

**Asterisk Business Edition C.1.6.2  
Customer Configuration Guide (CCG)  
For Use with  
AT&T IP Flexible Reach**

**Issue 1.0  
09/23/2008**

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## 1. Introduction

AT&T IP Flexible Reach provides SIP trunking of IP-PBXs. This document provides a configuration guide to assist Asterisk Business Edition C.1.6.2 administrators in connecting to AT&T IP Flexible Reach.

Traditional PSTN overflow as well as the individual station programming for Asterisk is included in the Appendices.

AT&T has tested Asterisk Business Edition C.1.6.2 with IP Flex Reach service in two network arrangements:

- An IP-PBX connecting to the PSTN for inbound and outbound traffic
- An IP-PBX as part of a private corporate network aka site to site.

This testing is functional in nature, typically one call at a time. **No load testing was performed.**

## 2. Special Notes

### 2.1. 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.

Asterisk Business Edition C.1.6.2 will be referred to as "ABE C.1.6.2" through the rest of the document.

### 2.2. Asterisk Setup and Programming

AT&T does not provide installation or configuration support for the Asterisk Business Edition. Support for Asterisk Business Edition is available from your Digium reseller or directly from Digium in accordance with the terms of the Digium Asterisk Business Edition subscription. For more information or configuration assistance, please contact your Digium reseller or call **Digium Technical Support at +1 256 428 6000**. Please have your Asterisk Business Edition subscription key available for expedited assistance.

## 2.3.Asterisk Operational Caveats

The Asterisk feature set does not include:

- G.729 Annex B (Voice Activity Detection, Comfort Noise) is not supported by Asterisk. The G.729A codec is supported and requires a license. The license can be purchased at [www.digium.com](http://www.digium.com) .

AT&T's certification testing of ABE C.1.6.2 has identified several operational caveats. These include:

1. Sending video codecs in the SIP messaging will create call failures. Make sure that video is not enabled as part of the signaling to the AT&T network.
2. When interacting with traditional voicemail, auto-attendant or similar application, repeated, rapid entry of an identical digit on a telephone keypad may result in lost digits. This caveat does **NOT** apply to operations such as dialing a telephone number to initiate a call.
3. Ringback is not available on blind transfer scenarios. The call rings at the correct station and completes as desired but no ringback is heard by the party being transferred.
4. The second station in a three-way calling arrangement, where one leg is an IP conferencing platform, will lose the ability to enter DTMF digits.
5. T.38 Fax is not supported. T.38 testing produced mixed results. Successful T.38 fax transmission was very dependent on the specific Analog Telephone Adapter (ATA). (The most consistent performance was obtained with an ATA that is now manufacturer discontinued.) See the Appendix E for a summary of the tested ATAs.
6. Fax calls from the ABE C.1.6.2 to a fax machine, where the **destination** fax machine is a super G3 fax machine **and** in the PSTN **and** in an AT&T Hosted Internet Protocol Communication Service (HIPCS) local serving area, will fail.
7. Placing station-to-station calls via ten digit dialing (routed through the AT&T network) to the same Asterisk server requires the parameter "directrtpsetup = yes" in sip.conf. AT&T notes that the directrtpsetup

- parameter is considered experimental. (This option sets up the call using the originating device's IP address so a re-INVITE is not invoked there.)
8. Only SIP trunking was tested. The H.323 protocol was not tested. The AT&T network does not support the Inter-Asterisk eXchange IAX protocol.
  9. Use of the SIP method UPDATE is not supported. Do **not** set "canreinvite=update" in sip.conf.
  10. Load testing was NOT performed. AT&T certification only addresses functional testing.

### 3. Certification Overview

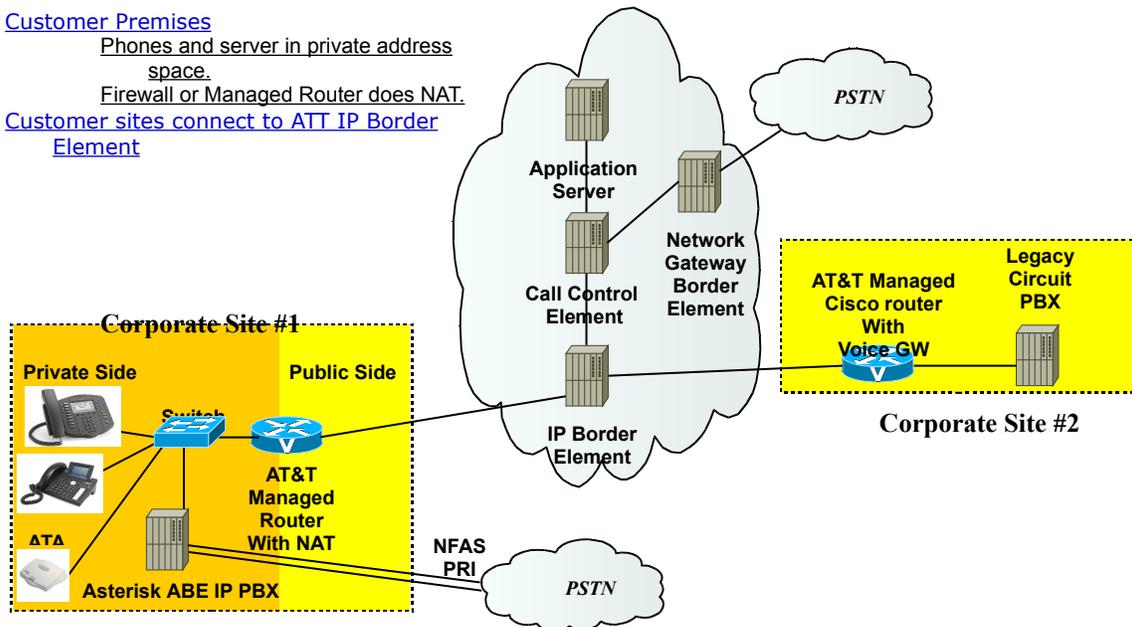
This section provides a service overview of the Asterisk Business Edition C.1.6.2 integration with AT&T IP Flexible Reach.

#### Customer Premises

Phones and server in private address space.

Firewall or Managed Router does NAT.

#### Customer sites connect to ATT IP Border Element



The test ABE C.1.6.2 customer premises consisted of the following components.

- **Polycom 600 IP phones** – These phones use the SIP protocol to communicate to the Asterisk IP PBX for call feature and routing support. These phones support Power over Ethernet (PoE) as well as Cisco in-line power and are powered accordingly. These phones used firmware version 3.0.1.0032.
- **Snom 360 IP phone** – These phones use the SIP protocol to communicate to the Asterisk IP PBX for call feature and routing support. These phones support Power over Ethernet (PoE) and are powered accordingly. This phone used firmware version 6.5.17.
- **Mediatrix 1102** – An ATA using the SIP protocol to communicate to the Asterisk IP PBX for call feature and routing support. This ATA provides an endpoint for T.38 fax but that feature was disabled. This ATA was using firmware version 5.0.23.153.
- **Fax machine** – These machines have an RJ-11 standard analog interface. The Mediatrix ATA was tested as a SIP adaptor for the analog interface.

- **Asterisk Business Edition C.1.6.2** – This is the central component of this IP-PBX.
  - Server – An IBM eServer running Linux CentOS 2.6.18-53.1.14.el5.smp.
  - PRI Interface – Digium TE410 PCI card. Supporting 4 T1s, this board was configured in a Non Facility Associated Signaling (NFAS) Primary Rate Interface (PRI) mode. This T1 voice card is used to connect to the local PSTN. This interface was only tested as a backup/alternative route for PSTN connectivity.
  - Codec support included G.711 mu law and G.729A. (The G.729A codec is not included with Asterisk Version 1.4.15 and must be purchased separately.) G.729B (Voice Activity Detection (VAD) and Comfort Noise) are **NOT** supported by ABE C.1.6.2.
- **AT&T Managed Router** – This is the router is provided and managed by AT&T.

The following routing scenarios are supported by the ABE C.1.6.2 and **DO NOT** use the AT&T Call Control. These scenarios shall use the G.711 or G.729A codecs.

- Intra ABE C.1.6.2 phone calls i.e. local phone to local phone

The following routing scenarios are supported by the ABE C.1.6.2 and **DO** use the AT&T Call Control. For voice calls, the G.729A codec is the first choice with G.711 mu law being a second offering.

- ABE C.1.6.2 phones to PSTN (domestic US and international).
- ABE C.1.6.2 phones to legacy PBX site via a Cisco gateway.
- Legacy PBX site via a Cisco gateway to Asterisk IP PBX phones.
- ABE C.1.6.2 phones at one ABE C.1.6.2 IP PBX site to ABE C.1.6.2 phones at another IP PBX site.

If the customer has subscribed to the AT&T IP Local service, then the following routing scenarios are supported by the ABE C.1.6.2 IP PBX and **DO** use the AT&T Call Control. For voice calls, the G.729A codec is the first choice with G.711 mu law being a second offering.

- Inbound PSTN to Asterisk IP PBX phone
- Outbound local PSTN calls from the Asterisk IP PBX phones.
- Outbound local N11 (i.e. 411, 911) calls from the Asterisk IP PBX phones.

## 4. How to Prepare for IP Flex Reach Service Turn-up

Preparation is crucial to a timely and successful implementation of IP Flex Reach. **AT&T expects that the customer or their designated representative is knowledgeable in Asterisk administration. The ABE C.1.6.2 software should have been installed and properly operating prior to the circuit turn-up.**

### 4.1. Information from Customer to AT&T

AT&T will interview the customer prior to the installation. In addition to the normal logistics, the ABE C.1.6.2 administrator should be able to provide:

- **IP address of the ABE C.1.6.2 server.** Be sure to tell AT&T the address of the IP interface on the Asterisk server that is designated to be connected to the AT&T managed router. This address is called the signaling address in AT&T terms.

Note: The customer can get an AT&T IP address (begins with 12.) for their Asterisk or they can use one of their own.

- **IP address range of any IP devices (e.g. SIP phones, ATAs, etc.)** expected to send RTP directly to the AT&T network. (This means that you will have "canreinvite=yes" in profiles for these devices.) If you plan to run the RTP through the ABE C.1.6.2 server (not recommended), these address ranges are not needed since the media IP address is the ABE C.1.6.2 server's IP address.

The customer should evaluate their desired dial plan. Two types of Direct Inward Dial (DIDs) numbers are available with AT&T IP Flex Reach: virtual Telephone Numbers (TNs) or non-virtual TNs.

- A virtual TN has an NPA that is different from the NPA nominally associated with the terminating customer site. In this case, AT&T will pass 10 digits to the ABE C.1.6.2. For example, if an ABE C.1.6.2 associated telephone is assigned a virtual TN, the number received from AT&T would be 10 digits (e.g. 732-320-4041).
- A non-virtual TN has same NPA as nominally associated customer site. For a non virtual TN, AT&T will pass the length of the phone extension plus some prefix (called steering digits) if needed (typically a 4 digit extension without a prefix). If an ABE C.1.6.2 telephone is associated with a non virtual TN, the number received from AT&T would be 4 digits (i.e. 4041 for a 732-320-4041 TN).

The customer should communicate their dial plan requirements to AT&T so that the proper numbers are assigned.

#### **4.2.Information to Customer from AT&T**

Sometime between the original interview and the turn-up, the customer should receive an email from AT&T containing the **IP addresses of two IP Border Elements, commonly called "I-P-B-Es"**. (An IPBE is called a Session Border Controller (SBC) in the general literature.) These addresses should be of the form 12.x.y.z where x,y,z represent the rest of the IP address. These addresses need to be included in your sip.conf configuration. Be sure to note the city associated with each address as a comment in your sip.conf file. **A successful implementation is not possible without these addresses.**

AT&T will provide the customer with the actual set of Direct Inward Dial (DID) numbers, prior to turn-up. These numbers must be assigned in the Asterisk's extensions.conf file, prior to service turn-up. Be sure to confirm the number of digits to be received from the AT&T network.

AT&T expects the customer to prepend a 1 (send 1+10 digits) to the AT&T network for domestic calls. The code "011" should be prepended (send 011 + digits) for international calls.

Originating calls to AT&T must have the calling party number (CPN) set to a 10 digit number regardless of the type of TN associated with the originating phone. This 10 digit AT&T TN is nominally set as part of the "callerid" parameter in the Asterisk's sip.conf file.

### **5. Configuration Guide**

This configuration guide specifies the Asterisk configuration files that are modified to support the AT&T IP Flexible Reach SIP trunking. **AT&T expects that the customer or their designated representative is knowledgeable in Asterisk administration. The ABE C.1.6.2 software should have been installed and properly operating prior to the circuit turn-up. AT&T will NOT provide information or guidance on any Asterisk C.1.6.2 programming not related to SIP trunking.**

The configuration files for SIP trunk programming are nominally found in the /etc/asterisk/ directory on the Asterisk server. (If the RealTime configuration is

used, many of these variables are found in the database.) Configuration modifications are performed using a text editor, vi, in this case.

The database configuration was not tested.

System control, status monitoring and high level debugging is performed through the Command Line Interface (CLI). The CLI is invoked by entering:

**asterisk -vvvrgc** at the root Linux command prompt. The results are displayed as follows:

```
[root@vicksburg ~]# asterisk -vvvrgc
```

```
Asterisk Business Edition C.1.6.2, Copyright (C) 1999 - 2008 Digium, Inc. and others.
```

```
Created by Mark Spencer
```

```
Thank you for using Business Edition. This Software is provided by Digium Inc under license. Please refer to the license agreement provided with the Software.
```

```
=====
```

```
== Parsing '/etc/asterisk/asterisk.conf': Found  
== Asterisk Business Edition Host-ID: 00:00:00:00  
== Found license "ABE123" providing 120 calls  
== Found total of 120 Business Edition calls  
== Parsing '/etc/asterisk/extconfig.conf': Found
```

```
Connected to Asterisk C.1.6.2 currently running on vicksburg (pid = 2684)
```

```
Verbosity is at least 3
```

```
vicksburg*CLI>
```

**Having the CLI open during service turn-up is very useful for diagnosing most common problems.**

### **5.1.Asterisk Version and Feature Requirements**

The Asterisk Business Edition C.1.6.2 is certified. The underlying Linux Operating System was CentOS 4.4 SMP with the latest patches applied.

The Asterisk software version can be verified by running the **show version** command from the CLI. The results are displayed as follows:

```
vicksburg*CLI> core show version
```

```
Asterisk C.1.6.2 built by root @ localhost.localdomain on a i686 running Linux on 2008-03-14 10:49:08 UTC
```

```
vicksburg*CLI>
```

The Asterisk CLI also provides a debugging interface, which is invoked by entering:

```
vicksburg*CLI> sip set debug  
SIP Debugging enabled
```

The debugging is disabled by entering:

```
vicksburg*CLI> sip set debug off  
SIP Debugging Disabled  
vicksburg*CLI>
```

In general, the SIP debugging mode should be off.

The G.729A codec licenses are not included with the Asterisk package. These licenses can be purchased separately, through the Digium, Inc. web site, [www.digium.com](http://www.digium.com). Note that installation of G.729A licenses requires that the server have access to the Internet. After the licenses are installed, the server can be moved to a private address space without Internet access.

To status the G.729A codecs, enter `show g729` at the CLI. The response should be similar to the following:

```
vicksburg*CLI> show g729  
0/0 encoders/decoders of 5 licensed channels are currently in use  
vicksburg*CLI>
```

This case show 5 licensed channels with none currently in use.

## 5.2.Asterisk Configuration Files

Asterisk configuration was performed with a series of "flat" files in the `/etc/asterisk/` directory. Note: ";" in the first column is used to designate a "comment" line. Significant settings are highlighted with yellow background.

The significant files for configuring Asterisk for IP Flex Reach are:

1. `extensions.conf` – sets dial plan
2. `rtp.conf` – sets RTP port numbers to be used
3. `sip.conf` – configures various SIP devices and AT&T's IPBEs

Some additional configuration files are included in the appendices for reference. These are:

1. zaptel.conf – Configures TDM PSTN basic DS1 parameters – Appendix A
2. zapata.conf – Configures TDM PSTN Q.921 and Q.931 PRI parameters (located in the /etc directory) – Appendix A
3. indications.conf – Sets US indications – Appendix B
4. modules.conf – Controls loading of functional modules – Appendix B

**Asterisk “contexts” are not specifically covered although some examples are included below. Be careful to include your contexts appropriately. Failure to do so may prevent successful call completion.**

### 5.1.1 extensions.conf

extensions.conf is used to set the dialing plan and all call routing. Call restrictions and calling capabilities are assigned in this file. Configuration shown here assigns the IPBEs to the variables SIPTRUNK1 and SIPTRUNK2. These variables are applied in the international, long distance, local and toll-free dialing commands.

For reference, an auto-attendant, voicemail retrieval, as well as the tested dial commands for the phones are included. Note that the extensions (four digit example for station-to-station dialing) and 10 digit numbers (DIDs) are specifically addressed. If the dialing plan features typical of IP Phones are used, this arrangement can be collapsed to the 10 digit only numbers.

The following **example** uses “mt\_vicksburg” as the context for the SIP phones. Three IP phones (polycom, polycom2 and snom) are configured as extensions 4041, 4042 and 4043 respectively. A single Mediatrix ATA (device 8177) is assigned to 732 320-4044. The voicemail retrieval extension is (732) 320-4045 while the auto-attendant number is (732) 320-4040.

Only a portion of the extensions.conf file is illustrated here.

```
; extensions.conf - the Asterisk dial plan
;
;
[globals]
;
SIPTRUNK1=SIP/first_gateway           ; SIP IPBE #1 Trunk interface
SIPTRUNK2=SIP/second_gateway         ; SIP IPBE #2 Trunk interface
ISDNTRUNK=Zap/G13                     ;Trunk interface
;
```

```
TRUNKMSD=0 ; MSD digits to strip (usually 1 or 0)
[trunkint]
```

```
;  
; International long distance through trunk  
;
```

```
exten => _011.,1,Dial(${SIPTRUNK1}/${EXTEN:${TRUNKMSD}})
exten => _011.,2,Dial(${SIPTRUNK2}/${EXTEN:${TRUNKMSD}})
```

```
[trunkld]
```

```
;  
; Long distance context accessed through trunk  
;
```

```
exten => _1NXXNXXXXXX,1,Dial(${SIPTRUNK1}/${EXTEN:${TRUNKMSD}})
exten => _1NXXNXXXXXX,2,Dial(${SIPTRUNK2}/${EXTEN:${TRUNKMSD}})
exten => _1NXXNXXXXXX,3,Dial(${ISDNTRUNK}/${EXTEN:0},30)
```

```
[trunklocal]
```

```
;  
; Local seven-digit dialing for the 732 area code accessed through trunk interface  
;
```

```
exten => _NXXXXXX,1,Dial(${SIPTRUNK1}/1732${EXTEN:${TRUNKMSD}})
exten => _NXXXXXX,2,Dial(${SIPTRUNK2}/1732${EXTEN:${TRUNKMSD}})
```

```
[trunktollfree]
```

```
;  
; Long distance context accessed through trunk interface  
;
```

```
exten => _1800NXXXXXX,1,Dial(${SIPTRUNK1}/${EXTEN:${TRUNKMSD}})
exten => _1800NXXXXXX,2,Dial(${SIPTRUNK2}/${EXTEN:${TRUNKMSD}})
```

```
exten => _1888NXXXXXX,1,Dial(${SIPTRUNK1}/${EXTEN:${TRUNKMSD}})
exten => _1888NXXXXXX,2,Dial(${SIPTRUNK2}/${EXTEN:${TRUNKMSD}})
```

```
exten => _1877NXXXXXX,1,Dial(${SIPTRUNK1}/${EXTEN:${TRUNKMSD}})
exten => _1877NXXXXXX,2,Dial(${SIPTRUNK2}/${EXTEN:${TRUNKMSD}})
```

```
exten => _1866NXXXXXX,1,Dial(${SIPTRUNK1}/${EXTEN:${TRUNKMSD}})
exten => _1866NXXXXXX,2,Dial(${SIPTRUNK2}/${EXTEN:${TRUNKMSD}})
```

```
[privatetrunkld]
```

```
;  
; Long distance context accessed anonymously through trunk  
;
```

```
exten => _11NXXNXXXXXX,1,SIPAddHeader(P-Preferred-
Identity:"${CALLERID(name)}"<sip\:${CALLERID(num)}\@172.16.4.5\5060\>)
exten => _11NXXNXXXXXX,n,SIPAddHeader(Privacy: id)
exten => _11NXXNXXXXXX,n,SetCallerID(Anonymous <restricted>,a)
exten => _11NXXNXXXXXX,n,Dial(${TRUNK}/${EXTEN:1})
;
; Arrange contexts as needed
;
[international]
;
; Master context for international long distance
;
include => longdistance
include => trunkint

[longdistance]
;
; Master context for long distance
;
include => local
include => trunkld
include => privatetrunkld

[local]
;
; Master context for local, toll-free, and stations
;
include => trunklocal
include => mt_vicksburg
include => trunktollfree
include => pri_from_als

[mainmenuVOIP]
include => mt_vicksburg ; Note include of SIP phone context
;
; Test VOIP ATT Auto-attendant - Auto Attendant "main menu" context with
submenus
;
exten => 7323204040,1,Answer
exten => 7323204040,2,Background(/var/lib/asterisk/sounds/directory)
exten => 7323204040,3,goto(Auto-Attendant,startwork,1)
exten => i,1,Hangup

[Auto-Attendant]
```

```
exten => startwork,1,Background(/var/lib/asterisk/sounds/enter-ext-of-person)
;
exten => startwork,2,WaitExten
exten => 1,1,Macro(stdexten,4041,SIP/polycom) ; "Call polycom"
exten => 2,1,Macro(stdexten, 4042,SIP/polycom2) ; "Call polycom2"
exten => 3,1,Macro(stdexten, 4043,SIP/snom) ;"Call snom"
exten => 4,1,Meetme,8600|M
exten => 5,1,Directory(mt_vicksburg)
exten => t,1,Dial(SIP/polycom,30,r) ; "Call default"
exten => i,1,Background,/var/lib/asterisk/sounds/num-not-in-db
exten => i,2,Hangup
```

[mt\_vicksburg]

```
include => longdistance
include => mainmenuVOIP
include => local
```

```
exten => 7323204045,1,VoicemailMain() ; General Voicemail pickup
exten => 7323204045,2,Hangup
```

```
exten => 4045,1,VoicemailMain(())@mt_vicksburg ; General Voicemail
pickup
; ; External DID
exten => 4045,2,Hangup
;
; Add local extensions to dial plan
;
exten => 4041,1,Macro(stdexten,4041,SIP/polycom) ; Polycom 600
exten => 4042,1,Macro(stdexten,4042,SIP/polycom2) ; Polycom 600
exten => 4043,1,Macro(stdexten,4043,SIP/snom) ; Polycom 600
;
; Add external numbers to dial plan
;
exten => 7323204041,1,Macro(stdexten,4041,SIP/polycom) ; Polycom 600
exten => 7323204042,1,Macro(stdexten,4042,SIP/polycom2) ; Polycom 600
exten => 7323204043,1,Macro(stdexten,4043,SIP/snom)
```

```
exten => 7323204044,1,Dial(SIP/8177,30) ; "FAX Line "
```

### 5.1.2 rtp.conf

This rtp.conf file has been modified to include the rtp ports used in the AT&T service.

```
;  
; RTP Configuration  
;  
[general]  
;  
; RTP start and RTP end configure start and end addresses  
;  
rtpstart=16384  
rtpend=32767  
;  
; Whether to enable or disable UDP checksums on RTP traffic  
;  
;rtpchecksums=no  
;  
; The amount of time a DTMF digit with no 'end' marker should be  
; allowed to continue (in 'samples', 1/8000 of a second)  
;  
;dtmftimeout=3000
```

### 5.1.3 sip.conf

This file assigns the IPBEs to variables (`first_gateway` and `second_gateway`) and assigns a SIP PING (NOTIFY Message) to be sent on a regular interval (2000 msec in the example). Other data like the IP address assignment is shown as well. Only an abbreviated version of the sip.conf file is provided here.

```
sip.conf  
;  
; SIP Configuration example for Asterisk with AT&T services  
;  
[general]  
context=pri_from_als ; Default context for incoming calls  
  
allowoverlap=yes  
  
realm=asterisk ; Realm for digest authentication  
; defaults to "asterisk"  
; Realms MUST be globally unique according to  
; RFC 3261  
; Set this to your host name or domain name  
bindport=5060 ; UDP Port to bind to (SIP standard port is 5060)  
  
bindaddr=172.16.4.5 ; IP address to bind to (0.0.0.0 binds to all)  
srvlookup=no ; Enable DNS SRV lookups on outbound calls
```

; Note: Asterisk only uses the first host  
; in SRV records  
; Disabling DNS SRV lookups disables the  
; ability to place SIP calls based on domain  
; names to some other SIP users on the Internet

**videosupport=no** ; Turn on support for SIP video

**disallow=all** ; First disallow all codecs  
**allow=g729** ; Allow codecs in order of preference  
**allow=ulaw**

;  
**musicclass=sip** ; Sets the default music on hold class for all SIP calls  
; This may also be set for individual users/peers  
**language=en** ; Default language setting for all users/peers  
; This may also be set for individual users/peers

**sendrpid = no** ; If Remote-Party-ID should be sent

**useragent=Vicksburg ABE PBX** ; Allows you to change the user agent string

**usereqphone = yes** ; If yes, ";user=phone" is added to uri that contains  
; a valid phone number

**dtmfmode = rfc2833** ; Set default dtmfmode for sending DTMF. Default.

;  
;----- T.38 FAX PASSTHROUGH SUPPORT -----  
;  
; This setting is available here, the [general] section, and in device configurations.  
; Setting this to yes, enables T.38 fax (UDPTL) passthrough on SIP to SIP calls, if  
; both parties have T.38 support enabled in their Asterisk configuration  
; This has to be enabled in the general section for all devices to work. You can  
; then disable it on a per device basis.

;  
; T.38 faxing only works in SIP to SIP calls, with no local or agent channel being  
used.

;  
**t38pt\_udptl = no** ; Asterisk T.38 not officially supported  
**t38pt\_rtp=no**  
**t38pt\_tcp=no**

;  
;----- MEDIA HANDLING -----  
; By default, Asterisk tries to re-invite the audio to an optimal path. If there's

```
; no reason for Asterisk to stay in the media path, the media will be redirected.  
; This does not really work with in the case where Asterisk is outside and have  
; clients on the inside of a NAT. In that case, you want to set canreinvite=nonat  
;
```

```
canreinvite=yes ; Asterisk by default tries to redirect the  
; RTP media stream (audio) to go directly from  
; the caller to the callee. Some devices do not  
; support this (especially if one of them is behind a NAT).  
; The default setting is YES. If you have all clients  
; behind a NAT, or for some other reason wants Asterisk to  
; stay in the audio path, you may want to turn this off.
```

```
; In Asterisk 1.4 this setting also affect direct RTP  
; at call setup (a new feature in 1.4 - setting up the  
; call directly between the endpoints instead of sending  
; a re-INVITE).
```

```
directrtpsetup=yes ; Enable the new experimental direct RTP setup. This sets up  
; the call directly with media peer-2-peer without re-invites.  
; Will not work for video and cases where the callee sends  
; RTP payloads and fntp headers in the 200 OK that does  
; not match the callers INVITE.
```

```
;  
; Heartbeat to AT&T gateways, one entry per IPBE  
;
```

```
[first_gateway]  
type=peer  
host=135.25.29.74 ; First AT&T IP Border Element (IPBE) Address  
qualify=2000 ; SIP Ping Frequency in msec.
```

```
;  
[second_gateway]  
type=peer  
host=135.25.29.1 ; Second AT&T IP Border Element (IPBE) Address  
qualify=2000 ; SIP Ping Frequency in msec.
```

```
;  
Note that these hosts are unsecured. All access to these IPBEs is strictly limited  
to assigned AT&T subscribers.
```

```
***** NOTE IP Addresses above are only EXAMPLES! An AT&T  
representative will provide the correct addresses for your circuit.  
*****
```

## 6. Troubleshooting

This section provides some tips about troubleshooting problems

- **Asterisk Command Line Interface (CLI)**  
The Asterisk Command Line Interface (CLI) is used to view the call flow as well as provide an overall debugging interface. From a terminal screen, logged in as root, enter **asterisk -vvvrgc** General debugging should be performed from this interface. Typing a "?" at the CLI prompt will show all commands. The TAB key can be used to complete the partial entry of any command
- **SIP debugging - sip debug**  
Detailed SIP message debugging is enabled at the CLI with the command: **sip debug** . The command: **sip no debug** will disable the debugging.
- **Status of SIP peers and friends**  
The status of all peers can be show via the CLI with the command: **sip show peers**

Some of the typical problems include:

- **None of my stations are registered** – Enable the sip debug as shown above.
  - If REGISTER messages are reaching the Asterisk, correct the device configuration or Asterisk peer profile.
  - If REGISTER messages are not reaching the Asterisk, check to see if the iptables and ip6tables are disabled and the services are not running. These firewalls may be enabled later and configured to pass the appropriate SIP / RTP traffic.
- **Calls from the AT&T Network aren't reaching the Asterisk** - Check to see if the iptables and ip6tables are disabled and the services are not running. These firewalls may be enabled later and set to pass the appropriate SIP / RTP traffic.
- **Calls seem to connect, but there is no audio** – Set the parameters **canreinvite = no** and **directrtpsetup = no** in sip.conf. Reload the Asterisk to make these settings active. This will force the RTP through the Asterisk server. Place another call. If the audio path is successfully established, an issue with NAT may exist within the customer's network.

- **Incoming calls don't a ring device but I see it fail on the CLI** – Review the CLI output carefully. Check to see that the called digits exactly match the extension of a device that is registered with your Asterisk. If the digits match, carefully check your context "include" statements.
- **Poor voice quality** – Check the Asterisk's ethernet interface settings to verify that they are at least 100 Mb, Full Duplex.

**Unable heartbeat (SIP Options) with IPBEs** – If the ABE Server has multiple ethernet interfaces, static routes will be required to insure that traffic for the IPBE is directed to the correct interface. These routes can be added at the Linux command line. Once the routes are working, adding them permanently can be done in the `/etc/rc.local` Linux file.

A good reference for general Asterisk information and programming is:

Van Meggelen, J., Madsen, L. and Smith, J. **Asterisk, The Future of Telephony, 2<sup>nd</sup> Edition**, O'Reilly Media, Inc 2007 **ISBN-10: 0-596-51048-9, ISBN-13: 978-0-596-51048-0**

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## APPENDICES

These appendices address several aspects of the Asterisk configuration that are not directly related to configuring for the AT&T IP-Flex Reach service. An Asterisk administrator maybe asked to include these capabilities. These files are provided as working examples.

### A. Appendix - PSTN / Primary Rate Interface (PRI) Overflow

The Asterisk server can be outfitted with analog or digital interfaces to the traditional PSTN. The configuration files, `/etc/zaptel.conf` and `/etc/asterisk/zapata.conf`, set the appropriate parameters.

This particular configuration provides for a three T1 Non Facility Associated Signaling (NFAS) Primary Rate Interface (PRI). This PRI has a primary D-channel in channel 24 on the first T1 (Facility ID 0) and a backup D-Channel in channel 48 (on the second T1 (Facility ID 1). The third T1 does not contain a D-channel. All channels are arranged in trunkgroup labeled "26". Note that the D-Channel configuration and Facility ID were selected to interface to a 4ESS. Assignment of the trunkgroup number is arbitrary.

#### A.1.zaptel.conf

This configuration file sets the physical layer parameters (Layer 1) associated with any TDM hardware in the Asterisk server. The file is nominally located in the `/etc` directory.

The values below are only those pertinent to the configuration of PRI interfaces in the `/etc/zaptel.conf`. Additional commentary and parameters will be contained in the actual file.

```
#  
# Zaptel Configuration File  
#  
# This file is parsed by the Zaptel Configurator, ztcfg  
#  
#  
# First come the span definitions, in the format  
#  
# span=<span num>,<timing source>,<line build out>,<framing>,<coding>[,yellow]  
#
```

```
span=1,1,0,esf,b8zs
span=2,0,0,esf,b8zs
span=3,0,0,esf,b8zs
#
#
# Next come the definitions for using the channels. The format is:
# <device>=<channel list>
#
# The channel list is a comma-separated list of channels or ranges
#
bchan=1-23
dchan=24
bchan=25-27,49
dchan=48
#
# Finally, you can preload some tone zones
#
# You may also specify a default zone with "defaultzone=<zone>" where zone
# is a two letter country code.
#
loadzone = us
defaultzone=us
#
```

## **A.2.zapata.conf**

This configuration file sets the link and signaling layer parameters (Layers 2 and 3) associated with the PRI hardware in the Asterisk server. The file is nominally located in the `/etc/asterisk` directory.

The values below are only those pertinent to the configuration of PRI interfaces in the `/etc/asterisk/zapata.conf`. Additional commentary and parameters will be contained in the actual file.

```
;  
; Zapata telephony interface  
;  
; Configuration file  
;  
[trunkgroups]  
;  
; Trunk groups are used for NFAS or GR-303 connections.  
;  
; Group: Defines a trunk group.
```

```
;      group => <trunkgroup>,<dchannel>[,<backupdchannel>...]
;
trunkgroup => 26,24,48
;
; Spanmap: Associates a span with a trunk group
;      spanmap => <zapspan>,<trunkgroup>[,<interfaceID>]
;
spanmap => 1,26,0
spanmap => 2,26,1
spanmap => 3,26,2

[channels]
;
; Default language
;
language=en
;
; Default context
;
context=pri_from_als
;
; Switchtype: Only used for PRI.
;
;
switchtype=4ess
;;
; PRI Dialplan: Only RARELY used for PRI.
;;
pridialplan=national
;
; PRI Local Dialplan: Only RARELY used for PRI (sets the calling number's
numbering plan)
;
prilocaldialplan=national
;
; PRI resetinterval: sets the time in seconds between restart of unused
; channels, defaults to 3600
;
resetinterval = 14400
;
; PRI Out of band indications.
; Enable this to report Busy and Congestion on a PRI using out-of-band
; notification
;
```

```
;
priindication = outofband
;
; If you need to override the existing channels selection routine and force all
; PRI channels to be marked as exclusively selected, set this to yes.
priexclusive = yes
;
; ISDN Timers
; All of the ISDN timers and counters that are used are configurable.
;
prtimer => t200,1000
prtimer => t313,4000
;
; To enable transmission of facility-based ISDN supplementary services (such
; as caller name from CPE over facility), enable this option.
facilityenable = yes
;
; Signalling method (default is fxs). Valid values:
; pri_cpe:      PRI signalling, CPE side
; pri_net:      PRI signalling, Network side
;
;
; Whether or not to use caller ID
;
usecallerid=yes
;
; Whether or not use the caller ID presentation for the outgoing call that the
; calling switch is sending.
;
usecallingpres=yes
;
; Support Caller*ID on Call Waiting
;
callwaitingcallerid=yes
;
; Support three-way calling
;
threewaycalling=yes
;
; Support flash-hook call transfer (requires three way calling)
; Also enables call parking (overrides the 'canpark' parameter)
;
transfer=yes
;
```

```
; Allow call parking
; ('canpark=no' is overridden by 'transfer=yes')
;
canpark=yes
;
; Support call forward variable
;
cancallforward=yes
;
; Whether or not to support Call Return (*69)
;
callreturn=yes
; Enable echo cancellation
; Use either "yes", "no", or a power of two from 32 to 256 if you wish to
; actually set the number of taps of cancellation.
;
echocancel=yes
;
; Generally, it is not necessary (and in fact undesirable) to echo cancel when
; the circuit path is entirely TDM. You may, however, reverse this behavior
; by enabling the echo cancel during pure TDM bridging below.
;
echocancelwhenbridged=yes
;
; In some cases, the echo canceller doesn't train quickly enough and there
; is echo at the beginning of the call. Enabling echo training will cause
; asterisk to briefly mute the channel, send an impulse, and use the impulse
; response to pre-train the echo canceller so it can start out with a much
; closer idea of the actual echo. Value may be "yes", "no", or a number of
; milliseconds to delay before training (default = 400)
;
echotraining=yes
;echotraining=800
; Logical groups can be assigned to allow outgoing rollover. Groups range
; from 0 to 63, and multiple groups can be specified.
;
group=13
;
; CallerID can be set to "asreceived" or a specific number if you want to
; override it.
;
callerid=asreceived
;
; Few zones are supported at the time of this writing, but may be selected
```

```
; with "progzone"  
;  
; This feature can also easily detect false hangups. The symptoms of this is  
; being disconnected in the middle of a call for no reason.  
;  
callprogress=yes  
progzone=us  
;  
; For fax detection, uncomment one of the following lines. The default is *OFF*  
;  
faxdetect=both  
;faxdetect=incoming  
;faxdetect=outgoing  
;faxdetect=no  
;  
; This option specifies a preference for which music on hold.  
;  
mohinterpret=sip  
;  
; This option specifies which music on hold class to suggest to the peer channel  
;  
mohsuggest=sip  
;  
; Sample PRI (CPE) config: Specify the switchtype, the signalling as either  
; pri_cpe or pri_net for CPE or Network termination, and generally you will  
; want to create a single "group" for all channels of the PRI.  
;  
signalling = pri_cpe  
channel => 1-23,25-27,49
```

## B. Appendix - Other Asterisk Configuration Files

These configuration files are provided for completeness.

### B.1.indications.conf

This indications.conf file is configured to support US locations. All other countries have been removed.

```
;  
; indications.conf  
; Configuration file for location specific tone indications  
; used by the pbx_indications module.  
;  
[general]  
country=us          ; default location  
;  
[us]  
description = United States / North America  
ringcadance = 2000,4000  
dial = 350+440  
busy = 480+620/500,0/500  
ring = 440+480/2000,0/4000  
congestion = 480+620/250,0/250  
callwaiting = 440/300,0/10000  
dialrecall = !  
350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,350+440  
record = 1400/500,0/15000  
info = !950/330,!1400/330,!1800/330,0  
stutter = !  
350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,  
!0/100,!350+440/100,!0/100,!350+440/100,!0/100,350+440
```

### B.2.modules.conf

The modules.conf file shown here is the one specifically used for AT&T testing. The settings described here do not directly affect the AT&T interoperability. This file's content is included for completeness and may differ for a given installation.

```
;  
; Asterisk configuration file  
;  
; Module Loader configuration file
```

```
;  
  
[modules]  
autoload=yes  
;;  
noload => pbx_gtkconsole.so  
noload => pbx_kdeconsole.so  
; Not using DUNDI  
noload => pbx_dundi.so  
;  
; Intercom application is obsoleted by chan_oss. Don't load it.  
;  
noload => app_intercom.so  
;  
; Load either OSS or ALSA, not both  
; By default, load OSS only (automatically) and do not load ALSA  
;  
noload => chan_alsa.so  
;  
; Don't load MGCP, Skinny, ooh323 and iax2 because we are not using them  
;  
noload => chan_mgcp.so  
noload => chan_skinny.so  
noload => chan_ooh323.so  
noload => chan_iax2.so  
;  
noload => app_enumlookup.so  
;  
noload => res_config_pgsql.so  
noload => res_config_mysql.so  
noload => res_config_odbc.so  
noload => cdr_addon_mysql.so  
noload => cdr_pgsql.so  
noload => cdr_odbc.so  
;  
; Don't load unused codecs  
;  
noload => codec_alaw.so  
noload => codec_speex.so  
noload => codec_g726.so  
noload => codec_adpcm.so  
noload => codec_lpc10.so  
;
```

## C. Appendix – Telephone Configuration in sip.conf

The Polycom 600, Snom 360 and the Mediatrix 1102 were explicitly tested. Their respective profiles in /etc/asterisk/sip.conf are included as examples.

### **Snom 360, Firmware 6.5.17**

```
[snom]
type=friend
username=snom
secret=att
host=dynamic
context=mt_vicksburg
subscribecontext=mt_vicksburg ; Only allow SUBSCRIBE for local extensions
nat=no
canreinvite=yes ; Asterisk by default tries to redirect the
videosupport=no
callerid=snom * * <7323204043>
qualify=200 ; Consider it down if it's 1 second to reply
qualifysmoothing=yes
mailbox=4043@mt_vicksburg ; Set MWI to extension, not device
callingpres=allowed_not_screened ; control presentation
dtmfmode=rfc2833 ; Choices are inband, rfc2833, or info
```

### **Polycom 600, Firmware 3.0.1.0032**

```
[polycom]
type=friend
username=polycom ; Polycom 600
secret=att
callerid=polycom <7323204041>
context=mt_vicksburg
subscribecontext=mt_vicksburg ; Only allow SUBSCRIBE for local extensions
host=dynamic
nat=no
canreinvite=yes ; Asterisk by default tries to redirect the
videosupport=no
qualify=200 ; Qualify peer is no more than 200ms away
qualifysmoothing=yes
mailbox=4041@mt_vicksburg ; Set MWI
callingpres=allowed_not_screened ; control presentation
dtmfmode=rfc2833 ; Choices are inband, rfc2833, or info
```

## **Mediatrix 1102, Firmware 5.0.23.153**

```
[8177]
type=friend
username=8177
secret=att
callerid=mediatrix_01 <7323204044>
context=mt_vicksburg
subscribecontext=mt_vicksburg ; Only allow SUBSCRIBE for local extensions
host=dynamic
nat=no
canreinvite=yes ; Asterisk by default tries to redirect the
t38pt_udptl=no ; Enable T.38 fax support
videosupport=no
qualify=300 ; Consider it down if it's 1 second to reply
disallow=all
allow=ulaw
callingpres=allowed_not_screened ; control presentation
dtmfmode=rfc2833 ; Choices are inband, rfc2833, or info
```

```
[8178]
type=friend
username=8178
secret=att
callerid=mediatrix_02 <7323204044>
context=mt_vicksburg
subscribecontext=mt_vicksburg ; Only allow SUBSCRIBE for local extensions
host=dynamic
nat=no
canreinvite=yes ; Asterisk by default tries to redirect the
t38pt_udptl=no ; Enable T.38 fax support
videosupport=no
qualify=300 ; Consider it down if it's 1 second to reply
disallow=all
allow=ulaw
callingpres=allowed_not_screened ; control presentation
dtmfmode=rfc2833 ; Choices are inband, rfc2833, or info
```

## D. Appendix – Use of Other SIP Devices

One of the significant features of Asterisk is that a variety of phones and ATAs can be utilized. The Polycom 600 and Snom 360 were explicitly tested by AT&T in this guide. Use of other devices is done at the risk of the Asterisk administrator. Information regarding the experiences of other users can be found at:

<http://www.voip-info.org/wiki-Asterisk+phones>

Various VOIP phone vendors provide guides for using their equipment with Asterisk. This material may provide guidance but there is **no guarantee** the device will work properly with the AT&T network.

## E. Appendix - G.711 and T.38 Analog Terminal Adapters

Several Analog Terminal Adapters (ATA) were tested with ABE C.1.6.2. AT&T has had partial success with T.38 when using two of these devices. All devices tested successfully when G.711 was used for fax transmission.

- **Mediatrix 1102** – An ATA using the SIP protocol to communicate to the Asterisk IP PBX for call feature and routing support. This ATA provides an endpoint for T.38 fax. This ATA was using firmware version 5.0.23.153.
- **UTStarcom iAN-02EX** – An analog terminal adaptor (ATA) using the SIP protocol to communicate to the Asterisk IP PBX for call feature and routing support. This ATA provides an endpoint for T.38 fax. This ATA was using firmware version V4.8.2.50b. This device is now manufacturer discontinued.
- **Sipura 3000** – An ATA using the SIP protocol to communicate to the Asterisk IP PBX for call feature and routing support. This device does not support T.38 fax protocol. This ATA was using firmware version 2.0.13 (GWg). This device is now manufacturer discontinued.