AVAYA

IP Office 6.0 Essential Edition PARTNER® Version SIP Trunking Configuration Guide

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

Issue 1.1 (05-17-2010)

Notice
While reasonable efforts were made to ensure that the information in this document was complete and accurate at the time of printing, Avaya Inc. can assume no liability for any errors. Changes and corrections to the information in this document may be incorporated in future releases.

Documentation Disclaimer
Avaya Inc. is not responsible for any modifications, additions, or deletions to the original published version of this documentation unless such modifications, additions, or deletions were performed by Avaya.

Link Disclaimer
Avaya Inc. is not responsible for the contents or reliability of any linked Web sites referenced elsewhere within this Documentation, and Avaya does not necessarily endorse the products, services, or information described or offered within them. We cannot guarantee that these links will work all of the time and we have no control over the availability of the linked pages.

Avaya Fraud Intervention
If you suspect that you are being victimized by toll fraud and you need technical assistance or support, call Technical Service Center Toll Fraud Intervention Hotline at +1-800-643-2353 for the United States and Canada. Suspected security vulnerabilities with Avaya Products should be reported to Avaya by sending mail to: securityalerts@avaya.com.

For additional support telephone numbers, see the Avaya Support web site (http://www.avaya.com/support).

Documentation information
For the most current versions of documentation, go to the Avaya Support web site (http://www.avaya.com/support) or the IP Office Knowledge Base (http://marketingtools.avaya.com/knowledgebase/).

Avaya Support
Avaya provides indirect and direct services for customer support, report problems or to ask questions about your product. These services are subject to your support agreement. Contact your local reseller / distributor for indirect support . Contact Avaya Global Services (AGS) for direct support. For additional information on support, see the Avaya Web site: http://www.avaya.com/support.

Contents

1. Introduction

This document provides a configuration guide to assist administrators in connecting Avaya IP Office 6.0 Essential Edition PARTNER Version 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 only the first option.

2. Special Notes

Emergency 911/E911 Services Limitations

Emergency 911/E911 Services Limitations and Restrictions - While AT&T IP Flexible 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 Essential Edition PARTNER Version does not support FAX transmission over SIP Trunks. However FAX transmission might work when the SIP Line is configured with G711 codec.

IP Office Advanced Edition supports fax transmission using SIP when T.38 is configured and successfully negotiated. This is supported in the IP500 platform with VCM10, VCM32 or VCM64.

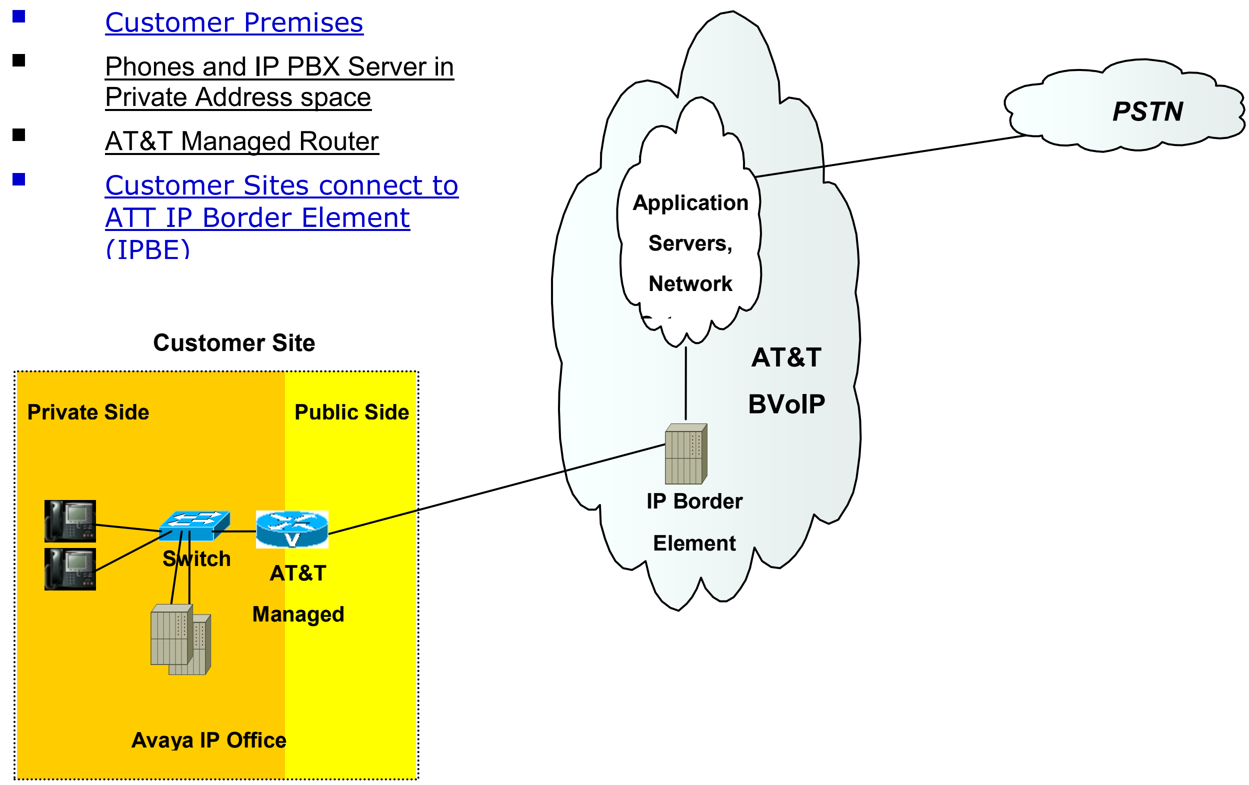
IP Office does not support fax transmission using SIP in any other configuration.

SIP Line failover

It is possible to configure several SIP Lines with different IP Addresses. However, IP Office Essential Edition PARTNER Version does not support the failover from one line to another. This is supported in the other Product Suite options.

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.



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

  This is an Avaya IP500v2 running in PARTNER Version mode. A Combination card is required.

- 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:

- 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

In order to support SIP trunks, the platform requires the use of a VCM module. This is provided by the Combo card. This card includes a VCM10 module, and provides 4 CO Lines ports, 6 digital station ports and 2 Analog station ports.

The maximum number of calls supported on the VCM card is specified by the VCM card number, that is, VCM 10 supports 10 calls. A maximum of two Combo cards are supported per platform.

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.

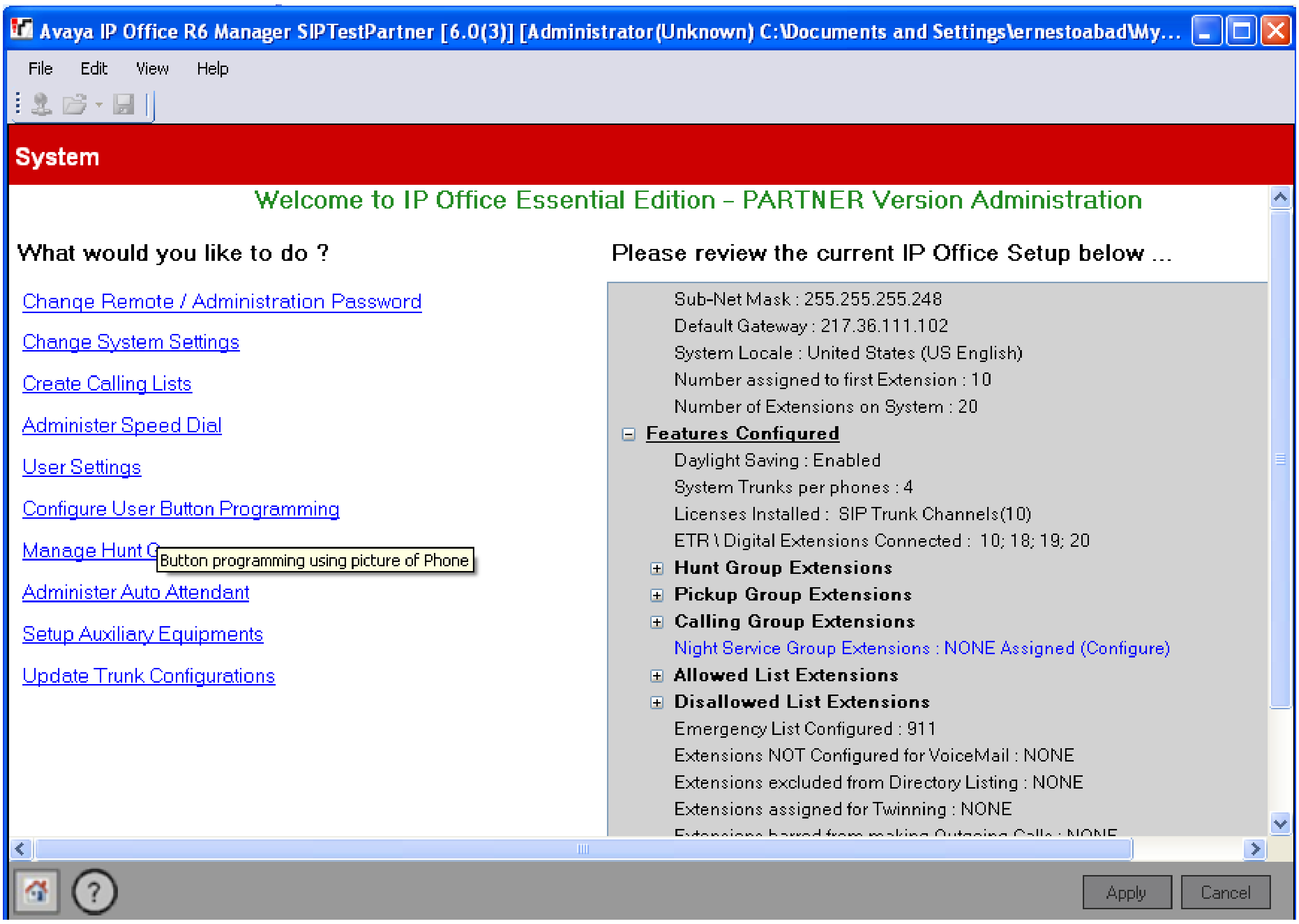
4. Customer Configuration Guide

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 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.

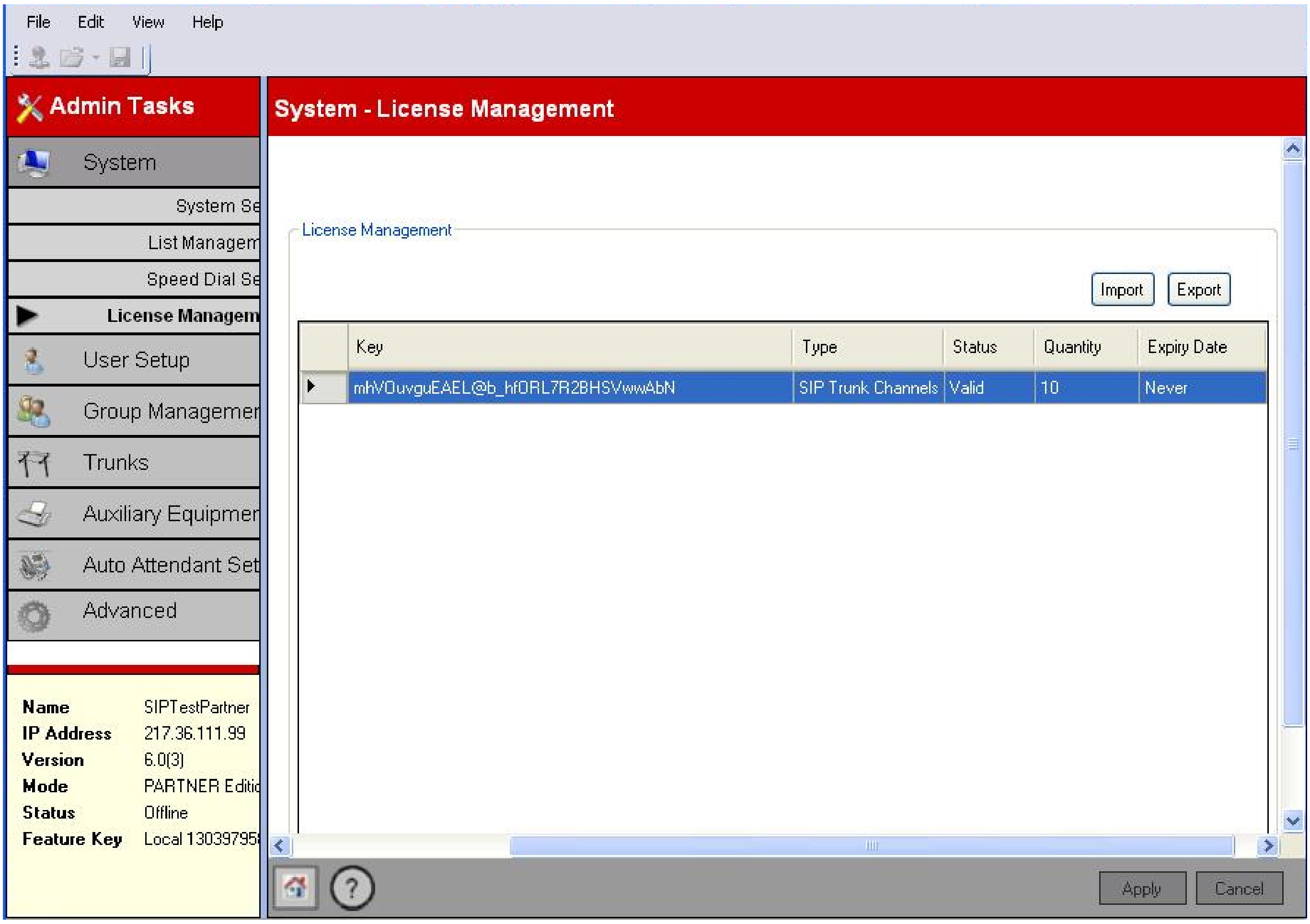


As shown above, the IP Office Core software version is 6. 0.(3).

The version will also be displayed at the bottom left of this window by deselecting the Hide Admin Tasks option from the View menu.

4.2 How to Check For SIP Trunking Licenses

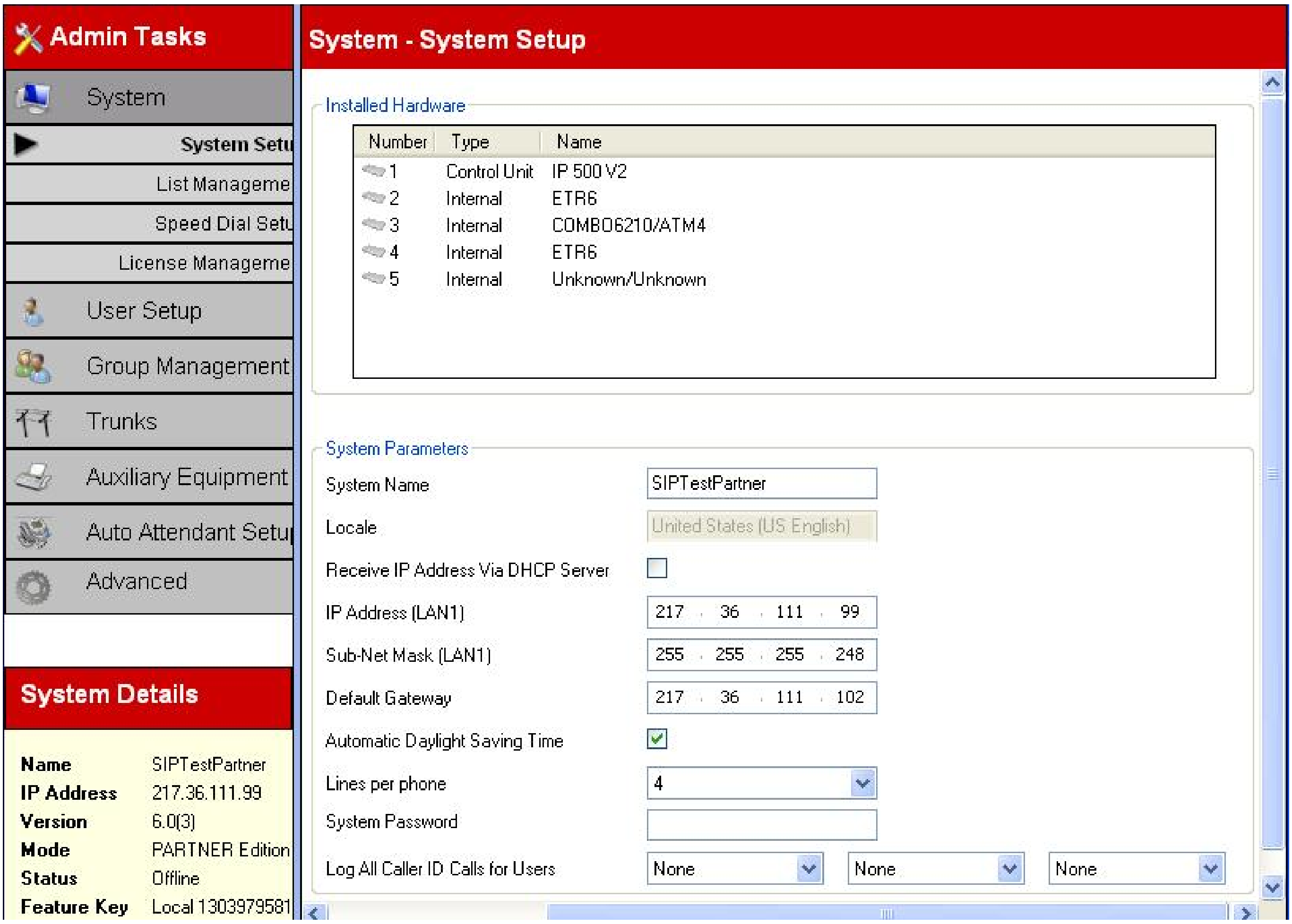
To make calls using SIP you must have a valid license that can be purchased through Avaya business partners. PARTNER Version offers 3 simultaneous SIP calls without the need of adding any SIP Licenses.



- *Key* is the license identifier that will be provided by Avaya business partners.
- *Type* must be set to *SIP Trunk Channel*. *License Status* should be set to *Valid*, if the acquired license is a valid one.
- *Status* reports whether the license is Valid or not.
- *Quantity* 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

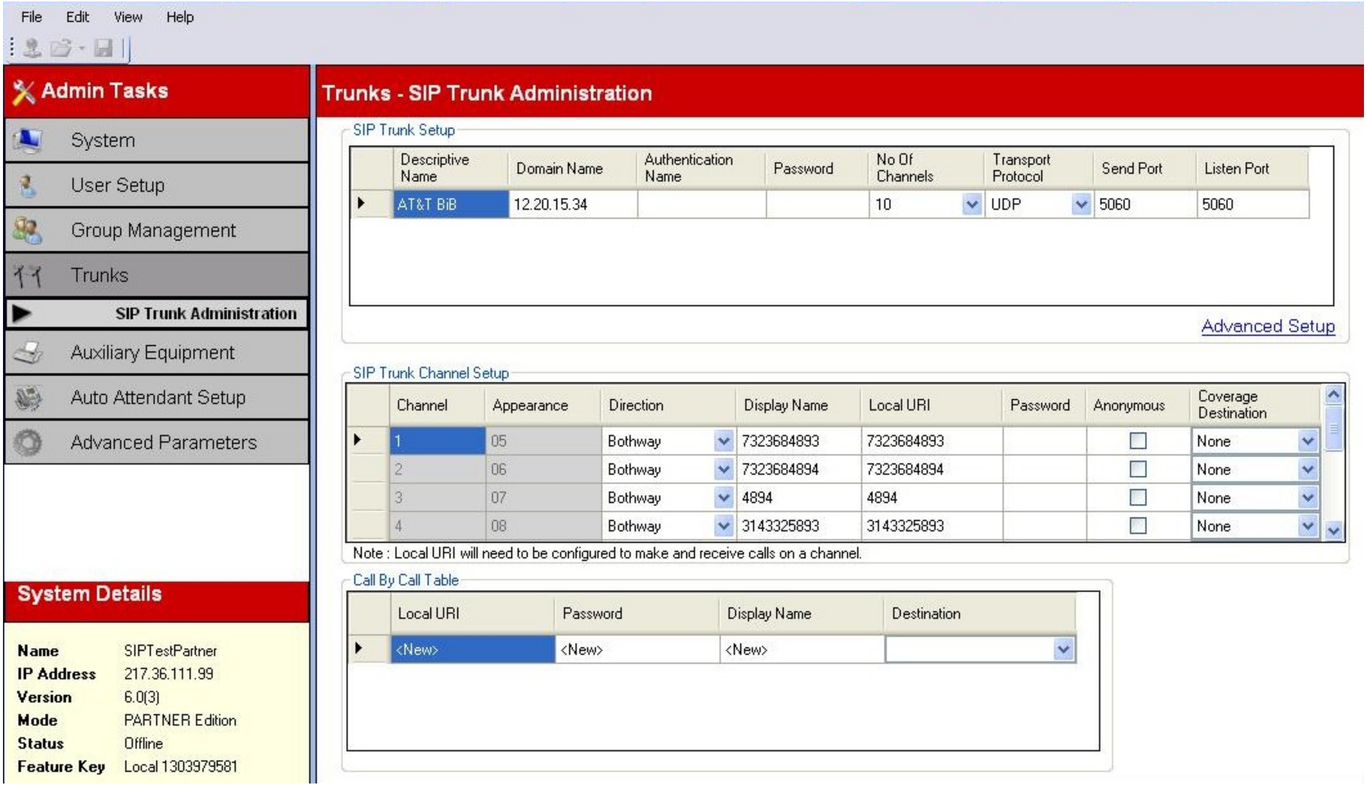
This section deals with the IP route configuration. The gateway IP address is the LAN side address of the AT&T managed router. Please contact AT&T customer care to get the correct address. (In PARTNER Version the WAN port is not enabled, only LAN port supported).



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

- Description Name: this is a free text entry to identify the trunk.
- Domain Name: Enter here the IP Address of 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
- Authentication Name and Password Name shall be left blank
- Number of channels: enter same number as number of licenses available.
- Send Port shall be set to 5060 (default)
- 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 five telephone numbers (10 digits) to configure with the following purpose:

Individual number: 732-123-1001

- **Incoming calls** are routed to a particular user.
- **Outgoing calls**: the user can dial out with this Caller Id.

Individual number, incoming calls only: 732-123-1002

- **Incoming calls** are routed to particular user.
- **Outgoing calls** using this DID are not possible.

Main line number to Auto Attendant: 732-123-1000

- **Incoming calls** are routed to an Auto-Attendant.
- **Outgoing calls**: users can dial out using this Caller Id.

Hunt Group number: 732-123-1003

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

Voicemail access number: 732-123-1004

- **Incoming calls** are routed to Voicemail so users can access their mailboxes externally.
- **Outgoing calls** using this DID are not possible.

5. Individual Number configuration

This example shows how to configure a SIP channel so calls to DID 732-123-1001 are routed to Extension 12. This is done in two steps:

- In *SIP Trunk Administration* configure a channel with DID 732-123-1001.
- In *User Setup, Button Programming*, configure a button for Extension 12 to the SIP channel/Line Appearance above.

5.1 Channel configuration for Individual Number

This section deals with the configuration of the SIP channel in the *Trunk – SIP Administration* screen.

SIP Trunk Channel Setup

Channel	Appearance	Direction	Display Name	Local URI	Password	Anonymous	Coverage Destination
1	05	Bothway	7321231001	7321231001		☐	12::

The fields on this form are described next.

- **Channel** indicates the channel number within the trunk. It is allocated by IP Office. Cannot be changed.
- **Appearance** indicates the Line Appearance number. It is allocated by IP Office. Cannot be changed.
- **Direction** indicates whether this channel is for incoming only or incoming and outgoing calls. Set it to Bothway.
- **Display Name** is used to populate the Display field of the From header for outgoing calls. Enter the DID for this channel, 7321231001.
- **Local URI** is used to populate the user part of the From header for outgoing calls. Enter the DID for this channel, 7321231001.
- **Password**. Leave it blank.
- **Anonymous** is used to hide the DID for outgoing calls. Do not select if you want the DID to be passed across the network.
- **Coverage Destination**. Select 12 from the drop down list so unanswered calls go to the Voicemail of Extension 12.

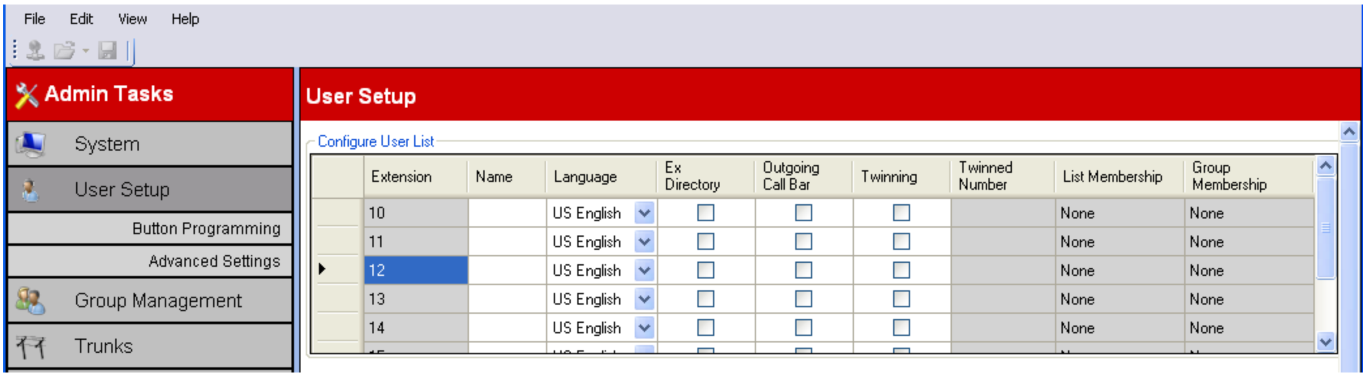
The value of the *Local URI* field is used in the From header of outgoing calls and also, for incoming calls, it is compared to the user part field of the To header to determine the channel selection. This example assume that the user part of the SIP URI in the R-URI field of the SIP INVITE arriving from the ATT network to IP Office is a 10 digit Telephone Number, 7321231001. 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, 7321231001.

5.2 Configure a button to a SIP channel/Line Appearance

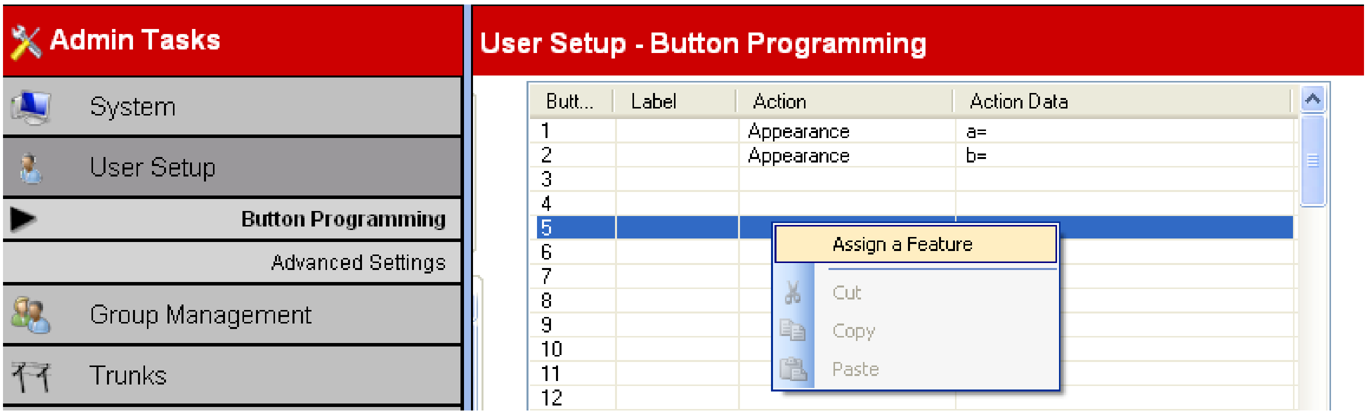
This section shows how to assign the DID configured above to a Button in Extension 12. By pressing this button the user will be able to make outgoing calls using DID 7321231001. Incoming calls to this DID will alert on this button and can be answered by pressing it.

Unanswered calls will be covered by the Voicemail of Extension 12.

First step is to go to the User Setup screen and select Extension 12.

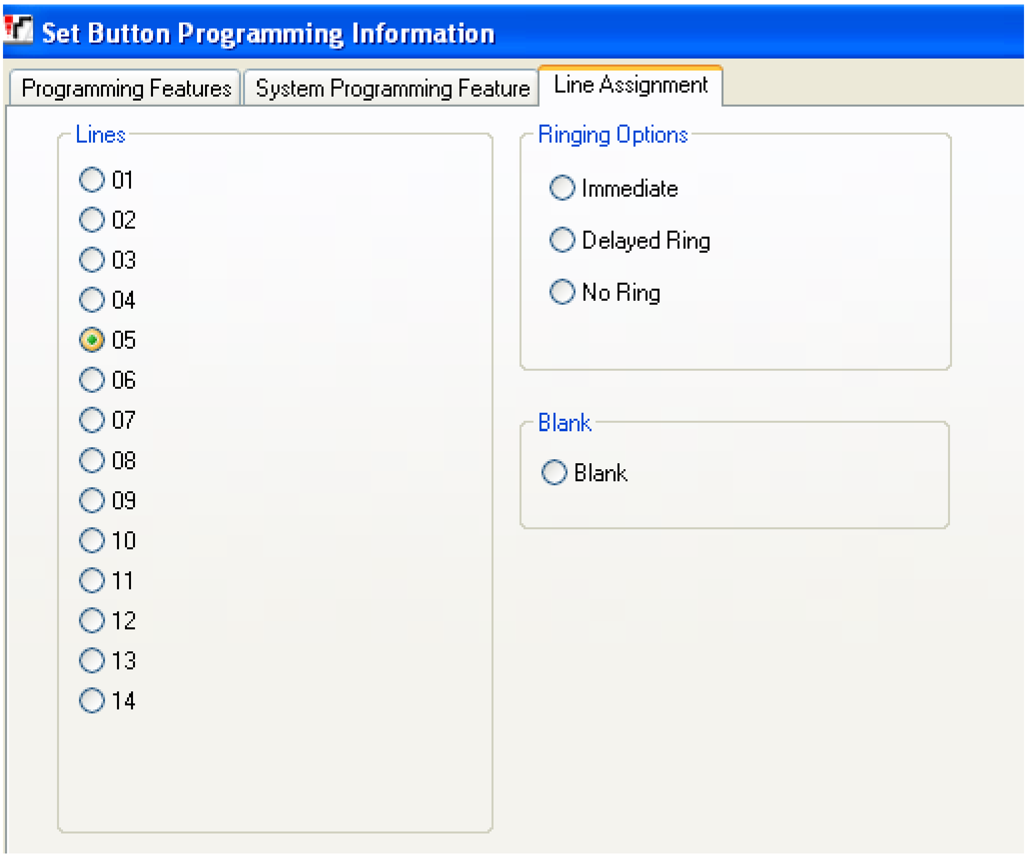


Then press on Button Programming to enter the button configuration screen. Once in that screen, select an available button by clicking anywhere on the line. See button 5 selected below. Then right click on the Action cell of that line and select Assign



Select the Line Assignment tab of the Set Button Programming Information screen. Select Line Appearance 05 to match that of the channel configured in the SIP Trunk Administration screen above.

5. Individual Number configuration



Click OK to close the window, click Apply and save the configuration.

No more steps required to configure the Individual Number for Extension 12.

5.3 Multiple calls

To receive more than one call on this number, create a new channel with this number as described above.

6. Individual Number, incoming calls only configuration

This example shows how to configure a SIP Trunk so calls to DID 732-123-1002 are routed to Extension 13 without using a Line Appearance. This user will not be able to make outgoing calls using that DID though. Configuration steps:

- Add a new SIP channel and point it to the Call-by-Call table.
- Add DID 732-123-1002 to the call-by-call table.

6.1 SIP Channel to call-by-call table

Go to the SIP Trunk administration screen and select Channel 2. Enter the values as in the picture below:

SIP Trunk Channel Setup

	Channel	Appearance	Direction	Display Name	Local URI	Password	Anonymous	Coverage Destination
	1	05	Bothway	7321231001	7321231001		☐	12::
▶	2		Incoming Call-...		<Not yet configured.>		☐	None

The fields on this form are described next.

- **Direction** indicates whether this channel is for incoming only or incoming and outgoing calls. Select Incoming Call-by-Call from the drop-down list.
- **Channel** indicates the channel number within the trunk. It is allocated by IP Office. Cannot be changed.
- **Appearance** indicates the Line Appearance number. No number assigned when Direction is set to Incoming Call-by-Call.
- **Display Name** is used to populate the Display field of the From header for outgoing calls. Leave it empty.
- **Local URI** is used to populate the user part of the From header for outgoing calls. Leave it empty.
- **Password**. Leave it blank.
- **Anonymous** is used to hide the DID for outgoing calls. Do not select if you want the DID to be passed across the network.
- **Coverage Destination**. Field is greyed out when Direction is set to Incoming Call-by-Call.

6.2 Add DID to Call-by-Call table

While in the SIP Trunk administration screen, select the first entry in the Call-by-Call table.

Call By Call Table

	Local URI	Password	Display Name	Destination
▶	7321231002	<New>	7321231002	13::

Configure the following:

- Local URI: Enter 7321231002
- Password: Leave it empty
- Display Name: enter 7321231002
- Destination: Select 13 from drop down list

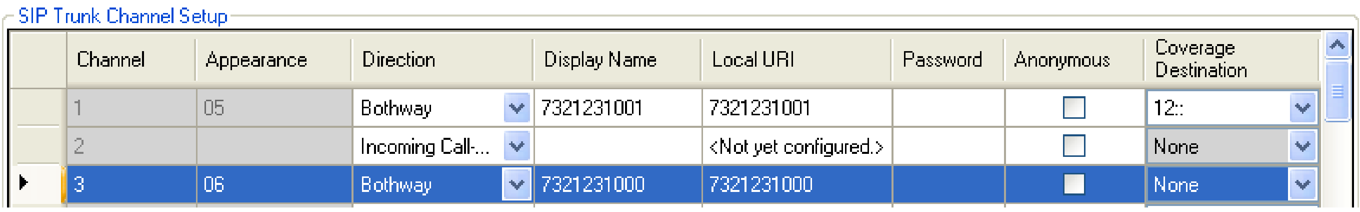
7. Calls to Auto Attendant

This example shows how to configure a SIP trunk so calls to DID 732-123-1000 are routed to the Auto Attendant. Steps:

- Add a SIP channel for DID 732-123-1000.
- Select a VMS schedule for Auto Attendant.

7.1 SIP channel for Auto Attendant's DID

In the SIP Trunk configuration screen, select Channel 3 and configure it like in the picture below:


SIP Trunk Channel Setup

Channel	Appearance	Direction	Display Name	Local URI	Password	Anonymous	Coverage Destination
1	05	Bothway	7321231001	7321231001			12::
2		Incoming Call-...		<Not yet configured.>			None
3	06	Bothway	7321231000	7321231000			None

Enter the following values:

- Channel: set by the system to 3, cannot be changed (value required for next step)
- Appearance: set by the system to 06, cannot be changed (value required for next step)
- Direction: Bothway
- Display Name: 7321231000
- Local URI: 7321231000
- Password: Leave it empty
- Anonymous: Leave it empty
- Coverage Destination: Select None from the Drop Down list

7.2 VMS Schedule for Auto Attendant

While in the SIP Trunk configuration screen, click on Advance Setup. In the new screen, inside the Channel Setup box, select Channel 3/ Line Appearance 06 (as per values given in previous step). Enter the values like in the picture below:

Channel Setup

Channel	Appearance	VMS Delay - Day	VMS Delay - Night	VMS Schedule
1	05	2	2	Never
3	06	0	0	Always

The fields on this form are:

- **Channel** Not configurable. Automatically assigned by system.
- **Appearance** Not configurable. Automatically assigned by system
- **VMS Delay - Day** is the number of rings before the call is sent to Auto Attendant during the Day time profile. Enter 0 so calls go to Auto Attendant immediately.
- **VMS Delay - Night** is the number of rings before the call is sent to Auto Attendant during the Night time profile. Enter 0 so calls go to Auto Attendant immediately.
- **VMS Schedule.** select Always from the drop down list. .

Note: The functionality of the Auto Attendant is configured in the Auto Attendant Setup screen. Its configuration is not covered in this guide.

7.3 Outgoing calls using Auto Attendant DID

Users can make SIP outgoing calls using the Auto Attendant DID by pressing the Intercom button, then pressing 8 followed by the SIP channel Line Appearance, in this example 06. At this point the user will receive dial tone.

8. Hunt Group number configuration

This example shows how to configure a SIP trunk so calls to DID 732-123-1003 are routed to Hunt Group 1. Steps:

- Add a SIP channel for DID 732-123-1003.
- Assign extensions to Hunt Group 1.

8.1 SIP channel for Hunt Group

In the SIP Trunk Administration screen, select Channel 4 and configure it like in the picture below:

SIP Trunk Channel Setup

Channel	Appearance	Direction	Display Name	Local URI	Password	Anonymous	Coverage Destination
1	05	Bothway	7321231001	7321231001		☐	12::
2		Incoming Call-...		<Not yet configured.>		☐	None
3	06	Bothway	7321231000	7321231000		☐	None
4	07	Bothway	7321231003	7321231003		☐	Hunt Group 1

Note : Local URI will need to be configured to make and receive calls on a channel

The fields on this form are described next.

- **Channel** indicates the channel number within the trunk. It is allocated by IP Office. Cannot be changed.
- **Appearance** indicates the Line Appearance number. No number assigned when Direction is set to Incoming Call-by-Call.
- **Direction** indicates whether this channel is for incoming only or incoming and outgoing calls. Select Bothway from the drop-down list.
- **Display Name** is used to populate the Display field of the From header for outgoing calls. Enter 7321231003.
- **Local URI** is used to populate the user part of the From header for outgoing calls. Enter 7321231003.
- **Password**. Leave it blank.
- **Anonymous** is used to hide the DID for outgoing calls. Do not select if you want the DID to be passed across the network.
- **Coverage Destination**. From the drop down list select Hunt Group 1.

Note: See the Group Management configuration screen for details about Hunt Group 1. Hunt Croup configuration is not covered in this guide.

8.2 Outgoing calls using Hunt Group DID

- Users can make SIP outgoing calls using the Auto Attendant DID by pressing the Intercom button, then pressing 8 followed by the SIP channel Line Appearance, in this example 07. At this point the user will receive dial tone.

9. Voicemail access number configuration

This example shows how to configure a SIP trunk so calls to DID 732-123-1004 are routed to Voicemails so users can access their mailboxes externally. Steps:

- Add a SIP channel for Voicemail access pointing to Incoming Call-By-Call table.
- Add entry for Voicemail in Call-By-Call table.

9.1 SIP channel for Voicemail access

In the SIP Trunk Administration screen, see channel 2 already configured for Incoming Call-By-Call in section 6.1. See picture below:

SIP Trunk Channel Setup

	Channel	Appearance	Direction	Display Name	Local URI	Password	Anonymous	Coverage Destination
	1	05	Bothway	7321231001	7321231001		☐	12::
▶	2		Incoming Call-...		<Not yet configured.>		☐	None

9.2 Entry for Voicemail in Call-By-Call table

While in the SIP Trunk administration screen, select the second entry in the Call-by-Call table.

Call By Call Table

	Local URI	Password	Display Name	Destination
▶	7321231002	<New>	7321231002	13::
	7321231004	<New>	7321231004	Voicemail

Configure the following:

- Local URI: Enter 7321231004
- Password: Leave it empty
- Display Name: enter 7321231004
- Destination: Voicemail

In order to access Voicemail externally, users must have a Voicemail code set.

10. 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
```