



IP Office Edition (Essential, Preferred, Advanced) Release 6.0 SIP Trunking Configuration Guide

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

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Documentation information

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1. Introduction

This document provides a configuration guide to assist administrators in connecting Avaya IP Office 6.0 SIP trunks to AT&T Flexible Reach and AT&T Business in a Box.

IP Office Release 6.0 offers the following Product Suite options:

- IP Office Essential Edition PARTNER Version
- IP Office Essential Edition
- IP Office Preferred Edition
- IP Office Advanced Edition

This configuration guide covers the last three Product Suite options. Partner Version will be posted on the DevConnect site.

2. Special Notes

Emergency 911/E911 Services Limitations

Emergency 911/E911 Services Limitations and Restrictions - While AT&T IPFlexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when that E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at <http://new.serviceguide.att.com><<http://new.serviceguide.att.com>>. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the 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 and IP500v2 control unit fitted with a VCM32, VCM64, or a combo card fitted with a VCM10.

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

Configuration Values

Configuration values used in this document are for reference and have local relevance to this configuration guide. Actual values used will be specific to the network used in a production implementation.

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)

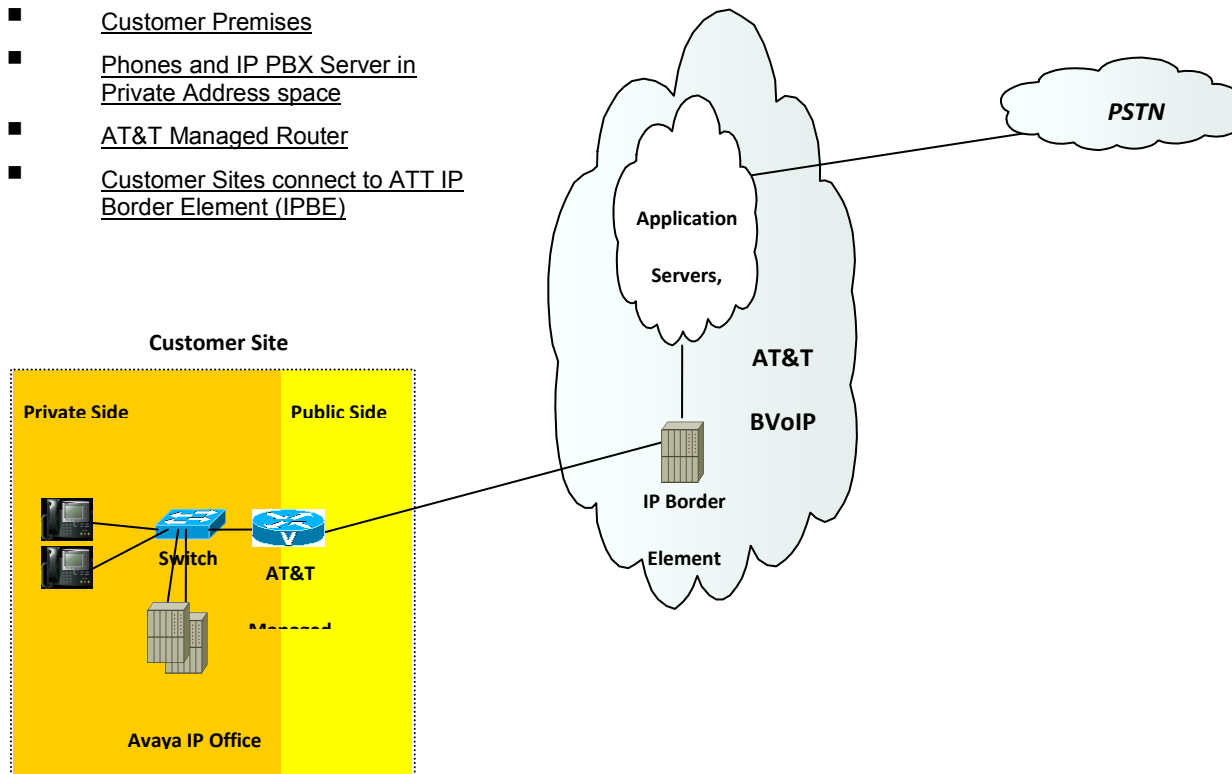


Figure 1 - Reference Configuration

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 digital phones and optionally T1 interfaces, wireless LAN interfaces etc.

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

3.2 Supported Platforms

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

- IP412
- IP500
- IP500 V2

Complete details regarding IP Office R6.0 can be found in the Avaya IP Office Knowledge Base at <http://marketingtools.avaya.com/knowledgebase>.

3.3 Supported Phone Types

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

- 1600/1600I, 4600, 5600, 9600 (non-SIP) series IPsets
- 1400, 2400, 5400, 6400 series digital sets
- 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.

- 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.
- IP 500 V2 supports:
 - Up to two VCM32 or VCM64 cards + up to 2 combination cards (each supports one VCM10) to yield a maximum of 148 VCM channels.

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

Local IP OFFICE phone to IP OFFICE phone

The following routing scenarios are supported by the IP OFFICE IP PBX and use the AT&T Call Control. For voice calls, the G.729 codec shall be used.

Note: 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.

The following routing scenarios are supported by the IP OFFICE IP PBX and 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. Licenses can be obtained in bundles of 1, 5, 10, or 20 SIP channels.

4.1 How to Identify You Are Running Version 6.0

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.

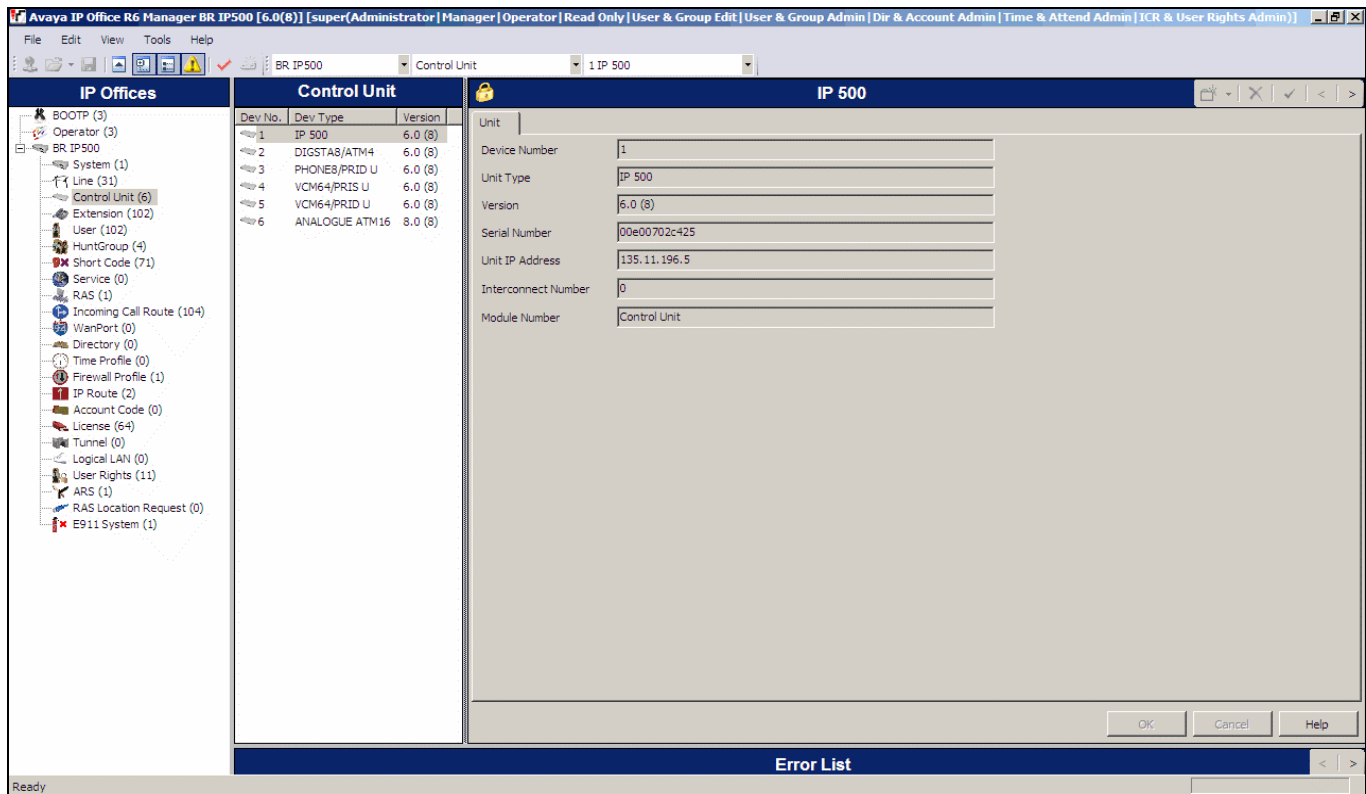


Figure 2 - Version Verification

As shown in the screen shown above, the IP Office Core software version is 6.0.(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. Licenses 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.

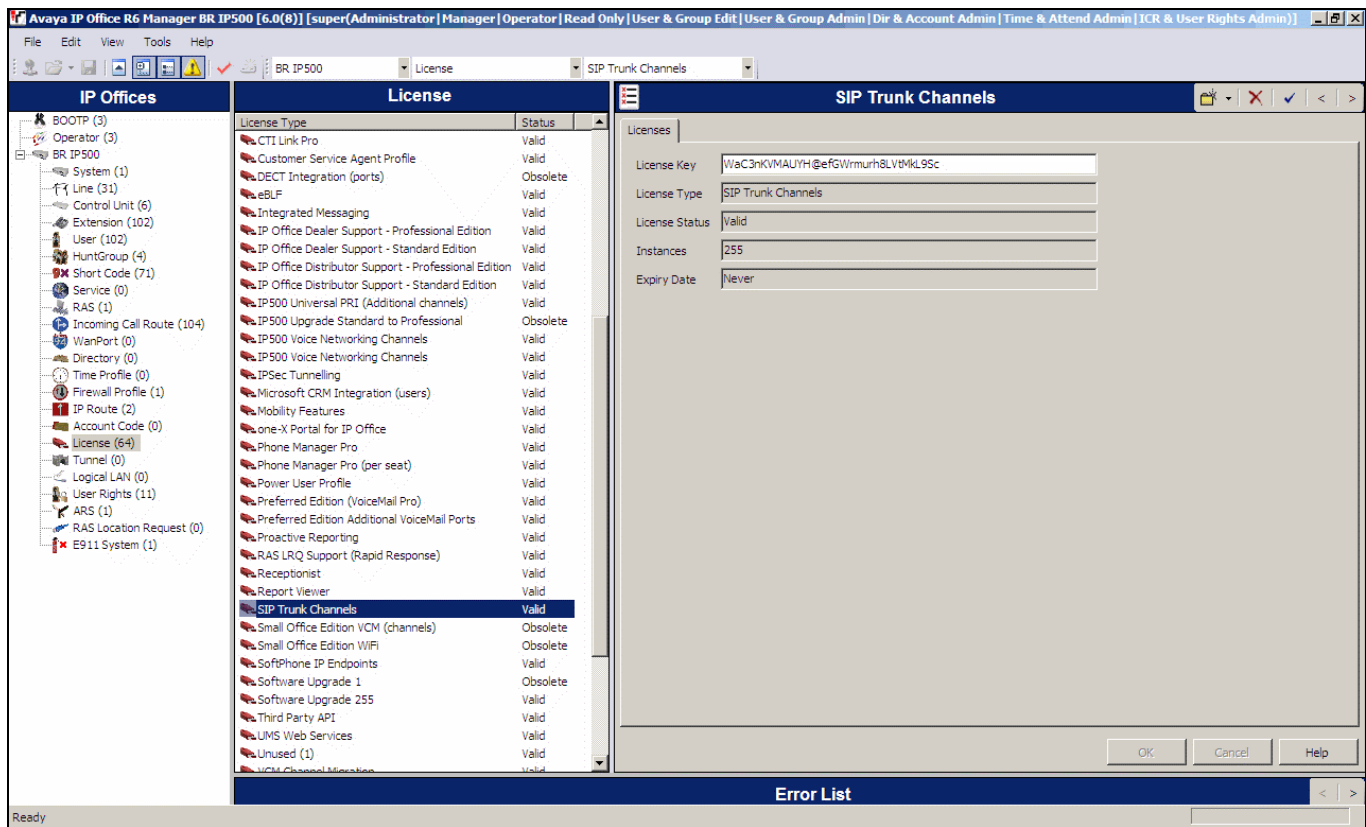


Figure 3 - License Verification

- **License Key** indicates the license identifier that has been provided by the Avaya business partner.
- **License Type** must be set to *SIP Trunk Channel*.
- **License Status** indicates the current validation status of the highlighted license type.
- **Instances** indicates the number of license instance that have been purchased.
- **Expiry Date** indicates 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 of the AT&T Border Element. Please contact AT&T customer care to get the correct address.

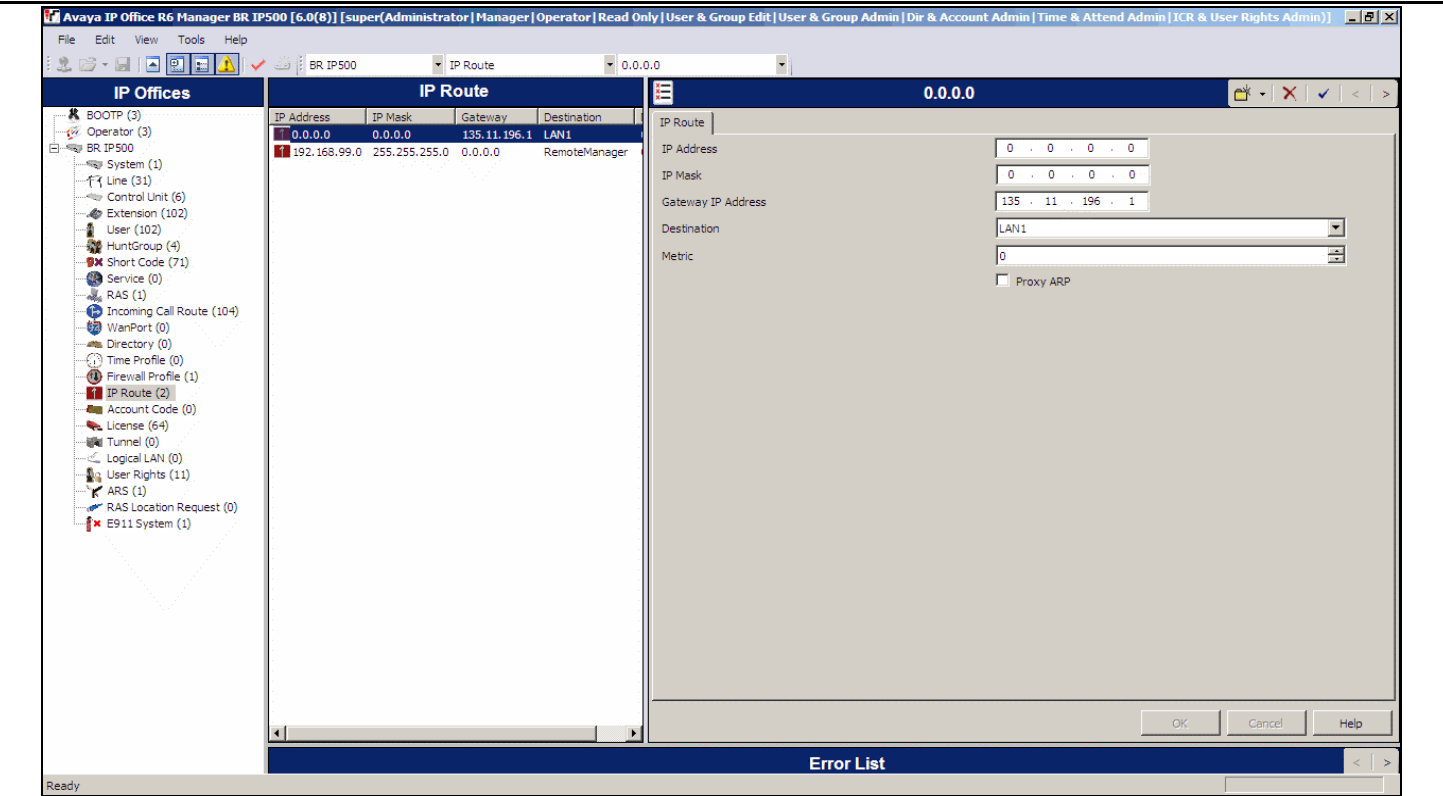


Figure 4 - IP Route Configuration

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

Note: 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.

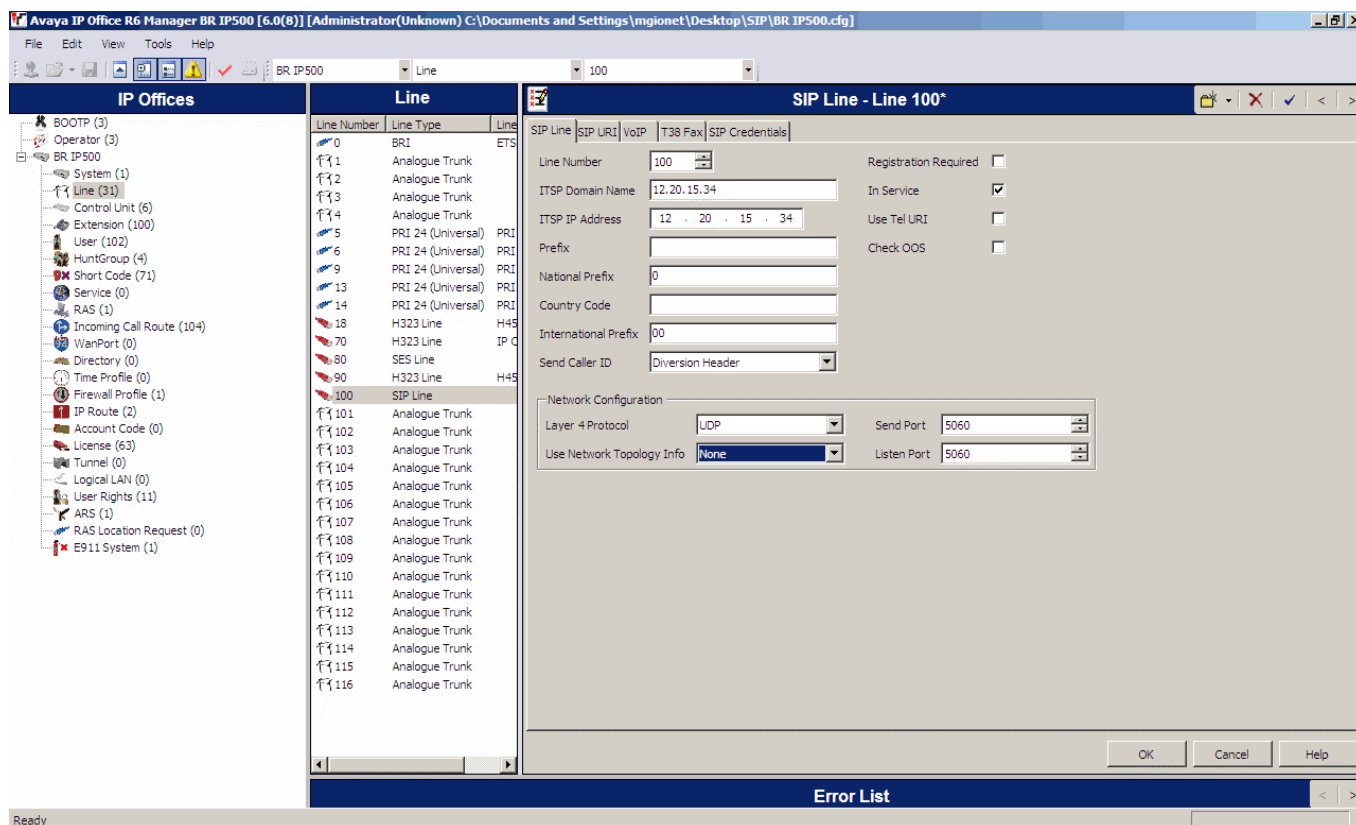


Figure 5 - SIP Line Configuration

- **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). This address will be provided by AT&T.
- **In Service** shall be ticked
- **Use TEL URI** shall be left un-ticked
- **Check OOS** shall be left un-ticked
- **Layer 4 Protocol** shall be set to UDP
- **Send Port** shall be set to 5060 (default)
- **Use Network Topology Info** set to None to disable STUN

- **Listen Port** shall be set to 5060 (default)

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

Click on the SIP URI tab and configure as follows. This section deals with the SIP URI tab on the SIP Line configuration.

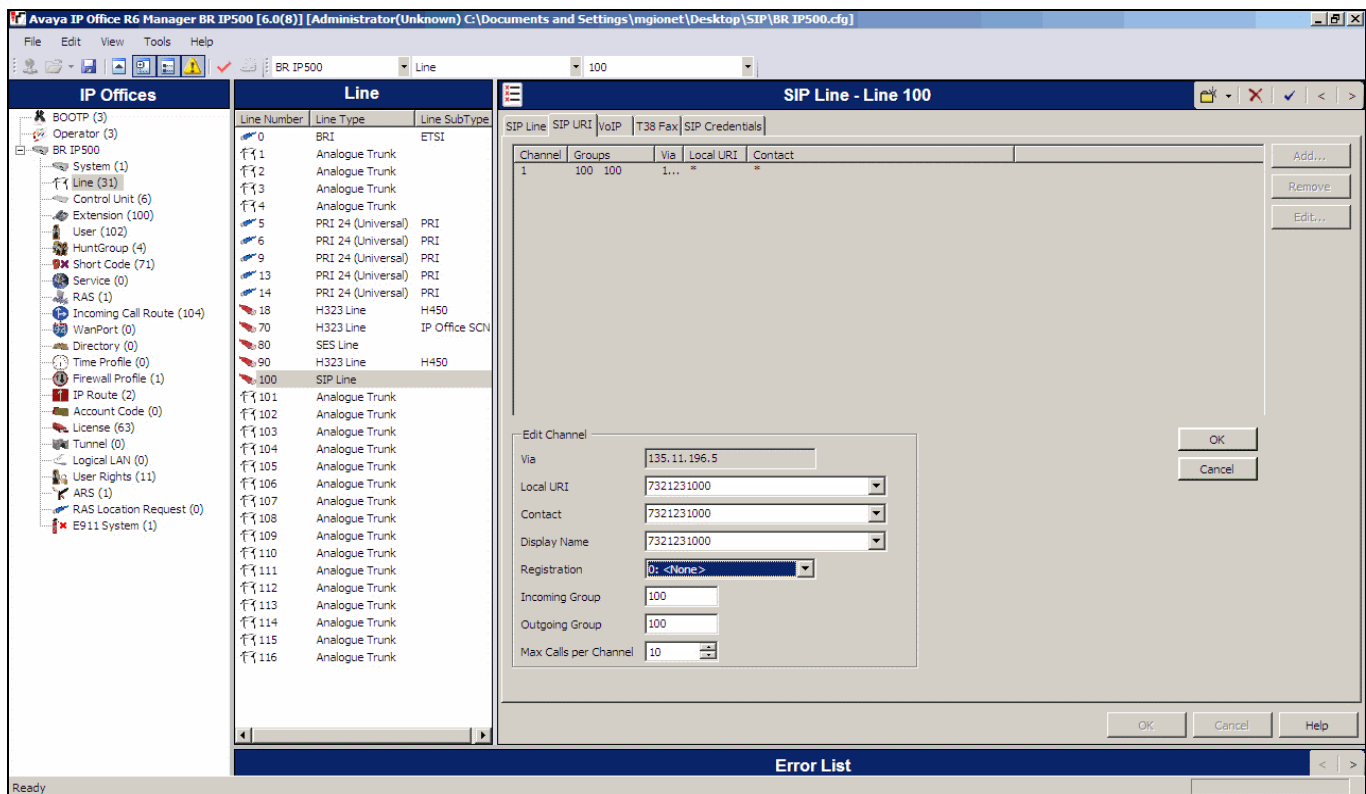


Figure 6 - Main Line Number Configuration

Configure as follows:

- **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.

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 header 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 header 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 at extension 300.

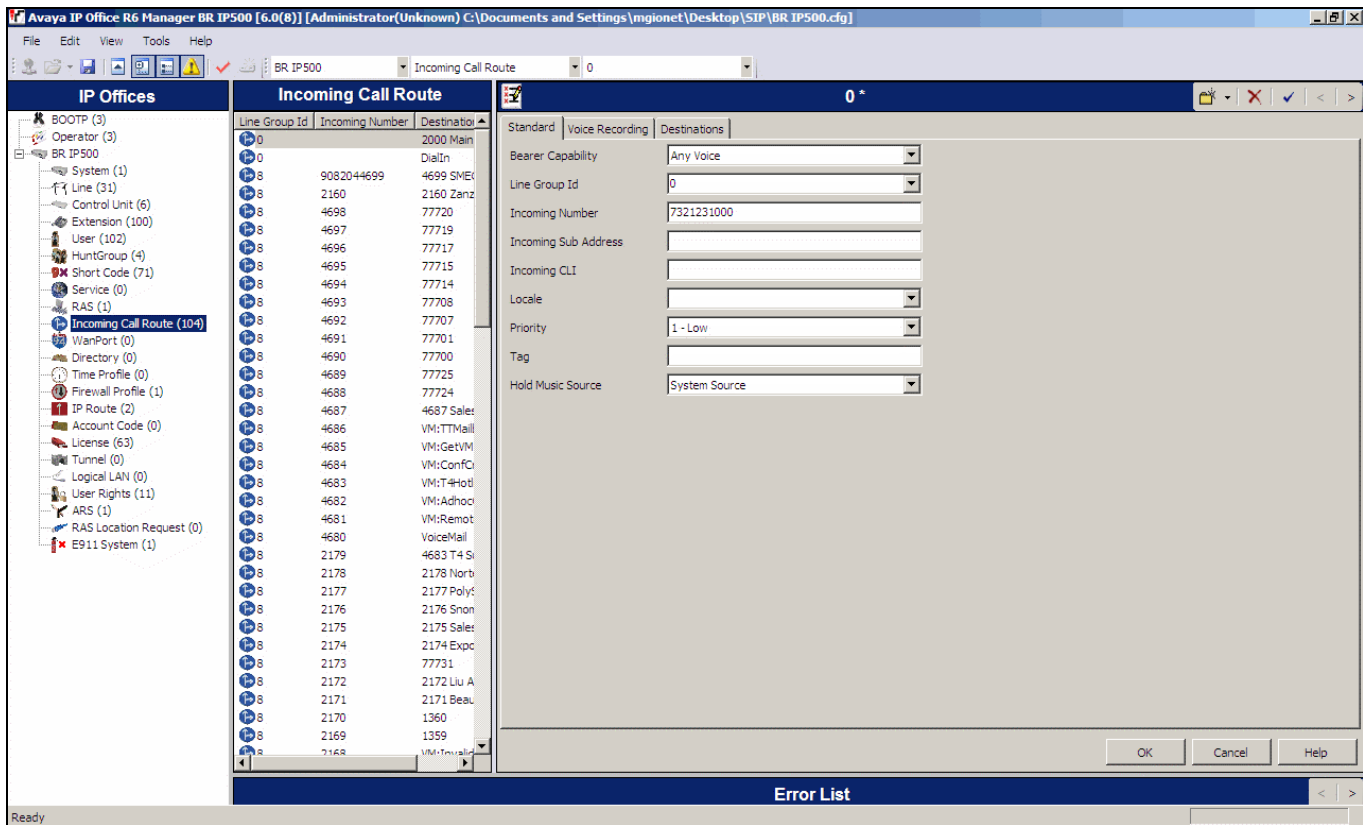


Figure 7 – Incoming Call Route Configuration

Configure the following in the Standard tab:

- **Bearer Capability** shall be set to Any Voice
- **Line Group ID:** shall be set to match the Incoming Group number configured in the SIP URI
- **Incoming Number:** 7321231000

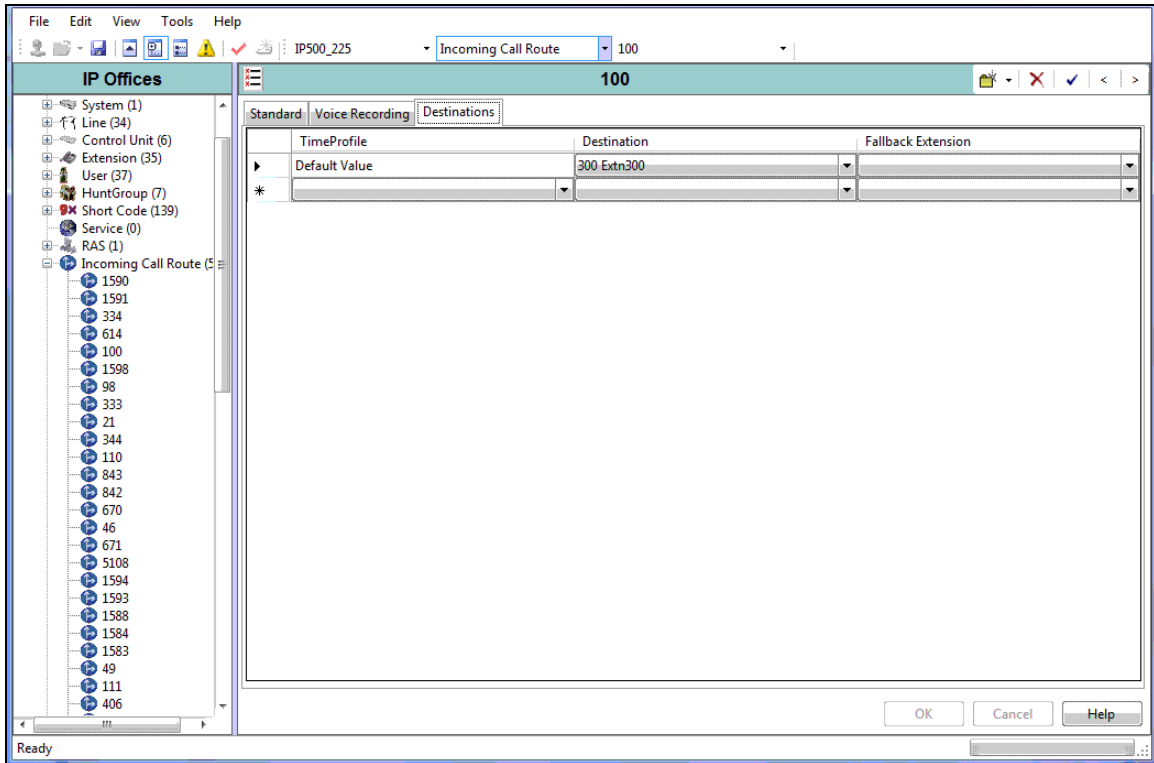


Figure 8 - Destination Tab Configuration

Configure the following in the Destinations tab:

- **Destination:** 300

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:

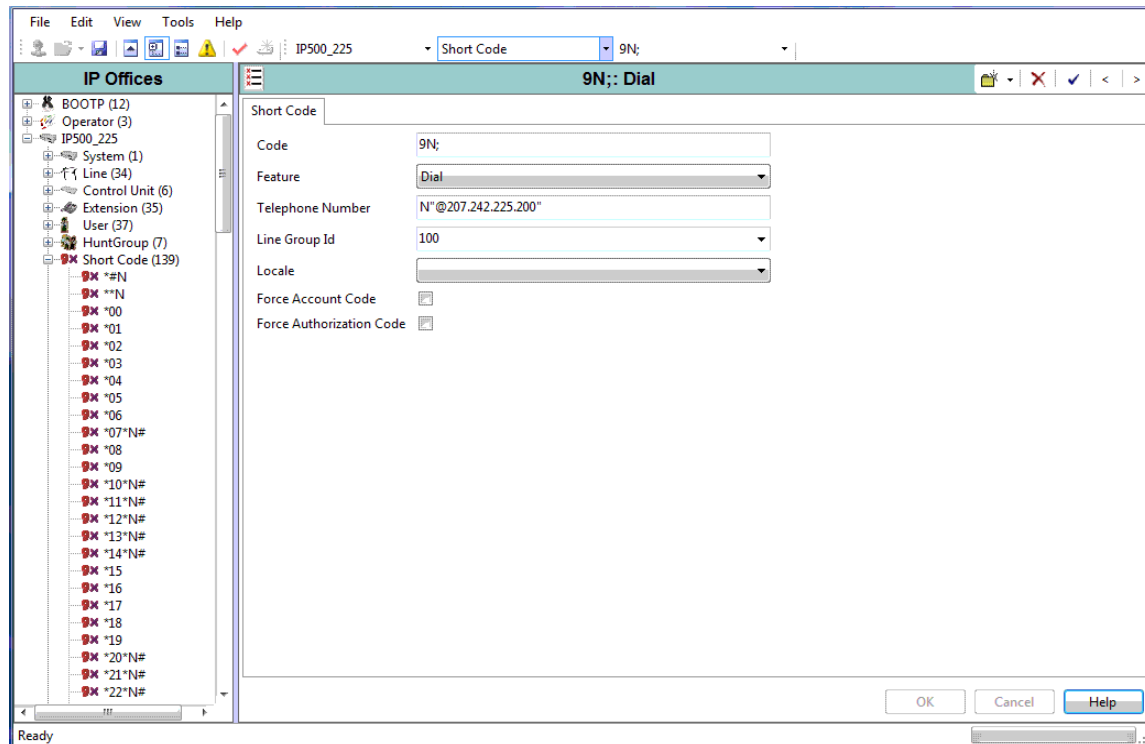


Figure 9 - Short Code Configuration

Field descriptions are as follows:

- **Short code:** 9N; 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.
- **Feature:** Set this field to "Dial".
- **Telephone number:** N"@207.242.225.200". 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.
- **Line Group ID:** 100 This should match the Outgoing Group Number configured in the SIP URI *Outgoing Group ID*, which is used to connect to the AT&T IP FLEXIBLE REACH network.

6. Hunt Group Number Configuration

Steps to configure the Hunt Group number:

- Add a SIP URI (Figure 6).
- Add an Incoming Call Route (Figure 7).
- 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"

This setting is required 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. Enter the following values:

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

7.2 SIP Form User Tab

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 to suppress Caller ID)

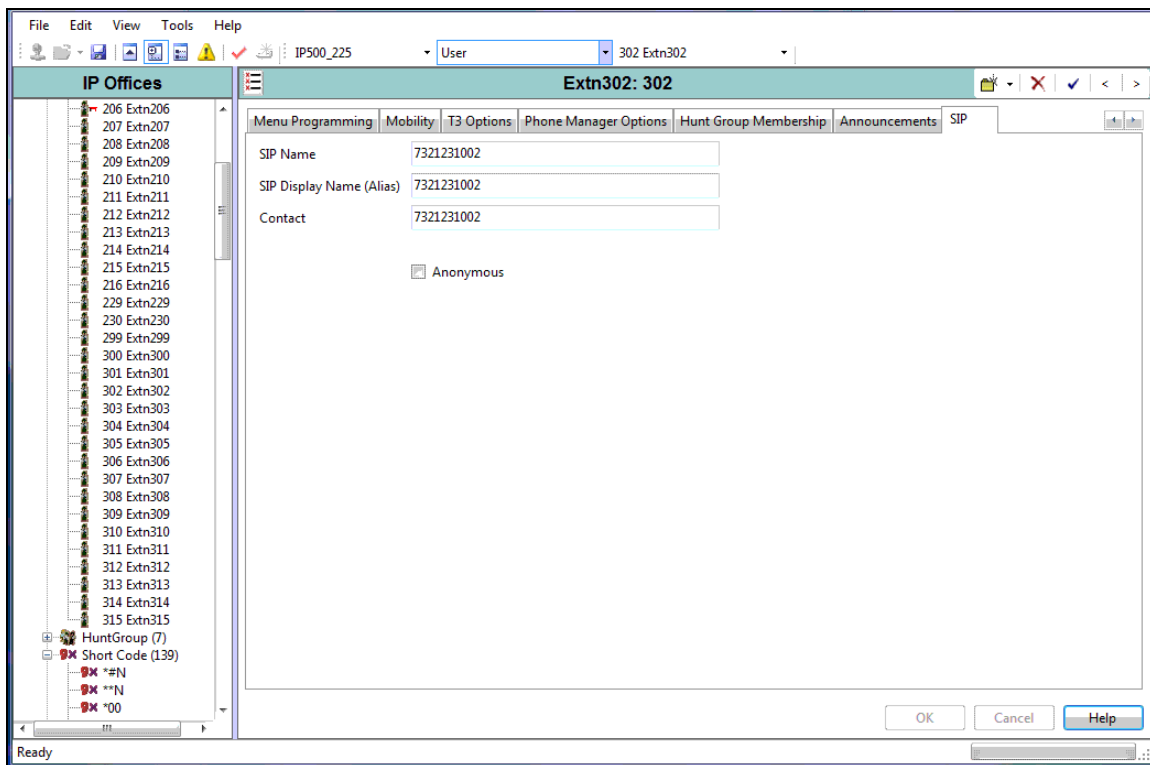


Figure 10 - SIP User Tab Configuration

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 (Figure 8):

- **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:

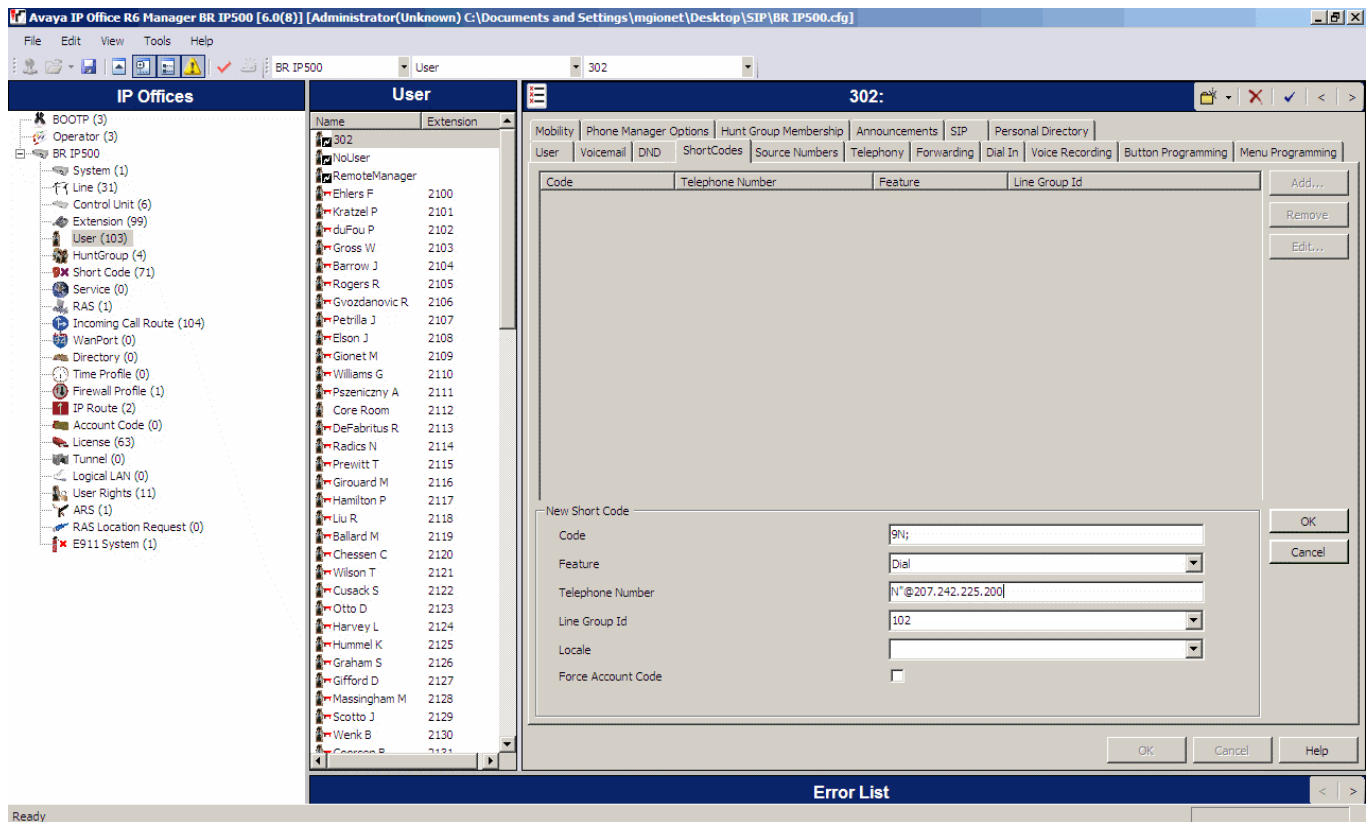


Figure 11 - SIP User Short Code Configuration

- **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.

- **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**.

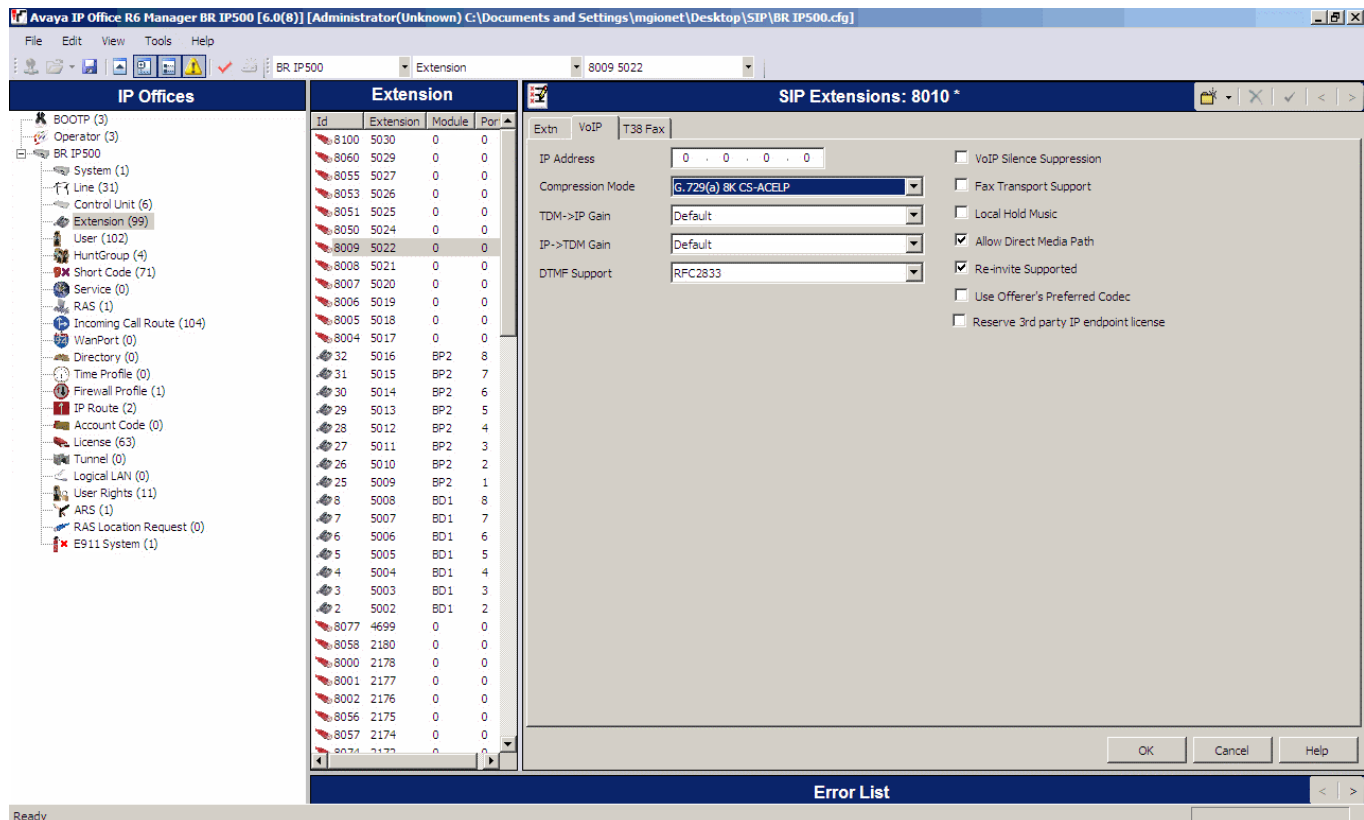


Figure 12 - SIP Extension VoIP Configuration

8.2 T.38 FAX Configuration

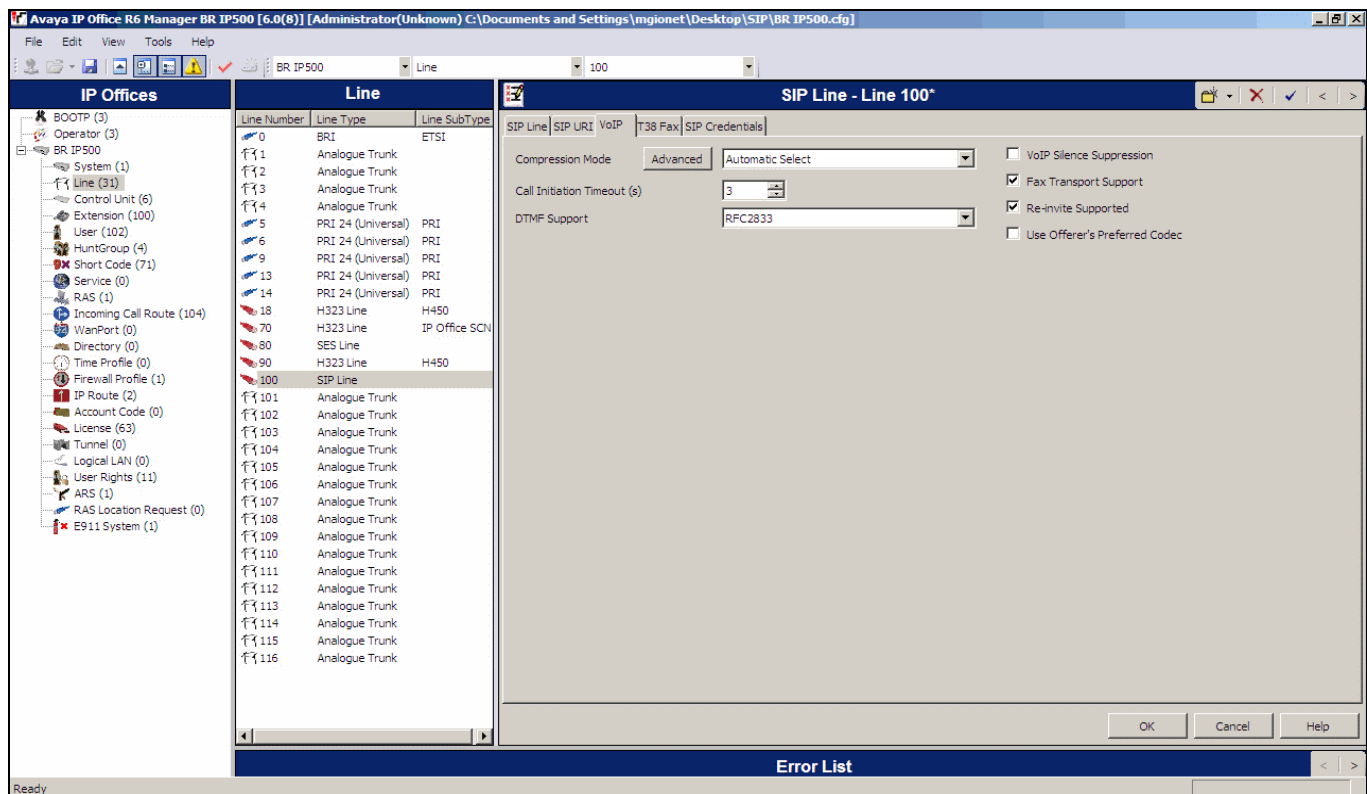


Figure 13 - SIP T.38 Fax Configuration

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

- **Call Initiation Timeout** shall be set to 3 seconds. This is the time IP Office will wait for a reply to an INVITE before attempting a new line via ARS
- **Re-INVITE Supported** shall be left ticked
- **Compression Mode** shall be set to G.729(a) 8K CS-ACELP

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:

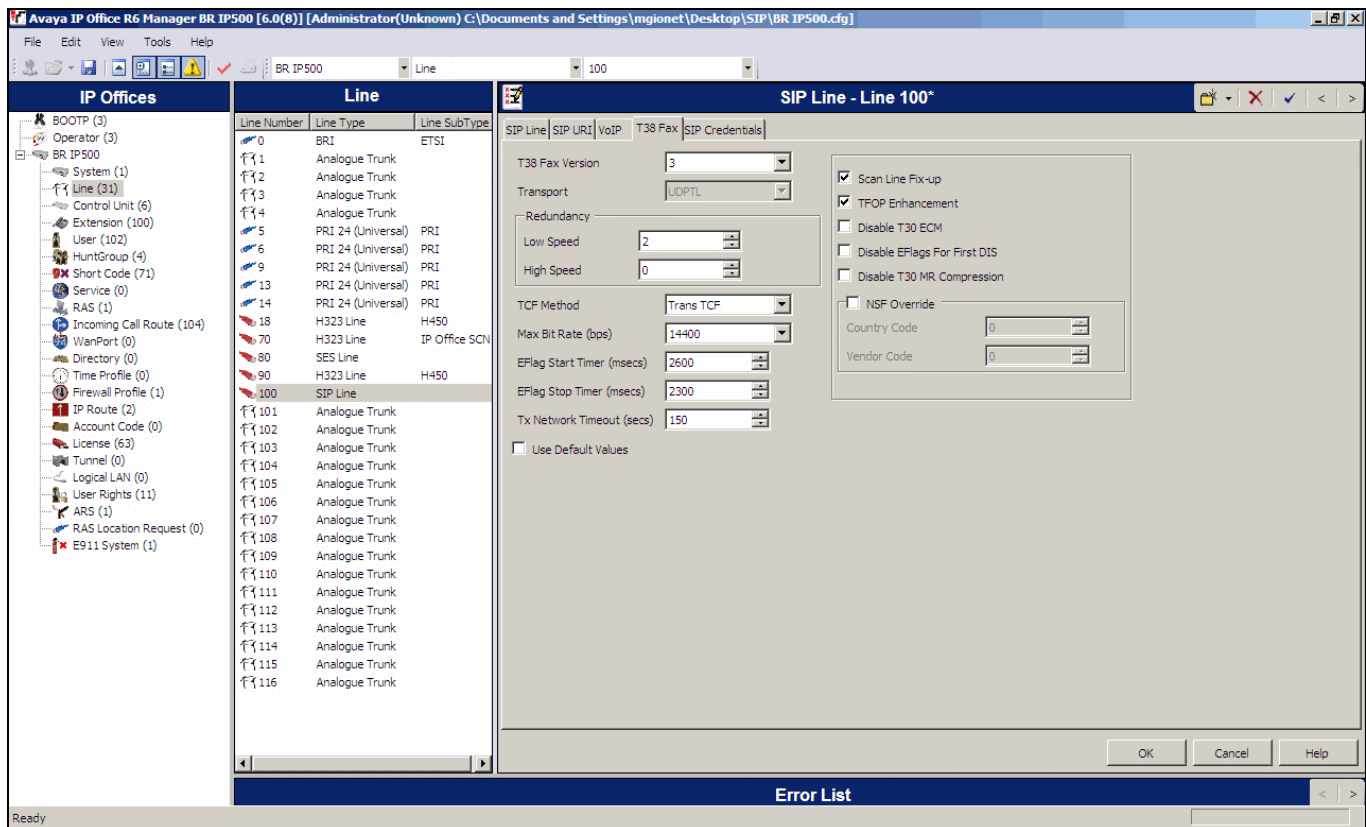


Figure 14 - Default SIP Line Options

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 is shown below:

```
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 Transition 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 Succeeded 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
IP Office Edition (Essential, Preferred, Advanced) Release 6.0 SIP Trunking Configuration Guide
AT&T Flexible Reach and AT&T Flexible Reach with Business in a Box(SM)
```

```

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->CMCSDialed
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: 5381584b34db8b076351cbea01c07aa@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=sendrec

```

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