Cisco 2811 FXO

Cisco Configuration Info

Configuration information for a Cisco 2811 running IOS IP Voice 12.4.24T

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Cisco Configuration Info for IOS 12.2

The following configuration file example is known to work when configuring sipX with a Cisco SIP Gateway FXO setup. You must replace parameters within the file (e.g., ip address 10.1.##.# 255.255.###.#) with your site specific parameters when you copy the file through a terminal window into the Cisco Gateway.

1. Copy the following example, starting with the first exclamation point ⚠️ and ending with end (end).
2. Open a terminal window/telnet to the Cisco gateway.
3. At a command prompt enter the Write to Screen command (e.g., Cisco#wr t) after gaining access through protected mode (enter the "enable" command at a command line prompt to enter protected mode and a # sign displays with the prompt) and a similar configuration displays.
4. Type the config command to enter an editable mode and copy the file example into the Cisco gateway.

```
!

! Last configuration change at 12:45:06 GMT Thu Apr 21 2005

! NVRAM config last updated at 12:45:29 GMT Thu Apr 21 2005

!

version 12.2

service timestamps debug datetime msec

service timestamps log datetime

no service password-encryption

!

hostname SIPxchangeGW

!

boot system flash:

logging console informational

logging monitor warnings

enable secret 5 $1$fzy9$AZFTI7pk7wB.eeH6ELL6R0

enable password P1ng-Restart!!

!

Enabling FXS interface registration on Cisco SIP Gateway:
clock timezone GMT -5

ip subnet-zero

!

!

ip tcp path-mtu-discovery

ip domain list pingtel.com

ip domain name pingtel.com

ip name-server 10.1.#.##

!

!

!
```

```
template mon
!
!
voice call carrier capacity active
!
voice service voip
sip
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g711alaw
codec preference 3 g729br8
!
!
!
!
!
!
!
!
voice source-group secured
access-list 1
disconnect-cause call-reject
!
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
!
!
!
interface Ethernet0/0
ip address 10.1.##.# 255.255.###.#
no ip mroute-cache
half-duplex
!
interface Ethernet0/1
ip address 10.1.##.# 255.255.###.#
no ip mroute-cache
shutdown
```

```
full-duplex
!
ip default-gateway 10.1.##.#
ip classless
ip route 0.0.0.0 0.0.0.0 10.1.##.#
ip route 10.1.#.# 255.255.###.# 10.1.##.#
no ip http server
!
!
logging 10.1.#.##
!
snmp-server packetsize 4096
snmp-server enable traps tty
call rsvp-sync
!
voice-port 1/0/0
input gain 8
no vad
connection plar 100
description SIPxchange1
caller-id enable
supervisory disconnect dualtone pre-connect
supervisory answer dualtone
!
voice-port 1/0/1
input gain 8
no vad
connection plar 100
description SIPxchange2
caller-id enable
supervisory disconnect dualtone pre-connect
supervisory answer dualtone
!
voice-port 1/1/0
input gain 8
no vad
connection plar 100
description SIPxchange3
caller-id enable
supervisory disconnect dualtone pre-connect
```

```
supervisory answer dualtone
!
voice-port 1/1/1
input gain 8
no vad
connection plar 100
description SIPxchange4
caller-id enable
supervisory disconnect dualtone pre-connect
supervisory answer dualtone
!
!
mgcp profile default
!
dial-peer cor custom
!
!
!
dial-peer voice 100 voip
huntstop
application session
destination-pattern ...
rtp payload-type nte 98
voice-class codec 1
session protocol sipv2
session target sip-server
dtmf-relay rtp-nte
!
dial-peer voice 10 pots
huntstop
application session
destination-pattern 1..........
port 1/0/0
forward-digits all
!
dial-peer voice 11 pots
huntstop
application session
destination-pattern 1..........
port 1/0/1
```

```
forward-digits all
!
dial-peer voice 12 pots
huntstop
application session
destination-pattern 1..........
port 1/1/0
forward-digits all
!
dial-peer voice 13 pots
huntstop
application session
destination-pattern 1..........
port 1/1/1
forward-digits all
!
dial-peer voice 30 pots
huntstop
application session
destination-pattern 911$
port 1/0/0
forward-digits all
!
dial-peer voice 31 pots
huntstop
application session
destination-pattern 911$
port 1/1/0
forward-digits all
!
dial-peer voice 32 pots
huntstop
application session
destination-pattern 911$
port 1/0/1
forward-digits all
!
dial-peer voice 33 pots
huntstop
application session
```

```
destination-pattern 911$
port 1/1/1
forward-digits all
!
dial-peer voice 40 pots
huntstop
application session
destination-pattern 411$
port 1/0/0
forward-digits all
!
dial-peer voice 41 pots
huntstop
application session
destination-pattern 411$
port 1/1/0
forward-digits all
!
dial-peer voice 50 pots
huntstop
application session
destination-pattern 9...$
port 1/0/0
forward-digits 3
!
dial-peer voice 51 pots
huntstop
application session
destination-pattern 9...$
port 1/0/1
forward-digits 3
!
dial-peer voice 52 pots
huntstop
application session
destination-pattern 9...$
port 1/1/0
forward-digits 3
!
dial-peer voice 53 pots
huntstop
```

```
application session
destination-pattern 9...$
port 1/1/1
forward-digits 3
!
dial-peer voice 20 pots
huntstop
application session
destination-pattern .......$
port 1/0/0
forward-digits all
prefix 1781
!
dial-peer voice 21 pots
huntstop
application session
destination-pattern .......$
port 1/0/1
forward-digits all
prefix 91781
!
dial-peer voice 22 pots
huntstop
application session
destination-pattern .......$
port 1/1/0
forward-digits all
prefix 91781
!
dial-peer voice 23 pots
huntstop
application session
destination-pattern .......$
no digit-strip
port 1/1/1
forward-digits all
prefix 91781
!
dial-peer voice 14 pots
huntstop
```

```
application session
destination-pattern ..........
port 1/0/0
forward-digits all
prefix 1
!
dial-peer voice 15 pots
huntstop
application session
destination-pattern ..........
port 1/0/1
forward-digits all
prefix 1
!
dial-peer voice 16 pots
huntstop
application session
destination-pattern ..........
port 1/1/0
forward-digits all
prefix 1
!
dial-peer voice 17 pots
huntstop
application session
destination-pattern ..........
port 1/1/1
forward-digits all
prefix 1
!
dial-peer voice 60 pots
huntstop
application session
destination-pattern 011.+
port 1/0/0
forward-digits all
!
dial-peer voice 61 pots
huntstop
application session
```

```
destination-pattern 011.+
port 1/0/1
forward-digits all
!
dial-peer voice 62 pots
huntstop
application session
destination-pattern 011.+
port 1/1/0
forward-digits all
!
dial-peer voice 63 pots
huntstop
application session
destination-pattern 011.+
port 1/1/1
forward-digits all
!
dial-peer voice 70 pots
huntstop
application session
destination-pattern 00$
port 1/0/0
forward-digits all
!
dial-peer voice 71 pots
huntstop
application session
destination-pattern 00$
port 1/1/0
forward-digits all
!
dial-peer voice 72 pots
huntstop
application session
destination-pattern 00$
port 1/0/1
forward-digits all
!
dial-peer voice 73 pots
```

```
huntstop
application session
destination-pattern 00$
port 1/1/1
forward-digits all
!
dial-peer voice 101 voip
huntstop
application session
destination-pattern A-Z....................
rtp payload-type nte 98
voice-class codec 1
session protocol sipv2
session target sip-server
dtmf-relay rtp-nte
!
dial-peer voice 130 pots
huntstop
application session
destination-pattern 0$
port 1/0/0
forward-digits all
!
dial-peer voice 131 pots
huntstop
application session
destination-pattern 0$
port 1/1/0
forward-digits all
!
dial-peer voice 132 pots
huntstop
application session
destination-pattern 0$
port 1/0/1
forward-digits all
!
dial-peer voice 133 pots
huntstop
application session
```

```
destination-pattern 0$
port 1/1/1
forward-digits all
!
gateway
timer receive-rtcp 5
!
sip-ua
max-forwards 15
no oli
sip-server dns:pingtel.com
!
!
line con 0
exec-timeout 0 0
line aux 0
exec-timeout 0 0
line vty 0 4
exec-timeout 0 0
password sip
login
line vty 5 15
login
!
ntp clock-period 17181250
ntp source Ethernet0/0
ntp server 192.43.244.18
!
end
```

Enabling FXS interface registration on Cisco SIP gateway:

> if you plan to use your Cisco device as both a gateway and for registered FXS users, you must create explicit custom rules in your dialplan for your FXS users otherwise sipXpbx authorization subsystem will not allow calls to your users on the Cisco device. That is because they share the same IP address as your gateway.sip-ua
> registrar dns:sipserver expires 3600

```
sip-ua
registrar dns:sipserver expires 3600
!

dial-peer voice 562 pots
!--- the following will be used as the registration name
destination-pattern 0562
port 0/0/2
!--- sipXpbx requires that the username and registration name must be the same
!--- Cisco IOS requires that the username must be at least 4 characters long
authentication username 0562 password 1234
!
```