 Fax** version 2
- Check **Scan Line Fix-up**
- Check **TFOP Enhancement**

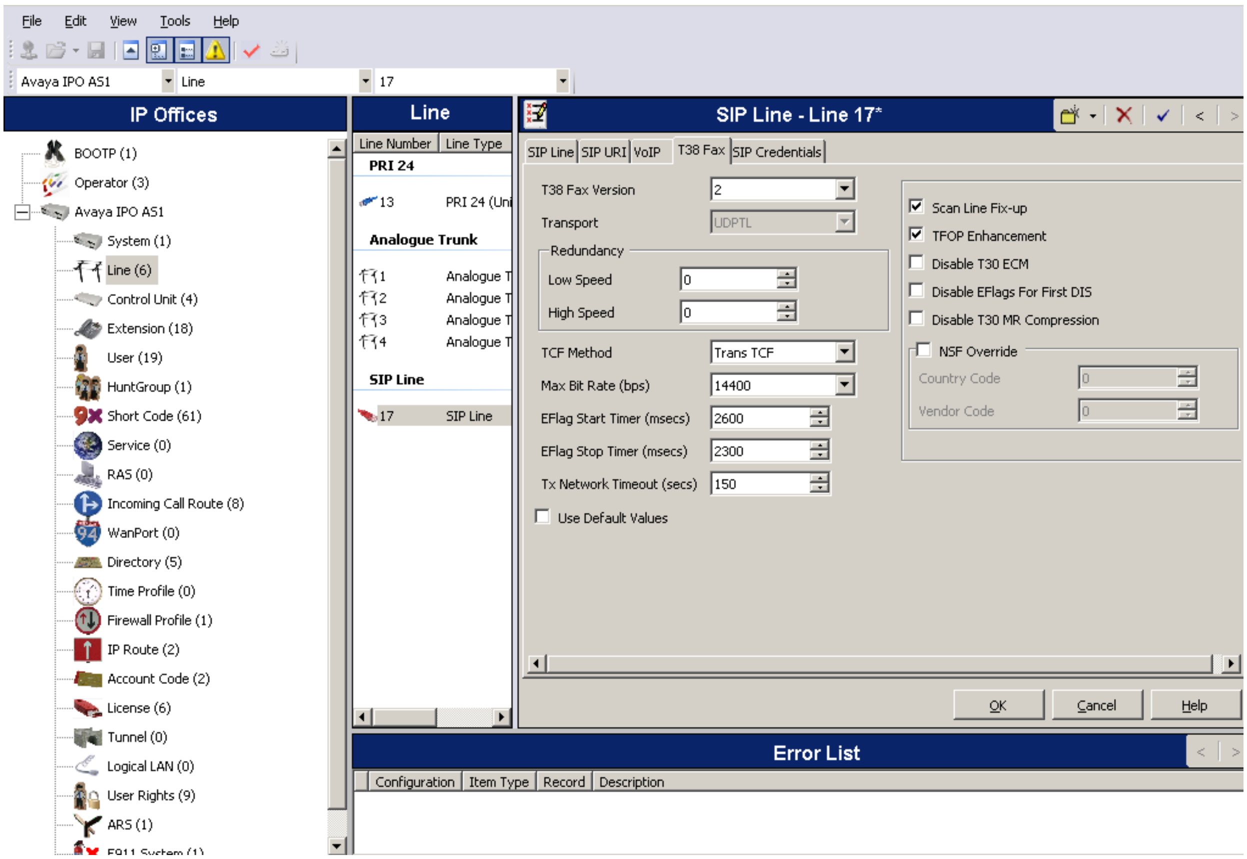

Figure 20 – SIP Lines: T.38 FAX

7.2.7 SIP Lines → SIP Credentials

In order to authentication the IPO with the Cox SIP Trunk service, the SIP Credentials must be configured in the PBX:

- **User Name**: must be the Trunk Group Pilot number, which in this example is “6782366000”.
- **Authentication**: must also be the Trunk Group Pilot number, e.g. “678236600”.
- **Password**: The randomly generated SIP Authentication password is provided by your Cox Account Representative and must be kept confidential. For this example only, we’ll make it simple and use the Trunk Group Pilot number “678236600”.
- **Expiry**: 60 seconds

Tip: Contact your Cox sales engineering representative to confirm the SIP Registration User Name and Password. Cox can confirm and/or change these values to match the PBX.

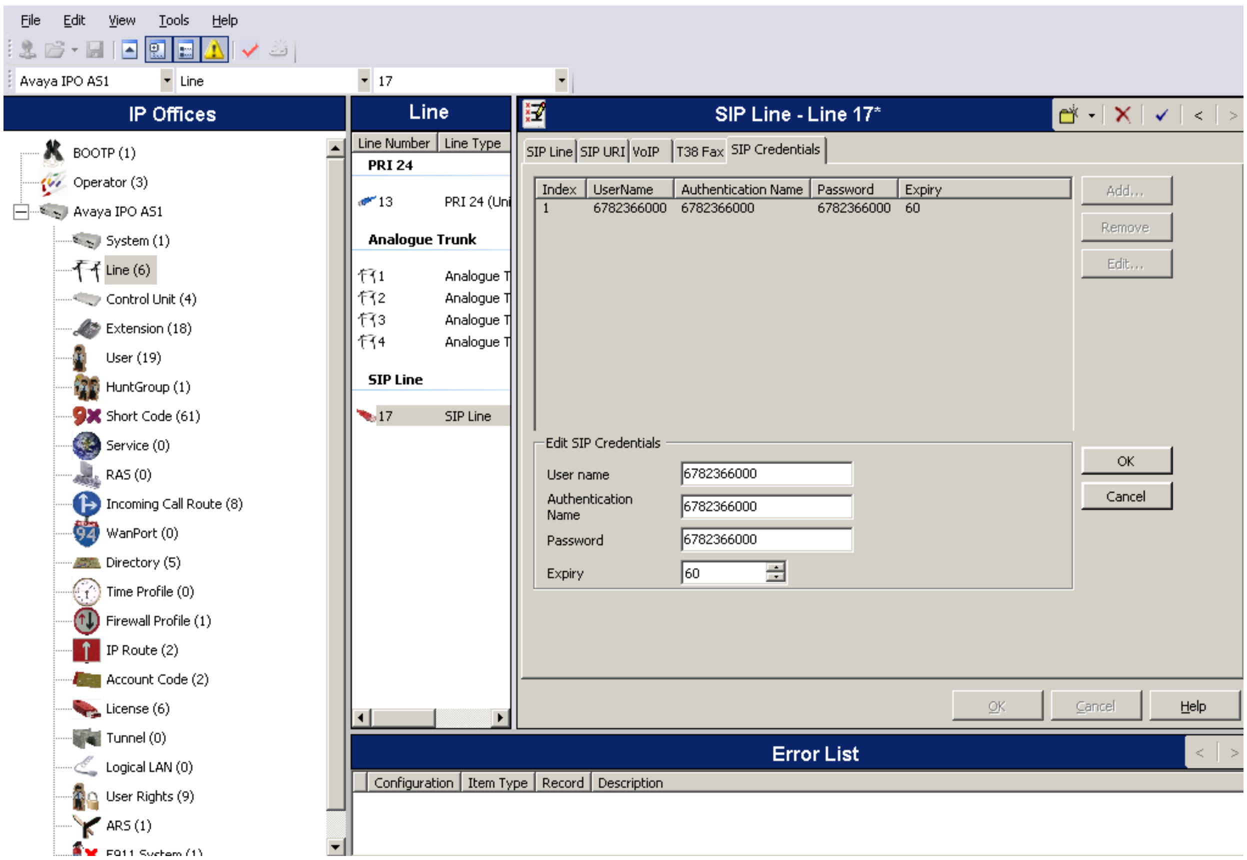

Figure 21 – SIP Lines: Credentials

7.2.8 User → SIP Aliasing (Optional)

Click the User → SIP tab and enter the required information as listed below.

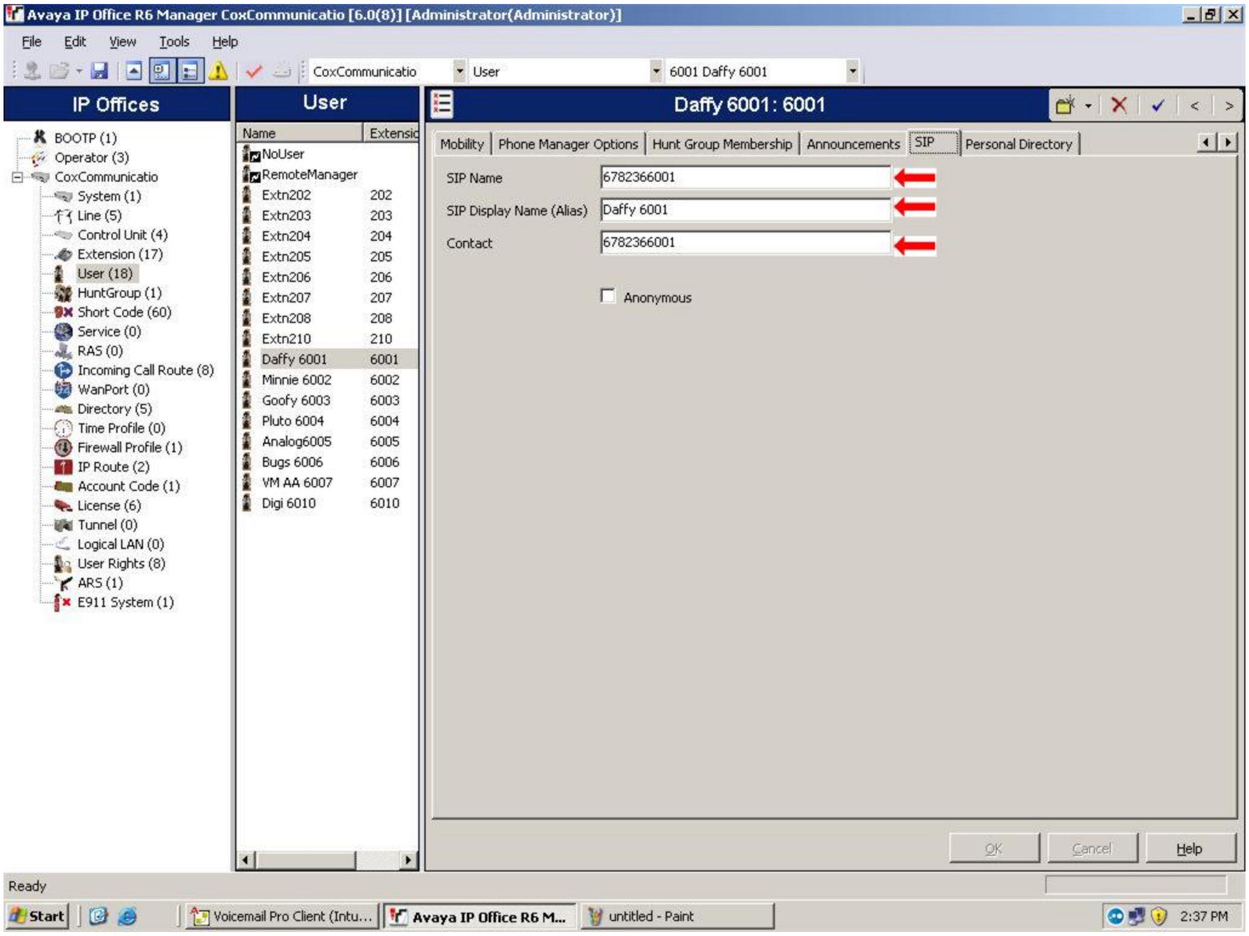

Figure 22 – SIP User Alias

7.3 Avaya Voice Mail Pro Configuration Guide

7.3.1 Setup Voicemail in System → Voicemail

1. In IP Office Manager, select “System”.
2. Click the “Voicemail” tab.

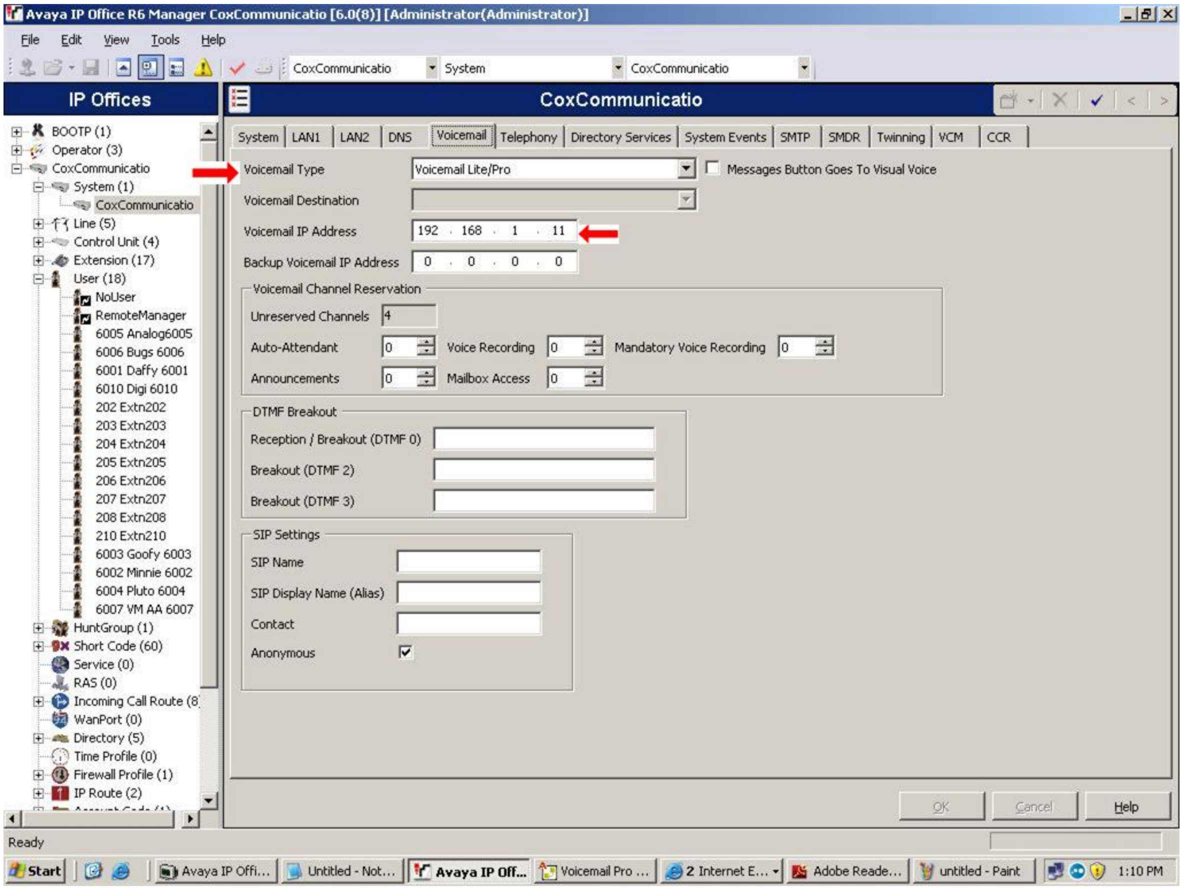

Figure 23 – Voicemail Configuration

3. Unless detailed otherwise, the option Voicemail Lite/Pro should be use with Voicemail Pro server. Additional options are displayed depending on the selected voicemail type.

   - Voicemail IP Address - By default the IP Office connects to the Voicemail Pro server by using the address 255.255.255.255 to broadcast for any server on the same LAN as itself. When it receives a response it will use that voicemail server. However it may be necessary or desired to set this access to an exact address. Change the default address (*255.255.255.255)* to the IP address of the PC on which the Voicemail Pro server is running.

7.3.2 User Voicemail Setup in System → User

1. Select the “Voicemail” tab.
2. By default the tick is checked for users’ voicemail to be active.

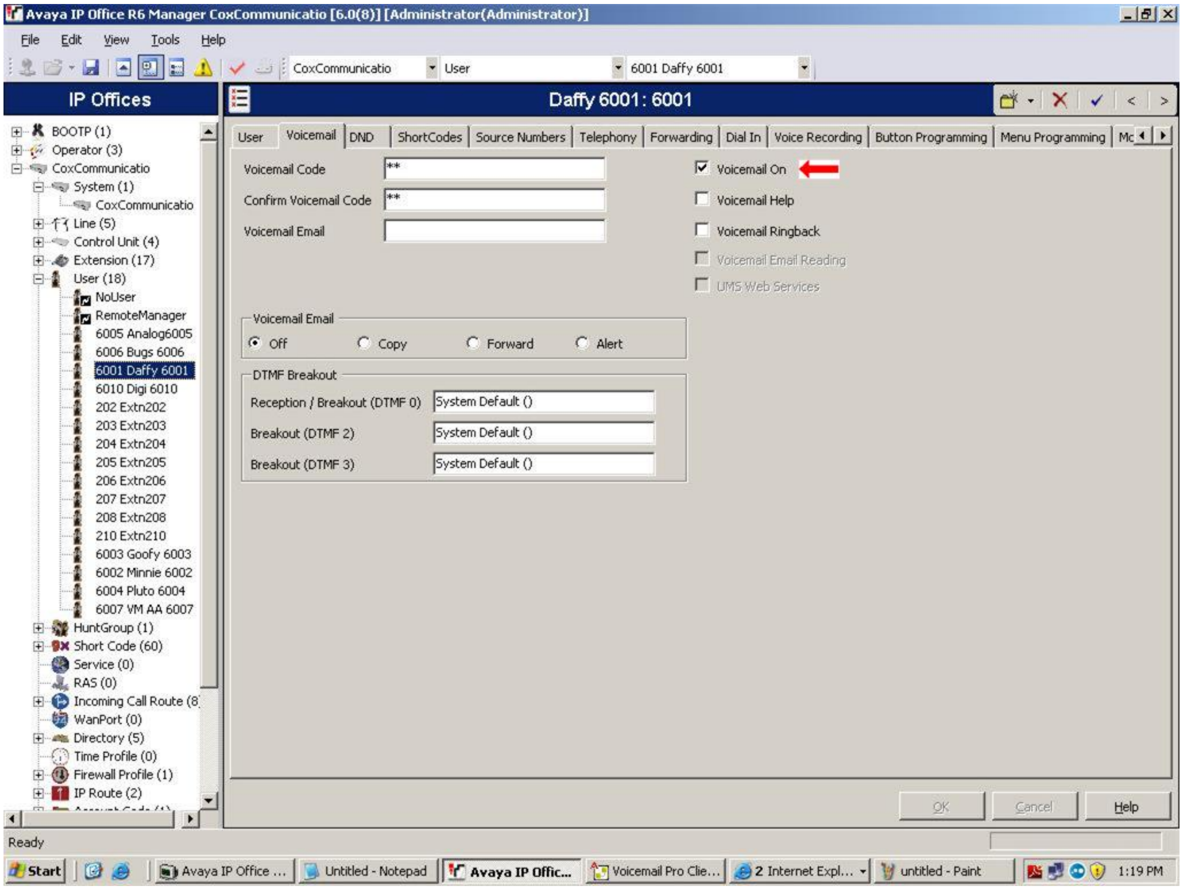

Figure 24 – Enable Voicemail

3. Voicemail will function normally with no other options selected but other options are available to enhance full feature capabilities of the voicemail system.

Appendix A: Troubleshooting Problems

A1 - DHCP Server and Default Settings (Information only)

When a new or defaulted IP Office control unit is switched on, it requests IP address information from a DHCP Server on the network. This operation will occur whether the LAN cable is plugged in or not.

- If a DHCP server responds within approximately 10 seconds, the control unit defaults to being a DHCP client and uses the IP address information supplied by the DHCP server.
- If no DHCP Server responds, the control unit defaults to being the DHCP server for the LAN using the following settings:
- For its LAN1 it allocates the IP address 192.168.42.1 and IP Mask 255.255.255.0. It supports 200 DHCP clients using the addresses range 192.168.42.2 and 192.168.42.201, the IP Mask 255.255.255.0 and default gateway address 192.168.42.1 (the Control Unit's LAN1 address).
- For its LAN2 if supported, it allocates the IP address 192.168.43.1 and IP Mask 255.255.255.0.
- Note that the IP Office does not check that these addresses are valid and or available on the network.

A2 – Running Avaya IP Office R6 SysMonitor

A2.1

Avaya IP Office R6 SysMonitor is a network tool installed alongside Avaya Office Manager for use with tracing inbound and outbound call processing. Any call tracing performed using this program will be actual messages and information from the LAN1 and/or LAN2/WAN ports. For specific use of program functions use the help file. For setting call trace parameters use <ctrl> + T for selection window. See example trace below.

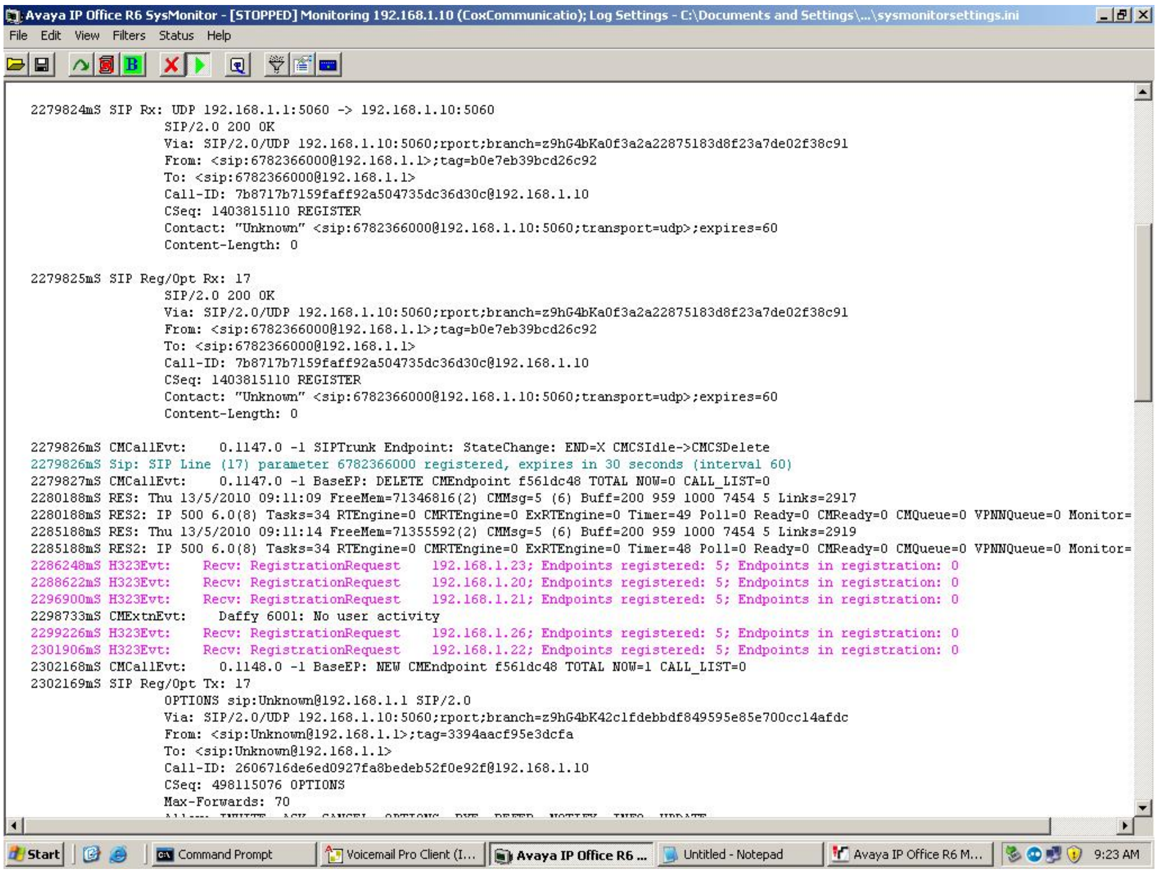

Figure 25 – SysMonitor Call Trace

A2.2

In order to view the proper capture of SIP messages in SysMonitor, select desired signal tracing from the configuration screen as shown below.

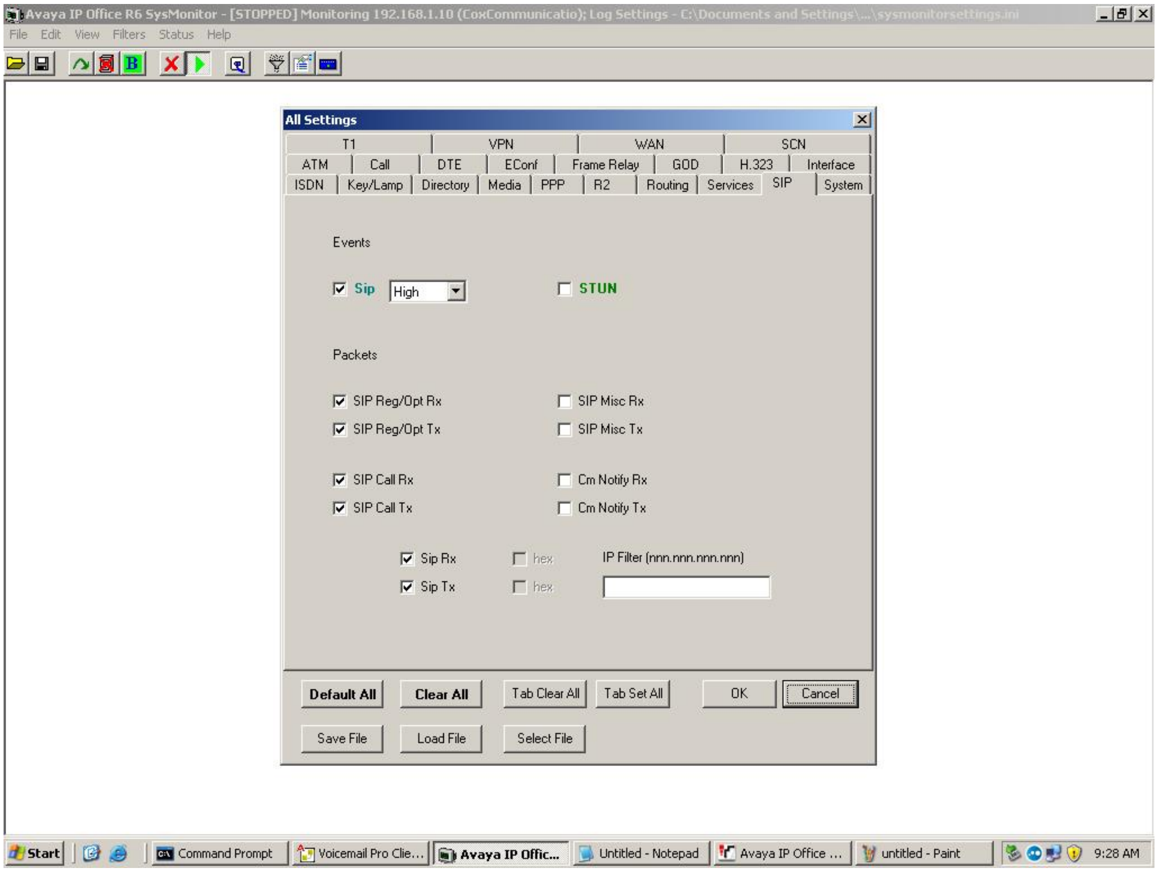

Figure 26 – SysMonitor SIP Signaling Capture

A2.3

In order to view the proper capture of CALL processes in SysMonitor, select desired signal tracing from the configuration screen as shown below.

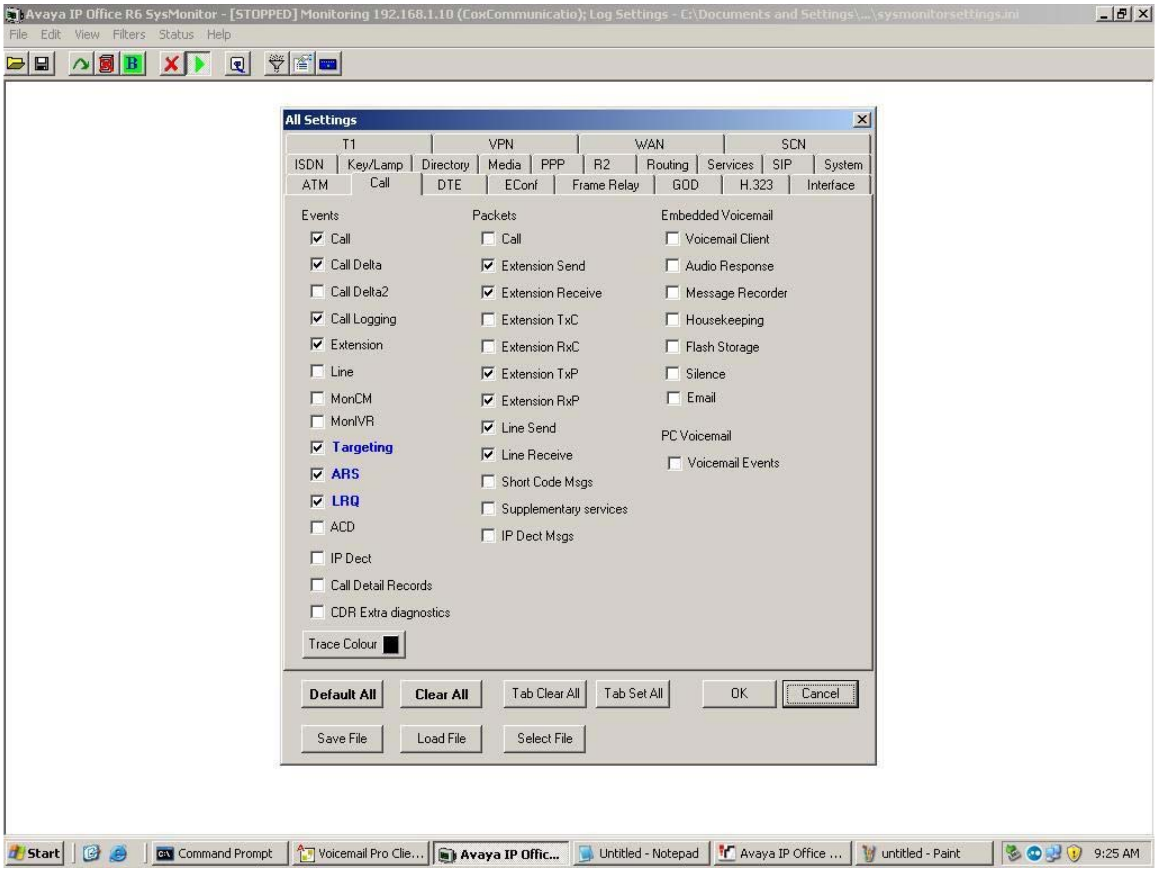

Figure 27 – SysMonitor Call Processes

A3 – Using Putty for call tracing

Settings for establishing SSH on port 22 with the EdgeMarc device, use the following options in Putty:

1. Host Name (or IP address) = 10.0.150.177 (see figure A3-1)
   - Port = 22
   - Protocol = SSH

2. In Session Logging = Log printable output only (see figure A3-2)
   - Select location to save log file of Putty session using the "browse" button.
   - The remaining can remain at default.

3. In Window settings = Control the scrollback = 9999 (see figure A3-3)

For examples please see below. Once you have your selections made you can save the particulars for a specific setup using the SAVE button on the "SESSION" page and provide a "Saved Sessions" name that corresponds to the connection you are setting up. All the parameters for this particular session will be remembered for future use of this session.

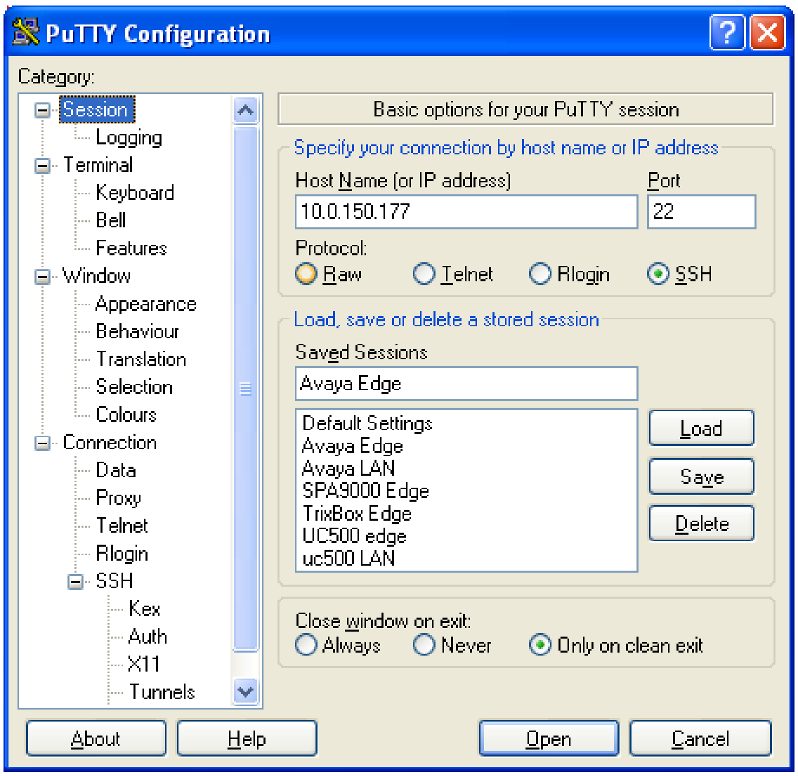

Figure 28 – PuTTY

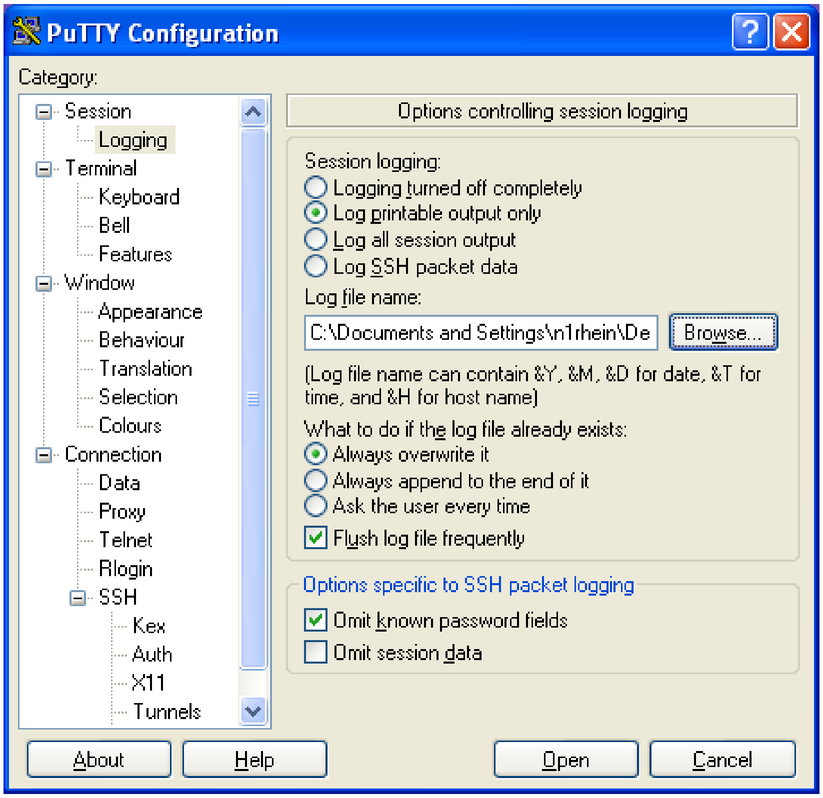

Figure 29 – Enable PuTTY Logging

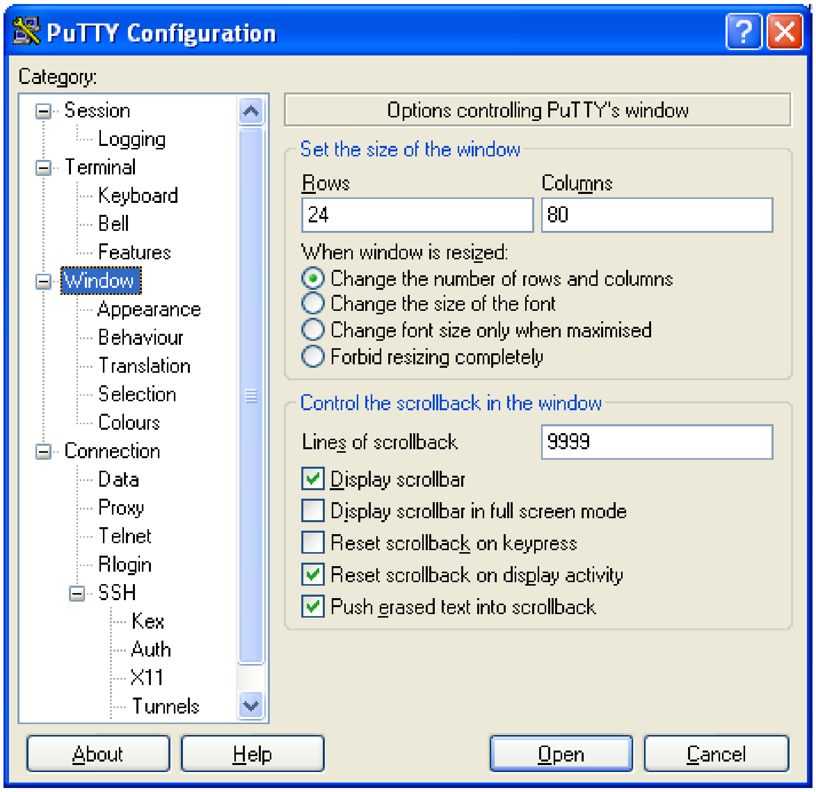

Figure 30 – PuTTY Scrollback Window

8 Appendix B: Sample SIP Traces between IPO and EdgeMarc 4550

```
/* SIP Registration examples call flow */
REGISTER sip:192.168.1.1 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.10:5060;rport;branch=z9hG4bKc8ba7a35c3b4734987c08f817e75a756
From: <sip:6782366000@192.168.1.1>;tag=aa37314b64a62d83
To: <sip:6782366000@192.168.1.1>
Call-ID: 80444c150a74e438091eb9e89f70b5e0@192.168.1.10
CSeq: 310040039 REGISTER
Contact: "Unknown" <sip:6782366000@192.168.1.10:5060;transport=udp>
Expires: 60
Max-Forwards: 70
User-Agent: IP Office 6.0 (8)
Supported: timer
Content-Length: 0


SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP
192.168.1.10:5060;received=10.0.150.177;rport=5060;branch=z9hG4bKc8ba7a35c3b4734987c08f817e75a756
From: <sip:6782366000@192.168.1.1>;tag=aa37314b64a62d83
To: <sip:6782366000@192.168.1.1>;tag=SDpjgh999-
Call-ID: 80444c150a74e438091eb9e89f70b5e0@192.168.1.10
CSeq: 310040039 REGISTER
WWW-Authenticate: DIGEST realm="BroadWorks", nonce="BroadWorksXg975k108Tri15vtBW", algorithm=MD5,
qop="auth"
Content-Length: 0


REGISTER sip:192.168.1.1 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.10:5060;rport;branch=z9hG4bKf447a8a104214f3e63992d27c49ea846
From: <sip:6782366000@192.168.1.1>;tag=aa37314b64a62d83
To: <sip:6782366000@192.168.1.1>
Call-ID: 80444c150a74e438091eb9e89f70b5e0@192.168.1.10
CSeq: 310040040 REGISTER
Contact: "Unknown" <sip:6782366000@192.168.1.10:5060;transport=udp>
Expires: 60
Authorization: Digest
username="6782366000",realm="BroadWorks",nonce="BroadWorksXg975k108Tri15vtBW",response="3644d5adb
06f20b03238b7ef6b85bfaf",uri="sip:192.168.1.1",algorithm=MD5,qop=auth,nc=00000001,cnonce="b08716c
bfedba082e328"
Max-Forwards: 70
User-Agent: IP Office 6.0 (8)
Supported: timer
Content-Length: 0


SIP/2.0 200 OK
Via: SIP/2.0/UDP
192.168.1.10:5060;received=10.0.150.177;rport=5060;branch=z9hG4bKf447a8a104214f3e63992d27c49ea846
Record-Route: <sip:6782366000@192.168.1.1;lr>
From: <sip:6782366000@192.168.1.1>;tag=aa37314b64a62d83
To: <sip:6782366000@192.168.1.1>;tag=SDpjgh999-1314960979-1273850262328
Call-ID: 80444c150a74e438091eb9e89f70b5e0@192.168.1.10
CSeq: 310040040 REGISTER
Contact: "Unknown" <sip:6782366000@192.168.1.10:5060;transport=udp>;expires=60;q=0.5
allow-events: call-info
allow-events: line-seize
allow-events: dialog
allow-events: message-summary
allow-events: as-feature-event

Content-Length: 0
```

/* Example of incoming call - SIP INVITE */

```
12:37:07.937566 192.168.1.1.5060 > 192.168.1.10.5060:
>>>>>>>>>>>>>>>sip header start>>>>>>>>>>>>>>>>>>>
INVITE sip:6782366000@192.168.1.10:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.1.1:5060;branch=z9hG4bKljci5c17ac1jpa484oqbg3kti6
Record-Route: <sip:6782366000@192.168.1.1;lr>
From: "INFOSYSTEST" <sip:4046691362@192.168.1.1:5060;user=phone>;tag=SDi4hne01-208868550-
1274099827942-
To: "Daffy Ext6001" <sip:6782366001@192.168.1.1:5060>
Call-ID: SDi4hne01-36c8a79d9b7659ed00d74e8e68cd0f63-vrvvfv3
CSeq: 321012340 INVITE
Contact: <sip:192.168.1.1:5060;transport=udp>
Supported: 100rel
Max-forwards: 69
Allow: ACK, BYE, CANCEL, INFO, INVITE, OPTIONS, PRACK, REFER, NOTIFY, UPDATE
Content-Type: application/sdp
Accept: multipart/mixed, application/media_control+xml, application/sdp
Content-Length:   249

v=0
o=BroadWorks 97948 1 IN IP4 192.168.1.1
s=-
c=IN IP4 192.168.1.1
t=0 0
m=audio 16388 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000/1
a=sqn: 0
a=cdsc: 1 image udptl t38
a=mptime:20
a=ptime:20
a=rtpmap:101 telephone-event/8000/1
a=fmtp:101 0-15

/* Example of outgoing call – SIP INVITE */
12:42:50.141368 192.168.1.10.5060 > 192.168.1.1.5060:
>>>>>>>>>>>>>>>sip header start>>>>>>>>>>>>>>>>>>>
INVITE sip:4046691362@192.168.1.1 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.10:5060;rport;branch=z9hG4bK056ecd8f51f774343b91ed00c3fea4a2
From: "Goofy 6003" <sip:6782366003@192.168.1.1>;tag=cf1ca948467dd728
To: <sip:4046691362@192.168.1.1>
Call-ID: 97c02e86c2a2cd1cd3ac7a5ffac0d10e@192.168.1.10
CSeq: 2024269192 INVITE
Contact: "Goofy 6003" <sip:6782366003@192.168.1.10:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Content-Type: application/sdp
Supported: timer
Content-Length: 275

v=0
o=UserA 1322397120 2756067711 IN IP4 192.168.1.10
s=Session SDP
c=IN IP4 192.168.1.10
t=0 0
m=audio 49154 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

/* Example of Call Forward – SIP INVITE */

```
14:17:26.622540 192.168.1.10.5060 > 192.168.1.1.5060:
>>>>>>>>>>>>>>>sip header start>>>>>>>>>>>>>>>>>>>
INVITE sip:4046691362@192.168.1.1 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.10:5060;rport;branch=z9hG4bK7617892393a677b8fae387b40918c93d
From: "Pluto 6004" <sip:6782366004@192.168.1.1>;tag=c563f361f1b95f2a
To: <sip:4046691362@192.168.1.1>
Call-ID: bb11b095453f93741cd57dff839ab375@192.168.1.10
CSeq: 1420493315 INVITE
Contact: "Pluto 6004" <sip:6782366004@192.168.1.10:5060;transport=udp>
Max-Forwards: 70
Authorization: Digest
username="6782366000",realm="BroadWorks",nonce="BroadWorksXg9bdq35iTkmppzfBW",response="744e0f077
3272f154ad36cf36fe57154",uri="sip:4046691362@192.168.1.1",algorithm=MD5,qop=auth,nc=00000001,cnon
ce="9437101590e08913307b"
Supported: timer
Content-Type: application/sdp
Content-Length: 204


v=0
o=UserA 2683301831 1130673826 IN IP4 192.168.1.10
s=Session SDP
c=IN IP4 192.168.1.10
t=0 0
m=audio 49152 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

/* Example of Call Transfer – SIP INVITE */

```
14:15:28.160969 192.168.1.10.5060 > 192.168.1.1.5060:
>>>>>>>>>>>>>>>sip header start>>>>>>>>>>>>>>>>>>>
INVITE sip:4046691362@192.168.1.1 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.10:5060;rport;branch=z9hG4bK908fd27aaa129bb2853e4b9f232b8f3d
From: "Pluto 6004" <sip:6782366004@192.168.1.1>;tag=d8b58eedb6d21c45
To: <sip:4046691362@192.168.1.1>
Call-ID: f17617f4ba7a1d53d92378e7e9248c49@192.168.1.10
CSeq: 1193567000 INVITE
Contact: "Pluto 6004" <sip:6782366004@192.168.1.10:5060;transport=udp>
Max-Forwards: 70
Authorization: Digest
username="6782366000",realm="BroadWorks",nonce="BroadWorksXg9bdnjqwT9sjmf1BW",response="c183ee235
ff1790ee48370156869b269",uri="sip:4046691362@192.168.1.1",algorithm=MD5,qop=auth,nc=00000001,cnon
ce="9450e0aec72632d1da3a"
Supported: timer
Content-Type: application/sdp
Content-Length: 274

v=0
o=UserA 3709671978 101872114 IN IP4 192.168.1.10
s=Session SDP
c=IN IP4 192.168.1.10
t=0 0
m=audio 49162 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
/* Example of *67 – Anonymous Caller ID */
INVITE sip:*67@192.168.1.1 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.10:5060;rport;branch=z9hG4bKf26fe8995d9a84bbee2019246279026f
From: "Goofy 6003" <sip:6782366003@192.168.1.1>;tag=eb743da82eeba7ec
To: <sip:*67@192.168.1.1>
Call-ID: 4c0259ee87aaea0ec60ed04612451b05@192.168.1.10
CSeq: 1297491428 INVITE
Contact: "Goofy 6003" <sip:6782366003@192.168.1.10:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Content-Type: application/sdp
Supported: timer
Content-Length: 275


v=0
o=UserA 3801011003 2142179999 IN IP4 192.168.1.10
s=Session SDP
c=IN IP4 192.168.1.10
t=0 0
m=audio 49154 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```