Cisco 7940

There are some things you need to do before your Cisco 7940/7960 IP phone is ready to work with SIP protocol. First, download the latest firmware from http://www.cisco.com/pcgi-bin/tablebuild.pl/sip-ip-phone7960 and create a config file using the template provided. Afterwards the phone will turn on and download the information from the TFTP server, at which point it will be ready to go.

Let's take a lot at the steps needed for setup.

When the phone initializes, it queries the TFTP server for the following information:

- The latest firmware file
- The dual-boot file (OS79XX.TXT)
- A config file created especially for this phone (the name includes the MAC address)
- The default config file
- The ring-list file
- The dial-plan file

The DHCP server should offer the following options (otherwise they need to be set up manually):

- dhcp option #1 (subnetwork mask)
- dhcp option #3 (default gateway)
- dhcp option #6 (DNS server address)
- dhcp option #15 (domain name)
- dhcp option #50 (IP address)
- dhcp option #66 (TFTP server address)

The initialization process for Cisco IP phones

1. The phone downloads the firmware file.

2. The phone gets its VLAN number so it can get parameters from the DHCP server. If it connects to Cisco Catalyst, it should also get a Voice-VLAN number configured for the commutator.

3. The phone gets an IP address from the DHCP server or from its settings.
4. The phone downloads the following files from memory or the TFTP server:
    - SEP.cnf.xml – created on the TFTP server, it includes the following (firmware version): <device><loadInformation>P0S3-08-2-00</loadInformation></device>. The phone checks its own firmware version and updates it as needed.
    - <Firmware version>.loads – if the versions match, the phone uses the file in its memory listed in the SEP.cnf.xml file.
    - OS79XX.TXT – used to ensure compatibility between SIP, MGCP, or SCCP and the TFTP server.
    - SIPDefault.cnf – default parameters for all phones (the settings for which are reviewed below).
    - SIP<MAC-адрес>.cnf – parameters for one particular phone (the structure of which is reviewed below).
    - RINGLIST.DAT – lists ringtone files and their locations.
    - dialplan.xml — contains a sample number plan. It can be sent to the phone using an NTFY message via the Event heading.

5. The phone checks its firmware version.
The filename should be SIPXXXXYYYYZZZZ.cnf, where XXXXYYYYZZZZ is the phone's uppercase MAC address.
Sample file name: SIP00503EFFD842.cnf
Each line in the file should be formatted as follows:
name_variable : value : optional comment
The following sample parameters would need to be changed:
    - line1_name – number or email address used during registration. The number is entered without dashes and the email without the host name.
    - proxy1_address – the IP address of the SIP proxy server used by the phone.
    - proxy1_port – the port number of the SIP proxy server used by the phone.

The rest of the parameters can be changed as needed. Here is a sample SIPDefault.cnf config file:

```
# SIP Default Configuration File
```

```
# Image Version

image_version: P0S3-08-3-00

# Proxy Server

proxy1_address: 172.16.255.255

proxy2_address: ""; Can be dotted IP or FQDN

proxy3_address: ""; Can be dotted IP or FQDN

proxy4_address: ""; Can be dotted IP or FQDN

proxy5_address: ""; Can be dotted IP or FQDN

proxy6_address: ""; Can be dotted IP or FQDN

# Proxy Server Port (default - 5060)

proxy1_port: 5060

proxy2_port: 5060

proxy3_port: 5060

proxy4_port: 5060

proxy5_port: 5060

proxy6_port: 5060

# Proxy Registration (0-disable (default), 1-enable)

proxy_register: 1

# Phone Registration Expiration [1-3932100 sec] (Default - 3600)

timer_register_expires: 3600

# Codec for media stream (g711ulaw (default), g711alaw, g729a)

preferred_codec: g711ulaw
```

```
# TOS bits in media stream [0-5] (Default - 5)

tos_media: 5

# Inband DTMF Settings (0-disable, 1-enable (default))

dtmf_inband: 1

# Out of band DTMF Settings

#(none-disable, avt-avt enable (default), avt_always-always avt)

dtmf_outofband: avt

# DTMF dB Level Settings

#(1-6dB down, 2-3db down, 3-nominal (default), 4-3db up, 5-6dB up)

dtmf_db_level: 3

# SIP Timers

timer_t1: 500; Default 500 msec

timer_t2: 4000; Default 4 sec

sip_retx: 10; Default 10

sip_invite_retx: 6; Default 6

timer_invite_expires: 180  ; Default 180 sec

####### New Parameters added in Release 2.0 #######

# Dialplan template (.xml format file relative to the TFTP root directory)

dial_template: dialplan

# TFTP Phone Specific Configuration File Directory

tftp_cfg_dir: ""; Example: ./sip_phone/

# Time Server
```

```
#(There are multiple values and configurations refer to Admin Guide for Specifics)

sntp_server: ""; SNTP Server IP Address

sntp_mode: anycast (default); unicast, multicast, or directedbroadcast

time_zone: EST; Time Zone Phone is in

dst_offset: 1; Offset from Phone's time when DST is in effect

dst_start_month: April; Month in which DST starts

dst_start_day: ""; Day of month in which DST starts

dst_start_day_of_week: Sun; Day of week in which DST starts

dst_start_week_of_month: 1; Week of month in which DST starts

dst_start_time: 02; Time of day in which DST starts

dst_stop_month: Oct; Month in which DST stops

dst_stop_day: ""; Day of month in which DST stops

dst_stop_day_of_week: Sunday; Day of week in which DST stops

dst_stop_week_of_month: 8; Week of month in which DST stops 8=last week of month

dst_stop_time: 2; Time of day in which DST stops

dst_auto_adjust: 1; Enable(1-Default)/Disable(0) DST automatic adjustment

time_format_24hr: 1; Enable(1 - 24Hr Default)/Disable(0 - 12Hr)

# Do Not Disturb Control

#(0-off (default), 1-on, 2-off with no user control, 3-on with no user control)

dnd_control: 0;

# Caller ID Blocking

#(0-disabled, 1-enabled, 2-disabled no user control, 3-enabled no user control)
```

```
callerid_blocking: 0; (Default is 0 - disabled and sending all calls as anonymous)

# Anonymous Call Blocking

#(0-disabled, 1-enabled, 2-disabled no user control, 3-enabled no user control)

anonymous_call_block: 0; (Default is 0 - disabled and blocking of anonymous calls)

# DTMF AVT Payload (Dynamic payload range for AVT tones - 96-127)

dtmf_avt_payload: 101; Default 101

# Sync value of the phone used for remote reset

sync: 1; Default 1

####### New Parameters added in Release 2.1 #######

# Backup Proxy Support

proxy_backup: ""; Dotted IP of Backup Proxy

proxy_backup_port: 5060; Backup Proxy port (default is 5060)

# Emergency Proxy Support

proxy_emergency: ""; Dotted IP of Emergency Proxy

proxy_emergency_port: 5060; Emergency Proxy port (default is 5060)

# Configurable VAD option

enable_vad: 0; VAD setting 0-disable (Default), 1-enable

####### New Parameters added in Release 2.2 ######

# NAT/Firewall Traversal

nat_enable: 0; 0-Disabled (default), 1-Enabled

nat_address: ""; WAN IP address of NAT box (dotted IP or DNS A record only)

voip_control_port: 5060; UDP port used for SIP messages (default - 5060)
```

```
start_media_port: 16384; Start RTP range for media (default - 16384)

end_media_port: 32766; End RTP range for media (default - 32766)

nat_received_processing: 0; 0-Disabled (default), 1-Enabled

# Outbound Proxy Support

outbound_proxy: ""; restricted to dotted IP or DNS A record only

outbound_proxy_port: 5060; default is 5060

####### New Parameter added in Release 3.0 #######

# Allow for the bridge on a 3way call to join remaining parties upon hangup

cnf_join_enable: 1; 0-Disabled, 1-Enabled (default)

####### New Parameters added in Release 3.1 #######

# Allow Transfer to be completed while target phone is still ringing

semi_attended_transfer: 1; 0-Disabled, 1-Enabled (default)

# Telnet Level (enable or disable the ability to Telnet into the phone)

telnet_level: 1; 0-Disabled (default), 1-Enabled, 2-Privileged

####### New Parameters added in Release 4.0 #######

# XML URLs

services_url: ""; URL for external Phone Services

directory_url: ""; URL for external Directory location

logo_url: ""; URL for branding logo to be used on phone display

# HTTP Proxy Support

http_proxy_addr: ""; Address of HTTP Proxy server

http_proxy_port: 80; Port of HTTP Proxy Server (80-default)
```

```
# Dynamic DNS/TFTP Support

dyn_dns_addr_1: ""; restricted to dotted IP

dyn_dns_addr_2: ""; restricted to dotted IP

dyn_tftp_addr: ""; restricted to dotted IP

# Remote Party ID

remote_party_id: 0; 0-Disabled (default), 1-Enabled
```

Setting up the config file for separate IP phones

The following parameters should be changed in the file: *anonymous_call_block, autocomplete, callerid_blocking, call_hold_ringback, call_waiting, dnd_control* – the rest only as needed. The file name should be configured as SIP<mac address>.cnf. Here is a sample config file for an individual phone:

```
# SIP Configuration Generic File

# Line 1 appearance

line1_name: 1234567

# Line 1 Registration Authentication

line1_authname: "UNPROVISIONED"

# Line 1 Registration Password

line1_password: "UNPROVISIONED"

# Line 2 appearance

line2_name: football

# Line 2 Registration Authentication

line2_authname: "UNPROVISIONED"

# Line 2 Registration Password

line2_password: "UNPROVISIONED"
```

```
####### New Parameters added in Release 2.0 #######

# Phone Label (Text desired to be displayed in upper right corner)

phone_label: ""; Has no effect on SIP messaging

# Line 1 Display Name (Display name to use for SIP messaging)

line1_displayname: "User ID"

# Line 2 Display Name (Display name to use for SIP messaging)

line2_displayname: ""

####### New Parameters added in Release 3.0 ######

# Phone Prompt (The prompt that will be displayed on console and Telnet)

phone_prompt: "SIP Phone"; Limited to 15 characters (Default - SIP Phone)

# Phone Password (Password to be used for console or Telnet login)

phone_password: "cisco"; Limited to 31 characters (Default - cisco)

# User classification used when Registering [ none (default), phone, ip ]

user_info: none

<strong>Setting of IP-phone menu Cisco</strong>
```

Some parameters can be set up manually using the Cisco IP phone menu. Settings for Cisco 7940/7960 IP phones are blocked by default. To unblock them, enter the password from the phone's config file by clicking **Settings => Unlock Config.** To block them, click *Lock Config* or *Exit*. After the changes are made, save them and the phone will restart with the new settings. Besides the main settings like the IP address and TFTP server address, you can also manually set up SIP parameters. Once the phone is unblocked, go to **Settings => SIP Configuration**. In the menu that opens set *line1_name, proxy1_address,* and *proxy1_port*—their format is listed below. If the phone is authorized via an SIP proxy server, also enter *line1_authname* and *line1_password*. Their values by default is *UNPROVISIONED*. We recommend setting up time using the config file. Here is a sample section set for Tallinn:

```
time_zone : BT

dst_offset : 01/00

dst_start_month : April

dst_start_day : 1

dst_start_time : 02/00

dst_stop_month : October

dst_stop_day : 1

dst_stop_time : 02/00

dst_stop_autoadjust : 1
```

DST is the date of time changes between summer and winter time, respectively.