Avaya IP Office 4.1 SIP
Customer Configuration Guide
For use with
AT&T IP Flexible Reach

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1 Introduction

This document provides a configuration guide to assist administrators in connecting Avaya IP Office Communication System to AT&T over SIP trunks.

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 is Not Supported

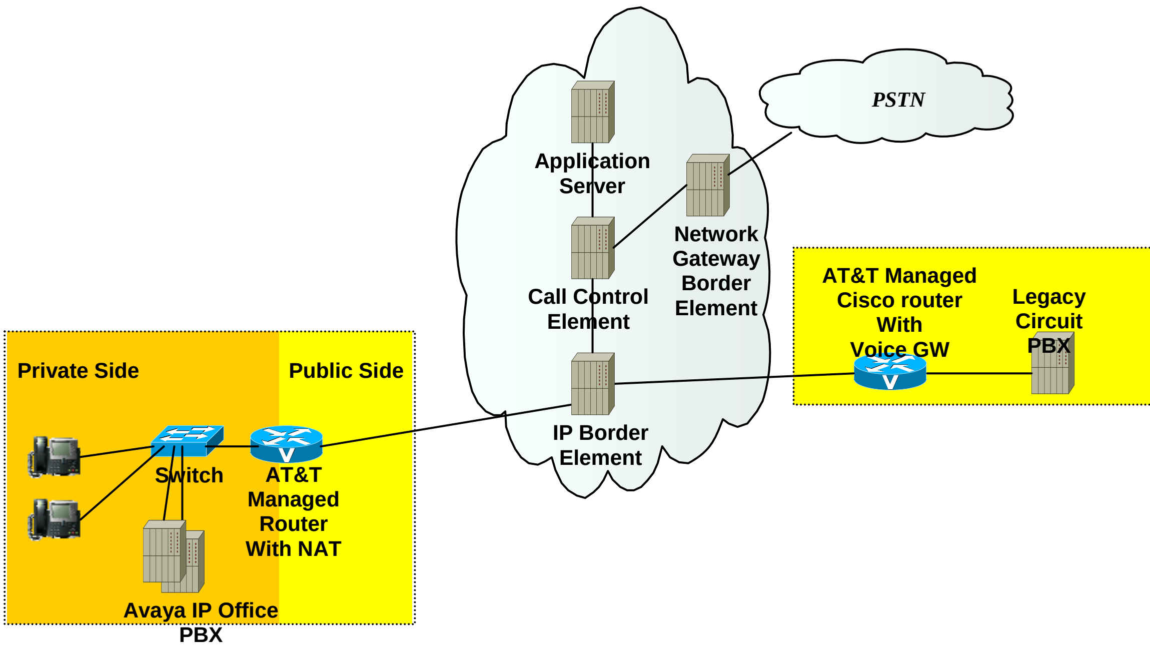
IP Office does not support fax transmission using SIP.

Failover to an Alternate AT&T Border Element

Currently, there is no tested configuration with the Avaya IP Office SIP trunk for connecting to an alternate AT&T Border Element if the primary border element is not available.

3 Overview

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



3.1 Configuration

The Avaya customer premises 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 platforms:

- Small Office Edition
- IP406v2
- IP412
- IP500

3.3 Supported Phone Types

A list of supported phones is provided below:

2400/5400 series digital
4600/5600 series IP
6400 series digital
T3 (IP and digital) –excluding Small Office
3701/3711 (IP DECT)
Analog phones

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 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 legacy VCM cards in number of two selected between the following options: VCM4, VCM 8, VCM 16, VCM 24 and VCM 30. Second option is new VCM cards in number of two from any VCM 32 and 64.

The 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 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.
- Outbound local N11 (i.e. 411, 911) calls from the IP OFFICE phones.

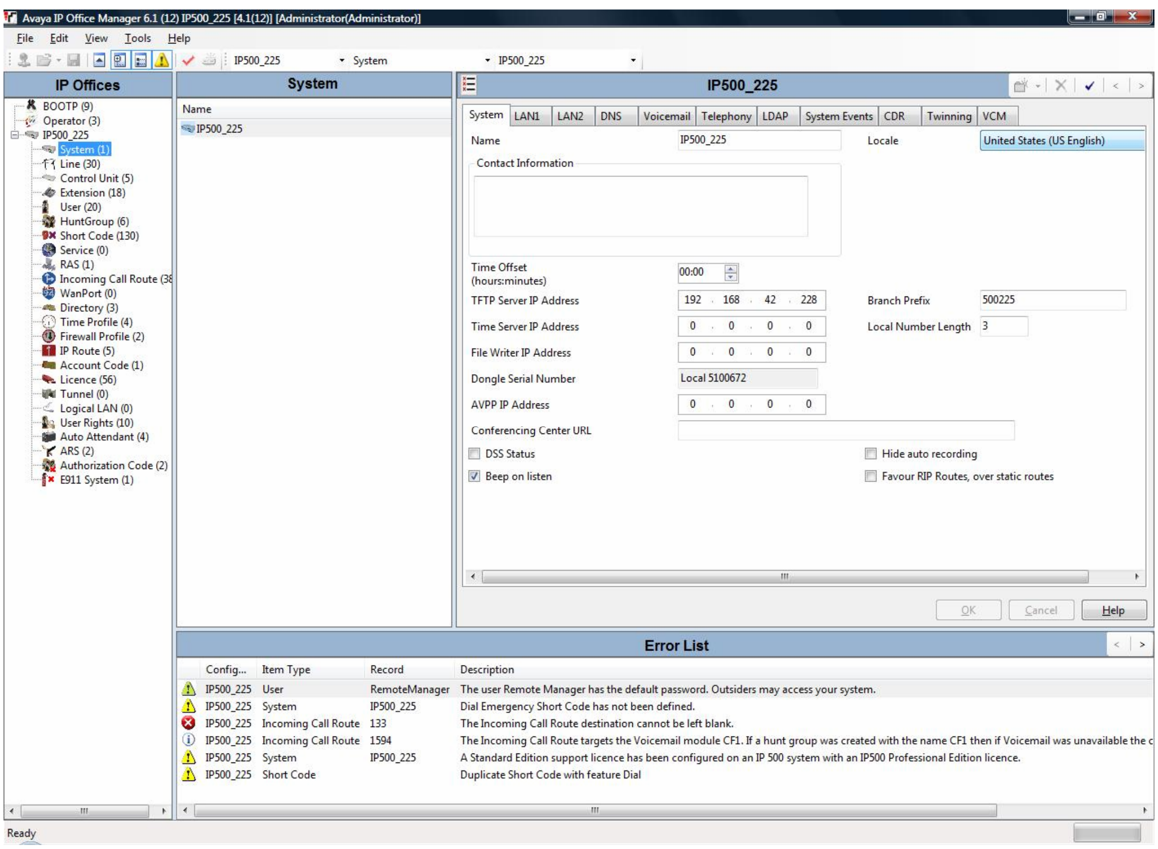
4 Customer Configuration Guide

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 with VCM cards.

4.1 How to identify you are running version 4.1

Users can identify the version number they are running by to looking at the top line of their manager screen and identify the Manager and Core version.



As shown above, the IP Office Core software version is 4.1 (12).

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.

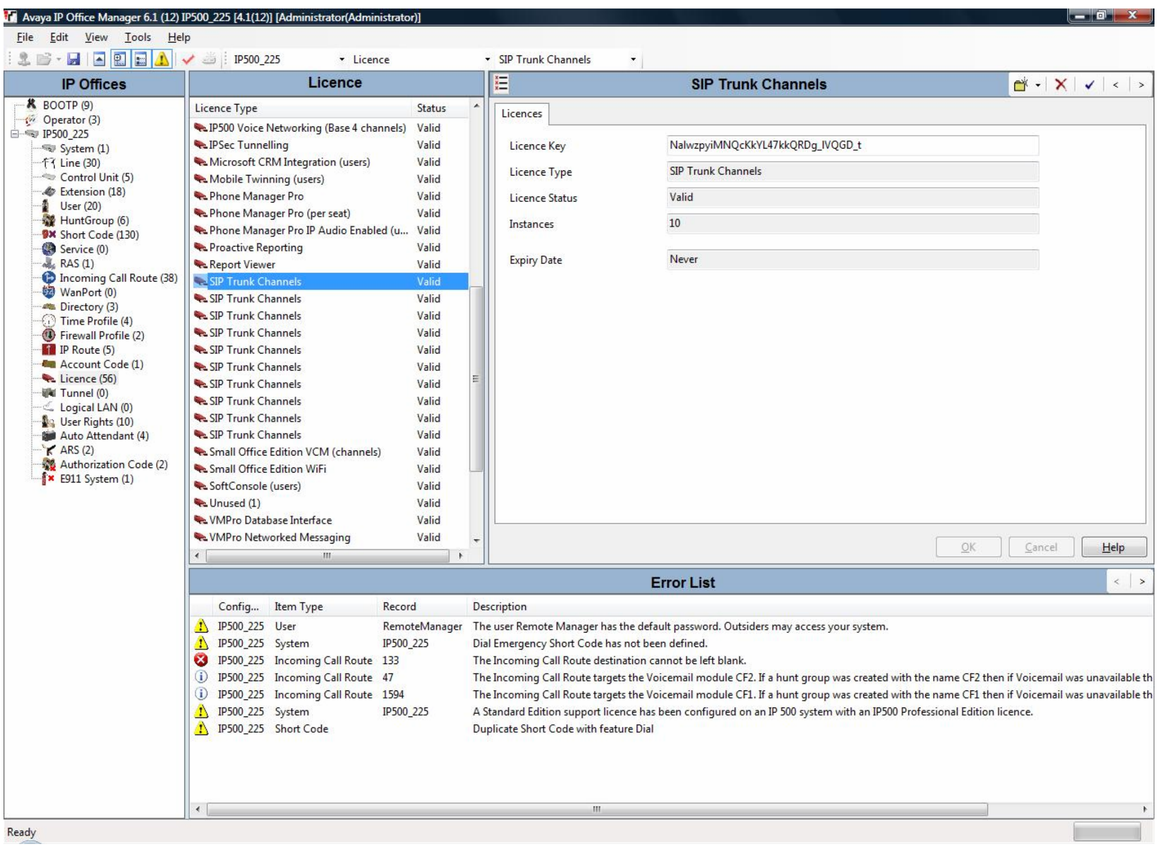


Figure 1: License From with valid SIP license

License Key is the license identifier that will be provided by Avaya business partners.

License Type must be set to *SIP Trunk Channel*.

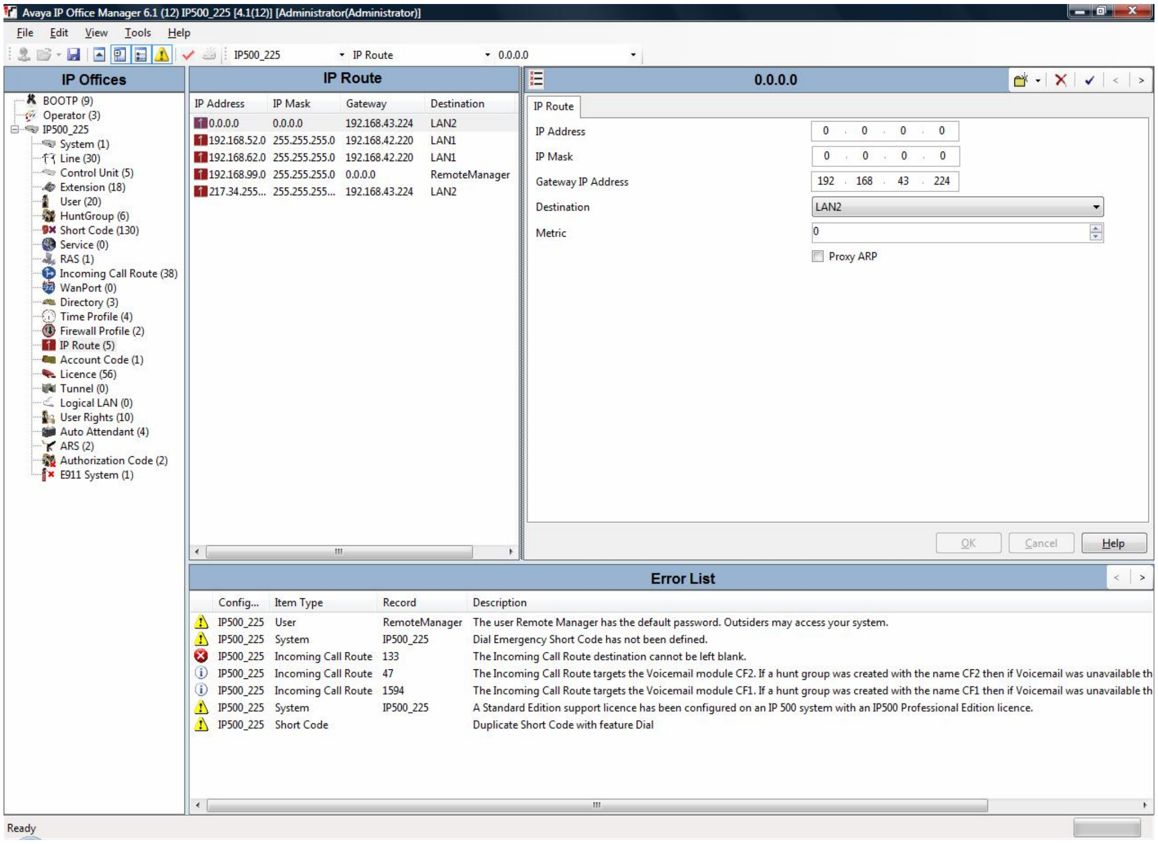
License Status should be set to *Valid*, if the acquired license is a valid one.

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 Main 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. **You must configure a SIP line for each of the 2 AT&T Border Elements provided by AT&T Customer care.**

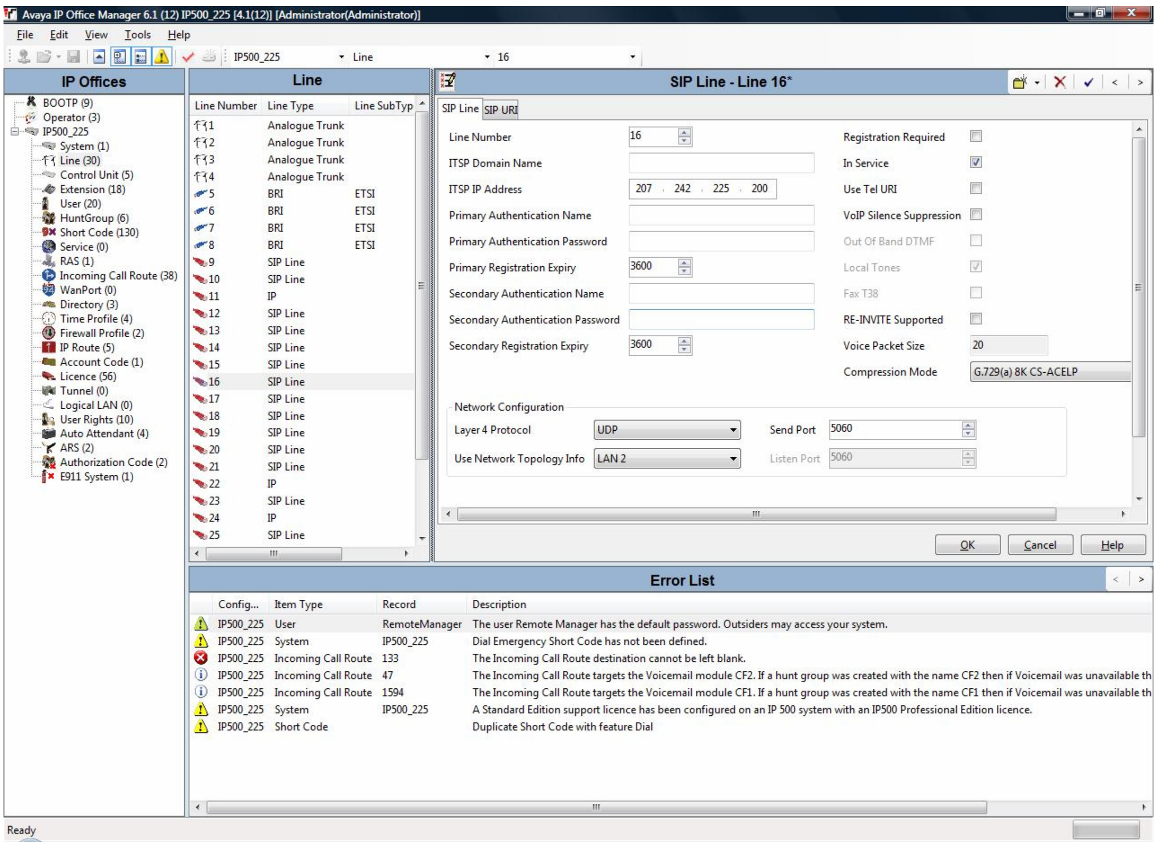


Figure 2: SIP Main Line

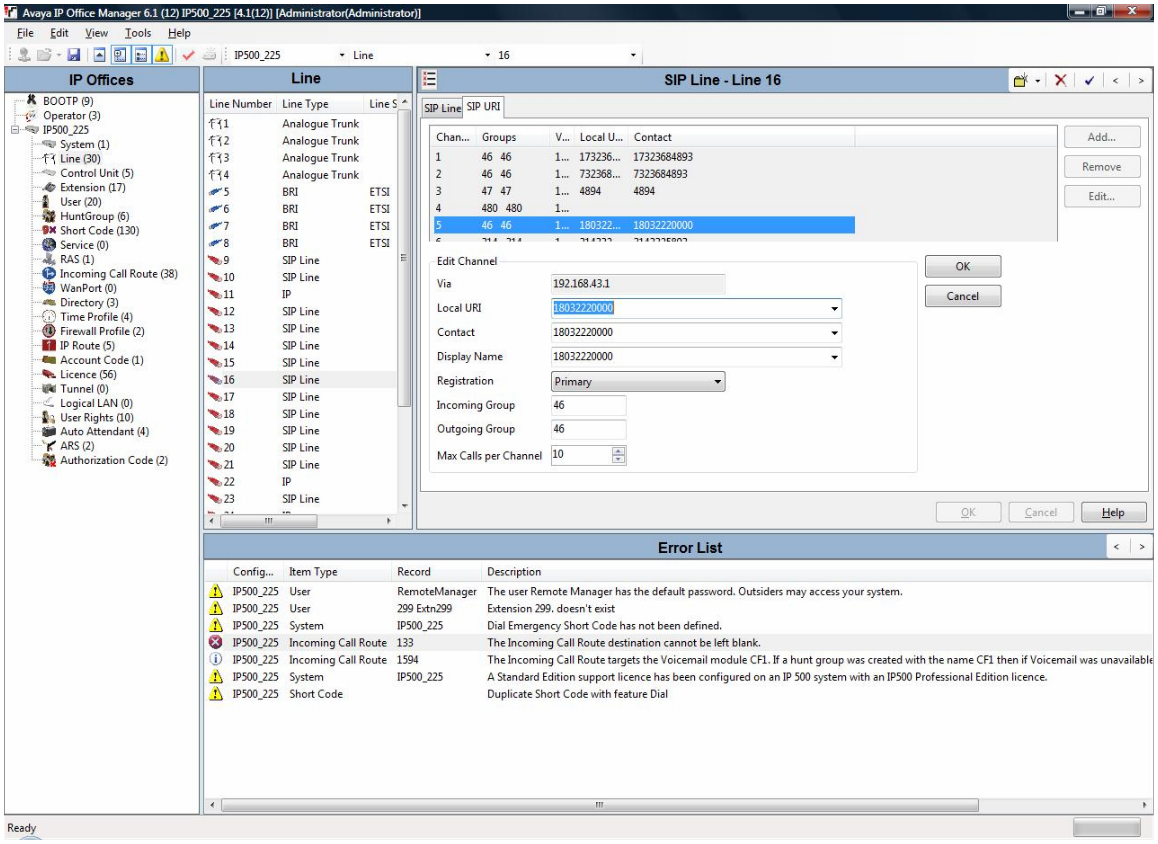
Here is a guideline for configuration of each field.

- Line number is automatically assigned by Manager in an incremental manner.
- 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.
- Primary Authentication Name shall be left blank
- Primary Authentication Password shall be left blank
- Primary Registration Expiry shall be left as default
- Secondary Authentication Name shall be left blank
- Secondary Authentication Password shall be left blank
- Secondary Registration Expiry shall be left as default
- In Service shall be ticked
- Use TEL URI shall be left un-ticked
- Re-INVITE Supported shall be left un-ticked
- Compression mode should be set to G.729(a) 8K CS-ACELP
- Send Port shall be set to 5060 (default)
- Listen Port shall be set to 5060 (default)

4.5 SIP Line: SIP URI tab

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



Local URI can be set in two different modes for this configuration type:

1. By editing each field individually. This mode allows setting each SIP URI to coincide with a given DID assigned by AT&T. With this mode, SIP URI settings are common for all users in the system.

2. By setting in to Use User Data (shown above). This setting allows differentiating each SIP URI to a given user in the system. The fields will be filled in using SIP tab in User form.

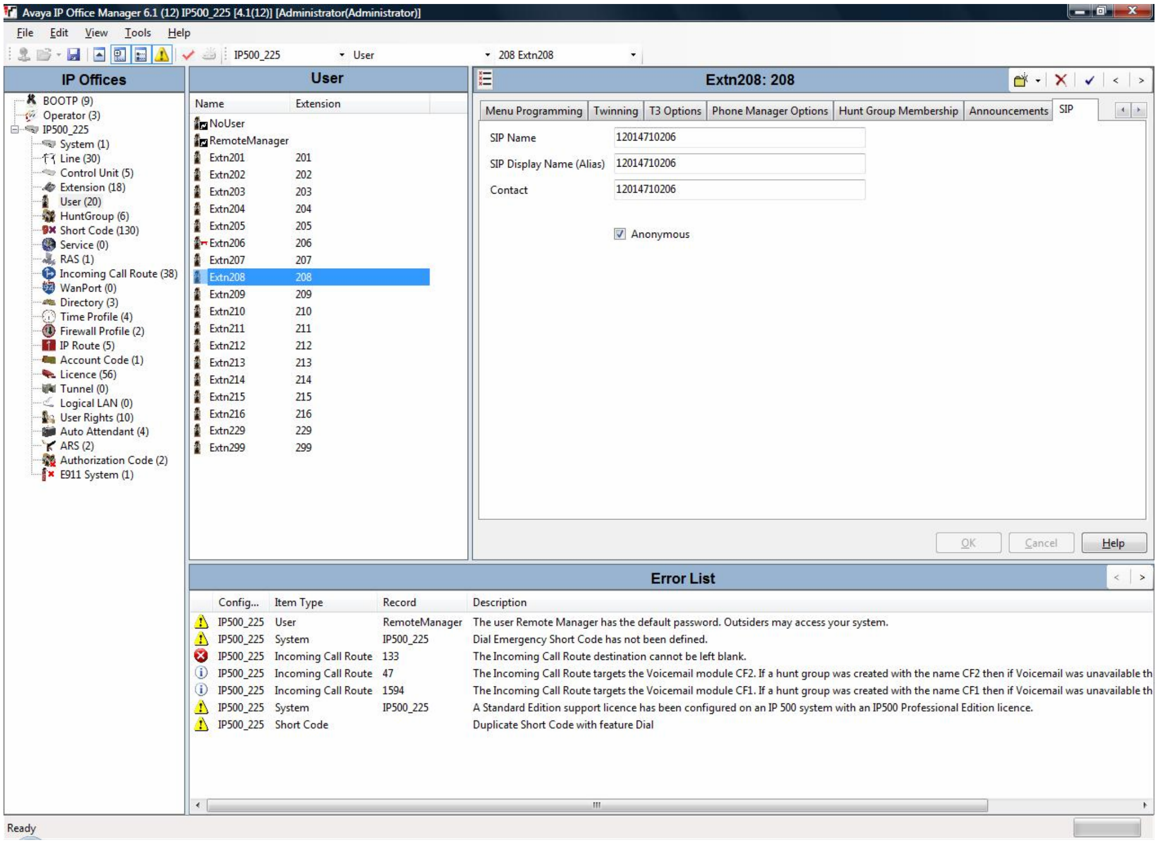
The fields on this form are Local URI, Contact, and Display Name and are described next.

- **Local URI** populates the user part of the FROM header for outbound calls.
- **Contact** sets the user part of the CONTACT header for outbound calls.
- **Display Name** sets the display name field in the FROM header for outbound calls.

The number of simultaneous calls may also dependent on the number of calls supported on the VCM cards.

4.6 Setting SIP tab in User Field

This section deals with the SIP tab on the user configuration.



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

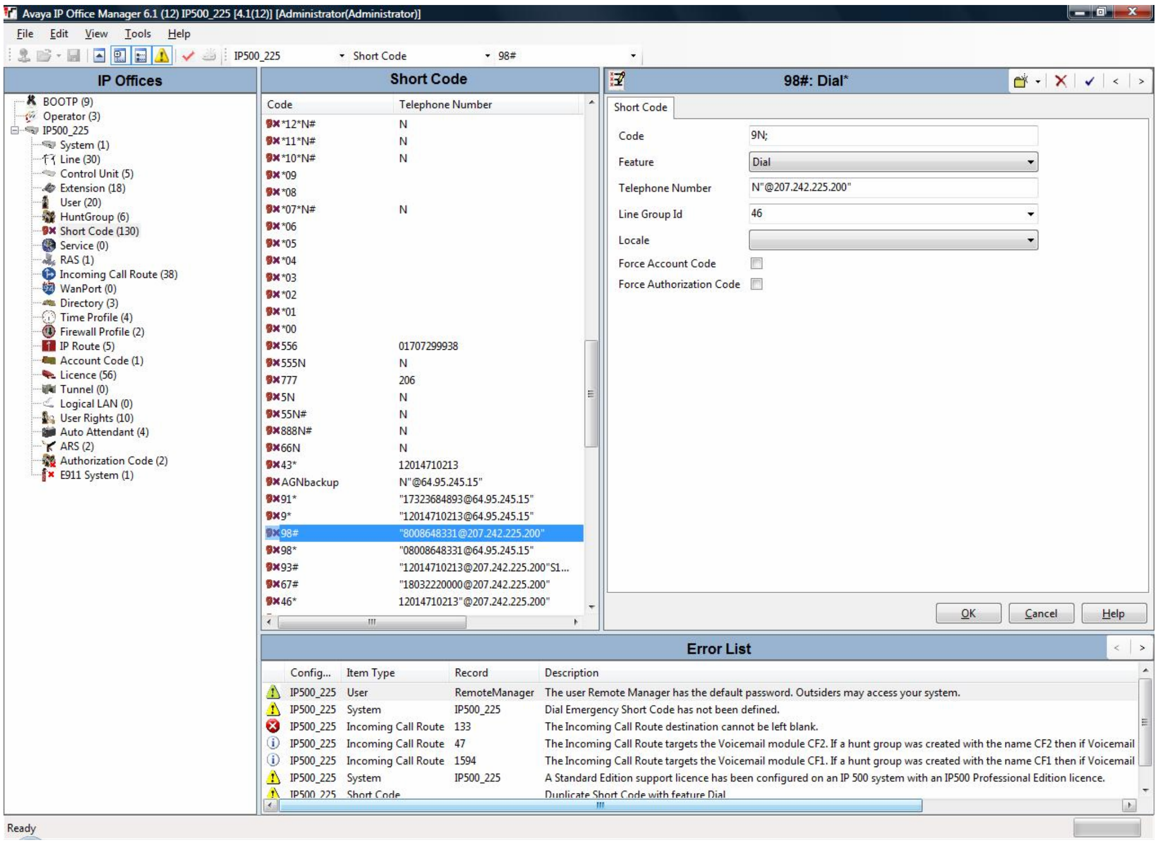
- **SIP Name** populates the user part of the FROM header for outbound calls.
- **SIP Display Name** sets the display name field in the FROM header for outbound calls.
- **Contact** sets the user part of the CONTACT header for outbound calls.

Anonymous enables the Privacy Mechanism for outbound calls according to RFC 3325.

4.7 Routing Calls to AT&T

This section describes the IP Office configuration required for sending calls to the AT&T IP FLEXIBLE REACH network.

The usual system of SHORT CODE object can be used to direct calls to the AT&T IP FLEXIBLE REACH network. The following screen shot gives an example in a fictitious IP Address.



The fields are populated as follows:

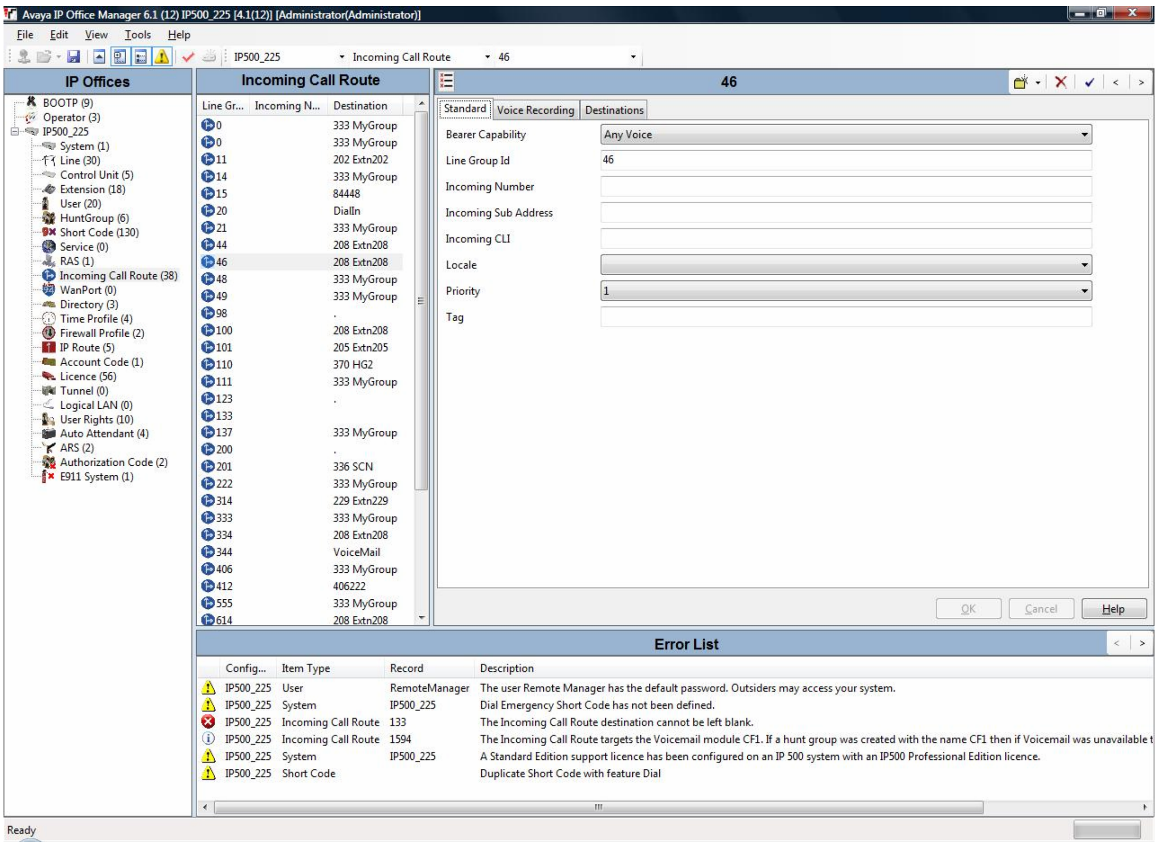
- **Short code** – This field matches the dialed number. The sample short code shown above will match on a "9" followed by any number of digits (represented by "N"). The short code should end with a semi colon.
- **Line Group ID** – Set this field to the SIP *URI Outgoing Group* ID that it is used to connect to the AT&T IP FLEXIBLE REACH network.
- **Feature** – Set this field to "Dial".
- **Telephone number** - is set to "N"@<IP Border Element address on the IP Flexible Reach Service>"". Note that IP Office can only send calls to

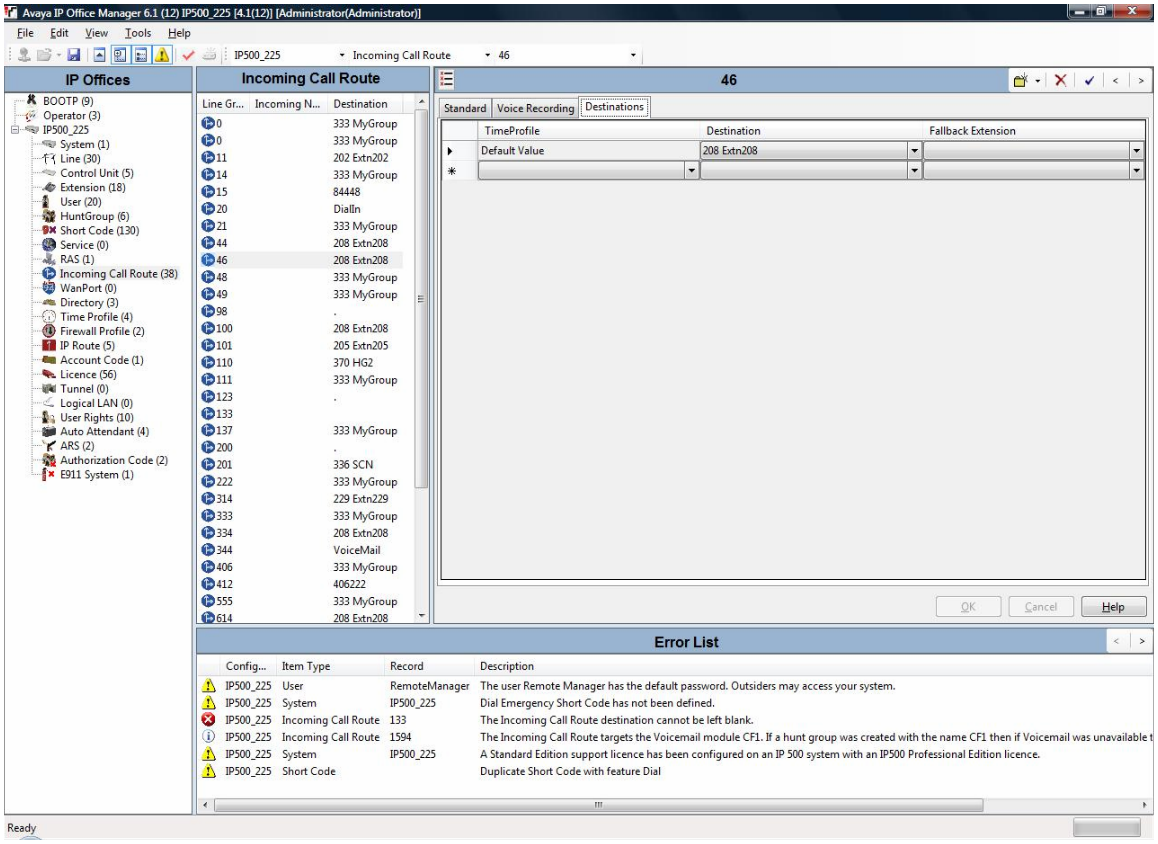
one AT&T Border Element. If that Border Element is not available, then the customer must manually re-configure this value to point to the secondary Border Element.

4.8 Receiving Calls from AT&T

The translation of each DID to an IP Office extension is done using ICR call route field as shown below.

You must configure Incoming Call Routes for each of the SIP lines associated with each of the 2 AT&T Border Elements provided by AT&T Customer care.





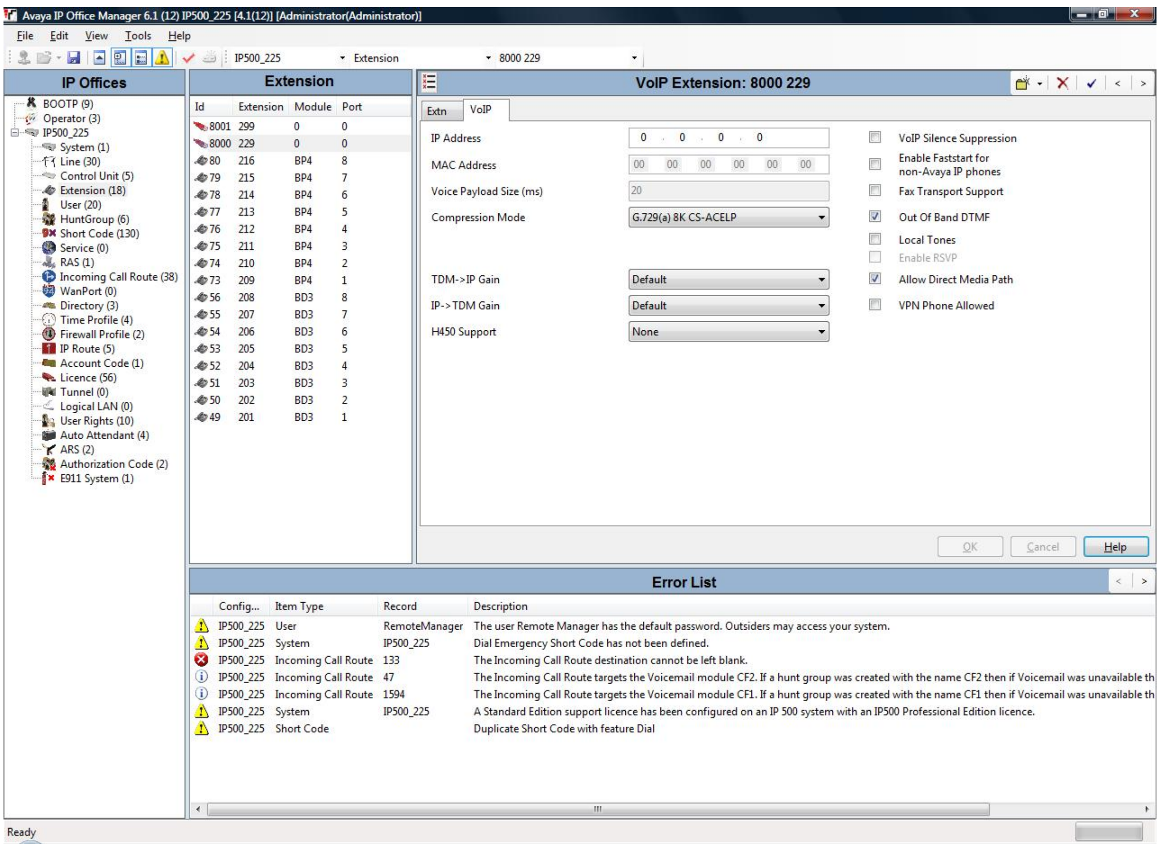
The fields are populated as follows:

- **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 or hunt-group.

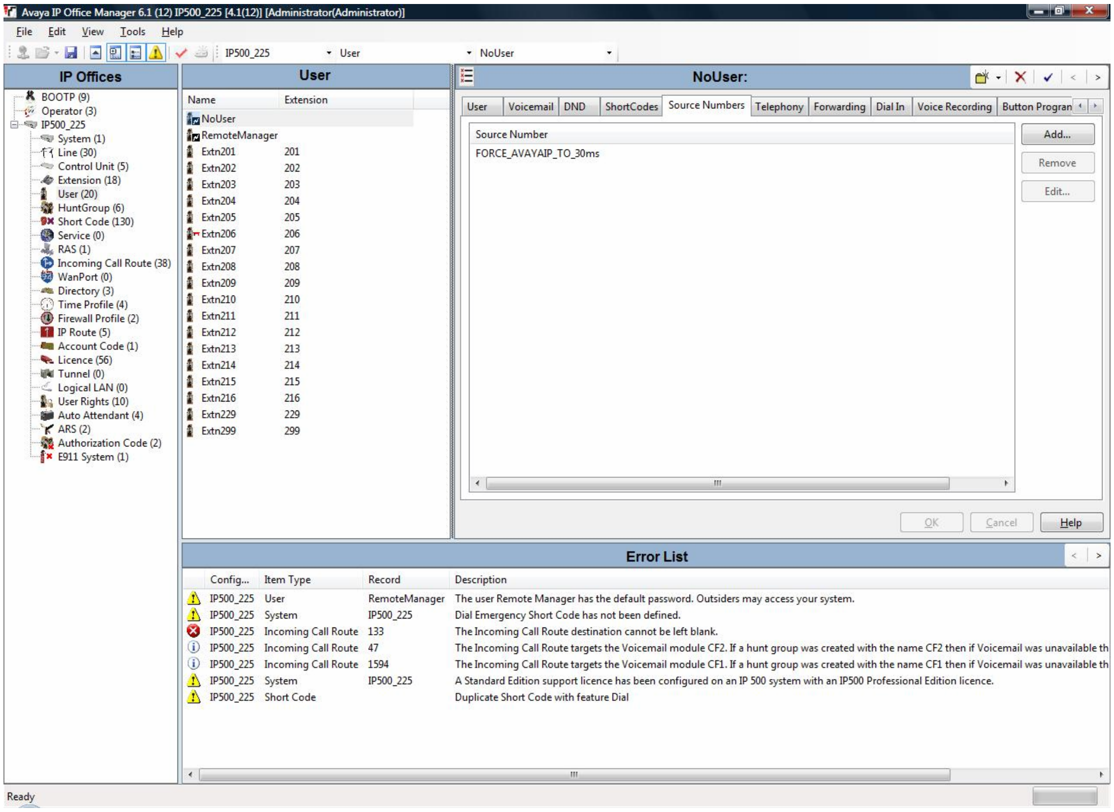
4.9 IP Phone Configuration

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.



An extra step is required to support 30 msec payloads for Avaya IP Phones. This is achieved by adding the SourceNumber *FORCE_AVAYAIP_TO_30ms* to the NoUser form:



5 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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