# IP Office 5.0 SIP Trunking Configuration Guide

## AT&T Flexible Reach and AT&T Flexible Reach with Business in a Box(SM)

Issue 1.2 (11th August 2009)

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# 1. Introduction

This document provides a configuration guide to assist administrators in connecting Avaya IP Office SIP trunks to AT&T Flexible Reach and AT&T Business in a Box.

# 2. Special Notes

### Emergency 911/E911 Services Limitations

While AT&T IP Flexible Reach Services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the AT&T IP Flexible Reach Service Guide in detail to understand these limitations and restrictions.

### Fax Support

IP Office supports fax transmission using SIP when T.38 is configured and successfully negotiated. This functionality is supported only in the IP500 control unit fitted with a VCM32 or a VCM64.

IP Office does not support fax transmission using SIP in any other configuration.

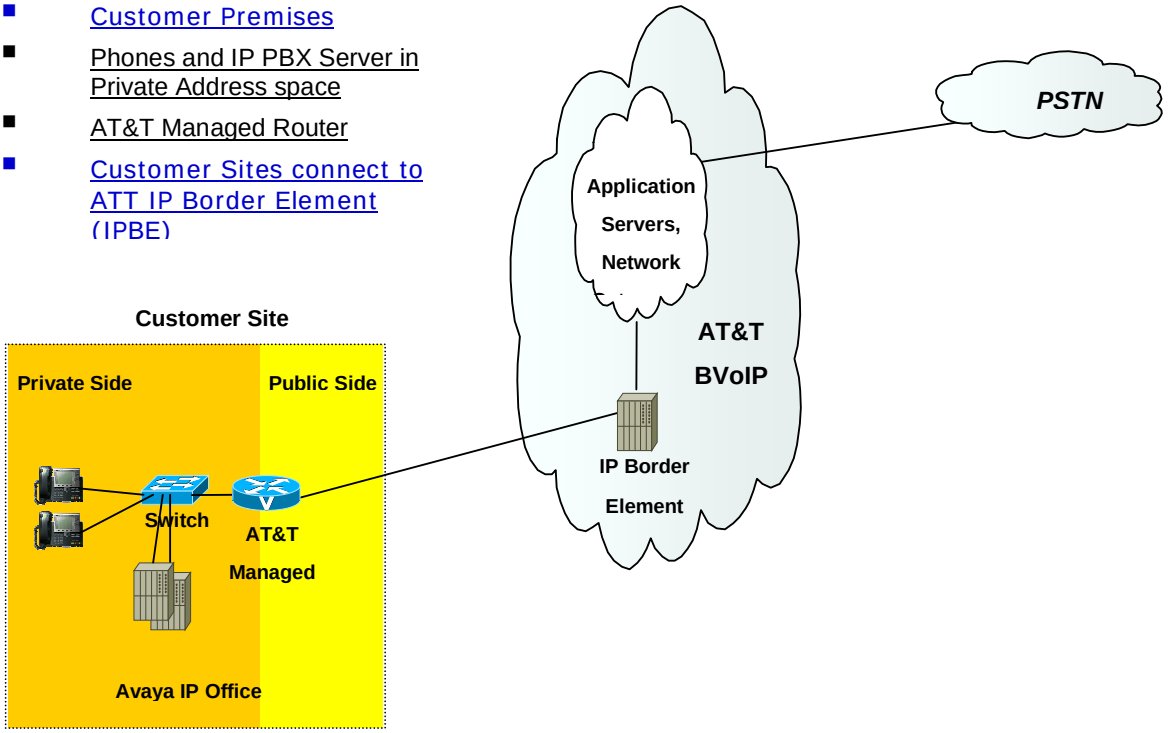
### Known Limitations

The solution documented in this application note supports FAX services over AT&T's Business-in-a-Box (IP Flexible Reach), SIP trunk service offer utilizing Group 3 Fax over T.38. However, there are some circumstances where Fax machines capable of supporting Supergroup 3 to Supergroup 3 Fax transmission will not work with this solution. Avaya and AT&T are aware of this issue and Avaya has developed a patch to address this problem. Please check with your Avaya support engineer for the availability of this patch.

# 3. Overview

This section provides a service overview of the integration of Avaya IP Office Communication System with the AT&T IP Flexible Reach or AT&T Business in a Box.

- Customer Premises
- Phones and IP PBX Server in Private Address space
- AT&T Managed Router
- Customer Sites connect to ATT IP Border Element (IPBE)




The diagram shows the IP Office system is deployed on the private side of the network and is linked to AT&T core network by an AT&T Managed router. This document contains configuration guidelines to the IP Office system only

## 3.1 Configuration

The Avaya customer premises equipment shall consist of the following components.

- Avaya IP Office Communication System

  Avaya offers various models and configurations for this equipment. Nominally the unit will include a built in processor unit, POTs line interfaces, analog ports, DCP ports for Avaya 6000 series digital phones and optionally T1 interfaces, wireless LAN interfaces etc.

- AT&T Managed Router (AT&T managed)

  This is the router managed by AT&T. The router shall perform network address translation, packet marking and QOS for voice

## 3.2 Supported Platforms

SIP trunks are supported in the following Avaya IP Office platforms:

- Small Office Edition (this unit is not supported in Release 5)
- IP406v2
- IP412
- IP500

## 3.3 Supported Phone Types

A list of Avaya IP Office supported phones is provided below:

- 1600 series IP
- 4600/5600 series IP
- 2400/5400 series digital
- 6400 series digital
- T3 (IP and digital) –excluding Small Office
- 3701/3711 (IP DECT)
- Analog phones
- Third party SIP Endpoints

### 3.4 Voice Coders Supported (VCM) per Platform

SIP trunks require the use of the VCM as described next.

- IP Office Communication System Small Office Edition: either VCM 3 or VCM 16.
- IP406v2 supports a single VCM chosen among the following types: VCM4, VCM 5, VCM 8, VCM 10, VCM 16, VCM 20, VCM 24 and VCM 30.
- IP412 supports up to any two cards VCM's of the following types; VCM4, VCM 5, VCM 8, VCM 10, VCM 16, VCM 20, VCM 24 and VCM 30.
- IP500 supports two variants:
    - First is up to two legacy VCM cards selected between the following options: VCM4, VCM 8, VCM 16, VCM 24 and VCM 30.
    - Second option (the only one that supports T.38 Fax transmission) is up to two new VCM cards from any VCM 32 and 64. Each card fitted provides 4 free channels. Additional channels, up to the card capacity, are enabled by the addition of IP500 VCM licenses to the IP Office system configuration.

The maximum number of calls supported on the VCM card is specified by the VCM card number (i.e. VCM 5 supports 5 calls, etc).

### 3.5 Basic Call Scenarios

The following routing scenarios are supported by the IP Office and **DO NOT** use the AT&T Call Control.

Local IP OFFICE phone to IP OFFICE phone

The following routing scenarios are supported by the IP OFFICE IP PBX and **DO** use the AT&T Call Control. For voice calls, the G.729 codec shall be used. **Fax other than using T.38 in an IP500 is not currently supported.**

IP OFFICE phones to PSTN (domestic US and international).
IP OFFICE phones to legacy PBX site with Cisco gateway.
Legacy PBX site with Cisco gateway to IP OFFICE phones.
IP OFFICE phones at one IP OFFICE IP PBX site to IP OFFICE phones at another IP PBX site.

If the customer has subscribed to Calling Plans B and C (Local), then the following routing scenarios are supported by the IP OFFICE IP PBX and **DO** use the AT&T Call Control. For voice calls, the G.729 or G.711 codec may be used. IP Office selects G.729 as the highest priority codec.

Inbound PSTN to IP OFFICE phone
Outbound local PSTN calls from the IP OFFICE phones.

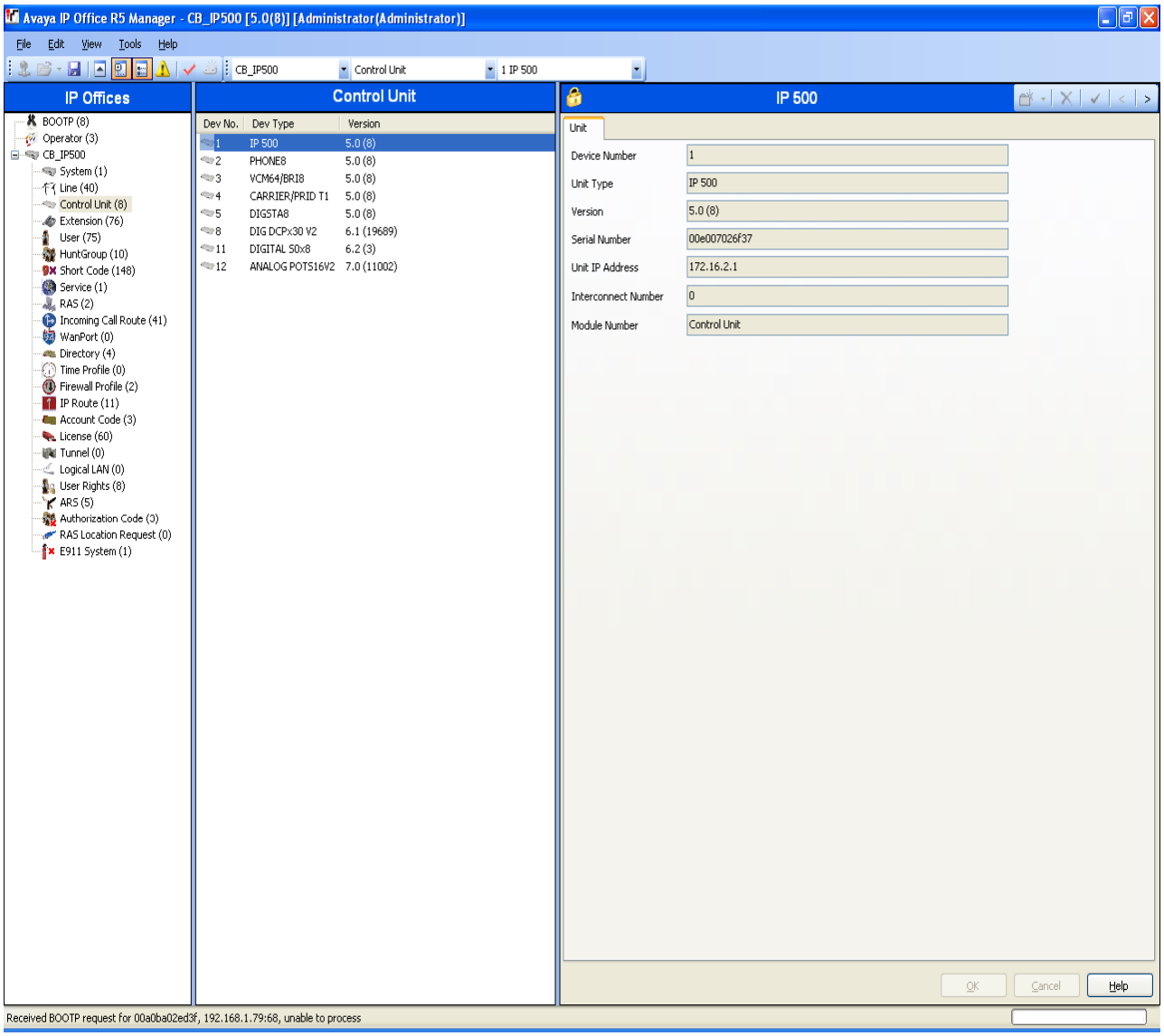
# 4. Customer Configuration Guide

This configuration guide specifies the Avaya IP Office Communication System screens that must be configured and updated to support the AT&T Voice over Managed Services.

In order to enable SIP communication you will need a valid SIP trunking license and IP Office system with VCM cards.

## 4.1 How to Identify You Are Running Version 5.0

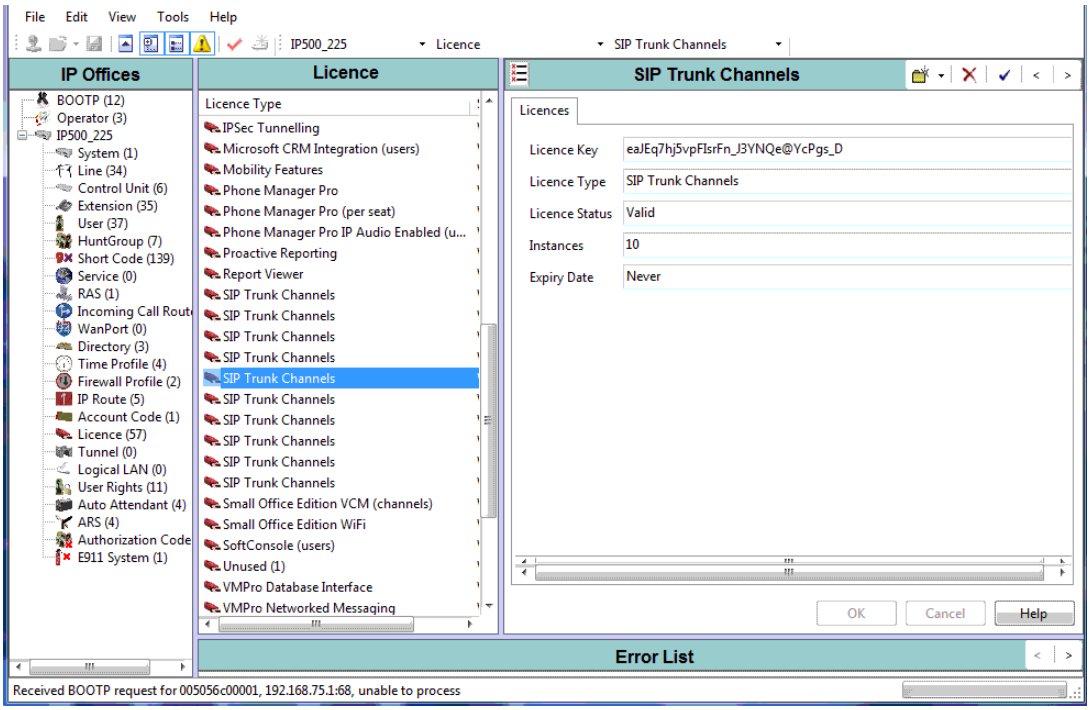
IP Office Administrators can identify the version number of the core software by opening up the running configuration using the IP Office Manager tool and looking at the number in the top bar. The version information is also displayed on the Control Unit item.




As shown above, the IP Office Core software version is 5. 0.(8)

## 4.2 How to Check For SIP Trunking Licenses

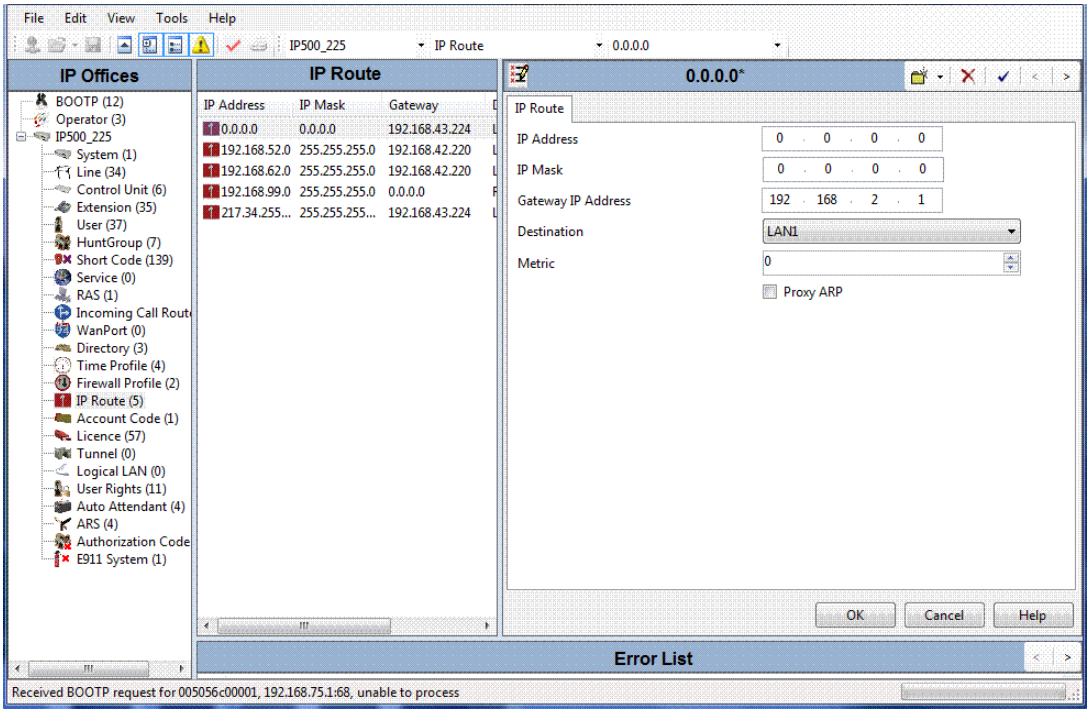
To make calls using SIP you must have a valid license that can be purchased through Avaya business partners, in the number of 1, 5, 10, 20 or a combination of all the above up to 128 instances of the same license. Avaya will provide a license that will need to be inserted into license form. An example is provided in Figure 3. 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.




- *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

### 4.3 Setting Up IP Routes to AT&T IP Network

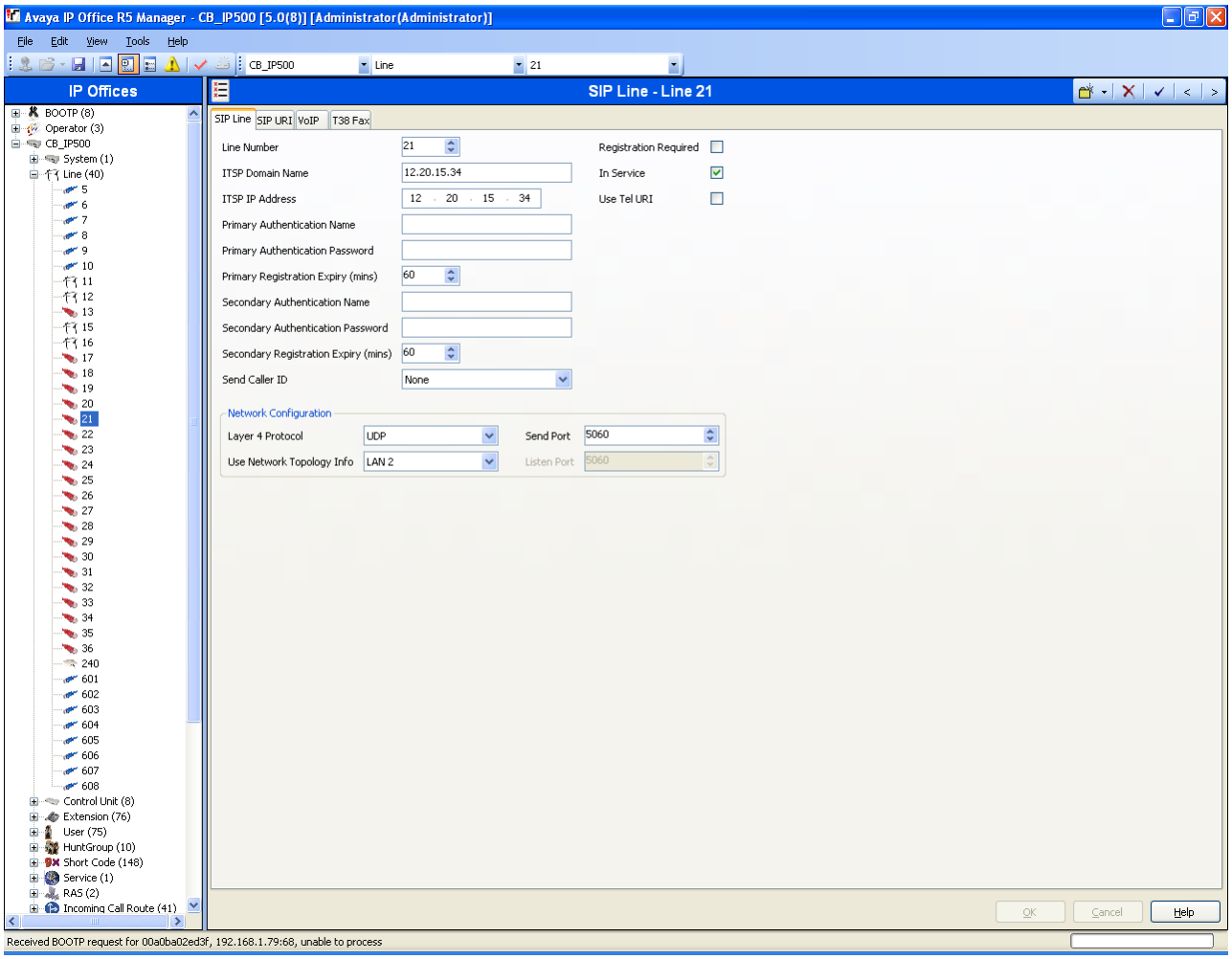
This section deals with the IP route configuration. The gateway IP address is the LAN side address of the AT&T managed router. Please contact AT&T customer care to get the correct address.



## 4.4 Primary SIP Line

This section deals with the SIP line tab on the SIP Line configuration.

PLEASE CONTACT YOUR AT&T CUSTOMER CARE REPRESENTATIVE FOR THE AT&T IPBE (IP BORDER ELEMENT) IP ADDRESSES FOR YOUR SPECIFIC PBX. For the Flexible Reach solution **you must configure a SIP line for each of the 2 AT&T Border Elements provided by AT&T Customer care. For Business in a Box (BIB) only one SIP Line is required.**




Set out below are the guidelines for configuration of each field in the SIP Line form

- Line number is automatically assigned by the Manager application when the SIP Line is created.
- ITSP Domain Name shall be left blank
- Registration Required, shall be left un-ticked
- ITSP IP Address shall be set to the AT&T border element associated with this SIP line. For Business in a Box (BIB), the ITSP IP address will be the IP address of the BIB which defaults to 192.168.2.1
- Primary (and Secondary) Authentication Name shall be left blank
- Primary (and Secondary) Authentication Password shall be left blank
- Primary (and Secondary) Registration Expiry shall be left as default
- In Service shall be ticked
- Use TEL URI shall be left un-ticked
- Send Port shall be set to 5060 (default)
- Listen Port shall be set to 5060 (default)

4. Customer Configuration Guide

- Use Network Topology Info: set it to None to disable STUN.
- Call Initiation Timeout: set it to 3 seconds. This is the time IP Office will wait for a reply to an INVITE before attempting a new line via ARS. (located in VoIP tab)
- Re-INVITE Supported shall be left un-ticked (located in VoIP tab)
- Compression mode should be set to G.729(a) 8K CS-ACELP (located in VoIP tab)

### 4.5 Telephone Numbers

Once the steps above have been completed the remaining configuration steps are related to the way telephone numbers (TNs) are assigned to users in the IP Office. These TNs have been provided by AT&T.

For simplicity, we will assume that there are three telephone numbers (10 digits) to configure with the following purpose:

**Main Line number**: 732-123-1000

- **Incoming calls** are routed to an Operator or Auto-Attendant.
- **Outgoing calls**: any user in the system can dial out with this Caller Id.

**Hunt Group number**: 732-123-1001

- **Incoming calls** are routed to the members of a Hunt Group.
- **Outgoing calls**: the members of the group can choose to dial out with this Caller Id or with the one of the Main Line number.

**Individual number**: 732-123-1002

- **Incoming calls** are routed to a particular user.
- **Outgoing calls**: the user can dial out with this Caller Id. Nobody else in the system can use this number

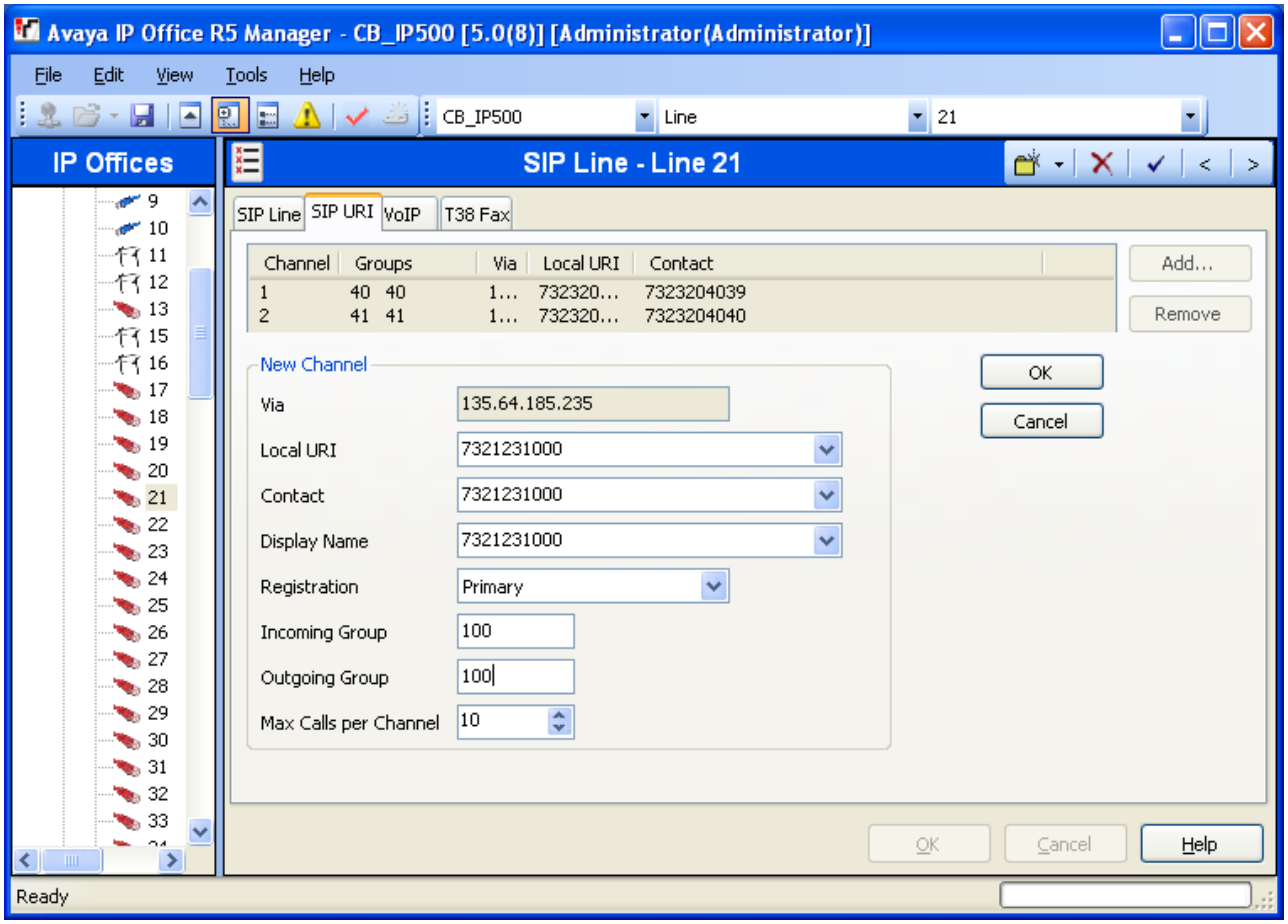
# 5. Main Line number configuration

Steps to configure the Main Line number:

- Add a SIP URI.
- Add an Incoming Call Route.
- Add a system Short Code

## 5.1 SIP URI for Main Line number

This section deals with the SIP URI tab on the SIP Line configuration.




The fields on this form are described next.

- **Local URI** populates the user part of the FROM header for outbound calls. Enter 7321231000.
- **Contact** sets the user part of the CONTACT header for outbound calls. Enter 7321231000.
- **Display Name** sets the display name field in the FROM header for outbound calls. Enter 7321231000
- **Incoming Group** is a number used internally by IP Office to make routing decisions for incoming calls. Enter 100.
- **Outgoing Group** is a number used internally by IP Office to make routing decisions for outgoing calls. Enter 100.
- The number of simultaneous calls allowed can be set using the **Max Calls per Channel** field but will also depend on the number of calls supported on the VCM cards and SIP licenses available.

5. Main Line number configuration

The value of the *Local URI* field will be used in the From header of outgoing calls and it will be compared to the To of incoming calls. This example assume that the user part of the SIP URI in the To field of the SIP INVITE arriving from the ATT network to IP Office is a 10 digit Telephone Number. By the same token, IP Office will populate the user part of the SIP URI in the From field with a 10 digit Telephone Number.

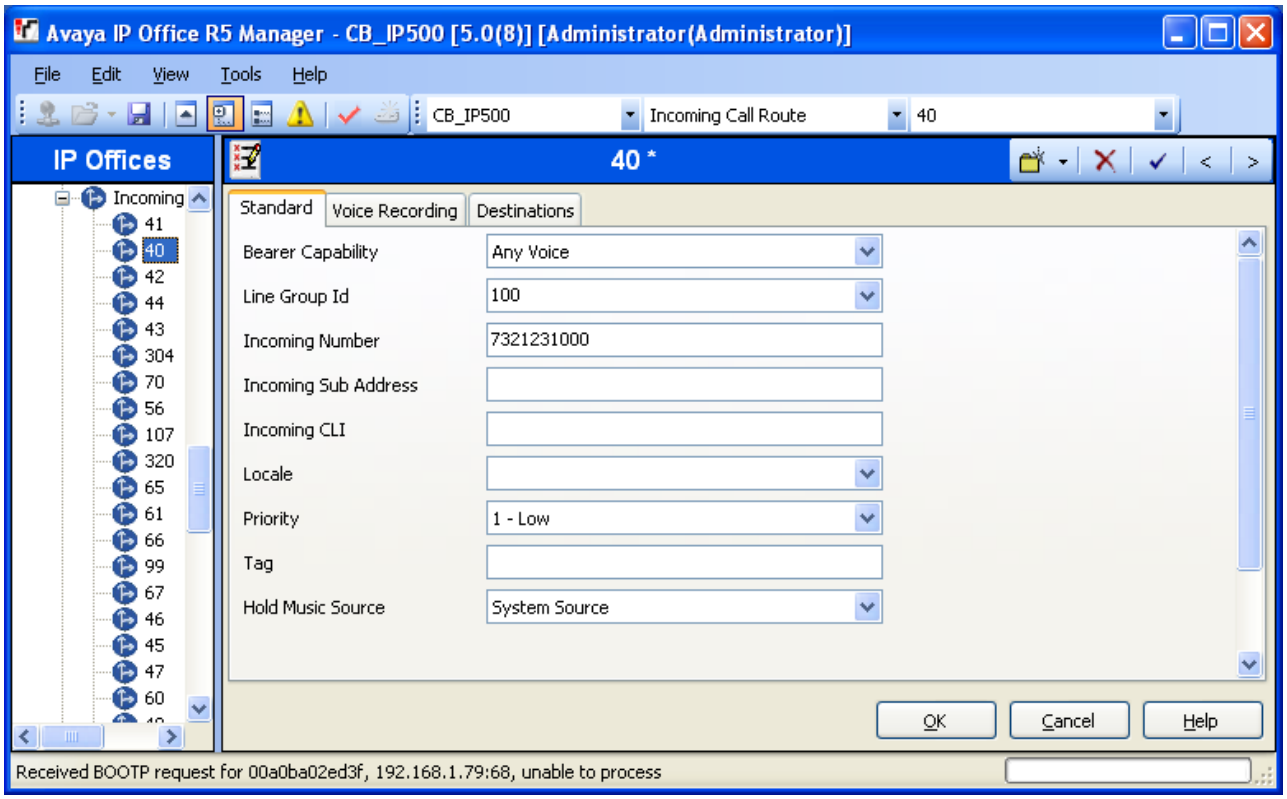
## 5.2 Incoming Call Route for Main Line Number

The translation of each DID to an IP Office extension is done using the Incoming Call Route (ICR) form as shown in pictures below.

In this example, number 7321231000 will be routed to the Operator in extension number 300.

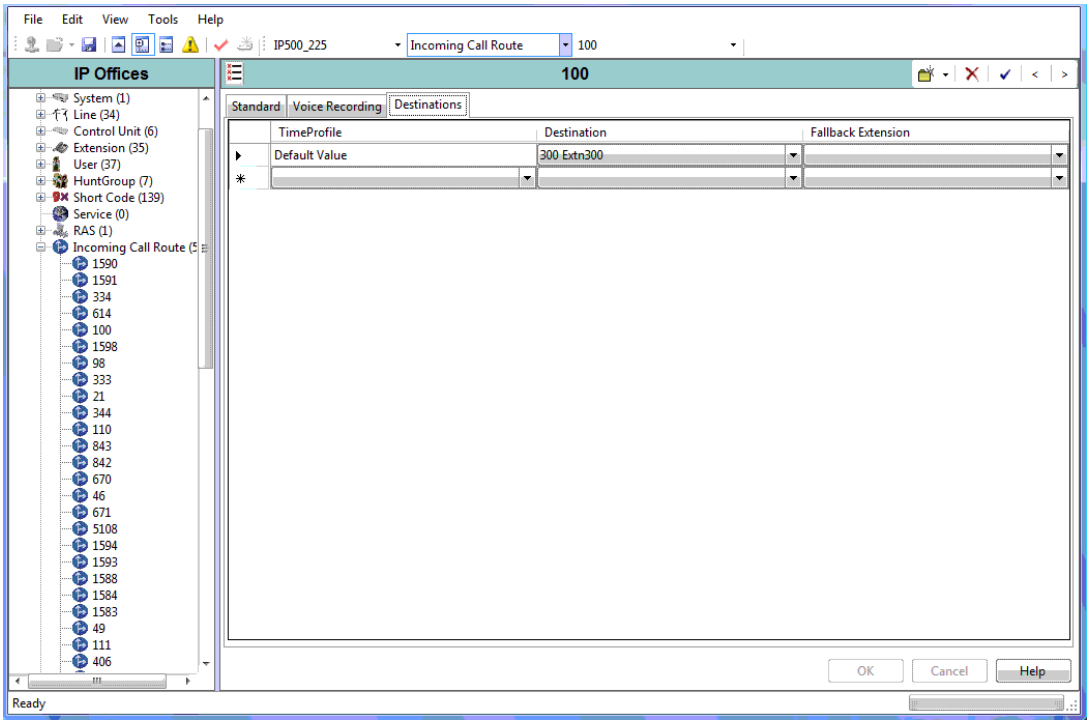
Configure the following in the Standard tab:

- Bearer Capability: Any
- Line Group Id: 100 (this matches the Incoming Group number configured in the SIP URI)
- Incoming Number: 7321231000



Configure the following in the Destinations tab:

- Destination: 300




The fields are populated as follows:

- **Bearer Capability** shall be set to *Any Voice.*
- **Line Group Id** shall be set to the *Incoming Group ID* of the *SIP URI* that is used to receive external phone calls.
- **Incoming Number** is the called number string sent by AT&T IP Flexible Reach. If the telephone number (i.e. TN) is a virtual TN, then AT&T will send the full 10 digit TN. If the TN is non-virtual, then AT&T will send the last 4 digits of the TN. In the example shown above there is no number entered so therefore all the calls are matched. In order to narrow call matching enter here the 10 or 4 digit TN.
- **Destination** shall be set to the desired target of incoming calls that can be an extension, hunt-group, Auto-Attendant or Voice Mail module.

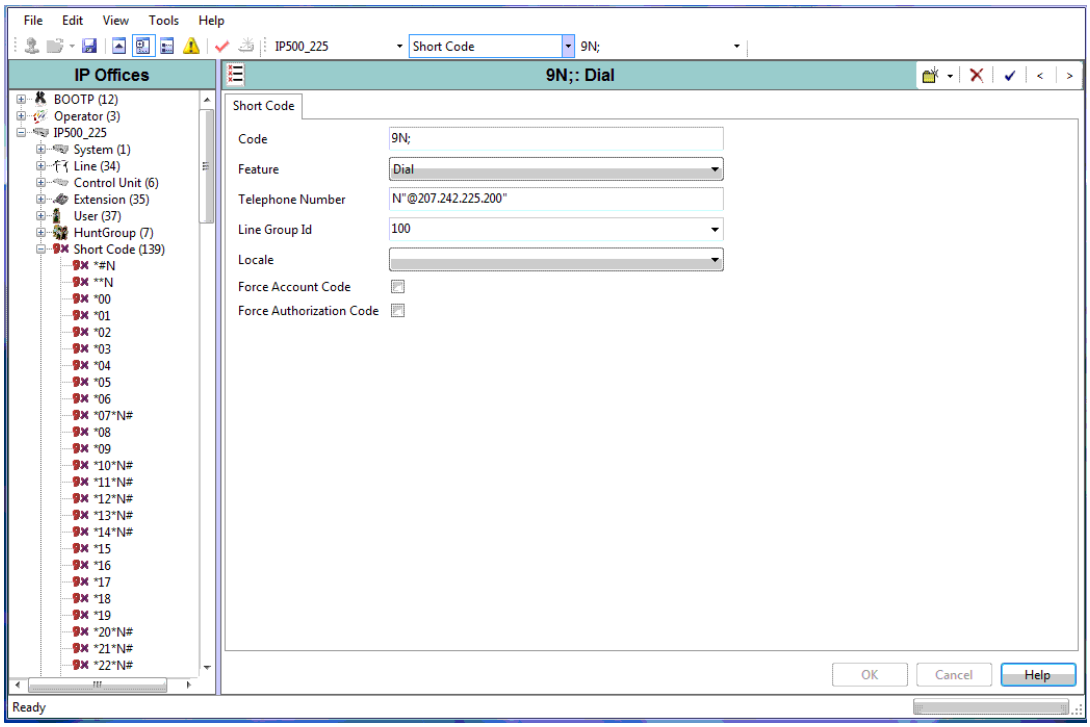
## 5.3 Short Code for Main Line Number

This section describes the IP Office configuration required for sending calls to the AT&T IP FLEXIBLE REACH or AT&T Business in a Box network.

The SHORT CODE form is used to direct calls to the AT&T SIP network. The following screen shot gives an example in a fictitious IP Address.

For the example under consideration we will assume that users will dial 9 followed by a number of digits to make outgoing calls. Assuming that the IP Address of the AT&T SIP server is 207.242.225.200, configure the following:

- Short code: 9N;
- Feature: Dial
- Line Group ID: 100 (this matches the Outgoing Group number configured in the SIP URI)
- Telephone Number: N"@207.242.225.200"




The fields are populated as follows:

- **Short code** – This field matches the dialled number. The sample short code shown above will match on a "9" followed by any number of digits (represented by "N"). The short code should end with a semi colon.
- **Line Group ID** – Set this field to the SIP *URI Outgoing Group* ID that it is used to connect to the AT&T IP FLEXIBLE REACH network.
- **Feature** – Set this field to "Dial".
- **Telephone number** – For AT&T IP Flexible Reach is set to "N"@<IP Border Element address on the IP Flexible Reach Service>"". Note that IP Office can only send calls to one AT&T Border Element. If that Border Element is not available, then the customer must manually re-configure this value to point to the secondary Border Element. For AT&T Business in a Box the IP Address is the LAN interface of the BIB. When using default values it will look like "N"@192.168.2.1""

# 6. Hunt Group Number Configuration

Steps to configure the Hunt Group number:

- Add a SIP URI.
- Add an Incoming Call Route.
- Add a User Rights form

## 6.1 SIP URI for Hunt Group Number

Add a new SIP URI entry in the SIP URI tab of the SIP Line (see picture in section 5.1 above). Enter the following values:

- Local, Display and Contact: 7321231001
- Incoming Group: 100
- Outgoing Group: 101

## 6.2 Incoming Call Route for Hunt Group Number

In this example, number 7321231001 will be routed to the extension number of the Hunt Group, for example, 399 (see picture of ICR form in section 5.2 above).

Configure the following in the Standard tab:

- Bearer Capability: Any
- Line Group Id: 100 (this matches the Incoming Group number configured in the SIP URI)
- Incoming Number: 7321231001

Configure the following in the Destinations tab:

- Destination: 399

## 6.3 Short Code for Hunt Group Number

For the example under consideration we will assume that users of the Hunt Group will dial 8 followed by a number of digits to make outgoing calls. Assuming that the IP Address of the AT&T SIP server is 207.242.225.200, configure the following short code in the Short Code tab of each member of the group:

- Short code: 8N;
- Feature: Dial
- Line Group ID: 101 (this matches the Outgoing Group number configured in the SIP URI)
- Telephone Number: N"@207.242.225.200"

As this setting needs to be done for each member of the Hunt Group, it is advisable to create a User Rights form and assign it to the members of the group.

# 7. Individual Number Configuration

Steps to configure the Individual number:

- Add a SIP URI.
- Configure SIP tab of User form
- Add an Incoming Call Route.

## 7.1 SIP URI for Individual Number

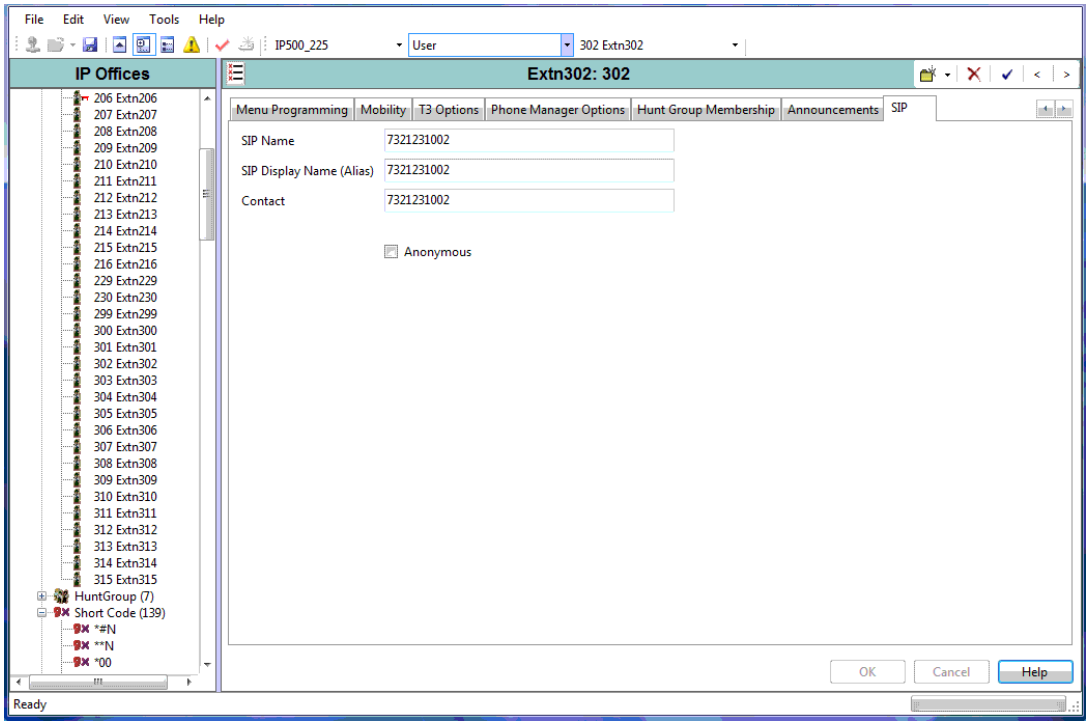
Add a new SIP URI entry in the SIP URI tab of the SIP Line (see picture in section 5.1 above). Enter the following values:

- Local: Use User Data
- Display: Use User Data
- Contact: Use User Data
- Incoming Group: 100
- Outgoing Group: 102

## 7.2 SIP Tab of User Form

In this example, number 7321231002 will be assigned to User 302. Go to the SIP tab of the User form and enter the following values:

- Local: 7321231002
- Display: 7321231002
- Contact: 7321231002
- Anonymous: (tick if you want to hide Caller Id)




The fields on this form are SIP Name, SIP Display Name (Alias), and Contact and are described next.

- **SIP Name** populates the user part of the FROM header for outbound calls (e.g. TN#1).
- **SIP Display Name** sets the display name field in the FROM header for outbound calls (e.g. TN#1).
- **Contact** sets the user part of the CONTACT header for outbound calls (e.g. TN#1).
- **Anonymous** enables the Privacy Mechanism for outbound calls according to RFC 3325

### 7.3 Incoming Call Route for Individual Number

In this example, number 7321231002 will be routed to extension number 302.

Configure the following in the Standard tab:

- Bearer Capability: Any
- Line Group Id: 102 (this matches the Incoming Group number configured in the SIP URI)
- Incoming Number: 7321231001

Configure the following in the Destinations tab:

- Destination: 302

### 7.4 Short Code for Individual Number

For the example under consideration we will assume that user 302 will dial 9 followed by a number of digits to make outgoing calls. Assuming that the IP Address of the AT&T SIP server is 207.242.225.200, configure the following short code in the Short Code tab of user 302:

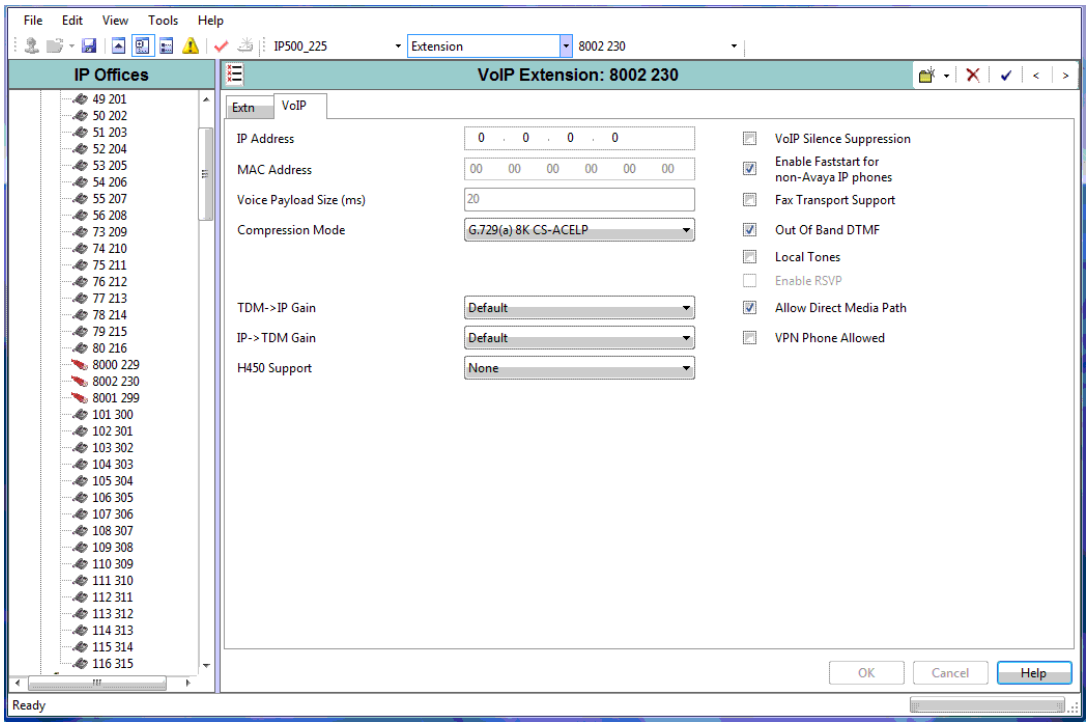
- Short code: 9N;
- Feature: Dial
- Line Group ID: 102 (this matches the Outgoing Group number configured in the SIP URI)
- Telephone Number: N"@207.242.225.200"

# 8. Additional Steps

## 8.1 IP Phone Configuration

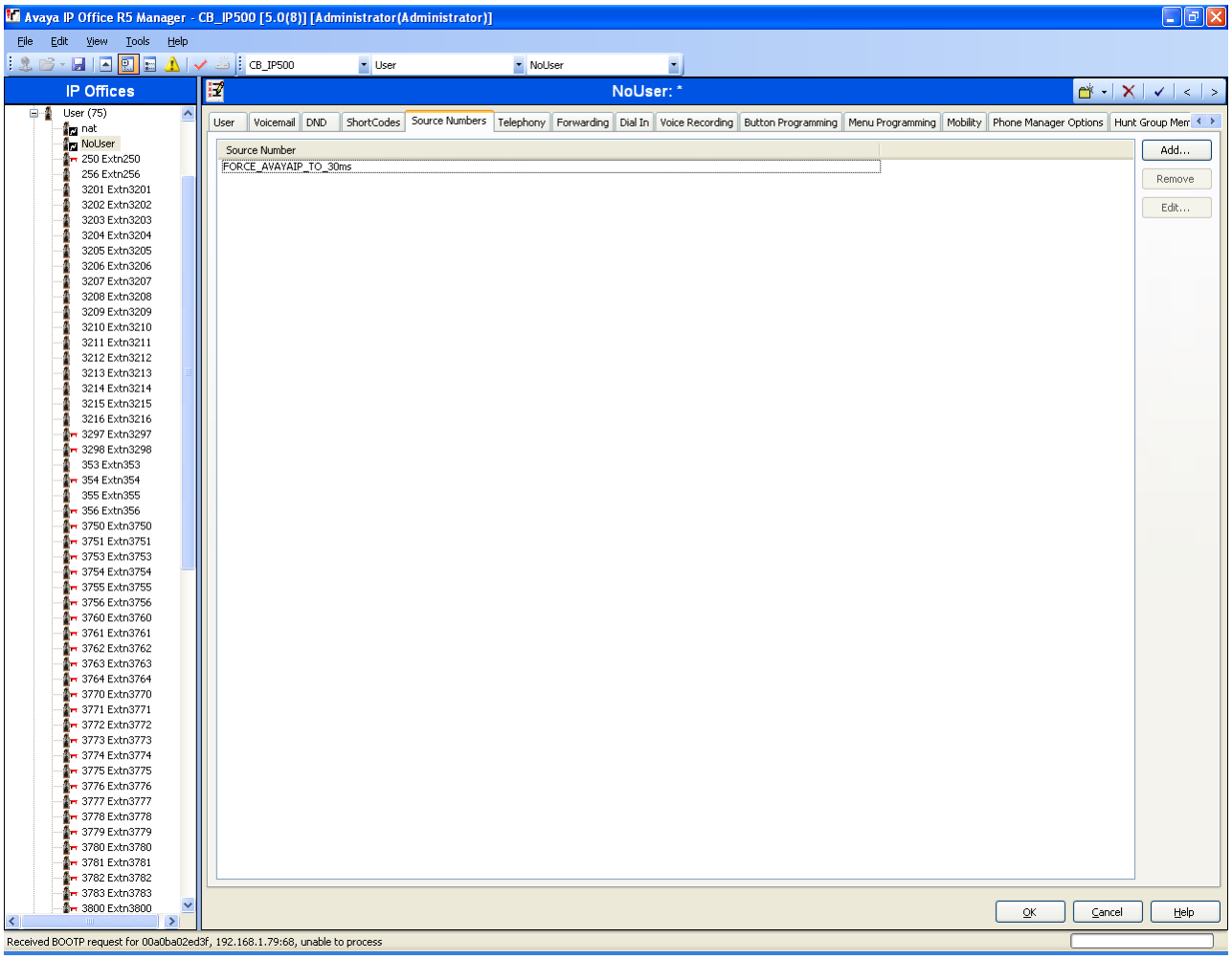
An example of an IP phone configuration is shown next.

- **Enable Fast Start for non Avaya phones** is enabled
- **Allow Direct Media Path** can be left ticked or un-ticked depending on user preference. SIP trunks will always use relay (no direct media path).
- To minimize the number of voice coder employed, it is recommended to set **Compression Mode** to ***G.729***.



## 8.2 NoUser Source Numbers

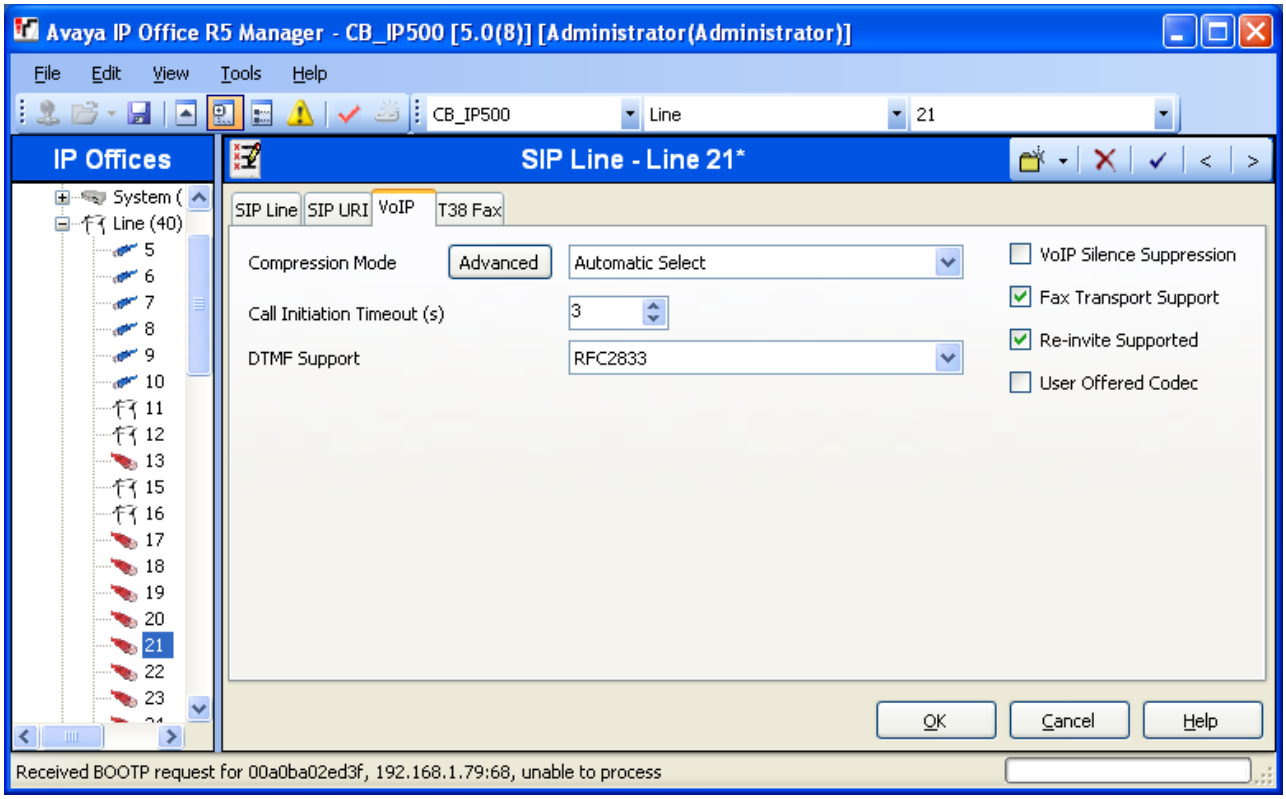
An extra step is required to support 30 msec payloads for Avaya IP Phones (except 1600 series phones). This is achieved by adding the Source Number ***FORCE_AVAYAIP_TO_30ms*** to the NoUser form.



## 8.2 T.38 FAX Configuration

To enable FAX T.38 in the SIP Line, check the following options in the VoIP Tab:

- **Re-invite supported**
- **Fax Transport Support**

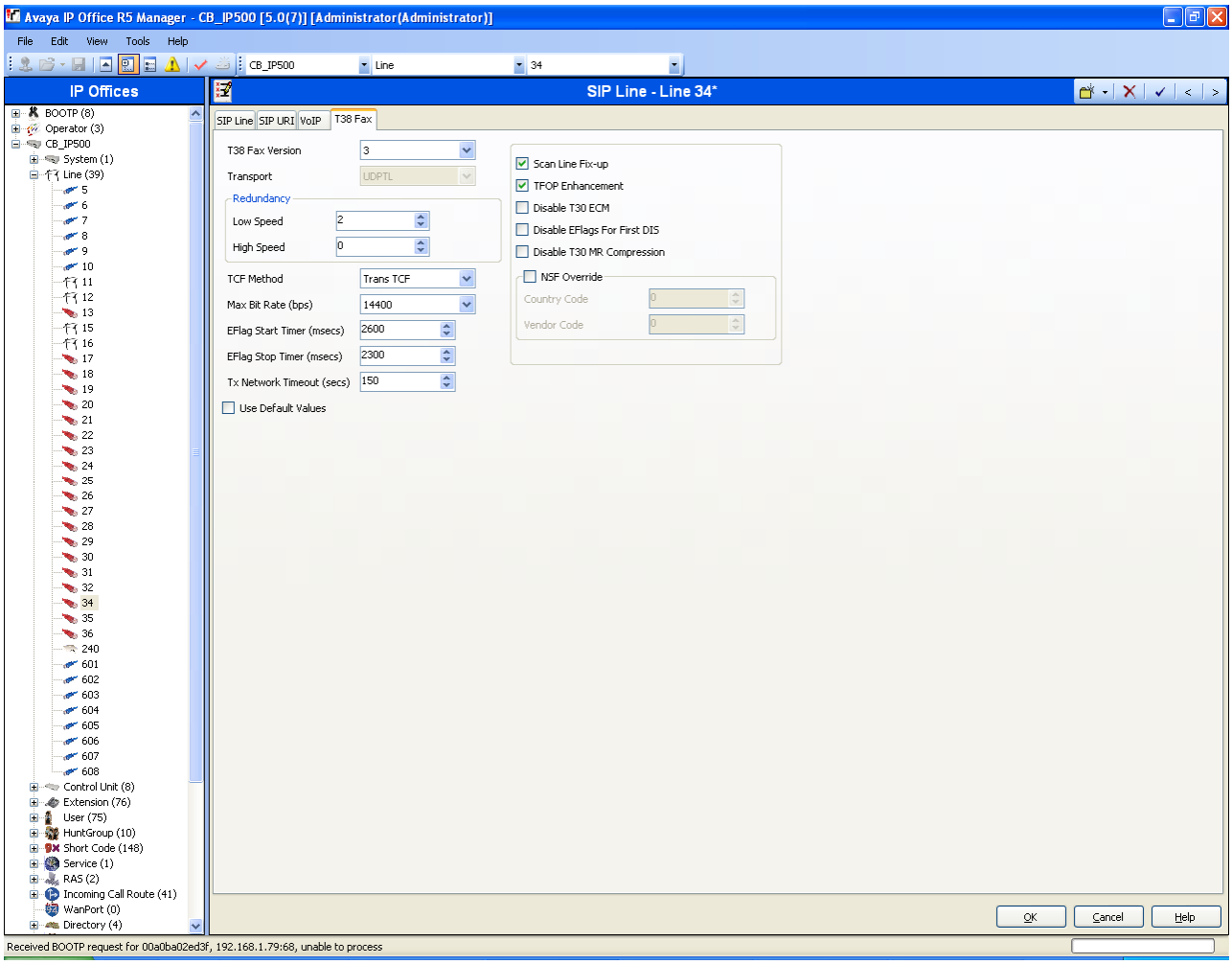



When Fax Transport Support is selected, the T38 Fax tab becomes editable. In the “T38 Fax” tab, by default, “Use Default Values” will be checked. The T38 configuration options are greyed out unless “Use Default Values” is un-checked.

For interoperability with AT&T it is advised that Low Speed Redundancy is enabled. To achieve this:

- Uncheck “Use Default Values”
- In the “Redundancy” area set “Low Speed” to “2

All the other options can be left as in default. See the example in the screenshot below.




## Placing Fax-Only outbound calls from IP Office

IP Office has the capability of setting up fax only calls. What this means is that in the original SIP INVITE method, only FAX (T38) capabilities are offered. This will force the call to start in T38 instead of starting in audio and then transition to fax by means of a SIP re-INVITE. However, this functionality is not supported in the AT&T network.

There are two ways of setting up a fax only call:

- Setting the “Analogue Extension” equipment classification to “Fax Machine”
- Using a “Dial Fax” short-code

These settings must not be used.

# 9. Trouble Shooting

IP Office has a protocol trace tool called “system monitor”. During trouble shooting, the customer may be asked to run this tool and provide traces to AT&T Customer Care. Sample output from this tool are shown next.

```
74693707mS SipDebugInfo: extension is dialing 8008648331@207.242.225.200

74693707mS SipDebugInfo: CMSetup receive, ep f5732238, dialog f5732d08
74693707mS SipDebugInfo: MZ extension is dialing 8008648331@207.242.225.200
74693707mS SipDebugInfo: ********************************************************
74693707mS SipDebugInfo: INVITE (method) SENT TO 207.242.225.200 5060
74693708mS SipDebugInfo: Registration Required is 0, Primary Status 0, Secondary Status 0
74693708mS SipDebugInfo: ********************************************************
74693710mS SipDebugInfo: ********************************************************
74693710mS SipDebugInfo: TxInvite: INVITE SENT TO 207.242.225.200  5060
74693710mS SipDebugInfo: ********************************************************
74693710mS SipDebugInfo:  Sending INVITE, ep f5732238, dialog f5732d08
74693710mS SipDebugInfo: Sip_sendToNetwork packet of length 830
74693711mS SipDebugInfo: SIP Line (16): SendToTarget  cff2e1c8, 5060
74693711mS SIP Trunk: 16:Tx
INVITE sip:8008648331@207.242.225.200 SIP/2.0
Via: SIP/2.0/UDP 217.36.111.99:5060;rport;branch=z9hG4bKfaf7bf23f5e3c2b636e28f7f567f7a56
From: ErnestoandPaul <sip:17323684893@217.36.111.99>;tag=023291dd0527aac4
To: <sip:8008648331@207.242.225.200>
Call-ID: 5381584b34db8b076351cbeca01c07aa@217.36.111.99
CSeq: 1472430449 INVITE
Contact: ErnestoandPaul <sip:17323684893@217.36.111.99:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO
Content-Type: application/sdp
Content-Length: 302

v=0
o=UserA 585867926 1856642184 IN IP4 217.36.111.99
s=Session SDP
c=IN IP4 217.36.111.99
t=0 0
m=audio 49152 RTP/AVP 18 4 8 0 101
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=fmtp:18 annexb = no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
74693711mS SipDebugInfo: initialising mTxnContext
74693712mS SipDebugInfo: ********************************************************
74693712mS SipDebugInfo: State Transtion form Old State 0 to New state 1
74693712mS SipDebugInfo: ********************************************************
74693712mS SipDebugInfo: SIPDialog::UpdateSDPState has just transitioned to state 1
74693867mS SIP Trunk: 16:Rx
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
217.36.111.99:5060;received=217.36.111.99;branch=z9hG4bKfaf7bf23f5e3c2b636e28f7f567f7a56;rport
=5060
From: ErnestoandPaul <sip:17323684893@217.36.111.99>;tag=023291dd0527aac4
To: <sip:8008648331@207.242.225.200>
Call-ID: 5381584b34db8b076351cbeca01c07aa@217.36.111.99
CSeq: 1472430449 INVITE

74693867mS SipDebugInfo: MZ SIPDialog: ReceiveFromTarget
74693869mS SipDebugInfo: MZ SIPDialog TXN : Decoding of message Succeded 1
74693869mS SipDebugInfo: SIP: ProcessInbound Message
74693869mS SipDebugInfo:  Find End Point 5381584b34db8b076351cbeca01c07aa@217.36.111.99
74693869mS SipDebugInfo: Process SIP response dialog f5732d08, method INVITE,CodeNum 100 in
state 1
74693870mS SipDebugInfo: MZ SIPDialog No Tag due to error
74693870mS SipDebugInfo: ExtractRouteFromRecord, entered
74693870mS CMTARGET:     TargetOnProgress: res: 1
74693871mS CMCallEvt:    0.8725.0 49 TargetingEP: RequestEnd 0.8726.0 49 SIPTrunk Endpoint
74693871mS CMTARGET:     0.8724.0 49 Extn208.0: CancelTimer CMTCNoAnswerTimeout
```

```
74693871mS CMCallEvt:    0.8725.0 -1 BaseEP: DELETE CMEndpoint f5734980 TOTAL NOW=2
CALL_LIST=1
74693871mS CMCallEvt:    0.8726.0 49 SIPTrunk Endpoint: StateChange: END=B CMCSOffering-
>CMCSAccept
74693872mS CMCallEvt:    0.8724.0 49 Extn208.0: StateChange: END=A CMCSDialling->CMCSDialled
74693872mS CMExtnEvt:    v=8 State, new=Proceeding  old=Dialling,0,0,Extn208
74693873mS SipDebugInfo: ******************************************************
74693873mS SipDebugInfo: State Transtion form Old State 1 to New state 5
74693873mS SipDebugInfo: ******************************************************
74693873mS SipDebugInfo: SIP Line (16): Cannot free Txn Key 2015
74694216mS SIP Trunk: 16:Rx
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP
217.36.111.99:5060;received=217.36.111.99;branch=z9hG4bKfaf7bf23f5e3c2b636e28f7f567f7a56;rport
=5060
From: ErnestoandPaul <sip:17323684893@217.36.111.99>;tag=023291dd0527aac4
To: <sip:8008648331@207.242.225.200>;tag=ds254196ac
Call-ID: 5381584b34db8b076351cbeca01c07aa@217.36.111.99
CSeq: 1472430449 INVITE
Content-Length: 228
Contact: <sip:8008648331@207.242.225.200:5060;transport=udp>
Allow: INVITE, BYE, ACK, CANCEL, PRACK, INFO
Content-Disposition: session; handling=required
Content-Type: application/sdp

v=0
o=Sonus_UAC 11634 6705 IN IP4 207.242.225.200
s=SIP Media Capabilities
c=IN IP4 207.242.225.200
t=0 0
m=audio 19196 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

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