Configuring the Cisco SPA8800 IP Telephony Gateway in an Asterisk® Environment

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Introduction
The Cisco® SPA8800 IP Telephony Gateway provides four RJ-11 FXS and four FXO ports, a 10/100BASE-T RJ-45 Ethernet interface to connect to either a router or multilayer switch, and an auxiliary port for local administration. It also provides a single multiport RJ-21 50-pin connector. The SPA8800 can in only a matter of minutes, be easily be configured as an Asterisk® FXO gateway.

Calls originating from the public switched telephone network (PSTN) can be terminated by the SPA8800's FXO ports and routed to analog or IP phones based on an Asterisk server's configuration. Analog phones connected to the SPA8800 can make low-cost VoIP calls via an Internet Telephony Service Provider (ITSP) or can make calls via the PSTN.

Many interesting call routing options are possible using Asterisk to control the SPA8800 gateway. Asterisk can be configured to trunk the SPA8800's four FXO ports together into a trunk group. A trunk group allows the PSTN lines connected to the FXO ports to be over-subscribed and shared among all configured analog and IP phones, effectively lowering telephony costs by not requiring a dedicated line per phone. For example, you can have 4 connected PSTN lines and share them with any number of phones. When all 4 lines are busy, the 5th user will hear a congestion tone.

Additionally, the SPA8800's FXS ports can be used in other ways, including connecting analog phones, door phones, and fax machines.

The SPA8800 supports fax with G.711 pass-through or real-time fax over IP via T.38 fax relay and also supports the G.711 A-law, G.711 µ-law, G.726, G.729A, G.723.1 voice codecs.

In the event that you only need FXS ports and do not need any FXO ports, consider using the Cisco SPA8000 8-Port IP Telephony Gateway. The SPA8000 configuration is very similar to the SPA8800.

Audience
This application note is targeted at anyone who needs an FXO gateway for their Asterisk server. It is expected that readers of this document are familiar with the administration tasks involved with configuring VoIP in an Asterisk environment.

Scope
This scope of this document is limited to configuring the SPA8800 in an Asterisk environment and does not address the following topics:

- Installing an Asterisk server
- Advanced Asterisk configuration
- SPA8800 localization
- Security

Refer to the Related Documents for additional configuration and background information.
Related Documents

- Asterisk: [http://www.asterisk.org](http://www.asterisk.org)
- Cisco SPA8800 Administration Guide
- Cisco Small Business IP Phones Admin Guide
- Cisco Small Business IP Phones User Guide
- Cisco Community Central: [Small Business Community ATA Support](http://www.cisco.com)

Overview

Configuring the SPA8800 is a relatively trivial task and is similar to configuring any of the Sipura / Linksys / Cisco ATA and SPA9000 Voice System devices. Troubleshooting configuration problems due to incorrectly typed information in configuration fields requires advanced network and Asterisk troubleshooting skills. This application note walks you through configuring a SPA8800 and also provides sample traces that may be of use to you when you are troubleshooting your SPA8800 in an Asterisk environment.

By the end of this document, an Asterisk phone user, analog or IP, will be able to pick up a phone and dial out via the PSTN or ITSP, depending on the steering digit they use.

Summary of Tasks in this Document

You must complete the following tasks in order to use the SPA8800 in an Asterisk environment:

1. Gather Basic Information
2. Configure the Asterisk Server
   a. Edit the sip.conf file
   b. Edit the extensions.conf file
   c. Connect to the Asterisk Server’s console
   d. Reload Asterisk modules
3. Connect the SPA8800
4. Configure the SPA8800
   a. Configure static IP address and related information
   b. Upgrade the SPA8800’s firmware
5. Configure phone extensions on the SPA8800 FXS Phone N ports
6. Configure FXO line parameters on the SPA8800 LINE N ports
7. Configure FXO line dial plans for inbound PSTN call routing
8. Test the phone system for appropriate behavior
Requirements
You need the following equipment and services:

- A functional Asterisk server
- A functional LAN with network connectivity to the Internet and an optional Internet Telephony Service Provider (ITSP)
- A SPA8800
- One to four analog phones
- Optional IP phones such as the SPA525G, SPA9x2, SPA9x1, or WIP310 (wireless) IP phones

Configuring Asterisk for a SPA8800
Before you configure your Asterisk server for the SPA8000, you need to gather some basic information:

1. Static IP address for the SPA8800. By default, this device is a DHCP client, but will be of no use to you if it is assigned a new dynamic IP address periodically. In this document, I use 192.168.2.237/24
2. Gateway / Default router, DNS, and NTP server IP addresses
3. Extension numbers for up to four analog phones to be connected to the SPA8800's PHONE FXS ports 1-4. In this document, I use 101, 102, 103, and 104.
4. Decide how many PSTN lines will be connected to the SPA8800's LINE FXO ports 1-4. In this document, I use two PSTN lines connected to FXO LINE ports 2 and 3 [UDP 5161 and 5261 respectively]
5. Decide what steering digits to use. In this document, I use 8 for PSTN and 9 for ITSP
6. Decide what phone or phones to route inbound PSTN and ITSP calls to. In this document, I will route all inbound calls to the analog phone associated with extension 101.
7. Decide what to name the SPA8800 context group in the extensions.conf file. In this document, I use the [fxsgroup] context.

Summary:

- SPA8800 static IP address: 192.168.2.237/24
- Gateway / Default router, DNS, and NTP server IP addresses
- Analog phones: 101-104
- PSTN lines: 2, LINE 2 [UDP 5161] and LINE 3 [UDP 5261]
- Steering digits: 8 for PSTN, 9 for ITSP
- Inbound PSTN call target: 101
- Inbound ITSP call target: 101
Configuring the Asterisk Server

Once you have gathered all of the basic information, you can begin configuring the Asterisk server.

*Editing the /etc/asterisk/sip.conf file*

```sh
# vim /etc/asterisk/sip.conf

[general]
...
...
;register => <DID>:@<ITSP>:@<password>:@<DID>:@<ITSP>/101
register =>
3615551212@sip.broadvoice.com:mypassword:3615551212@sip.broadvoice.com/101
...
...
;
;SPA8800 Changes
;define SPA8800 analog phone 1 extension 101
[101]
type=friend
secret=101
qualify=yes
nat=no
host=dynamic
canreinvite=no
context=fxsgroup
regext=101
;
;define SPA8800 analog phone 2 extension 102
[102]
type=friend
secret=102
qualify=yes
nat=no
host=dynamic
canreinvite=no
context=fxsgroup
regext=102
;
;define SPA8800 analog phone 3 extension 103
[103]
type=friend
secret=103
qualify=yes
nat=no
host=dynamic
canreinvite=no
context=fxsgroup
regext=103
```
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; ;define SPA8800 analog phone 4 extension 104
[104]
type=friend
secret=104
qualify=yes
nat=no
host-dynamic
canreinvite=no
context=fxsgroup
regext=104
;
;define SPA8800 pstn2 user
[pstn2]
type=friend
host=192.168.2.237 ;IP address of the SPA8800
port=5161  ;5161 is the default SIP port for line 2 on the SPA8800
dtmfmode=rfc2833
context=pstn2
insecure=very
;
;define SPA8800 pstn3 user
[pstn3]
type=friend
host=192.168.2.237 ;IP address of the SPA8800
port=5261  ;5261 is the default SIP port for line 2 on the SPA8800
dtmfmode=rfc2833
context=pstn3
insecure=very
;
[itsp1]
type=peer
user=phone
host=sip.broadvoice.com
fromdomain=sip.broadvoice.com
fromuser=3615551212
secret=MyITSPsecret
username=3615551212
insecure=very
authname=3615551212
dtmfmode=inband
dtmf-inband
canreinvite=no
qualify=yes
nat=yes
context=itssp1
;eof
Editing the /etc/asterisk/extensions.conf file

```bash
# vim /etc/asterisk/extensions.conf
...
; SPA8800 Changes
; outbound dialing
[fxsgroup]
;
;
; dial 7 to explicitly use FXO3
exten => _7.,1,Dial(SIP/${EXTEN:1}@pstn3,60,r)
;
; dial 8 as a steering digit:
; if FXO2 is not available, FXO3 will be used.
; if FXO3 is not available, the user hears congestion
exten => _8.,1,Dial(SIP/${EXTEN:1}@pstn2,60,r)
exten => _8.,2,Dial(SIP/${EXTEN:1}@pstn3,60,r)
;
; dial 9 to explicitly use ITSP
exten => _9.,1,Dial(SIP/${EXTEN:1}@itsp1,30,r)
;
; r causes ringing for calling party but audio is not
; passed until called party answers call
; T allows caller to transfer with #
exten => 101,1,Dial(SIP/101,60,rT)
exten => 102,1,Dial(SIP/102,60,rT)
exten => 103,1,Dial(SIP/103,60,rT)
exten => 104,1,Dial(SIP/104,60,rT)
exten => 200,1,Dial(SIP/200,60,rT)
exten => 201,1,Dial(SIP/201,60,rT)
;
; inbound from PSTN
[pstn2]
; t allows called person to transfer with a #
exten => 101,1,Dial(SIP/101,60,rt)
[pstn3]
exten => 201,1,Dial(SIP/201,60,rt)

; inbound calls from ITSP
[itsp1]
exten => 361551212,1,Answer
; enable ring group of extensions 101, 102, and 200
exten => 361551212,2,Dial(SIP/101&SIP/102&SIP/200,25,rt)
exten => 361551212,3,Hangup
; eof
...
...
```

Loading the Modified Configuration

1. Connect to the Asterisk console:

   ```bash
   $ sudo asterisk -r
   *CLI>
   ```

2. Use the reload command to load the changed configuration:
*CLI> module reload

This completes the Asterisk server configuration. You must now configure the SPA8800 to register to the Asterisk server.
Configuring the SPA8800

**SPA8800 Ports and Connections**

Figure 1 The photograph shows the rear of the SPA8800 and its connections

**Connect the SPA8800**

a. Connect the ETHERNET port of the SPA8800 to the LAN switch.

b. Connect analog phones to the PHONE 1-4 FXS ports or use the multiport RJ-21 50-pin connector.

c. Connect PSTN lines to the LINE 1-2 FXO ports or use the multiport RJ-21 50-pin connector.

d. Connect the power adapter.

e. Going off-hook with the analog phones will result in a fast-busy because the SPA8800 has not yet been configured.
Factory Resetting the SPA8800

You should factory reset the SPA8800 so that you start from a known starting point.

f. Connect an analog phone to the SPA8800 PHONE 1 port

g. Go off-hook, ignore the fast-busy or silence

h. Dial **** [four asterisks or stars]
i. Dial 73738# when prompted

j. Press 1 to confirm reset

k. Hang up when prompted

Configuring Static IP Address Addressing

You must configure the SPA8800 with a static IP address because this address is defined in the Asterisk Server's /etc/asterisk/sip.conf file.

l. Determine the SPA8800's Dynamically Assigned IP Address

i. Connect an analog phone to the SPA8800 PHONE 1 port

ii. Go off-hook, ignore the fast-busy or silence

iii. Dial **** [four asterisks or stars]

iv. Dial 110# when prompted

v. Document the IP address

vi. Hang up
m. Direct your browser to the SPA8800’s web user interface (web-ui)

   http://<IP_address_of_SPA8800>/admin/advanced

n. Change the following parameters:

   Network tab > Wan Status tab:
   
i. Internet Connection Settings > Connection Type: Static IP
   
ii. Static IP Settings:
   
   1. Static IP:
   2. Netmask:
   3. Gateway:
   
   iii. Optional Settings:
   
   1. Primary DNS
   2. Secondary DNS
   3. Primary NTP Server

o. Click Submit All Changes
Upgrading the SPA8800's Firmware

1. Direct your browser to the SPA8800's web user interface (web-ui)
   
   [http://<IP_address_of_SPA8800>/admin/advanced](http://<IP_address_of_SPA8800>/admin/advanced)

2. Verify that Compare Network tab > Status tab > Product Information > Software Version:
   is up to date with SPA8800 firmware available at the Cisco.com site. If newer firmware is
   available, save it to disk and upgrade the SPA8800 as follows:

3. Copy the downloaded firmware image to your TFTP server's root directory

4. Cause the SPA8800 to retrieve the firmware by TFTP and install the new image:
   

   Where:
   - **IPADDRESSofSPA** is the SPA8800 IP address
   - **TFTPADDRESS** is the TFTP server's IP address
   - **SPAFILE.bin** is the name of the downloaded firmware image

   Example:

Configuring Phone Extensions on the SPA8800 FXS Phone Ports

In this section, you will point the SPA8800 to the Asterisk Server as the SIP proxy and provide user credentials that you defined earlier in the `sip.conf` and `extensions.conf` Asterisk files. This configuration defines the characteristics of the analog phone connected to the SPA8800 FXS PHONE port.

1. Direct your browser to the SPA8800's web user interface (web-ui)
   
   \[http://<IP\_address\_of\_SPA8800>/admin/advanced\]

2. Voice tab > Phone 1 tab > Line Enable: yes

3. Voice tab > Phone 1 tab > Proxy and Registration > Proxy: 192.168.2.20
   Where this is the IP address of the Asterisk Server

4. Voice tab > Phone 1 tab > Subscriber Information >

5. Display Name: SPA8k8Phone1
   Where this name is assigned by you for easy identification

6. User ID: 101
   Where 101 [username] is defined in the `/etc/asterisk/sip.conf` and `/etc/asterisk/extensions.conf` files

7. Password: 101
   Where the password [secret] is defined in the `/etc/asterisk/sip.conf` file
8. Configure the remaining phones using the parameters that you defined in the 
/etc/asterisk/sip.conf and /etc/asterisk/extensions.conf files.

9. Click Submit All Changes if you do not intend to complete the next step at this time.

**Configuring FXO Line Ports on the SPA8800**

In this section, you will point the SPA8800 to the Asterisk Server as the SIP proxy for the FXO ports and the provide user credentials that you defined earlier in the sip.conf and extensions.conf Asterisk files. This configuration defines the characteristics of the FXO port connected to the PSTN line.

1. Direct your browser to the SPA8800's web user interface (web-ui)

   [http://<IP_address_of_SPA8800>/admin/advanced](http://<IP_address_of_SPA8800>/admin/advanced)

2. Voice tab > Line 2 tab > Line Enable: yes

3. Voice tab > Line N tab > Proxy and Registration >

   a. Proxy: 192.168.2.20 [This field does not need to be completed, but is a good reminder of which device is being used]

   b. Make Call Without Reg: yes

   c. Ans Call Without Reg: yes

4. Voice tab > Line N tab > Subscriber Information >

   d. Display Name: SPA8k8Line2

   e. User ID: pstn2

      Where pstn2 is defined in the /etc/asterisk/sip.conf and /etc/asterisk/extensions.conf files

   f. Password: pstn2

      Where the pstn2 password [secret] is defined in the /etc/asterisk/sip.conf file

      [This field does not need to be completed, the device does not need to register]

   g. Configure the remaining lines using the parameters that you defined in the /etc/asterisk/sip.conf and /etc/asterisk/extensions.conf files.

5. Click Submit All Changes if you do not intend to complete the next step at this time.
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Change these parameters

- NAT Settings
  - NAT Mapping Enable: Yes
  - NAT Keep Alive Enable: Yes
  - NAT Keep Alive Msg: NOTIFY
  - NAT Keep Alive Cnt: PROXY

- Network Settings
  - SIP TOS/CoS Serv Value: 0, 48
  - RTP TOS/CoS Serv Value: 0, 48
  - Network jitter Level: high
  - User Inactivity Adjustment: up and down

- SIP Settings
  - SIP Transport: TCP
  - SIP Port: 5061

- Proxy and Registration
  - Proxy: 102.110.2.20
  - Outbound Proxy: no
  - Use Outbound Proxy: no
  - Register: yes
  - Register Expires: 9000
  - Proxy Fallback Time: 3000

- Subscriber Information
  - Display Name: SPA8800
  - Username: admin
  - Password: password
  - Use Admin: no
Configuring FXO Line Dial Plans for inbound PSTN call Routing

In this section, you will configure FXO port dial plans. These dial plans affect inbound PSTN call routing and work in conjunction with definitions made in the /etc/asterisk/sip.conf and /etc/asterisk/extensions.conf files.

The [general] section of sip.conf contains the register directive which instructs the SIP proxy on where (101) to send inbound calls:

```
[general]
register => 3615551212@sip.broadvoice.com:mypassword:3615551212@sip.broadvoice.com/101
```

The [pstn2] and [pstn3] contexts of the extensions.conf file describes how inbound calls must be routed:

```
PSTN2
  exten => 101,1,Dial(SIP/101,60,rt)
PSTN3
  exten => 101,1,Dial(SIP/201,60,rt)
```

1. Direct your browser to the SPA8800’s web user interface (web-ui)

   `http://<IP_address_of_SPA8800>/admin/advanced`

2. Voice tab > Line 2 tab > Dial Plans>

   Edit any dial plan to route inbound calls from the PSTN line connected to this FXO line. In this example, Dial Plan 8 is edited with: `<:101@192.168.2.20>S0` where:
   
   All inbound calls will be routed to extension 101 of the Asterisk server [192.168.2.20].

3. Voice tab > Line 2 tab > PSTN-To-VoIP Gateway Setup > PSTN Caller Default DP: 8

   Where this number must match the dial plan used in the previous step.

   Refer to the Failing Inbound from PSTN in the Troubleshooting section to see a sample Asterisk console error message that results from an incorrect dial plan entry.
4. Repeat for Line 3.
5. Click Submit All Changes.
6. The entire configuration process is complete once the SPA8800 has rebooted.

**Testing the Phone System**

Test the phone system as follows for appropriate behavior:

1. Test internal calls:
   a. Verify that analog phones can call each other internally. For example call from 101 to 102.
   b. Optional: Verify that analog phones can call IP phones internally. For example, call from 101 to 200 [if configured]

2. Test inbound calls:
   a. From the PSTN, call a phone line attached to the SPA8800’s FXO line. Verify that the appropriate phone rings, analog phone 101 in this document's example.
   b. From the PSTN, call a DID associated with the Asterisk server. Verify that the appropriate phone rings, analog phone 101 in this document's example.

3. Test outbound calls:
   a. From an analog phone, make an outbound call using the PSTN by using the appropriate steering digit, 8 in this document's example.
   b. From an analog phone, make another simultaneous outbound call using the PSTN by using the appropriate steering digit, 8 in this document's example. Verify that both calls succeed, using both configured outbound PSTN lines.
   c. From an analog phone, make an outbound call using the ITSP by using the appropriate steering digit, 9 in this document's example.
T.38 Faxing

The SPA8800 supports fax with G.711 pass-through or real-time fax over IP via T.38 fax relay.

The only change from default setting for fax pass-through is to change from the default named signaling event (NSE) to ReINVITE:

Web-ui > Voice tab > Line N tab > Audio Configuration > FAX Passthru Method: ReINVITE

Optionally, you can change the FAX Line Toggle code from the default of #99. Predialing #99 as a prefix forces the call to be a fax call. This will cause the INVITE to indicate T.38 in the SDP without relying on a reINVITE to switch to T.38. The default can be changed from #99 with the web-ui > Voice tab > Regional tab > Vertical Service Activation Codes > FAX Line Toggle Code:

Refer to the
Trace of FAX Line Toggle Code #99 section showing the difference in the SDP in the INVITE.
## Codecs

The SPA8800 supports the G.711 A-law, G.711 µ-law, G.726, G.729A, G.723.1 voice codecs.

Use the Asterisk CLI to determine the codecs in use during a call:

```
*CLI> sip show channels
```

```
Peer    User/ANR    Call ID      Seq (Tx/Rx)  Format           Hold     Last Message
192.168.2.237 8138293     6a776c7438a  00102/00000  0x80004 (ulaw|h  No       Tx: ACK
192.168.2.237 101         5aed6b9b-9f  00101/00102  0x4 (ulaw)       No       Rx: ACK
111.222.111.222 3615551234 1e7c1653326  00110/00000  0x0 (nothing)    No
3 active SIP channels
```  

*CLI>

Change the SPA8800 web-ui > Voice tab > Line 2 tab > Audio Configuration > Preferred Codec: to G729a and toggle Use Pref Codec Only: yes, save settings, and make a new call.

The Asterisk server's `/etc/asterisk/sip.conf` must have the relevant codecs defined in the `[general]` section. For example:

```
[generic]
...
allow ulaw
allow g729
...```
Troubleshooting

Verify that the SPA8800’s analog phones attached to the PHONE ports are registered.

1. Direct your browser to the SPA8800’s web user interface (web-ui)
   
   \[
   \text{http://<IP\_address\_of\_SPA8800>}/\text{admin/advanced}
   \]

2. Voice tab > Info tab > Phone N Status > Registration State

3. Voice tab > Info tab > PSTN Line N Status > Line Voltage

   Verify that line voltage is present. A voltage of 0 indicates that the PSTN line is not properly connected.

   Registration state of Failed is normal for a properly configured system.
Troubleshooting Rejected Because Extension not Found

Failing Inbound from ITSP

`*CLI>

[May 20 10:06:45] NOTICE[16549]: chan_sip.c:13865 handle_request_invite: Call from '3615551212' to extension '3615551212' rejected because extension not found.

The above message can be a result of incorrect inbound call routing in the extensions.conf file when using Broadvoice.com as an ITSP. Instead of the following in the [itsp1] context:

exten => 101,1,Dial(SIP/101,25,Ttr)

Change to:

exten => 3615551212,1,Dial(SIP/101,25,Ttr)

Failing Inbound from PSTN

[Jun  3 17:34:37] NOTICE[31613]: chan_sip.c:14035 handle_request_invite: Call from 'pstin3' to extension '101' rejected because extension not found.

This indicates that an inbound call to the port named pstin3, is being routed to extension 101. Looking through extensions.conf and sip.conf, all is normal, yet the inbound call does not ring on extension 101. The problem in this scenario is that the SPA8800 Line N's dialplan has been incorrectly configured and is routing the call to extension 101 while extensions.conf is routing the call elsewhere. Correcting the dialplan on the SPA8800 and then saving the configuration solved the problem.
SPA8800 Debug and syslog

The SPA8800 supports writing debug and syslog messages to syslog servers. One server can be used, or separate servers can be used to receive messages. Four levels of verbosity are supported, 0 for no messages, 1 for terse, through 3 for verbose message output:

Following is an example of syslog information produced at Debug Level 3 [numbers changed]. This example shows when the phone goes off hook, digits dialed, calling information, a report that an unlisted codec is being requested, and when the phone goes back on hook:

M0: [0]Off Hook
M0: 2. Report digit 8 (1)(40 ms)
M0: 2. Report digit 5 (1)(40 ms)
M0: 2. Report digit 5 (1)(40 ms)
M0: 2. Report digit 5 (1)(40 ms)
M0: 2. Report digit 1 (1)(40 ms)
M0: 2. Report digit 2 (1)(40 ms)
M0: 2. Report digit 1 (1)(40 ms)
M0: Calling:85551212@192.168.2.20:0
M0: [0:0]AUD ALLOC CALL (port=16404)
M0: [0:0]RTP Rx Up
M0: CC:Ringback
M0: [0:0]RTP Rx Dn
M1: CC:pc(0)=18 not in codec list
M1: AUD:Stop PSTN Tone
M0: CC:Ringback
M2: [1:0]AUD ALLOC CALL (port=18457)
M2: [1:0]RTP Rx Up
M2: AUD:Stop PSTN Tone
M2: CC:Connected
M2: AUD:Stop PSTN Tone
M2: [1:0]ENC INIT 0
M2: [1:0]RTP Tx Up (pt=0->c0a80214:19926)
M2: [1:0]RTCP Tx Up
M0: [0:0]ENC INIT 0
M0: [0:0]RTP Tx Up (pt=0->c0a80214:19046)
M0: [0:0]RTCP Tx Up
M0: CC:Remote Resume
M2: FXO:Off Hook
M2: FXO:Stop CNDD
M0: CC:Connected
M0: [0:0]RTP Rx Up
M0: [0:0]RTP Rx 1st PKT @16404(2)
M2: [1:0]RTP Rx 1st PKT @18457(2)
M0: [0:0]DEC INIT 0
M2: [1:0]DEC INIT 0
M2: FXO:CPC
M2: AUD:Stop PSTN Tone
M2: FXO:On Hook
M2: FXO:Stop CNDD
M2: AUD:Stop PSTN Tone
M2: [0]FM Alert Stop RxTx {c=002b5f38;a=0}
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SPA8800 SIP Debugging

The SPA8800 can also supply SIP debug information to assist with troubleshooting. Enable SIP debugging as follows: SPA8800 web-ui > Voice tab > Line N tab > SIP Settings > SIP Debug Option:

Following is an example of syslog information produced at Debug Level 3 with SIP Debug Option set to full [numbers changed]. This example shows a SIP INVITE, the 100 trying, a report that an unlisted codec type is being requested, and the BYE:

M1: [1]<192.168.2.20:5060(826) INVITE sip:5551212@192.168.2.237:5161 SIP/2.0 Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK58d12649;rport From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as16cfeb3c To: <sip:5551212@192.168.2.237:5161> Contact: <sip:101@192.168.2.20> Call-ID: 4a713b136335955e38e0dd821f5c50dc8192.168.2.20 CSeq: 102 INVITE User-Agent: Asterisk PBX Max-Forwards: 70 Date: Thu, 28 May 2009 13:29:14 GMT Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY Supported: replaces Content-Type: application/sdp Content-Length: 285 v=0 o=root 7298 7298 IN IP4 192.168.2.20 s=session c=IN IP4 192.168.2.20 t=0 0 m=audio 10114 RTP/AVP 0 3 8 101 a=rtpmap:0 PCMU/8000 a=rtpmap:3 GSM/8000 a=rtpmap:8 PCMA/8000 a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-16 a=silenceSupp:off - - -
a=ptime:20
a=sendrecv
M1:
M1:
M0: CC:Ringback
M0: [0:0]RTP Rx Dn
M1: [1]->192.168.2.20:5060(305)
M1: [1]->192.168.2.20:5060(305)
SIP/2.0 100 Trying
To: <sip:5551212@192.168.2.237:5161>
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as16cfeb3c
Call-ID: 4a713b136335955e38e0dd821f5c50dc9192.168.2.20
CSeq: 102 INVITE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK58d12649
Server: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0

M1:
M1:
M0: CC:pc(0)=18 not in codec list
M0: AUD:Stop PSTN Tone
M1: [1]->192.168.2.20:5060(385)
M1: [1]->192.168.2.20:5060(385)
SIP/2.0 488 Not Acceptable Here
To: <sip:5551212@192.168.2.237:5161>;tag=e967475542dc2b0i1
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as16cfeb3c
Call-ID: 4a713b136335955e38e0dd821f5c50dc9192.168.2.20
CSeq: 102 INVITE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK58d12649
Warning: 304 spa "Media type not available"
Server: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0

M1:
M1:
M0: CC:Ringback
M0: [1]<<192.168.2.20:5060(398)
M1: [1]<<192.168.2.20:5060(398)
ACK sip:5551212@192.168.2.237:5161 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK58d12649;rport
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as16cfeb3c
Contact: <sip:101@192.168.2.20>
Call-ID: 4a713b136335955e38e0dd821f5c50dc9192.168.2.20
CSeq: 102 ACK
User-Agent: Asterisk PBX
Max-Forwards: 70
Content-Length: 0

M1:
M1:
M1: [1]<<192.168.2.20:5060(365)
M1: [1]<<192.168.2.20:5060(365)
BYE sip:5551212@192.168.2.237:5161 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK5886f277;rport
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as16cfeb3c
To: <sip:5551212@192.168.2.237:5161>;tag=e9674755542dc2b0i1
Call-ID: 4a713b136335955e38e0dd821f5c50dc9192.168.2.20
CSeq: 103 BYE
User-Agent: Asterisk PBX
Max-Forwards: 70
Content-Length: 0

M1:
M1:
M1: [1]->192.168.2.20:5060(353)
M1: [1]->192.168.2.20:5060(353)
SIP/2.0 481 Call Leg/Transaction Does Not Exist
To: <sip:5551212@192.168.2.237:5161>;tag=e9674755542dc2b0i1
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as16cfeb3c
Call-ID: 4a713b136335955e38e0dd821f5c50dc9192.168.2.20
CSeq: 103 BYE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK5886f277
Server: Cisco/SPA8800-6.1.7(GW)
Troubleshooting with Asterisk CLI Commands

The following Asterisk CLI commands are useful for troubleshooting the environment:

- core show version
- sip show peers
- sip show peer itsp1
- sip show peer 200
- sip show channels
- sip show settings
- sip show users
- sip show user 200
- sip show objects
- core show channels
- dialplan show

Connect to the Asterisk console with:

$ sudo asterisk -r

sip show peers

*CLI> sip show peers
Name/username  Host IP Addr  Dyn Nat ACL Port Status
itsp1/3615551212 <ITSP IP Addr> N 5060 OK (110 ms)
pstn2/pstn2   192.168.2.237        N      5161     Unmonitored
104/104        (Unspecified)    D   N      0        UNKNOWN
103/103        (Unspecified)    D   N      0        UNKNOWN
102/102        192.168.2.237    D   N      5160     OK (10 ms)
101/101        192.168.2.237    D   N      5060     OK (9 ms)
201/201        192.168.2.19     D   N      5060     Unmonitored
200/200        192.168.2.15     D   N      5060     Unmonitored
8 sip peers [Monitored: 3 online, 2 offline Unmonitored: 3 online, 0 offline]
*CLI>

sip show peer <PeerName>

This command is useful to verify the following:

Name, Context, and credentials. Following is an example where the DID and IP address have been sanitized.

*CLI> sip show peer itsp1
* Name : itsp1
Secret  : <Set>
MD5Secret : <Not set>
Context : itsp1
Subscr.Cont. : <Not set>
Language : 
AMA flags : Unknown
Transfer mode: open
### Configuring the Cisco SPA8800 IP Telephony Gateway in an Asterisk Environment

<table>
<thead>
<tr>
<th>Configuration</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>CallingPres</td>
<td>Presentation Allowed, Not Screened</td>
</tr>
<tr>
<td>FromUser</td>
<td>3615551212</td>
</tr>
<tr>
<td>FromDomain</td>
<td>sip.broadvoice.com</td>
</tr>
<tr>
<td>Callgroup</td>
<td></td>
</tr>
<tr>
<td>Pickupgroup</td>
<td></td>
</tr>
<tr>
<td>Mailbox</td>
<td></td>
</tr>
<tr>
<td>VM Extension</td>
<td>asterisk</td>
</tr>
<tr>
<td>LastMsgsSent</td>
<td>32767/65535</td>
</tr>
<tr>
<td>Call limit</td>
<td>0</td>
</tr>
<tr>
<td>Dynamic</td>
<td>No</td>
</tr>
<tr>
<td>Callerid</td>
<td>&quot;&quot; &lt;&gt;</td>
</tr>
<tr>
<td>MaxCallBR</td>
<td>384 kbps</td>
</tr>
<tr>
<td>Expire</td>
<td>-1</td>
</tr>
<tr>
<td>Insecure</td>
<td>port, invite</td>
</tr>
<tr>
<td>Nat</td>
<td>Always</td>
</tr>
<tr>
<td>ACL</td>
<td>No</td>
</tr>
<tr>
<td>T38 pt UDPTL</td>
<td>No</td>
</tr>
<tr>
<td>CanReinvite</td>
<td>No</td>
</tr>
<tr>
<td>FrommiscRedir</td>
<td>No</td>
</tr>
<tr>
<td>User=Phone</td>
<td>No</td>
</tr>
<tr>
<td>Video Support</td>
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</tr>
<tr>
<td>Trust RPID</td>
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<tr>
<td>Send RPID</td>
<td>No</td>
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<tr>
<td>Subscriptions</td>
<td>Yes</td>
</tr>
<tr>
<td>Overlap dial</td>
<td>Yes</td>
</tr>
<tr>
<td>DTMFmode</td>
<td>inband</td>
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<td>LastMsg</td>
<td>0</td>
</tr>
<tr>
<td>ToHost</td>
<td>sip.broadvoice.com</td>
</tr>
<tr>
<td>Addr-&gt;IP</td>
<td>&lt;ITSP IP Addr&gt; Port 5060</td>
</tr>
<tr>
<td>Defaddr-&gt;IP</td>
<td>0.0.0.0 Port 0</td>
</tr>
<tr>
<td>Def. Username</td>
<td>3615551212</td>
</tr>
<tr>
<td>SIP Options</td>
<td>100rel</td>
</tr>
<tr>
<td>Codecs</td>
<td>0x8000e gsm</td>
</tr>
<tr>
<td>Codec Order</td>
<td>(none)</td>
</tr>
<tr>
<td>Auto-Framing</td>
<td>No</td>
</tr>
<tr>
<td>Status</td>
<td>OK (113 ms)</td>
</tr>
<tr>
<td>Useragent</td>
<td></td>
</tr>
<tr>
<td>Reg. Contact</td>
<td></td>
</tr>
</tbody>
</table>
Configuring the Cisco SPA8800 IP Telephony Gateway in an Asterisk Environment

**sip show users**

The `sip show users` command is useful to display usernames, secrets [passwords], and context information. Following is an example:

```
*CLI> sip show users

Username       Secret       Accountcode       Def.Context       ACL  NAT
pstn2          pstn2        pstn2             pstn2            No   Always
104             104          fxsgroup          No Always
103             103          fxsgroup          No Always
102             102          fxsgroup          No Always
101             101          fxsgroup          No Always
201             201secret    fxsgroup          No Always
200             200secret    fxsgroup          No Always

*CLI>
```

**sip show user <UserName>**

The `sip show user <UserName>` command is useful for verifying context information. Following is an example:

```
*CLI>
  * Name       : 101
  Secret       : <Set>
  MD5Secret    : <Not set>
  Context      : fxsgroup
  Language     : 
  AMA flags    : Unknown
  Transfer mode: open
  MaxCallBR    : 384 kbps
  CallingPres  : Presentation Allowed, Not Screened
  Call limit   : 0
  Callgroup    : 
  Pickupgroup  :
  Callerid     : "" <>
  ACL          : No
  Codec Order  : (none)
  Auto-Framing: No

*CLI>
```

**Sample Traces**

Sometimes, the best way to troubleshoot Asterisk and SPA8800 interaction issues is to capture a trace and compare it against a similar transaction. Following are three traces showing in order:

1. A successful registration between the SPA8800 and the Asterisk server and a successful registration between the Asterisk server and the broadvoice.com SIP proxy. [IP addresses have been changed]
2. An inbound call from the PSTN routed to extension 101
3. An inbound call from the ITSP routed to extension 101
4. An outbound call from extension 101 via the PSTN
5. An outbound call from extension 101 via the ITSP
Trace of Asterisk Server Registering to ITSP

Frame 11 (453 bytes on wire, 453 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: Cisco-Li_9c:e3:2c (00:1d:7e:9c:e3:2c)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 147.135.32.221 (147.135.32.221)
Session Initiation Protocol
Request-Line: REGISTER sip:sip.broadvoice.com SIP/2.0
Method: REGISTER
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK355c454d;rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK355c454d
RPort: rport
From: <sip:3615551212@sip.broadvoice.com>;tag=as2aa95843
SIP from address: sip:3615551212@sip.broadvoice.com
SIP tag: as2aa95843
To: <sip:3615551212@sip.broadvoice.com>
SIP to address: sip:3615551212@sip.broadvoice.com
Call-ID: 5aca91fd5903c6562f020aab771422f4@127.0.0.1.1
CSeq: 104 REGISTER
Sequence Number: 104
Method: REGISTER
User-Agent: Asterisk PBX
Max-Forwards: 70
Expires: 120
Contact: <sip:101@192.168.2.20>
Contact Binding: <sip:101@192.168.2.20>
URI: <sip:101@192.168.2.20>
Event: registration
Content-Length: 0

Frame 12 (416 bytes on wire, 416 bytes captured)
Ethernet II, Src: Cisco-Li_9c:e3:2c (00:1d:7e:9c:e3:2c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
Internet Protocol, Src: 147.135.32.221 (147.135.32.221), Dst: 192.168.2.20 (192.168.2.20)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Status-Code: 200
[Resent Packet: False]
Message Header
Call-ID: 5aca91fd5903c6562f020aab771422f4@127.0.0.1.1
CSeq: 104 REGISTER
Sequence Number: 104
Method: REGISTER
From: <sip:3615551212@sip.broadvoice.com>;tag=as2aa95843
SIP from address: sip:3615551212@sip.broadvoice.com
SIP tag: as2aa95843
To: <sip:3615551212@sip.broadvoice.com>
SIP to address: sip:3615551212@sip.broadvoice.com
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK355c454d;rport=33579
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK355c454d
Received: 24.153.145.213
RPort: 33579
Contact: <sip:101@192.168.2.20>
Contact Binding: <sip:101@192.168.2.20>
URI: <sip:101@192.168.2.20>
SIP contact address: sip:101@192.168.2.20
Event: registration
Content-Length: 0
Trace of SPA922 Registering to an Asterisk Server

Frame 13 (683 bytes on wire, 683 bytes captured)
Ethernet II, Src: CiscoLin_db:51:d2 (00:0e:08:db:51:d2), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
Internet Protocol, Src: 192.168.2.13 (192.168.2.13), Dst: 192.168.2.20 (192.168.2.20)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: REGISTER sip:192.168.2.20 SIP/2.0
Method: REGISTER
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.13:5060;branch=z9hG4bK-d83df1c
Transport: UDP
Sent-by Address: 192.168.2.13
Sent-by port: 5060
Branch: z9hG4bK-d83df1c
From: "Asterisk201" <sip:201@192.168.2.20>;tag=2b2b435b2119d3aco0
SIP Display info: "Asterisk201"
SIP from address: sip:201@192.168.2.20
SIP tag: 2b2b435b2119d3aco0
To: "Asterisk201" <sip:201@192.168.2.20>
SIP Display info: "Asterisk201"
SIP to address: sip:201@192.168.2.20
Call-ID: a20ca248-9753bb6bc8192.168.2.13
CSeq: 52217 REGISTER
Sequence Number: 52217
Method: REGISTER
Max-Forwards: 70
Authorization: Digest
username="201",realm="asterisk",nonce="3e561ac2",uri=sip:192.168.2.20",algorithm=MD5,response ="728a48d3a084a509dfe29d3686e63317"
Authentication Scheme: Digest
Username: "201"
Realm: "asterisk"
Nonce Value: "3e561ac2"
Authentication URI: "sip:192.168.2.20"
Algorithm: MD5
Digest Authentication Response: "728a48d3a084a509dfe29d3686e63317"
Contact: "Asterisk201" <sip:201@192.168.2.20:5060;expires=0>
Contact Binding: "Asterisk201" <sip:201@192.168.2.20:5060;expires=0>
URL: "Asterisk201" <sip:201@192.168.2.20:5060>
SIP Display info: "Asterisk201"
SIP contact address: sip:201@192.168.2.20:5060
User-Agent: Asterisk PBX
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: replaces

Frame 14 (484 bytes on wire, 484 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: CiscoLin_db:51:d2 (00:0e:08:db:51:d2)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Status-Code: 100
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.13:5060;branch=z9hG4bK-d83df1c;received=192.168.2.13
Transport: UDP
Sent-by Address: 192.168.2.13
Sent-by port: 5060
Branch: z9hG4bK-d83df1c
Received: 192.168.2.13
From: "Asterisk201" <sip:201@192.168.2.20;tag=2b2b435b2119d3aco0>
SIP Display info: "Asterisk201"
SIP from address: sip:201@192.168.2.20
SIP tag: 2b2b435b2119d3aco0
To: "Asterisk201" <sip:201@192.168.2.20>
SIP Display info: "Asterisk201"
SIP to address: sip:201@192.168.2.20
Call-ID: a20ca248-9753bb6bc8192.168.2.13
CSeq: 52217 REGISTER
Sequence Number: 52217
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:201@192.168.2.20>
Contact Binding: <sip:201@192.168.2.20>
URI: <sip:201@192.168.2.20>
SIP contact address: sip:201@192.168.2.20
Content-Length: 0
Configuring the Cisco SPA8800 IP Telephony Gateway in an Asterisk Environment

Frame 15 (548 bytes on wire, 548 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: CiscoLin_db:51:d2 (00:0e:08:db:51:d2)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 401 Unauthorized
Status-Code: 401
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.13:5060;branch=z9hG4bK-d83df1c;r=192.168.2.13
Transport: UDP
Sent-by Address: 192.168.2.13
Sent-by port: 5060
Branch: z9hG4bK-d83df1c
Received: 192.168.2.13
From: "Asterisk201" <sip:201@192.168.2.20>;tag=2b2b435b2119d3aco0
SIP Display info: "Asterisk201"
SIP from address: sip:201@192.168.2.20
SIP tag: 2b2b435b2119d3aco0
To: "Asterisk201" <sip:201@192.168.2.20>;tag=as6eb2d1ad
SIP Display info: "Asterisk201"
SIP to address: sip:201@192.168.2.20
SIP tag: as6eb2d1ad
Call-ID: a20ca248-9753b6bc@192.168.2.13
CSeq: 52217 REGISTER
Sequence Number: 52217
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
WWW-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="4bb63f6e"
Authentication Scheme: Digest
Algorithm: MD5
Realm: "asterisk"
Nonce Value: "4bb63f6e"
Content-Length: 0
Frame 16 (684 bytes on wire, 684 bytes captured)
Ethernet II, Src: CiscoLin_db:51:d2 (00:0e:08:db:51:d2), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
Internet Protocol, Src: 192.168.2.13 (192.168.2.13), Dst: 192.168.2.20 (192.168.2.20)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: REGISTER sip:192.168.2.20 SIP/2.0
Method: REGISTER
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.13:5060;branch=z9hG4bK-8674aa69
Transport: UDP
Sent-by Address: 192.168.2.13
Sent-by port: 5060
Branch: z9hG4bK-8674aa69
From: "Asterisk201" <sip:201@192.168.2.20>;tag=2b2b435b2119d3aco0
SIP Display info: "Asterisk201"
SIP from address: sip:201@192.168.2.20
SIP tag: 2b2b435b2119d3aco0
To: "Asterisk201" <sip:201@192.168.2.20>
SIP Display info: "Asterisk201"
SIP to address: sip:201@192.168.2.20
SIP tag: 2b2b435b2119d3aco0
Call-ID: a20ca248-9753b6bc@192.168.2.13
CSeq: 52218 REGISTER
Sequence Number: 52218
Method: REGISTER
Max-Forwards: 70
Authorization: Digest
username="201",realm=asterisk,nonce="4bb63f6e",uri=sip:192.168.2.20,algorithm=MD5,response=9e3ff52ed50c8276650bd33d8aa347b
Authentication Scheme: Digest
Username: "201"
Realm: "asterisk"
Nonce Value: "4bb63f6e"
Authentication URI: sip:192.168.2.20
Algorithm: MD5
Digest Authentication Response: 9e3ff52ed50c8276650bd33d8aa347b
Contact: "Asterisk201" <sip:201@192.168.2.13:5060>;expires=0
Contact Binding: "Asterisk201" <sip:201@192.168.2.13:5060>;expires=0
URI: "Asterisk201" <sip:201@192.168.2.13:5060>
SIP Display info: "Asterisk201"
SIP contact address: sip:201@192.168.2.13:5060
User-Agent: Linksys/SPA922-6.1.3(a)
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: replaces
Content-Length: 0
Frame 17 (485 bytes on wire, 485 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: CiscoLin_db:51:d2
(00:0e:08:db:51:d2)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Status-Code: 100
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.13:5060;branch=z9hG4bK-8674aa69;received=192.168.2.13
Transport: UDP
Sent-by Address: 192.168.2.13
Sent-by port: 5060
Branch: z9hG4bK-8674aa69
Received: 192.168.2.13
From: "Asterisk201" <sip:201@192.168.2.20>;tag=2b2b435b2119d3aco0
SIP Display info: "Asterisk201"
SIP from address: sip:201@192.168.2.20
SIP tag: 2b2b435b2119d3aco0
To: "Asterisk201" <sip:201@192.168.2.20>
SIP Display info: "Asterisk201"
SIP to address: sip:201@192.168.2.20
Call-ID: a20ca248-9753bb6bc@192.168.2.13
CSeq: 52218 REGISTER
Sequence Number: 52218
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:201@192.168.2.20>
Contact Binding: <sip:201@192.168.2.20>
URI: <sip:201@192.168.2.20>
SIP contact address: sip:201@192.168.2.20
Content-Length: 0

Frame 18 (512 bytes on wire, 512 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: CiscoLin_db:51:d2
(00:0e:08:db:51:d2)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Status-Code: 200
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.13:5060;branch=z9hG4bK-8674aa69;received=192.168.2.13
Transport: UDP
Sent-by Address: 192.168.2.13
Sent-by port: 5060
Branch: z9hG4bK-8674aa69
Received: 192.168.2.13
From: "Asterisk201" <sip:201@192.168.2.20>;tag=2b2b435b2119d3aco0
SIP Display info: "Asterisk201"
SIP from address: sip:201@192.168.2.20
SIP tag: 2b2b435b2119d3aco0
To: "Asterisk201" <sip:201@192.168.2.20>
SIP Display info: "Asterisk201"
SIP to address: sip:201@192.168.2.20
Call-ID: a20ca248-9753bb6bc@192.168.2.13
CSeq: 52218 REGISTER
Sequence Number: 52218
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Expires: 0
Date: Fri, 05 Jun 2009 10:00:31 GMT
Content-Length: 0
Trace of SPA8800 Phone Ports Registering

Each enabled FXS port on the SPA8800 registers to the Asterisk server. The following trace shows phone ports 1 and 2 registering to the Asterisk server:

Frame 55 (545 bytes on wire, 545 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: REGISTER sip:192.168.2.20 SIP/2.0
Method: REGISTER
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-fa604ed5
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-fa604ed5
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=cfee310a1d38a6be00
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: cfee310a1d38a6be00
To: "SPA8k8Phone1"<sip:101@192.168.2.20>
SIP Display info: "SPA8k8Phone1"
SIP to address: sip:101@192.168.2.20
Call-ID: 662d034a1-65fd1530127.0.0.1
CSeq: 58444 REGISTER
Sequence Number: 58444
Method: REGISTER
Max-Forwards: 70
Contact: "SPA8k8Phone1" <sip:101@192.168.2.20;expires=3600
Contact Binding: "SPA8k8Phone1" <sip:101@192.168.2.237:5060;expires=3600
URI: "SPA8k8Phone1" <sip:101@192.168.2.237:5060
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: x-sipura, replaces
Content-Length: 0
Frame 56 (485 bytes on wire, 485 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Status-Code: 100
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-fa604ed5;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-fa604ed5
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20;tag=cfee310a1d38a6be00
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: cfee310a1d38a6be00
To: "SPA8k8Phone1"<sip:101@192.168.2.20>
SIP Display info: "SPA8k8Phone1"
SIP to address: sip:101@192.168.2.20
Call-ID: 662d034a1-65fd1530127.0.0.1
CSeq: 58444 REGISTER
Sequence Number: 58444
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:101@192.168.2.20>
Session Initiation Protocol
Status-Line: SIP/2.0 401 Unauthorized
Status-Code: 401
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-fa04ed5;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-fa04ed5
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=cfee310ad138a6be00
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: cfee310ad138a6be00
To: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as5523cb3f
SIP Display info: "SPA8k8Phone1"
SIP to address: sip:101@192.168.2.20
SIP tag: as5523cb3f
Call-ID: 66b034a1-e56d5e3127.0.0.1
CSeq: 58444 REGISTER
Sequence Number: 58444
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
WWW-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="76c8a749"
Authentication Scheme: Digest
Algorithm: MD5
Realm: "asterisk"
Nonce Value: "76c8a749"
Content-Length: 0

Frame 58 (697 bytes on wire, 697 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1
(00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: REGISTER sip:192.168.2.20 SIP/2.0
Method: REGISTER
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-1da4459
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-1da4459
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=cfee310ad138a6be00
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: cfee310ad138a6be00
To: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as5523cb3f
SIP Display info: "SPA8k8Phone1"
SIP to address: sip:101@192.168.2.20
Call-ID: 66b034a1-e56d5e3127.0.0.1
CSeq: 58445 REGISTER
Sequence Number: 58445
Method: REGISTER
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-1da4459
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-1da4459
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=cfee310ad138a6be00
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: cfee310ad138a6be00
To: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as5523cb3f
SIP Display info: "SPA8k8Phone1"
SIP to address: sip:101@192.168.2.20
Call-ID: 66b034a1-e56d5e3127.0.0.1
CSeq: 58445 REGISTER
Sequence Number: 58445
Method: REGISTER
Max-Forwards: 70
Authorization: Digest
username="101",realm="asterisk",nonce="76c8a749",uri="sip:192.168.2.20",algorithm=MD5,response =1142d87cafe17bd1b9e805fc26e9fbb1"
Authentication Scheme: Digest
Username: "101"
Realm: "asterisk"
Nonce Value: "76c8a749"
Authentication URI: "sip:192.168.2.20"
Algorithm: MD5
Digest Authentication Response: "1142d87cafe17bd1b9e805fc26e9fbb1"
Contact: "SPA8k8Phone1" <sip:101@192.168.2.237;5060;Expires=3600
Contact Binding: "SPA8k8Phone1" <sip:101@192.168.2.237:5060;Expires=3600
URI: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
SIP Display info: "SPA8k8Phone1"
SIP contact address: sip:101@192.168.2.237:5060
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces

Frame 59 (484 bytes on wire, 484 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Status-Code: 100
[Resent Packet: False]

Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-1-da4459;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-1-da4459
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=cfee310ad138a6be00
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: cfee310ad138a6be00
To: "SPA8k8Phone1" <sip:101@192.168.2.20>
SIP Display info: "SPA8k8Phone1"
SIP to address: sip:101@192.168.2.20
Call-ID: 6d6b034a1-6d1f53d270.0.0.1
CSeq: 58445 REGISTER
Sequence Number: 58445
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:101@192.168.2.20>
Contact Binding: <sip:101@192.168.2.20>
URI: <sip:101@192.168.2.20>
SIP contact address: sip:101@192.168.2.20
Content-Length: 0
Frame 60 (566 bytes on wire, 566 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: 00:24:97:f0:50:3c
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Status-Code: 200
[Resent Packet: False]

Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-1-da4459;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-1-da4459
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=cfee310ad138a6be00
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: cfee310ad138a6be00
To: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as5523cb3f
SIP Display info: "SPA8k8Phone1"
SIP to address: sip:101@192.168.2.20
SIP tag: as5523cb3f
Call-ID: 6d6b034a1-6d1f53d270.0.0.1
CSeq: 58445 REGISTER
Sequence Number: 58445
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Expires: 3600
Contact: <sip:101@192.168.2.237:5060>;expires=3600
Contact Binding: <sip:101@192.168.2.237:5060>;expires=3600
URI: <sip:101@192.168.2.237:5060>
SIP contact address: sip:101@192.168.2.237:5060
Date: Fri, 05 Jun 2009 10:15:55 GMT
Content-Length: 0
Frame 63 (546 bytes on wire, 546 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: 00:24:97:f0:50:3c
User Datagram Protocol, Src Port: 5160 (5160), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: REGISTER sip:192.168.2.20 SIP/2.0
Method: REGISTER
[Resent Packet: False]

Message Header
Via: SIP/2.0/UDP 192.168.2.237:5160;branch=z9hG4bK-53ffec42
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5160
Branch: z9hG4bK-53ffec42
From: "SPA8k8Phone2" <sip:102@192.168.2.20>;tag=a29b792271423dfe0
SIP Display info: "SPA8k8Phone2"
SIP from address: sip:102@192.168.2.20
SIP tag: a29b792271423dfe0
To: "SPA8k8Phone2" <sip:102@192.168.2.20>
SIP Display info: "SPA8k8Phone2"
SIP to address: sip:102@192.168.2.20
Call-ID: b628f5b9-f2e991a68127.0.0.1
CSeq: 24222 REGISTER
Sequence Number: 24222
Method: REGISTER
Max-Forwards: 70
Contact: "SPA8k8Phone2" <sip:102@192.168.2.237:5160>;expires=3600
Contact Binding: "SPA8k8Phone2" <sip:102@192.168.2.237:5160>;expires=3600
URI: "SPA8k8Phone2" <sip:102@192.168.2.237:5160>
SIP Display info: "SPA8k8Phone2"
SIP contact address: sip:102@192.168.2.237:5160
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces

Frame 64 (486 bytes on wire, 486 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5160 (5160)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Status-Code: 100
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5160;branch=z9hG4bK-53ffec42;received=192.168.2.237
Transport: UDP
Sent-by-Address: 192.168.2.237
Sent-by-port: 5160
Branch: z9hG4bK-53ffec42
Received: 192.168.2.237
From: "SPA8k8Phone2" <sip:102@192.168.2.20>;tag=a29b792271423dfe0
SIP Display info: "SPA8k8Phone2"
SIP from address: sip:102@192.168.2.20
SIP tag: a29b792271423dfe0
To: "SPA8k8Phone2" <sip:102@192.168.2.20>
SIP Display info: "SPA8k8Phone2"
SIP to address: sip:102@192.168.2.20
Call-ID: b628f5b9-f2e991a68127.0.0.1
CSeq: 24222 REGISTER
Sequence Number: 24222
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:102@192.168.2.20>
Contact Binding: <sip:102@192.168.2.20>
URI: <sip:102@192.168.2.20>
SIP contact address: sip:102@192.168.2.20
Content-Length: 0

Frame 65 (550 bytes on wire, 550 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5160 (5160)
Session Initiation Protocol
Status-Line: SIP/2.0 401 Unauthorized
Status-Code: 401
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5160;branch=z9hG4bK-53ffec42;received=192.168.2.237
Transport: UDP
Sent-by-Address: 192.168.2.237
Sent-by-port: 5160
Branch: z9hG4bK-53ffec42
Received: 192.168.2.237
From: "SPA8k8Phone2" <sip:102@192.168.2.20>;tag=a29b792271423dfe0
SIP Display info: "SPA8k8Phone2"
SIP from address: sip:102@192.168.2.20
SIP tag: a29b792271423dfe0
To: "SPA8k8Phone2" <sip:102@192.168.2.20>
SIP Display info: "SPA8k8Phone2"
SIP to address: sip:102@192.168.2.20
Call-ID: b628f5b9-f2e991a68127.0.0.1
CSeq: 24222 REGISTER
Sequence Number: 24222
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
WWW-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="1fc67977"
Authentication Scheme: Digest
Algorithm: MD5
Realm: "asterisk"
Nonce Value: "1fc67977"
Content-Length: 0

Frame 66 (699 bytes on wire, 699 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1
(00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: 5160 (5160), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: REGISTER sip:192.168.2.20 SIP/2.0
Method: REGISTER
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5160;branch=z9hG4bK-ededb26
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5160
Branch: z9hG4bK-ededb26
From: "SPA8k8Phone2" <sip:102@192.168.2.20>;tag=a29b792271423dfe0
SIP Display info: "SPA8k8Phone2"
SIP from address: sip:102@192.168.2.20
SIP tag: a29b792271423dfe0
To: "SPA8k8Phone2" <sip:102@192.168.2.20>
SIP Display info: "SPA8k8Phone2"
SIP to address: sip:102@192.168.2.20
Call-ID: b628f5b5-f2e991a6@127.0.0.1
CSeq: 24223 REGISTER
Sequence Number: 24223
Method: REGISTER
Max-Forwards: 70
Authorization: Digest
username="102", realm="asterisk", nonce="1fc67977", uri="sip:192.168.2.20", algorithm=MD5, response=
"lcbd8f1d8e70d514f0e494681ee4a40"
Authentication Scheme: Digest
Username: "102"
Realm: "asterisk"
Nonce Value: "1fc67977"
Authentication URI: "sip:192.168.2.20"
Algorithm: MD5
Digest Authentication Response: "lcbd8f1d8e70d514f0e494681ee4a40"
Contact: "SPA8k8Phone2" <sip:102@192.168.2.237:5160>;expires=3600
Contact Binding: "SPA8k8Phone2" <sip:102@192.168.2.237:5160>;expires=3600
URI: "SPA8k8Phone2" <sip:102@192.168.2.237:5160>
SIP Display info: "SPA8k8Phone2"
SIP contact address: sip:102@192.168.2.237:5160
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces

Frame 67 (486 bytes on wire, 486 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5160 (5160)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Status-Code: 100
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5160;branch=z9hG4bK-ededb26;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5160
Branch: z9hG4bK-ededb26
Received: 192.168.2.237
From: "SPA8k8Phone2" <sip:102@192.168.2.20>;tag=a29b792271423dfe0
SIP Display info: "SPA8k8Phone2"
SIP from address: sip:102@192.168.2.20
SIP tag: a29b792271423dfe0
To: "SPA8k8Phone2" <sip:102@192.168.2.20>
SIP Display info: "SPA8k8Phone2"
SIP to address: sip:102@192.168.2.20
Call-ID: b628f5b5-f2e991a6@127.0.0.1
CSeq: 24223 REGISTER
Sequence Number: 24223
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:102@192.168.2.20>
    Contact Binding: <sip:102@192.168.2.20>
    URI: <sip:102@192.168.2.20>
    SIP contact address: sip:102@192.168.2.20
Content-Length: 0

Frame 68 (568 bytes on wire, 568 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
    User Datagram Protocol, Src Port: sip (5060), Dst Port: 5160 (5160)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Status-Code: 200
    [Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5160;branch=z9hG4bK -edebcc26;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5160
Branch: z9hG4bK- edebcc26
Received: 192.168.2.237
From: "SPA8k8Phone2" <sip:102@192.168.2.20>;tag=a29b792271423dfeo0
    SIP Display info: "SPA8k8Phone2"
    SIP from address: sip:102@192.168.2.20
To: "SPA8k8Phone2" <sip:102@192.168.2.20>;tag=as674411f6
    SIP Display info: "SPA8k8Phone2"
    SIP to address: sip:102@192.168.2.20
    SIP tag: as674411f6
Call-ID: b628f5b9-f2e991a6@127.0.0.1
CSeq: 24223 REGISTER
    Sequence Number: 24223
Method: REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Expires: 3600
Contact: <sip:102@192.168.2.237:5160>;expires=3600
    Contact Binding: <sip:102@192.168.2.237:5160>;expires=3600
    URI: <sip:102@192.168.2.237:5160>
    SIP contact address: sip:102@192.168.2.237:5160
Date: Fri, 05 Jun 2009 10:02:02 GMT
Content-Length: 0
Trace of Call between SPA8800 FXS1 and FXS2

Frame 77 (1087 bytes on wire, 1087 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: INVITE sip:102@192.168.2.20 SIP/2.0
Method: INVITE
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-d558dcd5
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-d558dcd5
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=377ea678f1613b100
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 377ea678f1613b100
To: <sip:102@192.168.2.20>
SIP to address: sip:102@192.168.2.20
Remote-Party-ID: "SPA8k8Phone1" <sip:101@192.168.2.20>;screen=yes;party=calling
Call-ID: c677a744-91e29558@192.168.2.237
CSeq: 101 INVITE
Sequence Number: 101
Method: INVITE
Max-Forwards: 70
Contact: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
Contact Binding: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
URI: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
SIP Display info: "SPA8k8Phone1"
SIP contact address: sip:101@192.168.2.237:5060
Expires: 240
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 440
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces
Content-Type: application/sdp
Message Body

Frame 78 (553 bytes on wire, 553 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 407 Proxy Authentication Required
Status-Code: 407
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-d558dcd5;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-d558dcd5
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=377ea678f1613b100
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 377ea678f1613b100
To: <sip:101@192.168.2.20>
SIP to address: sip:101@192.168.2.20
SIP tag: as4380da58
Proxy-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="11dc491"
Supports: replaces
Supported: replaces
Proxy-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="11dc491"
Authentication Scheme: Digest
Algorithm: MD5
Realm: "asterisk"
Nonce Value: "11dc491"
Content-Length: 0

Frame 79 (434 bytes on wire, 434 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: ACK sip:102@192.168.2.20 SIP/2.0
Method: ACK
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-d558dc5
Transport: UDP
Sent-by: Address: 192.168.2.237
Sent-by: port: 5060
Branch: z9hG4bK-d558dc5
From: "SPA8kPhone1" <sip:101@192.168.2.20>;tag=377ea678f1613b100
SIP Display info: "SPA8kPhone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 377ea678f1613b100
To: <sip:102@192.168.2.20>;tag=as4380da58
SIP to address: sip:102@192.168.2.20
SIP tag: as4380da58
Call-ID: c677a744-912e2955@192.168.2.237
CSeq: 101 ACK
Sequence Number: 101
Method: ACK
Max-Forwards: 70
Contact: "SPA8kPhone1" <sip:101@192.168.2.237:5060>
Contact Binding: "SPA8kPhone1" <sip:101@192.168.2.237:5060>
URI: "SPA8kPhone1" <sip:101@192.168.2.237:5060>
SIP Display info: "SPA8kPhone1"
SIP contact address: sip:101@192.168.2.237:5060
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0

Frame 80 (1250 bytes on wire, 1250 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_c:33:a1
(00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: INVITE sip:102@192.168.2.20 SIP/2.0
Method: INVITE
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-e5988af5
Transport: UDP
Sent-by: Address: 192.168.2.237
Sent-by: port: 5060
Branch: z9hG4bK-e5988af5
From: "SPA8kPhone1" <sip:101@192.168.2.20>;tag=377ea678f1613b100
SIP Display info: "SPA8kPhone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 377ea678f1613b100
To: <sip:102@192.168.2.20>
SIP to address: sip:102@192.168.2.20
Remote-Party-ID: "SPA8kPhone1" <sip:101@192.168.2.20>;screen=yes;party=calling
Call-ID: c677a744-912e2955@192.168.2.237
CSeq: 102 INVITE
Sequence Number: 102
Method: INVITE
Max-Forwards: 70
Proxy-Authorization: Digest
username="101",realm="asterisk",nonce="111dc491",uri="sip:102@192.168.2.20",algorithm=MD5,resp onse="89c9bfa2d386061dad5b67972d33ec4"
Authentication Scheme: Digest
Username: "101"
Realm: "asterisk"
Nonce Value: "111dc491"
Authentication URI: "sip:102@192.168.2.20"
Algorithm: MD5
Digest Authentication Response: "89c9bfa2d386061dad5b67972d33ec4"
Contact: "SPA8kPhone1" <sip:101@192.168.2.237:5060>
Contact Binding: "SPA8kPhone1" <sip:101@192.168.2.237:5060>
URI: "SPA8kPhone1" <sip:101@192.168.2.237:5060>
SIP Display info: "SPA8kPhone1"
SIP contact address: sip:101@192.168.2.237:5060
Expires: 240
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 440
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces
Content-Type: application/sdp
Message Body
Frame 81 (470 bytes on wire, 470 bytes captured)
Ethernet II, Src: Internet_c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Configuring the Cisco SPA8800 IP Telephony Gateway in an Asterisk Environment

Status-Line: SIP/2.0 100 Trying
Status-Code: 100
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bk-e598af5;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bk-e598af5
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20;tag=377ea678f1613b1o0
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 377ea678f1613b1o0
To: <sip:102@192.168.2.20>
SIP to address: sip:102@192.168.2.20
Call-ID: c677a744-912e2955@192.168.2.237
CSeq: 102 INVITE
Sequence Number: 102
Method: INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:102@192.168.2.20>
Contact Binding: <sip:102@192.168.2.20>
URI: <sip:102@192.168.2.20>
SIP contact address: sip:102@192.168.2.20
Content-Length: 0

Frame 82 (1028 bytes on wire, 1028 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5160 (5160)
Session Initiation Protocol
Request-Line: INVITE sip:102@192.168.2.237:5160 SIP/2.0
Method: INVITE
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK05ac3a4e;rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK05ac3a4e
RPort: rport
From: "SPA8k8Phone1" <sip:101@192.168.2.20;tag=as57eae9e2
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: as57eae9e2
To: <sip:102@192.168.2.20>
Contact: <sip:101@192.168.2.20>
Contact Binding: <sip:101@192.168.2.20>
URI: <sip:101@192.168.2.20>
SIP contact address: sip:101@192.168.2.20
Call-ID: 3512588b465552ee7ec40bea27c3f72d@192.168.2.20
CSeq: 102 INVITE
Sequence Number: 102
Method: INVITE
User-Agent: Asterisk PBX
Max-Forwards: 70
Date: Fri, 05 Jun 2009 10:02:35 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: 453

Message Body

Frame 83 (486 bytes on wire, 486 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 180 Ringing
Status-Code: 180
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bk-e598af5;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bk-e598af5
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20;tag=377ea678f1613b1o0
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 377ea678f1613b10
To: <sip:102@192.168.2.20>;tag=as4fc6eca0
SIP to address: sip:102@192.168.2.20
SIP tag: as4fc6eca0
Call-ID: c677a744-912e2955@192.168.2.237
CSeq: 102 INVITE
Sequence Number: 102
Method: INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:102@192.168.2.20>
Contact Binding: <sip:102@192.168.2.20>
URI: <sip:102@192.168.2.20>
SIP contact address: sip:102@192.168.2.20
Content-Length: 0

Frame 84 (343 bytes on wire, 343 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: 5160 (5160), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Status-Code: 100
[Resent Packet: False]
Message Header
To: <sip:102@192.168.2.237:5160>
SIP to address: sip:102@192.168.2.237:5160
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as57eae9e2
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: as57eae9e2
Call-ID: 351258bb46555ee7ec40bea27c3f72d@192.168.2.20
CSeq: 102 INVITE
Sequence Number: 102
Method: INVITE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK05ac3a4e
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK05ac3a4e
Server: Cisco/SPA8800-6.1.7(GW)
Remote-Party-ID: "SPA8k8Phone1" <sip:102@192.168.2.20>;screen=yes;party=called
Content-Length: 0

Frame 85 (501 bytes on wire, 501 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: 5160 (5160), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 180 Ringing
Status-Code: 180
[Resent Packet: False]
Message Header
To: <sip:102@192.168.2.237:5160>;tag=cf247750d7f2b6e2i0
SIP to address: sip:102@192.168.2.237:5160
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as57eae9e2
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: as57eae9e2
Call-ID: 351258bb46555ee7ec40bea27c3f72d@192.168.2.20
CSeq: 102 INVITE
Sequence Number: 102
Method: INVITE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK05ac3a4e
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK05ac3a4e
Server: Cisco/SPA8800-6.1.7(GW)
Remote-Party-ID: "SPA8k8Phone2" <sip:102@192.168.2.20>;screen=yes;party=called
Content-Length: 0

Frame 86 (874 bytes on wire, 874 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: 5160 (5160), Dst Port: sip (5060)  
Session Initiation Protocol  
Status-Line: SIP/2.0 200 OK  
Status-Code: 200  
[Resent Packet: False]

Message Header  
To: <sip:102@192.168.2.237:5160>;tag=cff3a750f7f26e20  
SIP to address: sip:102@192.168.2.237:5160  
SIP tag: cff3a750f7f26e20  
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as57ea9e9e2  
SIP Display info: "SPA8k8Phone1"  
SIP from address: sip:101@192.168.2.20  
SIP tag: as57ea9e9e2  
Call-ID: 3512588b465552ee7ec40beb27c3f72d@192.168.2.20  
CSeq: 102 INVITE  
Sequence Number: 102  
Method: INVITE  
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK05ac3a4e  
Transport: UDP  
Sent-by Address: 192.168.2.20  
Sent-by port: 5060  
Branch: z9hG4bK05ac3a4e  
Contact: "SPA8k8Phone2" <sip:102@192.168.2.237:5160>  
Contact Binding: "SPA8k8Phone2" <sip:102@192.168.2.237:5160>  
URI: "SPA8k8Phone2" <sip:102@192.168.2.237:5160>  
SIP Display info: "SPA8k8Phone2"  
SIP contact address: sip:102@192.168.2.237:5160  
Server: Cisco/SPA8800-6.1.7(GW)  
Remote-Party-ID: "SPA8k8Phone2" <sip:102@192.168.2.20>;screen=yes;party=called  
Content-Length: 251  
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER  
Supported: x-sipura, replaces  
Content-Type: application/sdp

Message Body

Frame 87 (433 bytes on wire, 433 bytes captured)  
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)  
User Datagram Protocol, Src Port: 5060, Dst Port: 5060 (5060)  
Session Initiation Protocol  
Request-Line: ACK sip:102@192.168.2.237:5160 SIP/2.0  
Method: ACK  
[Resent Packet: False]  
Message Header  
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK041bb34;rport  
Transport: UDP  
Sent-by Address: 192.168.2.20  
Sent-by port: 5060  
Branch: z9hG4bK041bb34  
RPort: rport  
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as57ea9e9e2  
SIP Display info: "SPA8k8Phone1"  
SIP from address: sip:101@192.168.2.20  
SIP tag: as57ea9e9e2  
To: <sip:102@192.168.2.237:5160>;tag=cf247750f7f26e20  
SIP to address: sip:102@192.168.2.237:5160  
SIP tag: cf247750f7f26e20  
Call-ID: 3512588b465552ee7ec40beb27c3f72d@192.168.2.20  
CSeq: 102 ACK  
Sequence Number: 102  
Method: ACK  
User-Agent: Asterisk PBX  
Max-Forwards: 70  
Content-Length: 0

Frame 88 (894 bytes on wire, 894 bytes captured)  
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)  
User Datagram Protocol, Src Port: 5060, Dst Port: 5060 (5060)  
Session Initiation Protocol  
Status-Line: SIP/2.0 200 OK  
Status-Code: 200  
[Resent Packet: False]  
Message Header  
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-e5988af5;received=192.168.2.237  
Transport: UDP  
Sent-by Address: 192.168.2.237  
Sent-by port: 5060
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Branch: z9hG4bK-e5988af5
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:sip:101@192.168.2.20;tag=377e678f1613b160>
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 377e678f1613b160
To: <sip:102@192.168.2.20;tag=as4fc6eca0>
SIP to address: sip:102@192.168.2.20
SIP tag: as4fc6eca0
Call-ID: c677a744-912e20955@192.168.2.237
CSeq: 102 INVITE
Sequence Number: 102
Method: INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:102@192.168.2.20>
Contact Binding: <sip:102@192.168.2.20>
URI: <sip:102@192.168.2.20>
SIP contact address: sip:102@192.168.2.20
Content-Type: application/sdp
Content-Length: 380

Message Body

Frame 89 (597 bytes on wire, 597 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: ACK sip:102@192.168.2.20 SIP/2.0
Method: ACK
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-2deeb45a
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-2deeb45a
From: "SPA8k8Phone1" <sip:101@192.168.2.20;tag=377e678f1613b160>
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 377e678f1613b160
To: <sip:102@192.168.2.20;tag=as4fc6eca0>
SIP to address: sip:102@192.168.2.20
SIP tag: as4fc6eca0
Call-ID: c677a744-912e20955@192.168.2.237
CSeq: 102 ACK
Sequence Number: 102
Method: ACK
Max-Forwards: 70
Proxy-Authorization: Digest
username="101",realm="asterisk",nonce="111dc491",uri="sip:102@192.168.2.20",algorithm=MD5, response=89c9bfa2386061dad5b67972dd3ec44
Authentication Scheme: Digest
Username: "101"
Realm: "asterisk"
Nonce Value: "111dc491"
Authentication URI: "sip:102@192.168.2.20"
Algorithm: MD5
Digest Authentication Response: "89c9bfa2386061dad5b67972dd3ec44"
Contact: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
Contact Binding: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
URI: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
SIP Display info: "SPA8k8Phone1"
SIP contact address: sip:101@192.168.2.237:5060
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0

Frame 90 (396 bytes on wire, 396 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: BYE sip:101@192.168.2.20 SIP/2.0
Method: BYE
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5160;branch=z9hG4bK-6edcb597
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5160
Branch: z9hG4bK-6edcb597
From: <sip:102@192.168.2.237;tag=cf247750d7f2b62e10>
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SIP from address: sip:102@192.168.2.237
SIP tag: cf247750d7f26b6e10
To: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as57eae9e2
SIP Display info: "SPA8k8Phone1"
SIP to address: sip:101@192.168.2.20
SIP tag: as57eae9e2
Call-ID: 3512588b465552ee7ec40bea27c3f72d@192.168.2.20
CSeq: 101 BYE
Sequence Number: 101
Method: BYE
Max-Forwards: 70
User-Agent: Cisco/SPA8800-6.1.7(OW)
Content-Length: 0

Frame 91 (494 bytes on wire, 494 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5160 (5160)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Status-Code: 200
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5160;branch=z9hG4bK -6edcb597;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5160
Branch: z9hG4bK-6edcb597
Received: 192.168.2.237
From: <sip:102@192.168.2.237>;tag=cf247750d7f26b6e10
SIP from address: sip:102@192.168.2.237
SIP tag: cf247750d7f26b6e10
To: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as57eae9e2
SIP Display info: "SPA8k8Phone1"
SIP to address: sip:101@192.168.2.20
SIP tag: as57eae9e2
Call-ID: 3512588b465552ee7ec40bea27c3f72d@192.168.2.20
CSeq: 101 BYE
Sequence Number: 101
Method: BYE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:101@192.168.2.20>
Contact Binding: <sip:101@192.168.2.20>
URI: <sip:101@192.168.2.20>
SIP contact address: sip:101@192.168.2.20
Content-Length: 0

Frame 92 (379 bytes on wire, 379 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: BYE sip:101@192.168.2.237:5060 SIP/2.0
Method: BYE
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=zc9hG4bK7acb7fbb;rport
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK7acb7fbb
RPort: rport
From: <sip:102@192.168.2.237>;tag=as4fc6eca0
SIP from address: sip:102@192.168.2.237
SIP tag: as4fc6eca0
To: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=377ea678f1613b1o0
SIP Display info: "SPA8k8Phone1"
SIP to address: sip:101@192.168.2.20
SIP tag: 377ea678f1613b1o0
Call-ID: c677a744-912e2955@192.168.2.237
CSeq: 102 BYE
Sequence Number: 102
Method: BYE
User-Agent: Asterisk PBX
Max-Forwards: 70
Content-Length: 0

Frame 93 (338 bytes on wire, 338 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
Session Initiation Protocol

Status-Line: SIP/2.0 200 OK
Status-Code: 200

Message Header
To: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=377ea678f1613b10
SIP Display info: "SPA8k8Phone1"
SIP to address: sip:101@192.168.2.20
SIP tag: 377ea678f1613b10
From: <sip:102@192.168.2.20>;tag=as4fc6eca0
SIP from address: sip:102@192.168.2.20
SIP tag: as4fc6eca0
Call-ID: c677a744-912e2955@192.168.2.237
CSeq: 102 BYE
Sequence Number: 102
Method: BYE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK7acb7fbb
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK7acb7fbb
Server: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0

Trace of SPA8800 FXS Port Calling SPA922 IP Phone

Frame 1 (1086 bytes on wire, 1086 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: INVITE sip:201@192.168.2.20 SIP/2.0
Message Header
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 9088 9088 IN IP4 192.168.2.237
Session Name (s): -
Connection Information (c): IN IP4 192.168.2.237
Connection Network Type: IN
Connection Address Type: IP4
Connection Address: 192.168.2.237
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 16416 RTP/AVP 0 2 4 8 18 96 97 98
 100 101
  Media Attribute (a): rtpmap:0 PCMU/8000
  Media Attribute (a): rtpmap:2 G726-32/8000
  Media Attribute (a): rtpmap:4 G723/8000
  Media Attribute (a): rtpmap:8 PCMA/8000
  Media Attribute (a): rtpmap:18 G729a/8000
  Media Attribute (a): rtpmap:36 G726-40/8000
  Media Attribute (a): rtpmap:97 G726-24/8000
  Media Attribute (a): rtpmap:98 G726-16/8000
  Media Attribute (a): rtpmap:100 NSE/8000
  Media Attribute (a): fmtp:100 192-193
  Media Attribute (a): rtpmap:101 telephone-event/8000
  Media Attribute (a): fmtp:101 0-15
  Media Attribute (a): pttime:30
  Media Attribute (a): sendrecv

Frame 2 (552 bytes on wire, 552 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: ACK sip:201@192.168.2.20 SIP/2.0
Message Header
Message Body
Frame 3 (433 bytes on wire, 433 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: ACK sip:201@192.168.2.20 SIP/2.0
Message Header
Frame 4 (1249 bytes on wire, 1249 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c, Dst: Internet_1c:33:a1
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: INVITE sip:201@192.168.2.20 SIP/2.0
Message Header
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 9088 9088 IN IP4 192.168.2.237
Connection Information (c): IN IP4 192.168.2.237
Connection Network Type: IN
Connection Address Type: IP4
Connection Address: 192.168.2.237
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 16416 RTP/AVP 0 2 4 8 18 96 97 98 100 101
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:2 G726-32/8000
Media Attribute (a): rtpmap:4 G723/8000
Media Attribute (a): rtpmap:5 G722/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:9 G729a/8000
Media Attribute (a): rtpmap:10 G726-24/8000
Media Attribute (a): rtpmap:97 G726-24/8000
Media Attribute (a): rtpmap:98 G726-16/8000
Media Attribute (a): rtpmap:100 H323/8000
Media Attribute (a): fmtp:100 192-193
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmtp:101 0-15
Media Attribute (a): ptime:30
Media Attribute (a): sendrecv
Frame 5 (469 bytes on wire, 469 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Message Header
Frame 6 (1026 bytes on wire, 1026 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: CiscoLin_db:51:d2 (00:0e:08:db:51:d2)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: INVITE sip:201@192.168.2.13:5060 SIP/2.0
Message Header
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): root 7519 7519 IN IP4 192.168.2.20
Connection Information (c): IN IP4 192.168.2.20
Connection Network Type: IN
Connection Address Type: IP4
Connection Address: 192.168.2.20
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 13898 RTP/AVP 0 3 8 112 5 10 7 110 111 101
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:3 G728/8000
Media Attribute (a): rtpmap:5 G724/8000
Media Attribute (a): rtpmap:6 G729a/8000
Media Attribute (a): rtpmap:9 G723/8000
Media Attribute (a): rtpmap:10 G726-32/8000
Media Attribute (a): rtpmap:112 AAL2-G726-32/8000
Media Attribute (a): rtpmap:114 G722/8000
Media Attribute (a): rtpmap:115 G723/8000
Media Attribute (a): rtpmap:100 H323/8000
Media Attribute (a): fmtp:100 192-193
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmtp:101 0-16
Media Attribute (a): silenceSupp:off - - - -
Media Attribute (a): ptime:20
Media Attribute (a): sendrecv
Frame 7 (485 bytes on wire, 485 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 180 Ringing
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Message Header

Frame 8 (342 bytes on wire, 342 bytes captured)
Ethernet II, Src: CiscoLin_db:51:db (00:0e:08:db:51:db), Dst: Internet_1c:33:a1 (00:0e:08:db:51:db)
Internet Protocol, Src: 192.168.2.13 (192.168.2.13), Dst: 192.168.2.20 (192.168.2.20)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Message Header

Frame 9 (497 bytes on wire, 497 bytes captured)
Ethernet II, Src: CiscoLin_db:51:db (00:0e:08:db:51:db), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
Internet Protocol, Src: 192.168.2.13 (192.168.2.13), Dst: 192.168.2.20 (192.168.2.20)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 180 Ringing
Message Header

Frame 10 (813 bytes on wire, 813 bytes captured)
Ethernet II, Src: CiscoLin_db:51:db (00:0e:08:db:51:db), Dst: Internet_1c:33:a1 (00:0e:08:db:51:db)
Internet Protocol, Src: 192.168.2.13 (192.168.2.13), Dst: 192.168.2.20 (192.168.2.20)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 15796 15796 IN IP4 192.168.2.13
Session Name (s): -
Connection Information (c): IN IP4 192.168.2.13
Connection Network Type: IN
Connection Address Type: IP4
Connection Address: 192.168.2.13
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 16436 RTP/AVP 0 101
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmtp:101 0-15
Media Attribute (a): ptime:30
Media Attribute (a): sendrecv

Frame 11 (431 bytes on wire, 431 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: CiscoLin_db:51:db (00:0e:08:db:51:db)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: ACK sip:201@192.168.2.13:5060 SIP/2.0
Message Header
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): root 7519 7519 IN IP4 192.168.2.20
Session Name (s): session
Connection Information (c): IN IP4 192.168.2.20
Connection Network Type: IN
Connection Address Type: IP4
Connection Address: 192.168.2.20
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 11012 RTP/AVP 4 0 8 18 2 101
Media Attribute (a): rtpmap:4 annexa=no
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:18 G729/8000
Media Attribute (a): fmtp:18 annexb=nu
Media Attribute (a): rtpmap:2 G726-32/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmtp:101 0-16
Media Attribute (a): silenceSupp:o- - - -
Media Attribute (a): ptime:20
Media Attribute (a): sendrecv

Frame 12 (893 bytes on wire, 893 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): root 7519 7519 IN IP4 192.168.2.20
Session Name (s): session
Connection Information (c): IN IP4 192.168.2.20
Connection Network Type: IN
Connection Address Type: IP4
Connection Address: 192.168.2.20
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 11012 RTP/AVP 4 0 8 18 2 101
Media Attribute (a): rtpmap:4 annexa=no
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:18 G729/8000
Media Attribute (a): fmtp:18 annexb=nu
Media Attribute (a): rtpmap:2 G726-32/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmtp:101 0-16
Media Attribute (a): silenceSupp:o- - - -
Media Attribute (a): ptime:20
Media Attribute (a): sendrecv
Frame 13 (595 bytes on wire, 595 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1
(00:04:80:00:00:00)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: ACK sip:201@192.168.2.20 SIP/2.0
Message Header
Frame 14 (542 bytes on wire, 542 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1
(00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: BYE sip:201@192.168.2.20 SIP/2.0
Message Header
Frame 15 (477 bytes on wire, 477 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
Frame 16 (398 bytes on wire, 398 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: CiscoLin_db:51:d2
(00:0e:08:db:51:d2)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: BYE sip:201@192.168.2.13:5060 SIP/2.0
Message Header
Frame 17 (358 bytes on wire, 358 bytes captured)
Ethernet II, Src: CiscoLin_db:51:d2 (00:0e:08:db:51:d2), Dst: Internet_1c:33:a1
(00:04:80:00:00:00)
Internet Protocol, Src: 192.168.2.13 (192.168.2.13), Dst: 192.168.2.20 (192.168.2.20)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header

Trace of SPA8800 FXS 1 Making Outbound Call
Frame 1 (1104 bytes on wire, 1104 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1
(00:04:80:00:00:00)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: INVITE sip:85551313@192.168.2.20 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-61c59e9d
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-61c59e9d
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=432331171f7768d00
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 432331171f7768d00
To: <sip:85551313@192.168.2.20>
SIP to address: sip:85551313@192.168.2.20
Remote-Party-ID: "SPA8k8Phone1" <sip:101@192.168.2.20>;screen=yes;party=calling
Call-ID: c6e3a315f-e6a478e5@192.168.2.237
CSeq: 101 INVITE
Max-Forwards: 70
Contact: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
Contact Binding: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
Expires: 240
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 446
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces
Content-Type: application/sdp
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 3746361 3746361 IN IP4 192.168.2.237
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Session Name (s): -
Connection Information (c): IN IP4 192.168.2.237
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 16420 RTP/AVP 0 2 4 8 18 96 97 98

100 101
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:2 G726-32/8000
Media Attribute (a): rtpmap:4 G723/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:18 G729a/8000
Media Attribute (a): rtpmap:96 G726-40/8000
Media Attribute (a): rtpmap:97 G726-24/8000
Media Attribute (a): rtpmap:98 G726-16/8000
Media Attribute (a): rtpmap:100 NSE/8000
Media Attribute (a): fmtp:100 192-193
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmtp:101 0-15
Media Attribute (a): ptime:30
Media Attribute (a): sendrecv

Frame 2 (559 bytes on wire, 559 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 407 Proxy Authentication Required
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-61c59e9d;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-61c59e9d
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=432331171f77868do0
SIP Displays: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 432331171f77868do0
To: <sip:85551313@192.168.2.20>;tag=a0f213285
SIP to address: sip:85551313@192.168.2.20
SIP tag: a0f213285
Call-ID: c6e3a35f-0a478e5e@192.168.2.237
CSeq: 101 INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Proxy-Authorization: Digest algorithm=MD5, realm="asterisk", nonce="05celccc"
Authentication Scheme: Digest
Algorithm: MD5
Realm: "asterisk"
Nonce: Value: "05celccc"
Content-Length: 0

Frame 3 (445 bytes on wire, 445 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: ACK sip:85551313@192.168.2.20 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-61c59e9d
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-61c59e9d
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=432331171f77868do0
SIP Displays: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 432331171f77868do0
To: <sip:85551313@192.168.2.20>;tag=a0f213285
SIP to address: sip:85551313@192.168.2.20
SIP tag: a0f213285
Call-ID: c6e3a35f-0a478e5e@192.168.2.237
CSeq: 101 ACK
Max-Forwards: 70
Contact: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
Contact Binding: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
User-Agent: Cisco/SPA800-6.1.7(GW)
Content-Length: 0

Frame 4 (1272 bytes on wire, 1272 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: INVITE sip:85551313@192.168.2.20 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-2bd81ced
Transport: UDP
Sent-by: Address: 192.168.2.237
Sent-by: port: 5060
Branch: z9hG4bK-2bd81ced
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=43233171f77868d0
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 43233171f77868d0
To: <sip:85551313@192.168.2.20>
SIP to address: sip:85551313@192.168.2.20
Remote-Party-ID: "SPA8k8Phone1" <sip:101@192.168.2.20>;screen=yes;party=calling
Call-ID: c6e3a35f-e0a478e8@192.168.2.237
CSeq: 102 INVITE
Max-Forwards: 70
Proxy-AuthORIZATION: Digest
username="101",realm="asterisk",nonce="05celccoc",uri=sip:85551313@192.168.2.20”,algorithm=MD5
,response="18deca64e8376d23f2d9a7079b3eda9"
Authentication Scheme: Digest
Username: "101"
Realm: "asterisk"
Nonce Value: "05celccoc"
Authentication URI: "sip:85551313@192.168.2.20" Algorithm: MD5
Digest Authentication Response: "18deca64e8376d23f2d9a7079b3eda9"
Contact: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
Contact Binding: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
Expires: 240
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 446
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces
Content-Type: application/sdp
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): 3746361 3746361 IN IP4 192.168.2.237
Session Name (s): -
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 16420 RTP/AVP 0 2 4 8 18 96 97 98
10010
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:12 G726-12/8000
Media Attribute (a): rtpmap:4 G723/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:18 G729a/8000
Media Attribute (a): rtpmap:96 G726-40/8000
Media Attribute (a): rtpmap:97 G726-24/8000
Media Attribute (a): rtpmap:98 G726-16/8000
Media Attribute (a): rtpmap:100 NSE/8000
Media Attribute (a): ffmt:100 192-193
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): ffmt:101 0-15
Media Attribute (a):ptime:30
Media Attribute (a):sendrecv
Frame 5 (481 bytes on wire, 481 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-2bd81ced;received=192.168.2.237
Transport: UDP
Sent-by: Address: 192.168.2.237
Sent-by: port: 5060
Branch: z9hG4bK-2bd81ced
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=43233171f77868d0
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 43233171f77868d0
To: <sip:85551313@192.168.2.20>
SIP to address: sip:85551313@192.168.2.20
Call-ID: c6e3a35f-e0a478e8@192.168.2.237
CSeq: 102 INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:85551313@192.168.2.20>
Contact Binding: <sip:85551313@192.168.2.20>
Content-Length: 0

Frame 6 (1036 bytes on wire, 1036 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5161 (5161)
Session Initiation Protocol
Request-Line: INVITE sip:5551313@192.168.2.237:5161 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK18bc3f3d;rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK18bc3f3d
RPort: rport
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as7b52add2
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: as7b52add2
To: <sip:5551313@192.168.2.237:5161>
SIP to address: sip:5551313@192.168.2.237:5161
Contact: <sip:101@192.168.2.20>
Contact Binding: <sip:101@192.168.2.20>
Call-ID: 3b967441067a3be6785c0606e957a@192.168.2.20
CSeq: 102 INVITE
User-Agent: Asterisk PBX
Max-Forwards: 70
Date: Fri, 05 Jun 2009 20:26:11 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: 453
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): root 7519 7519 IN IP4 192.168.2.20
Session Name (s): session
Connection Information (c): IN IP4 192.168.2.20
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 13440 RTP/AVP 0 3 8 112 5 10 7 110

111 101
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:3 GSM/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:112 AAL2-G726-32/8000
Media Attribute (a): rtpmap:5 DVI4/8000
Media Attribute (a): rtpmap:10 L16/8000
Media Attribute (a): rtpmap:7 LPC/8000
Media Attribute (a): rtpmap:110 speex/8000
Media Attribute (a): rtpmap:111 G726-32/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmt:101 0-16
Media Attribute (a): silenceSupp:off - - - -
Media Attribute (a):ptime:20
Media Attribute (a): sendsrecv

Frame 7 (497 bytes on wire, 497 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 180 Ringing
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-2bd81ced;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-2bd81ced
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=432331171f77868d0
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 432331171f77868d0
To: <sip:5551313@192.168.2.20>
SIP to address: sip:5551313@192.168.2.20
Call-ID: c6e3a35f-e0a478e5@192.168.2.237
CSeq: 102 INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:85551313@192.168.2.20>
Contact Binding: <sip:85551313@192.168.2.20>
Content-Length: 0

Frame 8 (347 bytes on wire, 347 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1
(00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: 5161 (5161), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Message Header
To: <sip:85551313@192.168.2.237:5161>
SIP to address: sip:85551313@192.168.2.237:5161
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as7b52add2
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: as7b52add2
Call-ID: 3db97411067a5b6e75487f7a0e192.168.2.20
CSeq: 102 INVITE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK18bc3f3d
Transport: UD
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK18bc3f3d
Server: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0

Frame 9 (886 bytes on wire, 886 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1
(00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: 5161 (5161), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
To: <sip:85551313@192.168.2.237:5161>
SIP to address: sip:85551313@192.168.2.237:5161
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as7b52add2
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: as7b52add2
Call-ID: 3db97411067a5b6e75487f7a0e192.168.2.20
CSeq: 102 INVITE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK18bc3f3d
Transport: UD
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK18bc3f3d
Contact: "SPA8k8Line2" <sip:85551313@192.168.2.237:5161>
Contact Binding: "SPA8k8Line2" <sip:85551313@192.168.2.237:5161>
Server: Cisco/SPA8800-6.1.7(GW)
Remote-Party-ID: "SPA8k8Line2" <sip:psn@192.168.3.2>;screen=yes;party=called
Content-Length: 257
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces
Content-Type: application/sdp

Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 3745693 3745693 IN IP4 192.168.2.237
Session Name [s]: -
Connection Information (c): IN IP4 192.168.2.237
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 17475 RTP/AVP 0 100 101
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:100 NSE/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmt:100 192-193
Media Attribute (a): fmt:101 0-15
Media Attribute (a): pttime:30
Media Attribute (a): sendrecv

Frame 10 (440 bytes on wire, 440 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: ACK sip:85551313@192.168.2.237:5161 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK538bb7bc;rport
Transport: UD
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Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK538bb7bc
RPort: rport
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as7b52add2
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: as7b52add2
To: <sip:5551313@192.168.2.237:5161>;tag=7777c28b5e7542i1
SIP to address: sip:5551313@192.168.2.237:5161
SIP tag: e7777c28b5e7542i1
Contact: <sip:101@192.168.2.20>
Contact Binding: <sip:101@192.168.2.20>
Call-ID: 3db967441067a5b61785c06e9f7a08192.168.2.20
CSeq: 102 ACK
User-Agent: Asterisk PBX
Max-Forwards: 70
Content-Length: 0

Frame 11 (905 bytes on wire, 905 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-2bd81ced;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-2bd81ced
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=432331171f77868do0
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 432331171f77868do0
To: <sip:85551313@192.168.2.20>;tag=as62288bf1
SIP to address: sip:85551313@192.168.2.20
SIP tag: as62288bf1
Call-ID: c6e3a35f-e0a478e5@192.168.2.237
CSeq: 102 INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:85551313@192.168.2.20>
Contact Binding: <sip:85551313@192.168.2.20>
Content-Type: application/sdp
Content-Length: 380

Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): root 7519 7519 IN IP4 192.168.2.20
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 17642 RTP/AVP 4 0 8 10 1 0
Media Attribute (a): rtpmap:4 G723/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:18 G729/8000
Media Attribute (a): rtpmap:2 G726-12/8000
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:18 G729/8000
Media Attribute (a): rtpmap:18 annexb=no
Media Attribute (a): rtpmap:12 G726-12/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): rtpmap:101 annexb=no
Media Attribute (a): silenceSupp:off - - - -
Media Attribute (a): ptime:20
Media Attribute (a): sendrecv

Frame 12 (613 bytes on wire, 613 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: ACK sip:85551313@192.168.2.20 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-c4409cb5
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-c4409cb5
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=432331171f77868do0
SIP Display info: "SPA8k8Phone1"
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SIP from address: sip:101@192.168.2.20
SIP tag: 432331171f77868d0
To: <sip:85551313@192.168.2.20>;tag=as62288bf1
SIP to address: sip:85551313@192.168.2.20
SIP tag: as62288bf1
Call-ID: c6e3a35f-e0a478e5@192.168.2.237
CSeq: 102 ACK
Max-Forwards: 70
Proxy-Authorization: Digest
username="101",realm="asterisk",nonce="05celcc",url="sip:85551313@192.168.2.20",algorithm=MD5
,response="18deca64e8376d23f2d29a7079b0leda9" Authentication Scheme: Digest
Username: "101"
Realm: "asterisk"
Nonce Value: "05celcc"
Authentication URI: "sip:85551313@192.168.2.20"
Algorithm: MD5
Digest Authentication Response: "18deca64e8376d23f2d29a7079b0leda9"
Contact: "SPA8k8Phone1" <sip:101@192.168.2.237;5060>
Contact Binding: "SPA8k8Phone1" <sip:101@192.168.2.237;5060>
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0

Frame 13 (559 bytes on wire, 559 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: BYE sip:85551313@192.168.2.20 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-c568e4e6
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-c568e4e6
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=432331171f77868d0
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 432331171f77868d0
To: <sip:85551313@192.168.2.20>;tag=as62288bf1
SIP to address: sip:85551313@192.168.2.20
SIP tag: as62288bf1
Call-ID: c6e3a35f-e0a478e5@192.168.2.237
CSeq: 103 BYE
Max-Forwards: 70
Proxy-Authorization: Digest
username="101",realm="asterisk",nonce="05celcc",url="sip:85551313@192.168.2.20",algorithm=MD5
,response="1ff5a6e1706df31ebf7fbb3070a2438" Authentication Scheme: Digest
Username: "101"
Realm: "asterisk"
Nonce Value: "05celcc"
Authentication URI: "sip:85551313@192.168.2.20"
Algorithm: MD5
Digest Authentication Response: "1ff5a6e1706df31ebf7fbb3070a2438"
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0

Frame 14 (489 bytes on wire, 489 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK-c568e4e6;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5060
Branch: z9hG4bK-c568e4e6
Received: 192.168.2.237
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=432331171f77868d0
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: 432331171f77868d0
To: <sip:85551313@192.168.2.20>;tag=as62288bf1
SIP to address: sip:85551313@192.168.2.20
SIP tag: as62288bf1
Call-ID: c6e3a35f-e0a478e5@192.168.2.237
CSeq: 103 BYE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
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Contact: <sip:85551313@192.168.2.20>
Contact Binding: <sip:85551313@192.168.2.20>
Content-Length: 0

Frame 15 (407 bytes on wire, 407 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5161 (5161)
Session Initiation Protocol
Request-Line: BYE sip:5551313@192.168.2.237:5161 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK28e5ba05;rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK28e5ba05
RPort: rport
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as7b52add2
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: as7b52add2
To: <sip:5551313@192.168.2.237:5161>;tag=e7777c28b5e754211
SIP to address: sip:5551313@192.168.2.237:5161
SIP tag: e7777c28b5e754211
Call-ID: 3db67441067a5b617b5e695f7a0@192.168.2.20
CSeq: 103 BYE
User-Agent: Asterisk PBX
Max-Forwards: 70
Content-Length: 0

Frame 16 (362 bytes on wire, 362 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: 5161 (5161), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
To: <sip:5551313@192.168.2.237:5161>;tag=e7777c28b5e754211
SIP to address: sip:5551313@192.168.2.237:5161
SIP tag: e7777c28b5e754211
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as7b52add2
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.20
SIP tag: as7b52add2
Call-ID: 3db67441067a5b617b5e695f7a0@192.168.2.20
CSeq: 103 BYE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK28e5ba05
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK28e5ba05
Server: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0
Trace of SPA8800 FXS Receiving Inbound Call

Frame 1 (1093 bytes on wire, 1093 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1
(00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: 5161 (5161), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: INVITE sip:101@192.168.2.20 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5161|branch=z9hG4bK-ac9b45ce
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5161
Branch: z9hG4bK-ac9b45ce
From: "SPA8k8Line2" <sip:pstn2@192.168.3.2>;tag=26caa2f19d9f62e01
SIP Display info: "SPA8k8Line2"
SIP from address: sip:pstn2@192.168.2.237
SIP tag: 26caa2f19d9f62e01
To: <sip:101@192.168.2.20>
SIP to address: sip:101@192.168.2.20
Remote-Party-ID: "SPA8k8Line2" <sip:pstn2@192.168.3.2>;screen=yes;party=calling
Call-ID: ab1e6542-d868981e@192.168.3.2
CSeq: 101 INVITE
Sequence Number: 101
Method: INVITE
Max-Forwards: 70
Contact: "SPA8k8Line2" <sip:pstn2@192.168.2.237:5161>
Contact Binding: "SPA8k8Line2" <sip:pstn2@192.168.2.237:5161>
URI: "SPA8k8Line2" <sip:pstn2@192.168.2.237:5161>
SIP Display info: "SPA8k8Line2"
SIP contact address: sip:pstn2@192.168.2.237:5161
Expires: 240
User-Agent: Cisco/SPA8800-6.1.7(GW)
Content-Length: 446
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces
Content-Type: application/sdp
Message Body

Frame 2 (469 bytes on wire, 469 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5161 (5161)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK6dbb2197;rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5161
Branch: z9hG4bK-ac9b45ce
Received: 192.168.2.237
From: "SPA8k8Line2" <sip:pstn2@192.168.3.2>;tag=26caa2f19d9f62e01
SIP Display info: "SPA8k8Line2"
SIP from address: sip:pstn2@192.168.2.237
SIP tag: 26caa2f19d9f62e01
To: <sip:101@192.168.2.20>
SIP to address: sip:101@192.168.2.20
Call-ID: ab1e6542-d868981e@192.168.3.2
CSeq: 101 INVITE
Sequence Number: 101
Method: INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:101@192.168.2.20>
Contact Binding: <sip:101@192.168.2.20>
URI: <sip:101@192.168.2.20>
SIP contact address: sip:101@192.168.2.20
Content-Length: 0

Frame 3 (1031 bytes on wire, 1031 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: INVITE sip:101@192.168.2.237:5060 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK6dbb2197;rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK6dbb2197
RPort: rport
From: "SPA8k8Line2" <sip:psn2@192.168.2.20>;tag=as7100dbb6
SIP Display info: "SPA8k8Line2"
SIP from address: sip:psn2@192.168.2.20
SIP tag: as7100dbb6
To: <sip:101@192.168.2.237:5060>
SIP to address: sip:101@192.168.2.237:5060
Contact: <sip:psn2@192.168.2.20>
Contact Binding: <sip:psn2@192.168.2.20>
URI: <sip:psn2@192.168.2.20>
SIP contact address: sip:psn2@192.168.2.20
Call-ID: 1b7c71b81593c9cf71d3add64041e610@192.168.2.20
CSeq: 102 INVITE
Sequence Number: 102
Method: INVITE
User-Agent: Asterisk PBX
Max-Forwards: 70
Date: Fri, 05 Jun 2009 20:28:20 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: 453
Message Body

Frame 4 (485 bytes on wire, 485 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5161 (5161)
Session Initiation Protocol
Status-Line: SIP/2.0 180 Ringing
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5161;branch=z9hG4bK-ac9b45ce;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5161
Branch: z9hG4bK-ac9b45ce
Received: 192.168.2.237
From: "SPA8k8Line2" <sip:psn2@192.168.3.2>;tag=26caa2f219d9f620e1
SIP Display info: "SPA8k8Line2"
SIP from address: sip:psn2@192.168.3.2
SIP tag: 26caa2f219d9f620e1
To: <sip:101@192.168.2.20>;tag=as7788891d
SIP to address: sip:101@192.168.2.20
SIP tag: as7788891d
Call-ID: ab1e6542-d86981e8@192.168.3.2
CSeq: 101 INVITE
Sequence Number: 101
Method: INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:101@192.168.2.20>
Contact Binding: <sip:101@192.168.2.20>
URI: <sip:101@192.168.2.20>
SIP contact address: sip:101@192.168.2.20
Content-Length: 0

Frame 5 (344 bytes on wire, 344 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 100 Trying
Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK6dbb2197
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK6dbb2197
Server: Cisco/SPA8800-6.1.7 (GW)
Content-Length: 0
Configuring the Cisco SPA8800 IP Telephony Gateway in an Asterisk Environment

Frame 6 (502 bytes on wire, 502 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 180 Ringing
Message Header
To: <sip:101@192.168.2.237:5060>;tag=ec5f76c3013369e10
SIP to address: sip:101@192.168.2.237:5060
SIP tag: ec5f76c3013369e10
From: "SPA8k8Line2" <sip:pstn2@192.168.2.20>;tag=as7100db6
SIP Display info: "SPA8k8Line2"
SIP from address: sip:pstn2@192.168.2.20
SIP tag: as7100db6
Call-ID: 1b7c71b1593c9cf71d3add640f41e610192.168.2.20
CSeq: 102 INVITE
Sequence Number: 102
Method: INVITE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK6dbb2197
Transport: UDPha: 192.168.2.20:5060
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK6dbb2197
Contact: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
Contact Binding: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
URI: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.237:5060
SIP tag: as7100db6
Remote-Party-ID: "SPA8k8Phone1" <sip:101@192.168.2.20>;screen=yes;party=called
Content-Length: 0
Frame 7 (881 bytes on wire, 881 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
To: <sip:101@192.168.2.237:5060>;tag=ec5f76c3013369e10
SIP to address: sip:101@192.168.2.237:5060
SIP tag: ec5f76c3013369e10
From: "SPA8k8Line2" <sip:pstn2@192.168.2.20>;tag=as7100db6
SIP Display info: "SPA8k8Line2"
SIP from address: sip:pstn2@192.168.2.20
SIP tag: as7100db6
Call-ID: 1b7c71b1593c9cf71d3add640f41e610192.168.2.20
CSeq: 102 INVITE
Sequence Number: 102
Method: INVITE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK503cc10c;rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK503cc10c
Contact: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
Contact Binding: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
URI: "SPA8k8Phone1" <sip:101@192.168.2.237:5060>
SIP Display info: "SPA8k8Phone1"
SIP from address: sip:101@192.168.2.237:5060
SIP tag: as7100db6
Remote-Party-ID: "SPA8k8Phone1" <sip:101@192.168.2.20>;screen=yes;party=called
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces
Content-Type: application/sdp
Message Body
Frame 8 (436 bytes on wire, 436 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: ACK sip:101@192.168.2.237:5060 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK503cc10c;rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK503cc10c
RPort: rport
From: "SPA8k8Line2" <sip:pstn2@192.168.2.20>;tag=as7100db6
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SIP Display info: "SPA8k8Line2"
SIP from address: sip:pstn2@192.168.2.20
SIP tag: as7100bd6
To: <sip:101@192.168.2.237:5060>;tag=ecf57f6c3013369e10
SIP to address: sip:101@192.168.2.237:5060
SIP tag: ecf57f6c3013369e10
Contact: <sip:pstn2@192.168.2.20>
Contact Binding: <sip:pstn2@192.168.2.20>
URI: <sip:pstn2@192.168.2.20>
SIP contact address: sip:pstn2@192.168.2.20
Call-ID: 1b7c71b81593c9cf71d3add64df41e610192.168.2.20
CSeq: 102 ACK
Sequence Number: 102
Method: ACK
User-Agent: Asterisk PBX
Max-Forwards: 70
Content-Length: 0
Frame 9 (893 bytes on wire, 893 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5161 (5161)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5161;branch=z9hG4bK-ac9b45ce
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5161
Branch: z9hG4bK-ac9b45ce
Received: 192.168.2.237
From: "SPA8k8Line2" <sip:pstn2@192.168.2.20:5161>;tag=26caa2f219d962e01
SIP Display info: "SPA8k8Line2"
SIP from address: sip:pstn2@192.168.2.20
SIP tag: 26caa2f219d962e01
To: <sip:101@192.168.2.20>;tag=as7788891d
SIP to address: sip:101@192.168.2.20
SIP tag: as7788891d
Call-ID: able6542-d868901e@192.168.3.2
CSeq: 101 INVITE
Sequence Number: 101
Method: INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:101@192.168.2.20>
Contact Binding: <sip:101@192.168.2.20>
URI: <sip:101@192.168.2.20>
Content-Type: application/sdp
Content-Length: 380
Message Body
Frame 10 (434 bytes on wire, 434 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1
(00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5161 (5161)
Session Initiation Protocol
Request-Line: ACK sip:101@192.168.2.20 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5161;branch=z9hG4bK-8be8524e
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5161
Branch: z9hG4bK-8be8524e
From: "SPA8k8Line2" <sip:pstn2@192.168.2.20:5161>;tag=26caa2f219d962e01
SIP Display info: "SPA8k8Line2"
SIP from address: sip:pstn2@192.168.2.20
SIP tag: 26caa2f219d962e01
To: <sip:101@192.168.2.20>;tag=as7788891d
SIP to address: sip:101@192.168.2.20
SIP tag: as7788891d
Call-ID: able6542-d868901e@192.168.3.2
CSeq: 101 ACK
Sequence Number: 101
Method: ACK
Max-Forwards: 70
Contact: "SPA8k8Line2" <sip:pstn2@192.168.2.237:5161>
Contact Binding: "SPA8k8Line2" <sip:pstn2@192.168.2.237:5161>
URI: "SPA8k8Line2" <sip:pstn2@192.168.2.237:5161>
SIP Display info: "SPA8k8Line2"
SIP contact address: sip:pstn2@192.168.2.237:5161
User-Agent: Cisco/SPA8800-6.1.7(GW)
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Frame 11 (550 bytes on wire, 550 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5160 (5160)
Session Initiation Protocol
Request-Line: OPTIONS sip:102@192.168.2.237:5160 SIP/2.0

Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bkK44611a08;rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9H4bkK44611a08
RPort: rport
From: "asterisk" <sip:asterisk@192.168.2.20>;tag=as5841881b
SIP Display info: "asterisk"
SIP from address: sip:asterisk@192.168.2.20
SIP tag: as5841881b
To: <sip:102@192.168.2.237:5160>
SIP to address: sip:102@192.168.2.237:5160
Contact: <sip:asterisk@192.168.2.20>
Contact Binding: <sip:asterisk@192.168.2.20>
URI: <sip:asterisk@192.168.2.20>
SIP contact address: sip:asterisk@192.168.2.20
Call-ID: 755faa340f75586929e34da01c4470b3@192.168.2.20
CSeq: 102 OPTIONS
Sequence Number: 102
Method: OPTIONS
User-Agent: Asterisk PBX
Max-Forwards: 70
Date: Fri, 05 Jun 2009 20:28:34 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Length: 0

Frame 12 (458 bytes on wire, 458 bytes captured)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK

Message Header
To: <sip:102@192.168.2.237:5160>;tag=3ac5a2f21de19e6ei0
SIP to address: sip:102@192.168.2.237:5160
SIP tag: 3ac5a2f21de19e6ei0
From: "asterisk" <sip:asterisk@192.168.2.20>;tag=as5841881b
SIP Display info: "asterisk"
SIP from address: sip:asterisk@192.168.2.20
SIP tag: as5841881b
Call-ID: 755faa340f75586929e34da01c4470b3@192.168.2.20
CSeq: 102 OPTIONS
Sequence Number: 102
Method: OPTIONS
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bkK44611a08
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9H4bkK44611a08
Server: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces

Frame 13 (550 bytes on wire, 550 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: OPTIONS sip:101@192.168.2.237:5060 SIP/2.0

Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bkK143a03f0;rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9H4bkK143a03f0
RPort: rport
From: "asterisk" <sip:asterisk@192.168.2.20>;tag=as4f0ca42b
SIP Display info: "asterisk"
SIP from address: sip:asterisk@192.168.2.20
SIP tag: as4f0ca42b
To: <sip:101@192.168.2.237:5060>
SIP to address: sip:101@192.168.2.237:5060
Contact: <sip:asterisk@192.168.2.20>
Contact Binding: <sip:asterisk@192.168.2.20>
URI: <sip:asterisk@192.168.2.20>
SIP contact address: sip:asterisk@192.168.2.20
Call-ID: 7614b5463341f49f6ef923c247589ef0192.168.2.20
CSeq: 102 OPTIONS
Sequence Number: 102
Method: OPTIONS
User-Agent: Asterisk PBX
Max-Forwards: 70
Date: Fri, 05 Jun 2009 20:28:34 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Length: 0

Frame 14 (458 bytes on wire, 458 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
To: <sip:101@192.168.2.237:5060>;tag=a302c7faf51b7c8ei0
SIP to address: sip:101@192.168.2.237:5060
SIP tag: a302c7faf51b7c8ei0
From: "asterisk" <sip:asterisk@192.168.2.20>;tag=as4f0ca42b
SIP Display info: "asterisk"
SIP from address: sip:asterisk@192.168.2.20
SIP tag: as4f0ca42b
Call-ID: 7614b5463341f49f6ef923c247589ef0192.168.2.20
CSeq: 102 OPTIONS
Sequence Number: 102
Method: OPTIONS
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK143a03f0
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK143a03f0
Server: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura, replaces

Frame 15 (542 bytes on wire, 542 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: Cisco-Li_9c:e3:2c (00:1d:7e:9c:e3:2c)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 147.135.32.221 (147.135.32.221)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: OPTIONS sip:sip.broadvoice.com SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK178d807d/rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK178d807d
RPort: rport
From: "asterisk" <sip:asterisk@192.168.2.20>;tag=as584485d8
SIP Display info: "asterisk"
SIP from address: sip:asterisk@192.168.2.20
SIP tag: as584485d8
To: <sip:sip.broadvoice.com>
SIP to address: sip:sip.broadvoice.com
Contact: <sip:asterisk@192.168.2.20>
Contact Binding: <sip:asterisk@192.168.2.20>
URI: <sip:asterisk@192.168.2.20>
SIP contact address: sip:asterisk@192.168.2.20
Call-ID: 657fd4a435b5be15c73d1346250c50@192.168.2.20
CSeq: 102 OPTIONS
Sequence Number: 102
Method: OPTIONS
User-Agent: Asterisk PBX
Max-Forwards: 70
Date: Fri, 05 Jun 2009 20:28:34 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Length: 0

Frame 16 (476 bytes on wire, 476 bytes captured)
Ethernet II, Src: Cisco-Li_9c:e3:2c (00:1d:7e:9c:e3:2c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
Internet Protocol, Src: 147.135.32.221 (147.135.32.221), Dst: 192.168.2.20 (192.168.2.20)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Configuring the Cisco SPA8800 IP Telephony Gateway in an Asterisk Environment

Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
Call-ID: 65f7d74a435b5be15c73d1341625c0c5@192.168.2.20
CSeq: 102 OPTIONS
Sequence Number: 102
Method: OPTIONS
From: "asterisk" <sip:asterisk@192.168.2.20>;tag=as584485d8
SIP Display info: "asterisk"
SIP from address: sip:asterisk@192.168.2.20
SIP tag: as584485d8
To: <sip:sip.broadvoice.com>
SIP to address: sip:sip.broadvoice.com
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK178d807d;received=24.153.145.213;rport=33579
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK178d807d
Received: 24.153.145.213
RPort: 33579
Supported: 100rel
Allow: INVITE, BYE, ACK, OPTIONS, CANCEL, PRACK
Accept: application/sdp
Accept-Encoding:
Accept-Language: en
Content-Length: 0

Frame 17 (379 bytes on wire, 379 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: 5161 (5161), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: BYE sip:101@192.168.2.20 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5161;branch=z9hG4bK-cdeab552
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5161
Branch: z9hG4bK-cdeab552
From: "SPA8k8Line2" <sip:pstn2@192.168.3.2>;tag=26caa2f219d9f62e01
SIP Display info: "SPA8k8Line2"
SIP from address: sip:pstn2@192.168.3.2
SIP tag: 26caa2f219d9f62e01
To: <sip:101@192.168.2.20>;tag=as7788891d
SIP to address: sip:101@192.168.2.20
SIP tag: as7788891d
Call-ID: ab1e6542-d868981e@192.168.3.2
CSeq: 102 BYE
Sequence Number: 102
Method: BYE
Max-Forwards: 70
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:101@192.168.2.20>

Frame 18 (477 bytes on wire, 477 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5161 (5161)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
Via: SIP/2.0/UDP 192.168.2.237:5161;branch=z9hG4bK-cdeab552;received=192.168.2.237
Transport: UDP
Sent-by Address: 192.168.2.237
Sent-by port: 5161
Branch: z9hG4bK-cdeab552
Received: 192.168.2.237
From: "SPA8k8Line2" <sip:pstn2@192.168.3.2>;tag=26caa2f219d9f62e01
SIP Display info: "SPA8k8Line2"
SIP from address: sip:pstn2@192.168.3.2
SIP tag: 26caa2f219d9f62e01
To: <sip:101@192.168.2.20>;tag=as7788891d
SIP to address: sip:101@192.168.2.20
SIP tag: as7788891d
Call-ID: ab1e6542-d868981e@192.168.3.2
CSeq: 102 BYE
Sequence Number: 102
Method: BYE
User-Agent: Cisco/SPA8800-6.1.7(GW)
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:101@192.168.2.20>
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Contact Binding: <sip:101@192.168.2.20>
URI: <sip:101@192.168.2.20>
SIP contact address: sip:101@192.168.2.20
Content-Length: 0

Frame 19 (401 bytes on wire, 401 bytes captured)
Ethernet II, Src: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c (00:24:97:f0:50:3c)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: BYE sip:101@192.168.2.237:5060 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK6006b167;rport
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK6006b167
RPort: rport
From: "SPA8k8Line2" <sip:pstn2@192.168.2.20>;tag=as7100bd6
SIP Display info: "SPA8k8Line2" SIP from address: sip:pstn2@192.168.2.20
SIP tag: as7100bd6
To: <sip:101@192.168.2.237:5060>;tag=ecf57f6c3013369e10
SIP to address: sip:101@192.168.2.237:5060
SIP tag: ecf57f6c3013369e10
Call-ID: 1b7c71b185193c9cf1d3add640f41e610192.168.2.20
CSeq: 103 BYE
Sequence Number: 103
Method: BYE
User-Agent: Asterisk PBX
Max-Forwards: 70
Content-Length: 0

Frame 20 (360 bytes on wire, 360 bytes captured)
Ethernet II, Src: 00:24:97:f0:50:3c (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1 (00:e0:4d:1c:33:a1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
To: <sip:101@192.168.2.237:5060>;tag=ecf57f6c3013369e10
SIP to address: sip:101@192.168.2.237:5060
SIP tag: ecf57f6c3013369e10
From: "SPA8k8Line2" <sip:pstn2@192.168.2.20>;tag=as7100bd6
SIP Display info: "SPA8k8Line2" SIP from address: sip:pstn2@192.168.2.20
SIP tag: as7100bd6
Call-ID: 1b7c71b185193c9cf1d3add640f41e610192.168.2.20
CSeq: 103 BYE
Sequence Number: 103
Method: BYE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK6006b167
Transport: UDP
Sent-by Address: 192.168.2.20
Sent-by port: 5060
Branch: z9hG4bK6006b167
Server: Cisco/SPA8800-6.1.7(GW)
Content-Length: 0
Trace of FAX Line Toggle Code #99

Following are two trace segments of a SIP INVITE showing the difference in the INVITE when #99 is dialed.

This trace shows the SDP information from a call made where 85551212 was dialed where 8 is a steering digit. Notice that the media type is audio and that audio codecs are listed in the media formats.

```
| Media Type: audio |
| Media Port: 16418 |
| Media Protocol: RTP/AVP |
| Media Format: ITU-T G.711 PCMU |
| Media Format: ITU-T G.721 |
| Media Format: ITU-T G.723 |
| Media Format: ITU-T G.711 PCMA |
| Media Format: ITU-T G.729 |
| Media Format: 96 |
| Media Format: 97 |
| Media Format: 98 |
| Media Format: 100 |
| Media Format: 101 |
```

This second trace shows the SDP information from a call made where #9985551212 was dialed where 8 is a steering digit. Notice that the media type is image and the media format is t38 etc.

```
| Media Type: image |
| Media Port: 16422 |
| Media Protocol: udp-tpl |
| Media Format: t38 |
| Media Attribute (a): T38FaxVersion:0 |
| Media Attribute Fieldname: T38FaxVersion |
| Media Attribute Value: 0 |
| Media Attribute (a): T38MaxBitRate:14400 |
| Media Attribute Fieldname: T38MaxBitRate |
| Media Attribute Value: 14400 |
| Media Attribute (a): T38FaxRateManagement:transferredTCF |
| Media Attribute Fieldname: T38FaxRateManagement |
| Media Attribute Value: transferredTCF |
| Media Attribute (a): T38FaxMaxBuffer:200 |
| Media Attribute Fieldname: T38FaxMaxBuffer |
| Media Attribute Value: 200 |
| Media Attribute (a): T38FaxMaxDatagram:200 |
| Media Attribute Fieldname: T38FaxMaxDatagram |
| Media Attribute Value: 200 |
```

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Gathering Information for Support

In the event that you need to reach out for support, collect the following information first:

A. SPA8800's configuration

   Web-UI > Admin Login > Advanced >

   Voice tab

   Browser > File > Save As > [save entire page as SPA8800Voice.html]

   Network tab:

   Browser > File > Save As > [save entire page as SPA8800Network.html]

B. SPA8800 syslog log from debug output:

   Web-UI > Admin Login > Advanced >

   System tab > Syslog & Debug Server: and Debug Level: 3

   Line N > SIP Debug Option:

   Configuring this is described fully at: https://www.myciscocommunity.com/docs/DOC-5405

C. Voice tab

D. Asterisk sip.conf

E. Asterisk extensions.conf

F. WireShark trace