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

Table of Contents

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
- Asterisk Book from O'Reilly: http://www.asteriskdocs.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

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

```
# 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

username=101

secret=101

qualify=yes

nat=no

host=dynamic

canreinvite=no

context=fxsgroup

regext=101

;

;define SPA8800 analog phone 2 extension 102

[102]

type=friend

username=102

secret=102

qualify=yes

nat=no

host=dynamic

canreinvite=no

context=fxsgroup

regext=102

;

;define SPA8800 analog phone 3 extension 103

[103]

type=friend

username=103

secret=103

qualify=yes

nat=no

host=dynamic
```



```
canreinvite=no
context=fxsgroup
regext=103
;
;define SPA8800 analog phone 4 extension 104
[104]
type=friend
username=104
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=3612887272
secret=TitSSTQqpp
username=3612887272
insecure=very
authname=3612887272
dtmfmode=inband
dtmf=inband
canreinvite=no
qualify=yes
```

```
nat=yes
context=itsp1
;eof
```

Editing the /etc/asterisk/extensions.conf file

```
# vim /etc/asterisk/extensions.conf
…
…
; SPA8800 Changes
;outbound dialing
[fxsgroup]
;
;
; dial 7 to explicitly use FXS3
exten => _7.,1,Dial(SIP/${EXTEN:1}@pstn3,60,r)
;
; dial 8 as a steering digit:
; if FXS2 is not available, FXS3 will be used.
; if FXS3 is not available, the user hears congestion
exten => _8.,1,Dial(SIP/${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 => 3615551212,1,Answer
;enable ring group of extensions 101, 102, and 200
exten => 3615551212,2,Dial(SIP/101&SIP/102&SIP/200,25,rt)
exten => 3615551212,3,Hangup
;eof
…
…
```

Loading the Modified Configuration

1. Connect to the Asterisk console:

```
$ 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

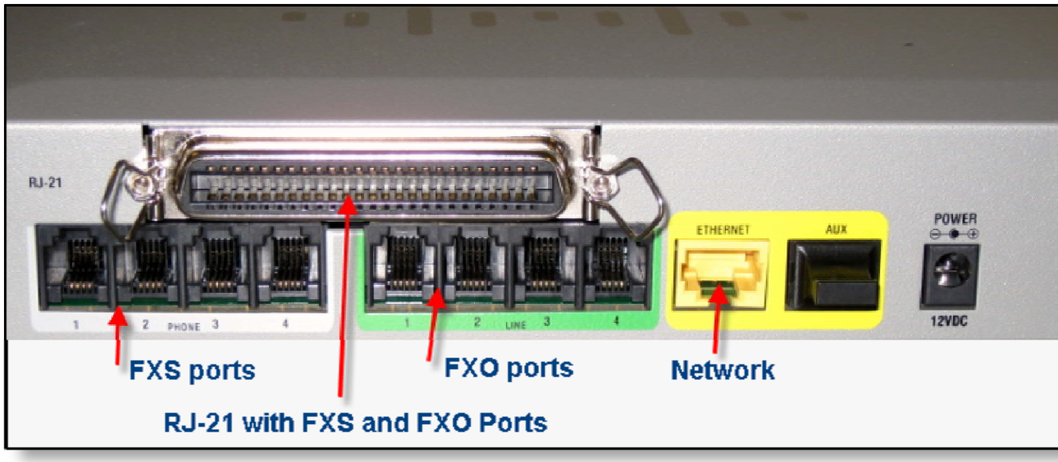


Figure 1The photograph shows the rear of the SPA8800 and its connections

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.

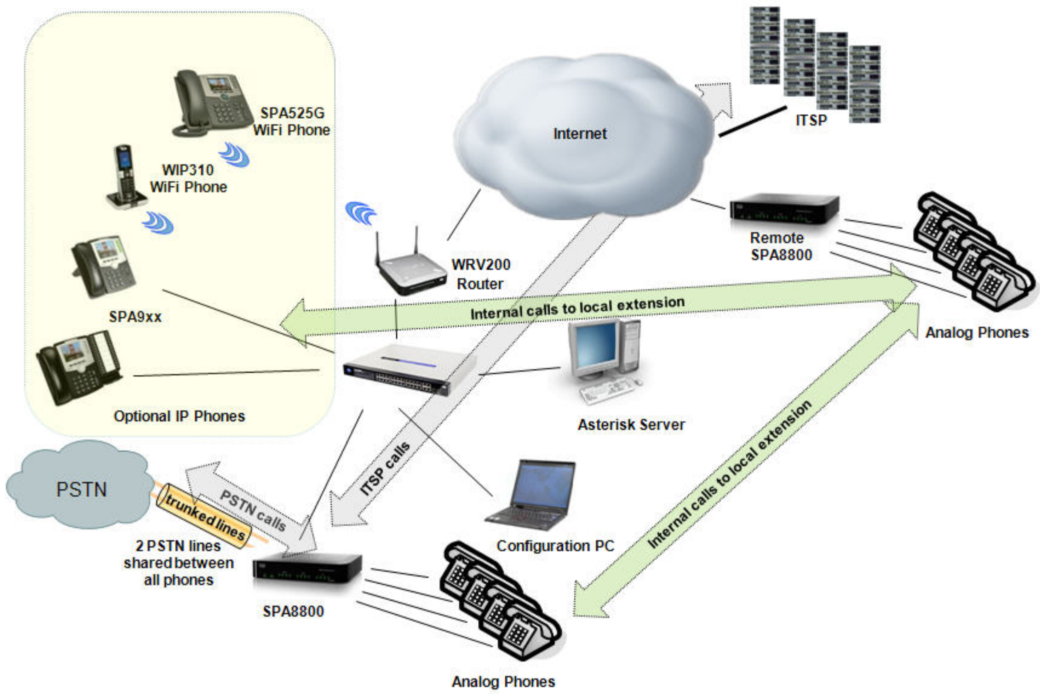


Figure 2: An example of deploying 2 SPA8800s in an Asterisk Environment

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:

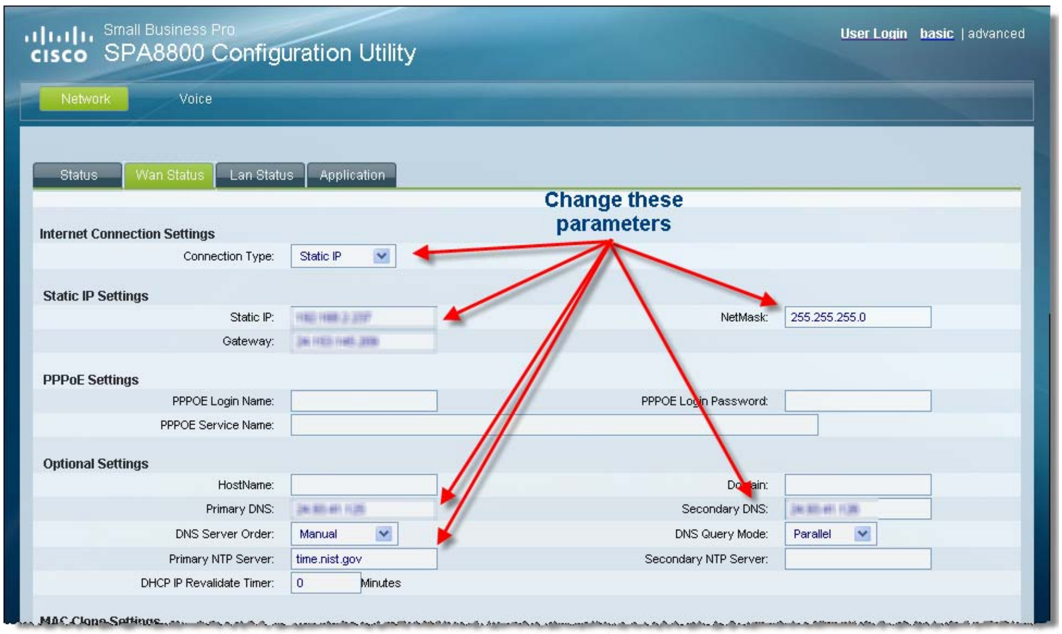
i. Internet Connection Settings > Connection Type: Static IP

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**

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:

`http://`*`IPADDRESSofSPA`*`/upgrade?tftp://TFTPADDRESS/SPAFILE.bin`

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:

`http://192.168.2.237/upgrade?tftp://192.168.2.20/spa8800-6-1-7-GW.bin`

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 the 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

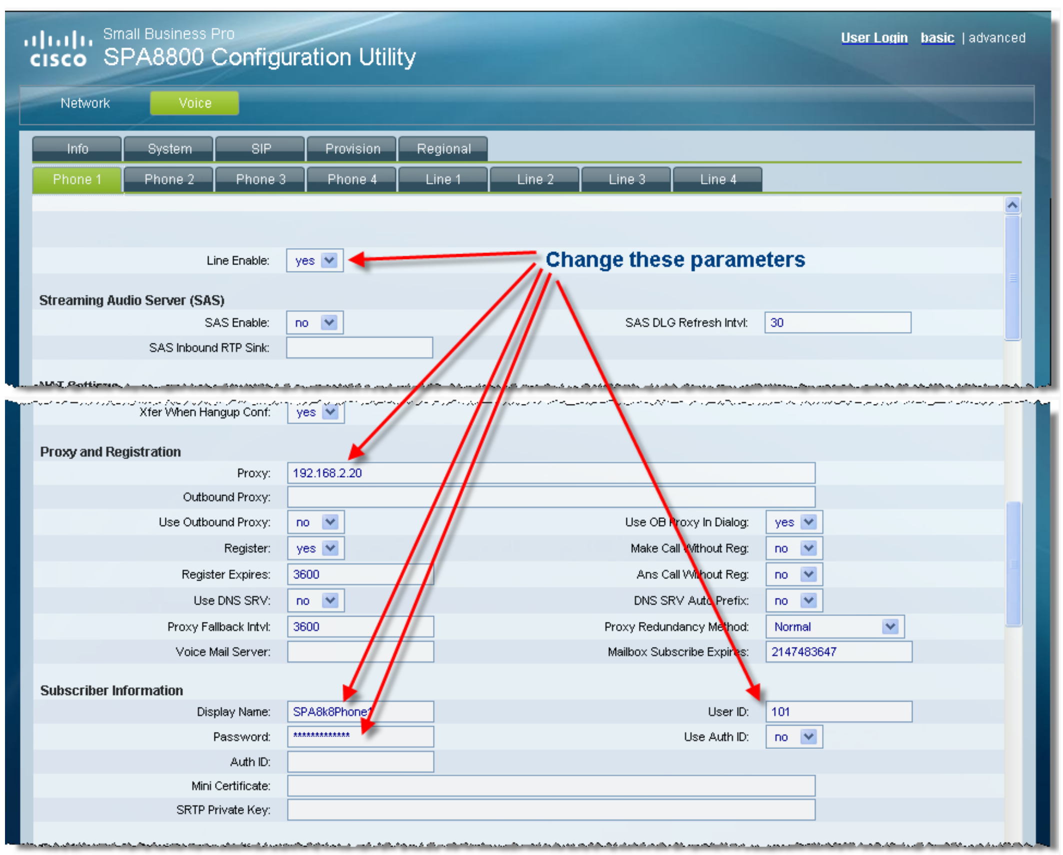
3. Voice tab > Phone 1 tab > Proxy and Registration > Proxy: 192.168.2.20
   Where this is the IP address of the Asterisk Server

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`**

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 >

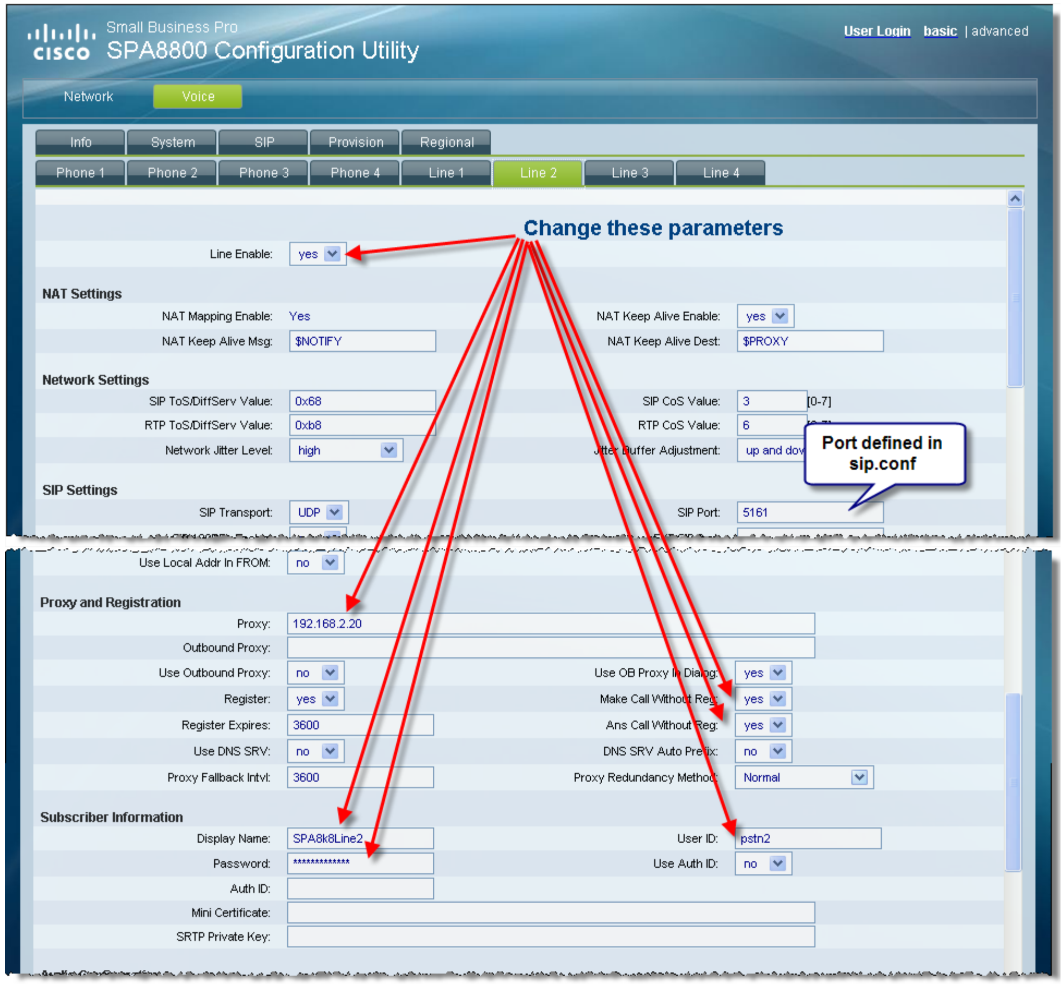
   d. Display Name: SPA8k8Line2

   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.


Small Business Pro
SPA8800 Configuration Utility
User Login
basic
advanced
Network
Voice
Info
System
SIP
Provision
Regional
Phone 1
Phone 2
Phone 3
Phone 4
Line 1
Line 2
Line 3
Line 4
Change these parameters
Line Enable: yes
NAT Settings
NAT Mapping Enable: Yes
NAT Keep Alive Enable: yes
NAT Keep Alive Msg: $NOTIFY
NAT Keep Alive Dest: $PROXY
Network Settings
SIP ToS/DiffServ Value: 0x68
SIP CoS Value: 3 [0-7]
RTP ToS/DiffServ Value: 0xb8
RTP CoS Value: 6
Network Jitter Level: high
Jitter Buffer Adjustment: up and dov
Port defined in sip.conf
SIP Settings
SIP Transport: UDP
SIP Port: 5161
Use Local Addr In FROM: no
Proxy and Registration
Proxy: 192.168.2.20
Outbound Proxy:
Use Outbound Proxy: no
Use OB Proxy In Dialog: yes
Register: yes
Make Call Without Reg: yes
Register Expires: 3600
Ans Call Without Reg: yes
Use DNS SRV: no
DNS SRV Auto Prefix: no
Proxy Fallback Intvl: 3600
Proxy Redundancy Method: Normal
Subscriber Information
Display Name: SPA8k8Line2
User ID: pstn2
Password: *************
Use Auth ID: no
Auth ID:
Mini Certificate:
SRTP Private Key:

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:

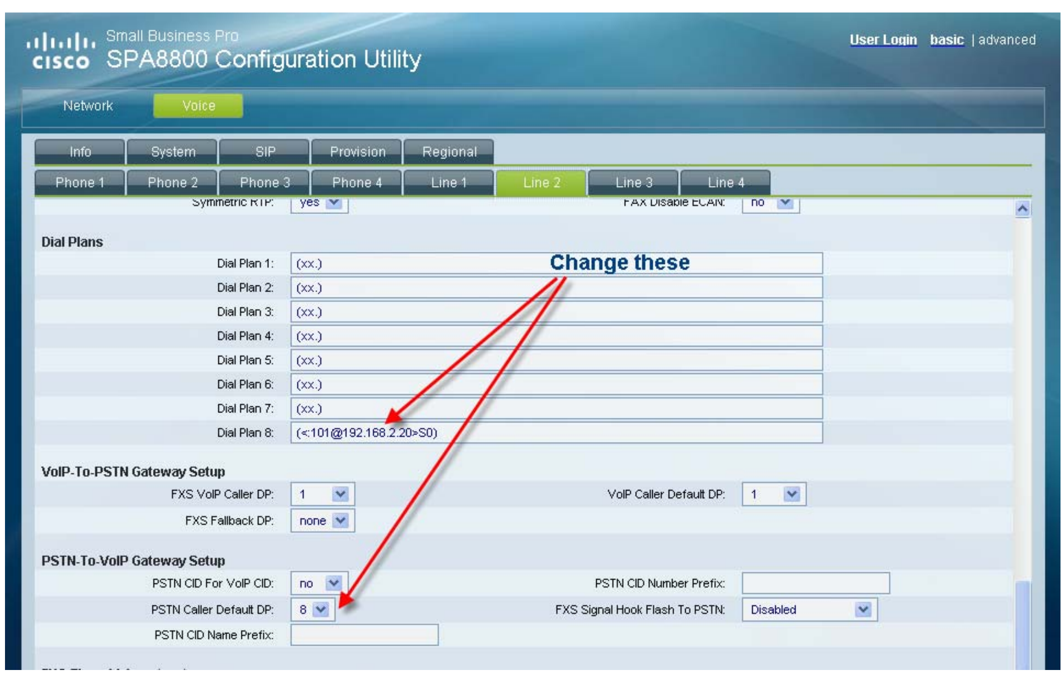
```
[pstn2]
exten => 101,1,Dial(SIP/101,60,rt)
[pstn3]
exten => 101,1,Dial(SIP/201,60,rt
```

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:

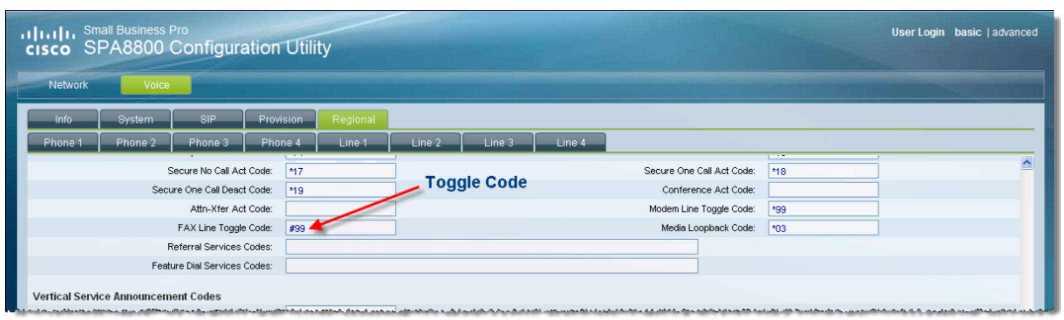
1. Test internal calls:
    a. Verify that analog phones can call each other internally. For example call from 101 to 102.
    b. Optional: Verify that analog phones can call IP phones internally. For example, call from 101 to 200 [if configured]
2. Test inbound calls:
    a. From the PSTN, call a phone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

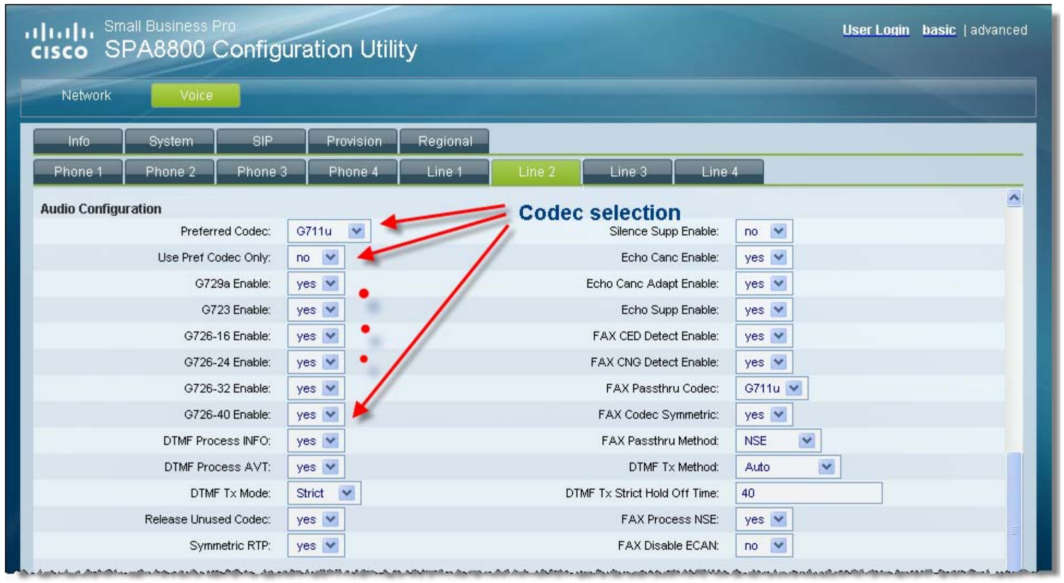


Refer to the

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     6e776c7438a  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:

```
[general]
…
…
allow ulaw
allow g729
…
…
```

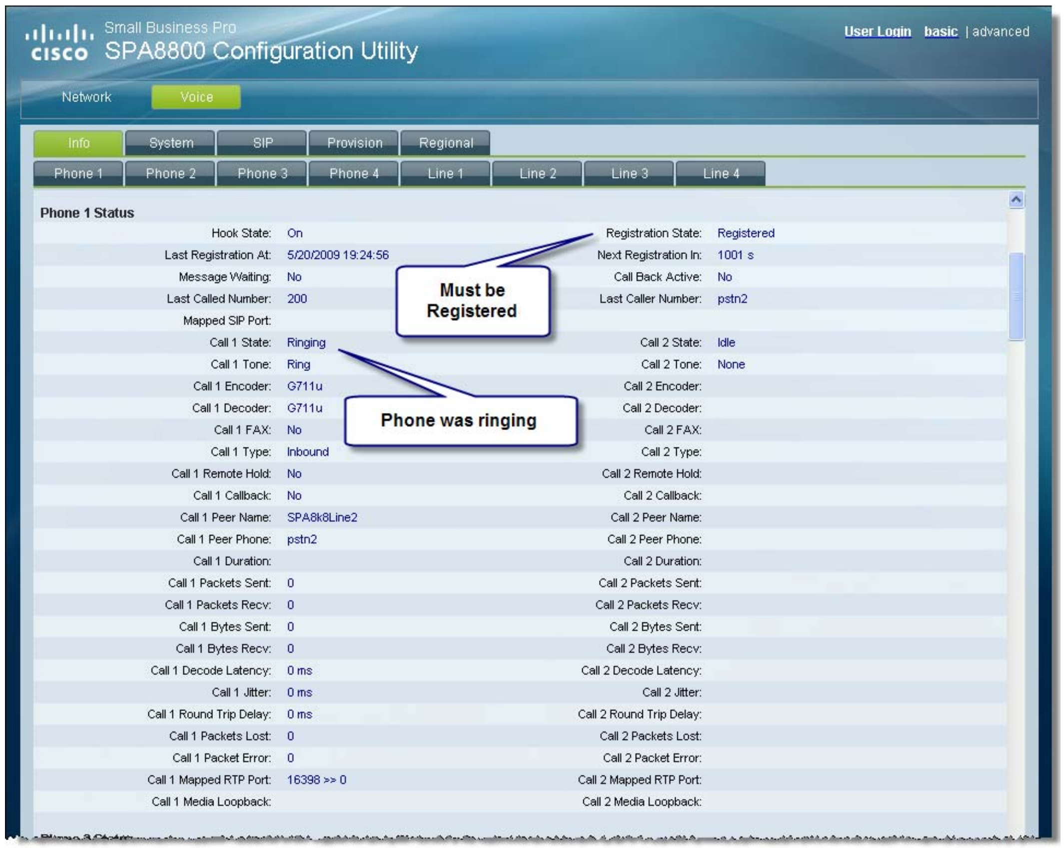

Troubleshooting

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

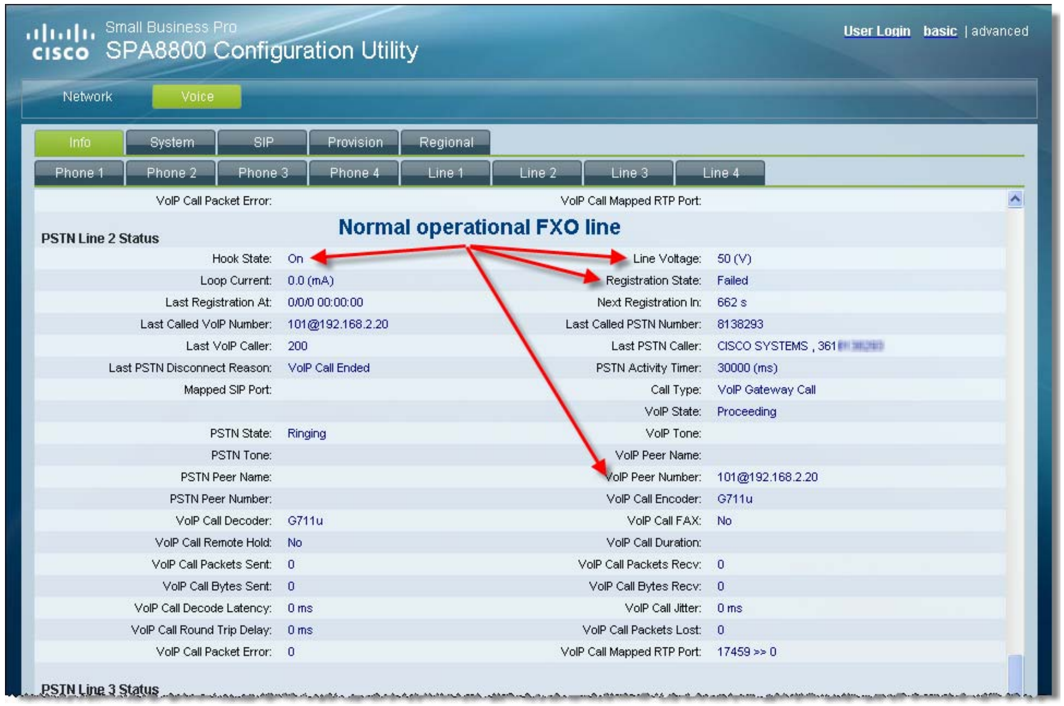
1. Direct your browser to the SPA8800's web user interface (web-ui)

   http://<*IP_address_of_SPA8800*>/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)
```

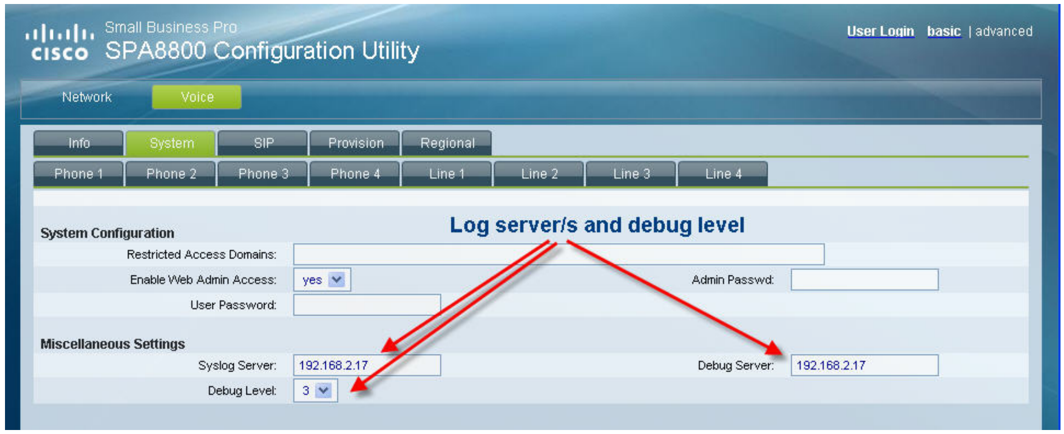
Failing Inbound from PSTN

```
[Jun  3 17:34:37] NOTICE[31613]: chan_sip.c:14035 handle_request_invite: Call from
'pstn3' to extension '101' rejected because extension not found.
```

This indicates that an inbound call to the port named pstn3, 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:



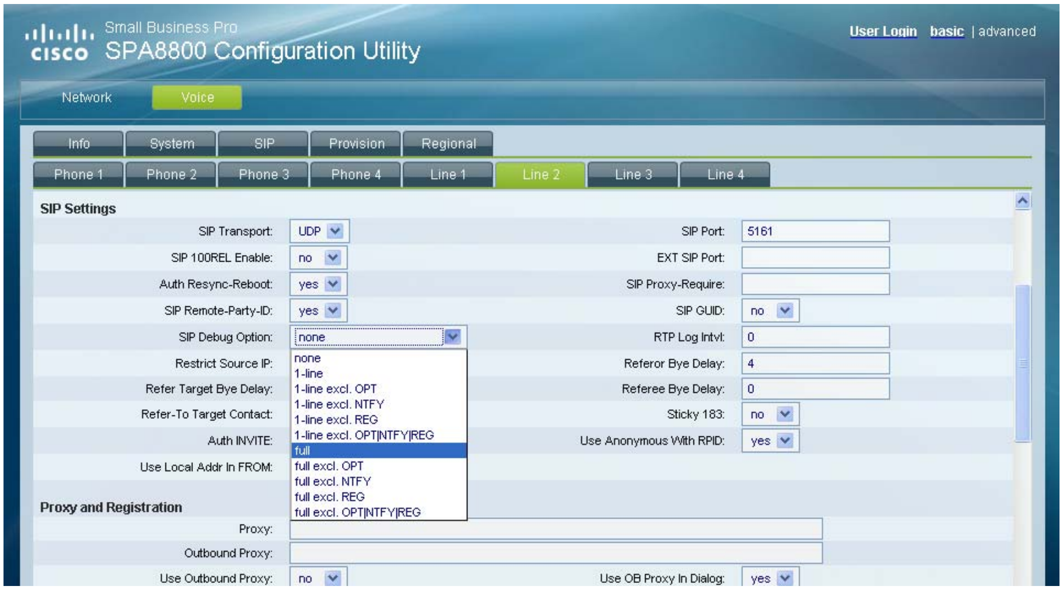
Following is an example of syslog information produced at Debug Level 3 [numbers changed]. This example shows when the phone goes off hook, digits dialed, calling information, a report that an unlisted codec is being requested, and when the phone goes back on hook:

```
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: 2. Report digit 2 (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)
```

```
M2: [1:0]AUD Rel Call
M0: CC:Ended
M2: DLG Terminated 345e80
M2: Sess Terminated
M0: [0]FM Alert Stop RxTx (c=002b02d8;a=0)
M0: [0:0]AUD Rel Call
M0: [0]On Hook
```

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: 4a713b136335955e38e0dd821f5c50dc@192.168.2.20
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: 4a713b136335955e38e0dd821f5c50dc@192.168.2.20
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:
M1: CC:pc(0)=18 not in codec list
M1: 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: 4a713b136335955e38e0dd821f5c50dc@192.168.2.20
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
M1: [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
To: <sip:5551212@192.168.2.237:5161>;tag=e967475542dc2b0i1
Contact: <sip:101@192.168.2.20>
Call-ID: 4a713b136335955e38e0dd821f5c50dc@192.168.2.20
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=e967475542dc2b0i1
Call-ID: 4a713b136335955e38e0dd821f5c50dc@192.168.2.20
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=e967475542dc2b0i1
From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=as16cfeb3c
Call-ID: 4a713b136335955e38e0dd821f5c50dc@192.168.2.20
CSeq: 103 BYE
Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK5886f277
Server: Cisco/SPA8800-6.1.7(GW)
```



```
Content-Length: 0
```

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            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
```



```
CallingPres  : Presentation Allowed, Not Screened
FromUser     : 3615551212
FromDomain   : sip.broadvoice.com
Callgroup    :
Pickupgroup  :
Mailbox      :
VM Extension : asterisk
LastMsgsSent : 32767/65535
Call limit   : 0
Dynamic      : No
Callerid     : "" <>
MaxCallBR    : 384 kbps
Expire       : -1
Insecure     : port,invite
Nat          : Always
ACL          : No
T38 pt UDPTL : No
CanReinvite  : No
PromiscRedir : No
User=Phone   : No
Video Support: No
Trust RPID   : No
Send RPID    : No
Subscriptions: Yes
Overlap dial : Yes
DTMFmode     : inband
LastMsg      : 0
ToHost       : sip.broadvoice.com
Addr->IP     : <ITSP IP Addr> Port 5060
Defaddr->IP  : 0.0.0.0 Port 0
Def. Username: 3615551212
SIP Options  : 100rel
Codecs       : 0x8000e (gsm|ulaw|alaw|h263)
Codec Order  : (none)
Auto-Framing:  No
Status       : OK (113 ms)
Useragent    :
Reg. Contact :
```


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            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)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
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@127.0.1.1
        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>
                    SIP contact address: 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)
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
        Call-ID: 5aca91fd5903c6562f020aab771422f4@127.0.1.1
        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;received=24.153.145.213;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
        Expires: 30
        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: a20ca248-9753b6bc@192.168.2.13
        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@192.168.2.13:5060>;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)
        Content-Length: 0
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.13 (192.168.2.13)
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: a20ca248-9753b6bc@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
        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 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.13 (192.168.2.13)
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;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>;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
        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
="9ee3ff52ed50c8276650bd33d8aa347b"
            Authentication Scheme: Digest
            Username: "201"
            Realm: "asterisk"
            Nonce Value: "4bb63f6e"
            Authentication URI: "sip:192.168.2.20"
            Algorithm: MD5
            Digest Authentication Response: "9ee3ff52ed50c8276650bd33d8aa347b"
        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)
        Content-Length: 0
        Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
        Supported: replaces
```



```
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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.13 (192.168.2.13)
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: a20ca248-9753b6bc@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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.13 (192.168.2.13)
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>;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: 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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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 192.168.2.237:5060;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=cfee310ad138a6beo0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: cfee310ad138a6beo0
        To: "SPA8k8Phone1" <sip:101@192.168.2.20>
            SIP Display info: "SPA8k8Phone1"
            SIP to address: sip:101@192.168.2.20
        Call-ID: 66b034a1-65d1f53@127.0.0.1
        CSeq: 58444 REGISTER
            Sequence Number: 58444
            Method: REGISTER
        Max-Forwards: 70
        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 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=cfee310ad138a6beo0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: cfee310ad138a6beo0
        To: "SPA8k8Phone1" <sip:101@192.168.2.20>
            SIP Display info: "SPA8k8Phone1"
            SIP to address: sip:101@192.168.2.20
        Call-ID: 66b034a1-65d1f53@127.0.0.1
        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>
            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 57 (549 bytes on wire, 549 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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 192.168.2.237:5060;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=cfee310ad138a6beo0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: cfee310ad138a6beo0
        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: 66b034a1-65d1f53@127.0.0.1
        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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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 192.168.2.237:5060;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=cfee310ad138a6beo0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: cfee310ad138a6beo0
        To: "SPA8k8Phone1" <sip:101@192.168.2.20>
            SIP Display info: "SPA8k8Phone1"
            SIP to address: sip:101@192.168.2.20
        Call-ID: 66b034a1-65d1f53@127.0.0.1
        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
="1142d87cae17b1d1b98c805fc26e9fb1"
            Authentication Scheme: Digest
            Username: "101"
            Realm: "asterisk"
            Nonce Value: "76c8a749"
            Authentication URI: "sip:192.168.2.20"
            Algorithm: MD5
            Digest Authentication Response: "1142d87cae17b1d1b98c805fc26e9fb1"
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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-1da4459;received=192.168.2.237
            Transport: UDP
            Sent-by Address: 192.168.2.237
            Sent-by port: 5060
            Branch: z9hG4bK-1da4459
            Received: 192.168.2.237
        From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=cfee310ad138a6beo0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: cfee310ad138a6beo0
        To: "SPA8k8Phone1" <sip:101@192.168.2.20>
            SIP Display info: "SPA8k8Phone1"
            SIP to address: sip:101@192.168.2.20
        Call-ID: 66b034a1-65d1f53@127.0.0.1
        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: Internet_1c:33:a1 (00:e0:4d:1c:33:a1), Dst: 00:24:97:f0:50:3c
(00:24:97:f0:50:3c)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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-1da4459;received=192.168.2.237
            Transport: UDP
            Sent-by Address: 192.168.2.237
            Sent-by port: 5060
            Branch: z9hG4bK-1da4459
            Received: 192.168.2.237
        From: "SPA8k8Phone1" <sip:101@192.168.2.20>;tag=cfee310ad138a6beo0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: cfee310ad138a6beo0
        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: 66b034a1-65d1f53@127.0.0.1
        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 10:01:55 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: Internet_1c:33:a1
(00:e0:4d:1c:33:a1)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
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=a29b792271423dfeo0
            SIP Display info: "SPA8k8Phone2"
            SIP from address: sip:102@192.168.2.20
            SIP tag: a29b792271423dfeo0
        To: "SPA8k8Phone2" <sip:102@192.168.2.20>
            SIP Display info: "SPA8k8Phone2"
            SIP to address: sip:102@192.168.2.20
        Call-ID: b628f5b9-f2e991a6@127.0.0.1
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=a29b792271423dfeo0
            SIP Display info: "SPA8k8Phone2"
            SIP from address: sip:102@192.168.2.20
            SIP tag: a29b792271423dfeo0
        To: "SPA8k8Phone2" <sip:102@192.168.2.20>
            SIP Display info: "SPA8k8Phone2"
            SIP to address: sip:102@192.168.2.20
        Call-ID: b628f5b9-f2e991a6@127.0.0.1
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=a29b792271423dfeo0
            SIP Display info: "SPA8k8Phone2"
            SIP from address: sip:102@192.168.2.20
            SIP tag: a29b792271423dfeo0
        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: 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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
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-edebcc26
            Transport: UDP
            Sent-by Address: 192.168.2.237
            Sent-by port: 5160
            Branch: z9hG4bK-edebcc26
        From: "SPA8k8Phone2" <sip:102@192.168.2.20>;tag=a29b792271423dfeo0
            SIP Display info: "SPA8k8Phone2"
            SIP from address: sip:102@192.168.2.20
            SIP tag: a29b792271423dfeo0
        To: "SPA8k8Phone2" <sip:102@192.168.2.20>
            SIP Display info: "SPA8k8Phone2"
            SIP to address: sip:102@192.168.2.20
        Call-ID: b628f5b9-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
="1cbd8f1d81e70d514f0e494681eea440"
            Authentication Scheme: Digest
            Username: "102"
            Realm: "asterisk"
            Nonce Value: "1fc67977"
            Authentication URI: "sip:192.168.2.20"
            Algorithm: MD5
            Digest Authentication Response: "1cbd8f1d81e70d514f0e494681eea440"
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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-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
            SIP tag: a29b792271423dfeo0
        To: "SPA8k8Phone2" <sip:102@192.168.2.20>
            SIP Display info: "SPA8k8Phone2"
            SIP to address: sip:102@192.168.2.20
        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
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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
            SIP tag: a29b792271423dfeo0
        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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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=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
        Remote-Party-ID: "SPA8k8Phone1" <sip:101@192.168.2.20>;screen=yes;party=calling
        Call-ID: c677a744-912e2955@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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=377ea678f1613b1o0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: 377ea678f1613b1o0
        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 INVITE
            Sequence Number: 101
            Method: INVITE
        User-Agent: Asterisk PBX
        Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
        Supported: replaces
        Proxy-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="111dc491"
            Authentication Scheme: Digest
            Algorithm: MD5
            Realm: "asterisk"
            Nonce Value: "111dc491"
        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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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-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=377ea678f1613b1o0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: 377ea678f1613b1o0
        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: "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 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_1c:33:a1
(00:e0:4d:1c:33:a1)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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: "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
        Remote-Party-ID: "SPA8k8Phone1" <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="89c9bfa2d386061dad5b67972ddd3ec4"
            Authentication Scheme: Digest
            Username: "101"
            Realm: "asterisk"
            Nonce Value: "111dc491"
            Authentication URI: "sip:102@192.168.2.20"
            Algorithm: MD5
            Digest Authentication Response: "89c9bfa2d386061dad5b67972ddd3ec4"
        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 81 (470 bytes on wire, 470 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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-e5988af5;received=192.168.2.237
            Transport: UDP
            Sent-by Address: 192.168.2.237
            Sent-by port: 5060
            Branch: z9hG4bK-e5988af5
            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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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@192.168.2.237:5160>
            SIP to address: sip:102@192.168.2.237:5160
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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-e5988af5;received=192.168.2.237
            Transport: UDP
            Sent-by Address: 192.168.2.237
            Sent-by port: 5060
            Branch: z9hG4bK-e5988af5
            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>;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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
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: 3512588b465552ee7ec40bea27c3f72d@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)
        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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
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=cf247750d7f2b62ei0
            SIP to address: sip:102@192.168.2.237:5160
            SIP tag: cf247750d7f2b62ei0
        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: 3512588b465552ee7ec40bea27c3f72d@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: 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)
```

```
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
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=cf247750d7f2b62ei0
            SIP to address: sip:102@192.168.2.237:5160
            SIP tag: cf247750d7f2b62ei0
        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: 3512588b465552ee7ec40bea27c3f72d@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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5160 (5160)
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=z9hG4bK0141bb36;rport
            Transport: UDP
            Sent-by Address: 192.168.2.20
            Sent-by port: 5060
            Branch: z9hG4bK0141bb36
            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@192.168.2.237:5160>;tag=cf247750d7f2b62ei0
            SIP to address: sip:102@192.168.2.237:5160
            SIP tag: cf247750d7f2b62ei0
        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 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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-e5988af5;received=192.168.2.237
            Transport: UDP
            Sent-by Address: 192.168.2.237
            Sent-by port: 5060
```



```
            Branch: z9hG4bK-e5988af5
            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>;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-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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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=377ea678f1613b1o0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: 377ea678f1613b1o0
        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 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,resp
onse="89c9bfa2d386061dad5b67972ddd3ec4"
            Authentication Scheme: Digest
            Username: "101"
            Realm: "asterisk"
            Nonce Value: "111dc491"
            Authentication URI: "sip:102@192.168.2.20"
            Algorithm: MD5
            Digest Authentication Response: "89c9bfa2d386061dad5b67972ddd3ec4"
        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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
User Datagram Protocol, Src Port: 5160 (5160), 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=cf247750d7f2b62ei0
```



```
            SIP from address: sip:102@192.168.2.237
            SIP tag: cf247750d7f2b62ei0
        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(GW)
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=cf247750d7f2b62ei0
            SIP from address: sip:102@192.168.2.237
            SIP tag: cf247750d7f2b62ei0
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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 192.168.2.20:5060;branch=z9hG4bK7acb7fbb;rport
            Transport: UDP
            Sent-by Address: 192.168.2.20
            Sent-by port: 5060
            Branch: z9hG4bK7acb7fbb
            RPort: rport
        From: <sip:102@192.168.2.20>;tag=as4fc6eca0
            SIP from address: sip:102@192.168.2.20
            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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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
        Status-Code: 200
        [Resent Packet: False]
    Message Header
        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
        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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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: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 (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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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

Frame 3 (433 bytes on wire, 433 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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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 (00:24:97:f0:50:3c), Dst: Internet_1c:33:a1
(00:e0:4d:1c:33:a1)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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: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 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.13 (192.168.2.13)
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
            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 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 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): 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
    Status-Line: SIP/2.0 180 Ringing
```

```
    Message Header

Frame 8 (342 bytes on wire, 342 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
    Status-Line: SIP/2.0 100 Trying
    Message Header

Frame 9 (497 bytes on wire, 497 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
    Status-Line: SIP/2.0 180 Ringing
    Message Header

Frame 10 (813 bytes on wire, 813 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
    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:d2
(00:0e:08:db:51:d2)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.13 (192.168.2.13)
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

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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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 G723/8000
            Media Attribute (a): fmtp:4 annexa=no
            Media Attribute (a): rtpmap:0 PCMU/8000
            Media Attribute (a): rtpmap:8 PCMA/8000
            Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute (a): fmtp:18 annexb=no
            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:off - - - -
            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:e0:4d:1c:33:a1)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.13 (192.168.2.13)
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: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/2.0 200 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:e0:4d:1c:33:a1)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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=432331171f77868do0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: 432331171f77868do0
        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-e0a478e5@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
```

```
            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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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 Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: 432331171f77868do0
        To: <sip:85551313@192.168.2.20>;tag=as0f213285
            SIP to address: sip:85551313@192.168.2.20
            SIP tag: as0f213285
        Call-ID: c6e3a35f-e0a478e5@192.168.2.237
        CSeq: 101 INVITE
        User-Agent: Asterisk PBX
        Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
        Supported: replaces
        Proxy-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="05ce1ccc"
            Authentication Scheme: Digest
            Algorithm: MD5
            Realm: "asterisk"
            Nonce Value: "05ce1ccc"
        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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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 Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: 432331171f77868do0
        To: <sip:85551313@192.168.2.20>;tag=as0f213285
            SIP to address: sip:85551313@192.168.2.20
            SIP tag: as0f213285
        Call-ID: c6e3a35f-e0a478e5@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/SPA8800-6.1.7(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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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=432331171f77868do0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: 432331171f77868do0
        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-e0a478e5@192.168.2.237
        CSeq: 102 INVITE
        Max-Forwards: 70
        Proxy-Authorization: Digest
username="101",realm="asterisk",nonce="05ce1ccc",uri="sip:85551313@192.168.2.20",algorithm=MD5
,response="18deca64e8376d23f2d9a70790b1eda9"
            Authentication Scheme: Digest
            Username: "101"
            Realm: "asterisk"
            Nonce Value: "05ce1ccc"
            Authentication URI: "sip:85551313@192.168.2.20"
            Algorithm: MD5
            Digest Authentication Response: "18deca64e8376d23f2d9a70790b1eda9"
        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): -
            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 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=432331171f77868do0
            SIP Display info: "SPA8k8Phone1"
            SIP from address: sip:101@192.168.2.20
            SIP tag: 432331171f77868do0
        To: <sip:85551313@192.168.2.20>
            SIP to address: sip:85551313@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 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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: 3db967441067a5b61785c0606e95f7a0@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): fmtp:101 0-16
            Media Attribute (a): silenceSupp:off - - - -
            Media Attribute (a): ptime:20
            Media Attribute (a): sendrecv

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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=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-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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
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:5551313@192.168.2.237:5161>
            SIP to address: sip:5551313@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: 3db967441067a5b61785c0606e95f7a0@192.168.2.20
        CSeq: 102 INVITE
        Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK18bc3f3d
            Transport: UDP
            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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
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=e7777c28b5e7542i1
            SIP to address: sip:5551313@192.168.2.237:5161
            SIP tag: e7777c28b5e7542i1
        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: 3db967441067a5b61785c0606e95f7a0@192.168.2.20
        CSeq: 102 INVITE
        Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK18bc3f3d
            Transport: UDP
            Sent-by Address: 192.168.2.20
            Sent-by port: 5060
            Branch: z9hG4bK18bc3f3d
        Contact: "SPA8k8Line2" <sip:5551313@192.168.2.237:5161>
            Contact Binding: "SPA8k8Line2" <sip:5551313@192.168.2.237:5161>
        Server: Cisco/SPA8800-6.1.7(GW)
        Remote-Party-ID: "SPA8k8Line2" <sip:pstn2@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): 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 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
User Datagram Protocol, Src Port: sip (5060), Dst Port: 5161 (5161)
Session Initiation Protocol
    Request-Line: ACK sip:5551313@192.168.2.237:5161 SIP/2.0
    Message Header
        Via: SIP/2.0/UDP 192.168.2.20:5060;branch=z9hG4bK538bb7bc;rport
            Transport: UDP
```



```
            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=e7777c28b5e7542i1
            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: 3db967441067a5b61785c0606e95f7a0@192.168.2.20
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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
            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 17642 RTP/AVP 4 0 8 18 2 101
            Media Attribute (a): rtpmap:4 G723/8000
            Media Attribute (a): fmtp:4 annexa=no
            Media Attribute (a): rtpmap:0 PCMU/8000
            Media Attribute (a): rtpmap:8 PCMA/8000
            Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute (a): fmtp:18 annexb=no
            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: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), Dst: Internet_1c:33:a1
(00:e0:4d:1c:33:a1)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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"
```

```
            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 ACK
        Max-Forwards: 70
        Proxy-Authorization: Digest
username="101",realm="asterisk",nonce="05ce1ccc",uri="sip:85551313@192.168.2.20",algorithm=MD5
,response="18deca64e8376d23f2d9a70790b1eda9"
            Authentication Scheme: Digest
            Username: "101"
            Realm: "asterisk"
            Nonce Value: "05ce1ccc"
            Authentication URI: "sip:85551313@192.168.2.20"
            Algorithm: MD5
            Digest Authentication Response: "18deca64e8376d23f2d9a70790b1eda9"
        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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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: 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=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: 103 BYE
        Max-Forwards: 70
        Proxy-Authorization: Digest
username="101",realm="asterisk",nonce="05ce1ccc",uri="sip:85551313@192.168.2.20",algorithm=MD5
,response="1ff5a6e1706fd31ebfeffbb3070a2438"
            Authentication Scheme: Digest
            Username: "101"
            Realm: "asterisk"
            Nonce Value: "05ce1ccc"
            Authentication URI: "sip:85551313@192.168.2.20"
            Algorithm: MD5
            Digest Authentication Response: "1ff5a6e1706fd31ebfeffbb3070a2438"
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=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: 103 BYE
        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 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=e7777c28b5e7542i1
            SIP to address: sip:5551313@192.168.2.237:5161
            SIP tag: e7777c28b5e7542i1
        Call-ID: 3db967441067a5b61785c0606e95f7a0@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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
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=e7777c28b5e7542i1
            SIP to address: sip:5551313@192.168.2.237:5161
            SIP tag: e7777c28b5e7542i1
        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: 3db967441067a5b61785c0606e95f7a0@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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
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=26caa2f219d9f62eo1
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.3.2
            SIP tag: 26caa2f219d9f62eo1
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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 192.168.2.237:5161;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:pstn2@192.168.3.2>;tag=26caa2f219d9f62eo1
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.3.2
            SIP tag: 26caa2f219d9f62eo1
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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:pstn2@192.168.2.20>;tag=as7100bdb6
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.2.20
            SIP tag: as7100bdb6
        To: <sip:101@192.168.2.237:5060>
            SIP to address: sip:101@192.168.2.237:5060
        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: 1b7c71b81593c9cf71d3add640f41e61@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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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:pstn2@192.168.3.2>;tag=26caa2f219d9f62eo1
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.3.2
            SIP tag: 26caa2f219d9f62eo1
        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: 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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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
        To: <sip:101@192.168.2.237:5060>
            SIP to address: sip:101@192.168.2.237:5060
        From: "SPA8k8Line2" <sip:pstn2@192.168.2.20>;tag=as7100bdb6
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.2.20
            SIP tag: as7100bdb6
        Call-ID: 1b7c71b81593c9cf71d3add640f41e61@192.168.2.20
        CSeq: 102 INVITE
            Sequence Number: 102
            Method: INVITE
        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
```

```
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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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
        To: <sip:101@192.168.2.237:5060>;tag=ecf57f6c3013369ei0
            SIP to address: sip:101@192.168.2.237:5060
            SIP tag: ecf57f6c3013369ei0
        From: "SPA8k8Line2" <sip:pstn2@192.168.2.20>;tag=as7100bdb6
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.2.20
            SIP tag: as7100bdb6
        Call-ID: 1b7c71b81593c9cf71d3add640f41e61@192.168.2.20
        CSeq: 102 INVITE
            Sequence Number: 102
            Method: INVITE
        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
        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
        Server: Cisco/SPA8800-6.1.7(GW)
        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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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
        To: <sip:101@192.168.2.237:5060>;tag=ecf57f6c3013369ei0
            SIP to address: sip:101@192.168.2.237:5060
            SIP tag: ecf57f6c3013369ei0
        From: "SPA8k8Line2" <sip:pstn2@192.168.2.20>;tag=as7100bdb6
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.2.20
            SIP tag: as7100bdb6
        Call-ID: 1b7c71b81593c9cf71d3add640f41e61@192.168.2.20
        CSeq: 102 INVITE
            Sequence Number: 102
            Method: INVITE
        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
        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
        Server: Cisco/SPA8800-6.1.7(GW)
        Remote-Party-ID: "SPA8k8Phone1" <sip:101@192.168.2.20>;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

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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=as7100bdb6
```

```
                SIP Display info: "SPA8k8Line2"
                SIP from address: sip:pstn2@192.168.2.20
                SIP tag: as7100bdb6
            To: <sip:101@192.168.2.237:5060>;tag=ecf57f6c3013369ei0
                SIP to address: sip:101@192.168.2.237:5060
                SIP tag: ecf57f6c3013369ei0
            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: 1b7c71b81593c9cf71d3add640f41e61@192.168.2.20
            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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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;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:pstn2@192.168.3.2>;tag=26caa2f219d9f62eo1
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.3.2
            SIP tag: 26caa2f219d9f62eo1
        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: 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-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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
User Datagram Protocol, Src Port: 5161 (5161), Dst Port: sip (5060)
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@192.168.3.2>;tag=26caa2f219d9f62eo1
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.3.2
            SIP tag: 26caa2f219d9f62eo1
        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: 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)
```

```
        Content-Length: 0

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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=z9hG4bK44611a08;rport
            Transport: UDP
            Sent-by Address: 192.168.2.20
            Sent-by port: 5060
            Branch: z9hG4bK44611a08
            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)
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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
User Datagram Protocol, Src Port: 5160 (5160), 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=z9hG4bK44611a08
            Transport: UDP
            Sent-by Address: 192.168.2.20
            Sent-by port: 5060
            Branch: z9hG4bK44611a08
        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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=z9hG4bK143a03f0;rport
            Transport: UDP
            Sent-by Address: 192.168.2.20
            Sent-by port: 5060
            Branch: z9hG4bK143a03f0
            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: 7614b5463341f49f69ef923c247589ef@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 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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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
        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: 7614b5463341f49f69ef923c247589ef@192.168.2.20
        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: 65f7d74a435b5be15c73d1341625c0c5@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)
```

```
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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), Dst: 192.168.2.20 (192.168.2.20)
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=26caa2f219d9f62eo1
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.3.2
            SIP tag: 26caa2f219d9f62eo1
        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: Cisco/SPA8800-6.1.7(GW)
        Content-Length: 0

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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=26caa2f219d9f62eo1
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.3.2
            SIP tag: 26caa2f219d9f62eo1
        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: 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 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)
Internet Protocol, Src: 192.168.2.20 (192.168.2.20), Dst: 192.168.2.237 (192.168.2.237)
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=as7100bdb6
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.2.20
            SIP tag: as7100bdb6
        To: <sip:101@192.168.2.237:5060>;tag=ecf57f6c3013369ei0
            SIP to address: sip:101@192.168.2.237:5060
            SIP tag: ecf57f6c3013369ei0
        Call-ID: 1b7c71b81593c9cf71d3add640f41e61@192.168.2.20
        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)
Internet Protocol, Src: 192.168.2.237 (192.168.2.237), 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
        To: <sip:101@192.168.2.237:5060>;tag=ecf57f6c3013369ei0
            SIP to address: sip:101@192.168.2.237:5060
            SIP tag: ecf57f6c3013369ei0
        From: "SPA8k8Line2" <sip:pstn2@192.168.2.20>;tag=as7100bdb6
            SIP Display info: "SPA8k8Line2"
            SIP from address: sip:pstn2@192.168.2.20
            SIP tag: as7100bdb6
        Call-ID: 1b7c71b81593c9cf71d3add640f41e61@192.168.2.20
        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.

```
…
Session Initiation Protocol
    Request-Line: INVITE sip:85551212@192.168.2.236 SIP/2.0
    Message Header
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 225352 225352 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 16418 RTP/AVP 0 2 4 8 18 96 97 98
100 101
                Media Type: audio
                Media Port: 16418
                Media Proto: 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
            Media Attribute (a): rtpmap:0 PCMU/8000
                Media Attribute Fieldname: rtpmap
                Media Format: 0
                MIME Type: PCMU
…
```

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.

```
…
Session Initiation Protocol
    Request-Line: INVITE sip:85551212@192.168.2.236 SIP/2.0
    Message Header
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 230398 230398 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): image 16422 udptl t38
                Media Type: image
                Media Port: 16422
                Media Proto: udptl
                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
            Media Attribute (a): T38FaxUdpEC:t38UDPRedundancy
                Media Attribute Fieldname: T38FaxUdpEC
                Media Attribute Value: t38UDPRedundancy
…
```


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