



Application Notes

IDT / Net2phone SIP Trunking Configuration Guide for Cisco Business Edition 3000 (BE3000) Release 8.6.4 with Cisco Unified Border Element Release 8.8.

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Table of Contents

1. Introduction	3
2. SIP Trunking Network Components	3
2.1 Hardware Components	4
2.2 Software Requirements	4
3. Features	4
3.1 SIP Registration Method	4
4. Caveats and Limitations	5
4.1 Configuration considerations	5
5. Configuration	6
5.1 Configuring the Business Edition 3000 (Cisco BE3000)	6
5.2 Configuring Cisco BE3000 Dial Plan	7
5.3 Configuring Cisco BE-3000 Hub/Central Site	8
5.4 Configure Cisco BE3000 PSTN Connections	10
5.5 Configuring SIP Trunk to IDT / Net2phone (via CUBE)	11
5.6 Configuring Cisco BE3000 Devices (SPA88000 IP telephony gateway)	12
5.7 Configuring SPA8800 Advanced Voice Settings	13
5.8 Configuring Cisco BE30000 Usage Profile	14
5.9 Configuring Cisco BE3000 Users	15
5.10 Configuring Cisco BE3000 Phones	16
5.11 Configuring CISCO CUBE	17



1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider IDT / Net2phone and Cisco Business Edition 3000 IP-PBX.

IDT / Net2Phone SIP Trunking Service (IDT) referenced within these Application Notes is designed for business customers. The service enables PSTN calling via a broadband WAN connection using SIP protocol. This converged network solution is a cost effective alternative to traditional PSTN trunks such as analog and/or ISDN-PRI.

2. SIP Trunking Network Components

The network for the SIP Trunk reference configuration is illustrated below.

Network Topology

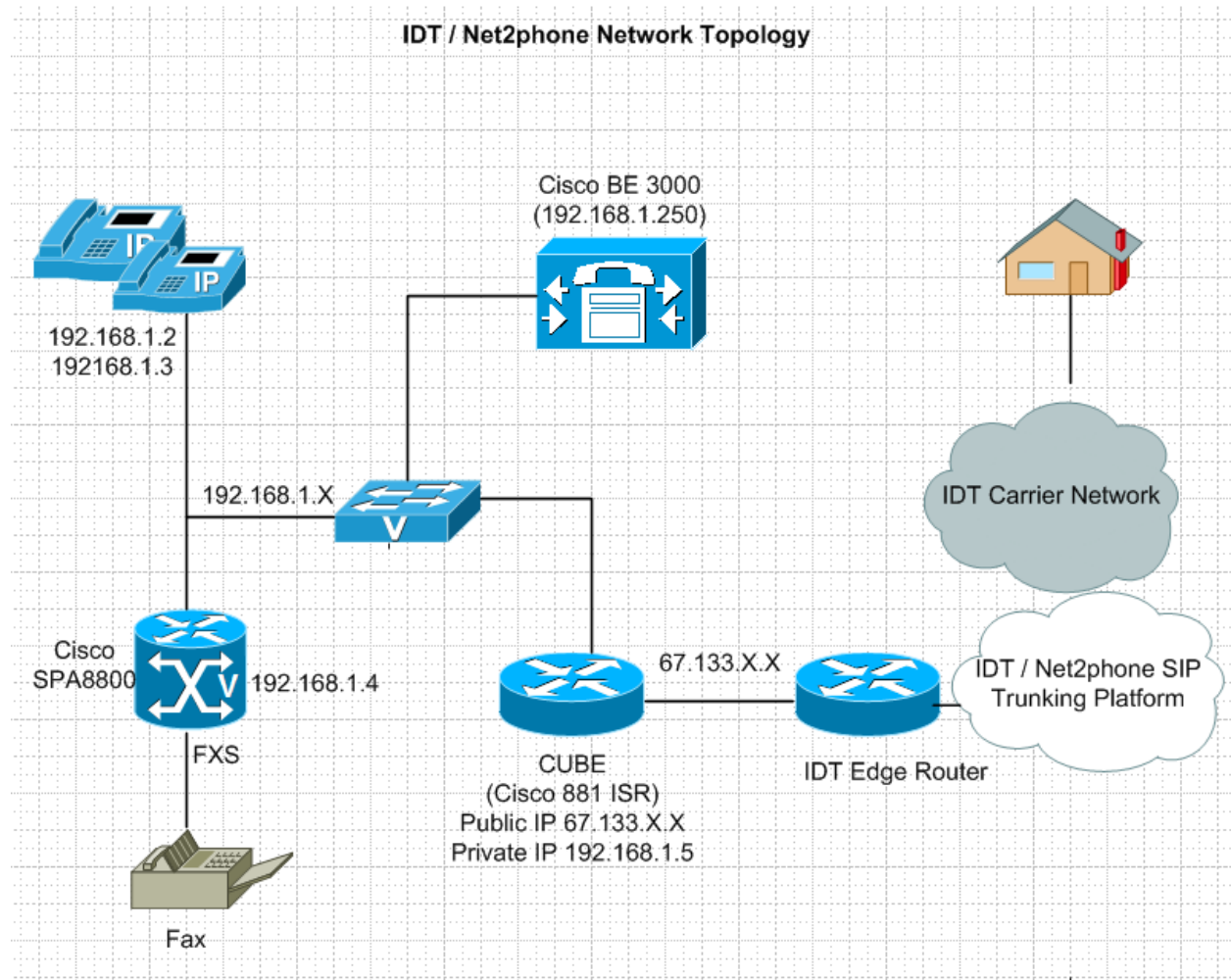


Figure 1. Basic Call Setup



Note: The Cisco Unified Border Element depicted in Figure 1 is not an IDT managed device. It is recommended that the group responsible for the administration, management and configuration of the Cisco Unified Communications Manager also manage and configure the Cisco Unified Border Element.

2.1 Hardware Components

- Cisco Business Edition 3000
- Cisco 881 Series Integrated Services Access Device
- Cisco SG300-28 switch with PoE.
- Cisco SPA8800 IP Telephony Gateway (FXS ports required for Faxing)
- Cisco IP phones (CP-3905 phones were used during testing).
- Analog fax machine

2.2 Software Requirements

- Cisco Business Edition 3000 software version 8.6.4.100000-15
- Cisco IOS version 15.1(4)M4. This configuration was validated using software version :c880voice-universalk9-mz.151-4.M4.bin.
- The documented CISCO CUBE configuration can be supported with the following IOS feature sets: UNIVERSAL
- SPA8800 software version 6.1.9

3. Features

3.1 SIP Registration Method

To establish the SIP Trunk, IDT requires a SIP REGISTER with digest authentication. IDT provides the customer with the username and password which needs to be configured on the Cisco CUBE. Please refer to the CUBE configuration at the end of this guide.

IDT also implements Digest Authentication for outgoing calls. This call authentication scheme as specified in SIP RFC 3261 provides security and integrity protection for SIP signaling.



Features Supported

- Basic call using G.729 /G.711 codec
- Calling Party Number Presentation
- Anonymous call
- Intra-Site Call Transfer
- Intra-Site Call Conference
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- Call Waiting
- Call Park
- Hunt groups (Simultaneous and Sequential Ring)
- DTMF (RFC2833)
- Fax using G.711 pass-through
- Incoming DNIS translation and Routing.
- Outbound calls via IDT's IP and TDM networks.
- CPE voicemail managed service, leave and retrieve voice messages via incoming IDT SIP Trunk (voicemail on BE3000)
- Auto-attendant transfer-to-service on BE3000.
- PBX-Defined Caller ID (spoofing)

4. Caveats and Limitations

- SPA8800 does not support T.38 fax on SG3.
- Cisco IPPBX to Cisco IPPBX was not tested.

4.1 Configuration considerations

- For outbound calls, IDT recommends that customers set G.729 codec as the preferred code and G.711 as the secondary codec .
- When setting up faxing using G.711 pass-through, outbound calls should be prefixed by a dedicated outside dial access code (in this document digit “8” was used) matching a separate outbound dial-peer configured to “strip” the outside dial access code and use G.711 as the only available codec. Similarly, inbound calls to fax endpoints must match a dial-peer based on the called number, configured to only provide G.711 as the only codec to the Cisco BE3000.



5. Configuration

5.1 Configuring the Business Edition 3000 (Cisco BE3000)

Cisco BE3000 Software version

The screenshot displays the Cisco Business Edition 3000 Administrative Interface. The page title is "Cisco Business Edition 3000 Administrative Interface". The breadcrumb navigation is "Maintenance > Installed Software". The main content area is titled "Installed Software" and contains the following information:

- System Software**
Active Version: 8.8.4.10000-15
- Optional Software Packages**
Listed below are the optional software packages installed on the system. Use the Maintenance > Upgrade page to install or upgrade optional software package files.
- Installed Files**
A table with columns "Name" and "Installation Date". The table is currently empty, displaying "No data to display".

The left sidebar contains a navigation menu with the following items: Monitoring, Users/Phones, Connections, System Settings, Maintenance (highlighted), Manage Licenses, Installed Software (highlighted), Upgrade, Backup, Restore, Configuration Export, and Restart/Shutdown. The bottom of the interface shows the copyright notice "©2013, Cisco Systems, Inc. All rights reserved." and a taskbar with an Internet browser icon and a 100% zoom level.



5.2 Configuring Cisco BE3000 Dial Plan

The screenshot shows the Cisco Business Edition 3000 Administrative Interface. The left sidebar contains navigation options: Monitoring, Users/Phones, Connections, System Settings (selected), and Maintenance. Under System Settings, there are sub-options: Administrator, Auto Attendant, Call Detail Offloading Data/Time, Dial Plan (selected), Music on Hold, Reach Me Anywhere, System Notifications, Voice Feature Settings, and Voicemail Notification. The main content area is titled 'System Settings > Dial Plan' and 'Dial Plan'. It has tabs for General, Translation Rules, Blocking Rules, Abbreviated Dialing, and Application Dial Rules. The 'General' tab is active, showing the 'Configure Dial Plan' section. The 'Business Number' section has a 'Main Number' field with the value '1954567202'. The 'Extensions' section includes 'Extension Length' (set to 4), 'Default Allowed Extensions' (7000-7099), and 'Additional Allowed Extensions' (6000-6999). The 'Dialing Prefixes' section includes 'Operator Dial Code' (0), 'Rings extension' (7202 (pberson)), 'Outside Dial Codes' (9,8), and 'Feature Dial Code' (3). The 'Feature Extensions' section is a table with the following data:

Feature	Extension
Call Park	3000-3099
Call Pickup Groups	3200-3299
Meet Me	3400-3499
Message Waiting Indicator Off	3113
Message Waiting Indicator On	3112
Mobile Transfer Numbers	3500-3599

Note: In this configuration, the customer's main telephone number is defined. Also please note that 2 outside dial codes are configured: normal outbound calls (G.729 codec) will be placed dialing "9" as the outside dial access code. When dialing the "8" as the outside access code for faxing, the codec used on the calls will be G.711 based on a specific dial-peer setup on the Cisco CUBE.



5.3 Configuring Cisco BE-3000 Hub/Central Site

The screenshot displays the Cisco Business Edition 3000 Administrative Interface. The main content area is titled "Edit Site - CentralSite" and is divided into three tabs: "General", "Call Settings", and "Call Quality". The "General" tab is active, showing the following configuration details:

- Basic Site Information:**
 - Name: CentralSite
 - Description: Hub / Headquarters Site
 - Local Area Codes: 954 (Note: Separate multiple area codes with commas.)
- Internal Networks:**
 - Subnet: 127.0.0.1, Mask: 32
 - Subnet: 192.168.1.0, Mask: 24
- Media Access Allowed:**
 - Access to Media Resources: Allow Access to Conference Bridge
 - Allow Access to Music On Hold Server
 - Allow Access to Transcoders

Buttons for "Save" and "Reset" are located at the bottom of the configuration area.

Note: Using the General Settings tab, you can define the Local Area Codes as well as the internal network subnets.

The screenshot displays the Cisco Business Edition 3000 Administrative Interface, showing the "Call Settings" tab for the "Edit Site - CentralSite" configuration. The configuration details are as follows:

- Call Privileges:** Highest Privilege Allowed: International Calls
- Emergency Calling:**
 - Allow Emergency Calls from this Site
 - Additional Emergency Services Number: [Empty field]
 - Emergency Location ID Numbers: [Empty field]

Note: For the emergency services to correctly identify the site location as the source of emergency calls, you must specify at least one ELIN. This ELIN phone number must be unique to a site and be registered with its location.
- PSTN Access:**
 - PSTN Call Routing:
 - Route all calls from this site through the same connection group. View Dial Plan Patterns. Connection Group: All Connections
 - Route calls from this site through different connection groups based on Outside Dial Code. View Dial Plan Patterns. Outside Dial Code 8: All Connections. Outside Dial Code 9: All Connections
 - Route calls from this site through connection groups based on Outside Dial Code and call type. View Dial Plan Patterns. Outside Dial Code 8: Emergency Calls: All Connections, Local Calls: All Connections, Toll Free Calls: All Connections, Long Distance Calls: All Connections, International Calls: All Connections, Service Calls: All Connections

A "PSTN Access" button is visible at the bottom right of the configuration area.

Note: Using the Call Settings tab, you have to define the trunks used for PSTN access.



The screenshot displays the Cisco Business Edition 3000 Administrative Interface. The main content area is titled "Edit Site - CentralSite" and is divided into three tabs: "General", "Call Settings", and "Call Quality". The "Call Quality" tab is active, showing the following settings:

- Bandwidth Between Sites:** 11 (1.5 Mbit/s)
- Video:** Enable Video Calls Between Sites
- Bandwidth Allocated for Audio & Video:** 0% to 100% slider
- Audio Quality/Call Quantity Tradeoff:** Slider from "Most Calls" to "Best Quality", currently set to "Best Quality".
- Approximate Call Capacity To Other Sites:** 19 audio only calls, 0 video calls

Below these settings, there is a section for "Calls Within Sites" with a similar "Audio Quality/Call Quantity Tradeoff" slider, also set to "Best Quality". At the bottom of the settings area are "Save" and "Reset" buttons.

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Note: Using the Call Quality tab, you can configure the BE3000 to support both G.729 and G.711 calls. This is done by setting the Audio Quality / Tradeoff to Best Quality.



5.4 Configure Cisco BE3000 PSTN Connections

The screenshot displays the Cisco Business Edition 3000 Administrative Interface. The page title is "Cisco Business Edition 3000 Administrative interface". The user is logged in as "admin". The navigation menu on the left includes "Monitoring", "Users/Phones", "Connections", "Network", "PSTN Connections", "Connection Groups", "Devices", and "Sites". The "PSTN Connections" page shows a table of existing connections and an "Add PSTN Connection..." button.

Name	Description	Connection Type	Device Name	Actions
IDT-CUBE	SIP Trunk to IDT Lab	SIP Trunk	IDT-CUBE	Edit Delete...
S0/SU/0/0/1-0@internal-gateway	Slot 0/ Sub Unit 0/ Data Signaling 1-0	T1 PRI	internal-gateway	Edit Delete...
SPAB821F88349002	SPA gateway port 1	FXO	SPAB821F88349002	Edit Delete...
SPAB21F88349004	SPA gateway port 2	FXO	SPAB821F88349004	Edit Delete...
SPAB21F88349006	SPA gateway port 3	FXO	SPAB821F88349006	Edit Delete...
SPAB21F88349008	SPA gateway port 4	FXO	SPAB821F88349008	Edit Delete...

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5.5 Configuring SIP Trunk to IDT / Net2phone (via CUBE)

The screenshot shows the Cisco Business Edition 3000 Administrative Interface. The main content area is titled "Edit IDT-CUBE" and is divided into several sections:

- General:** Connection Name: IDT-CUBE, Description: SIP Trunk to IDT Lab, Connection Type: SIP Trunk, Device Type: Cisco Unified Border Element(CUBE), Device Name: IDT-CUBE.
- Connection Settings:** Provider IP Address: 192.168.1.5, Provider Port: 5060, BE3000 Port: 5060.
- Hide Advanced Settings:**
 - Presentation and Identification:** Include Remote-Party-Id: Yes, Calling Line ID Presentation - Outgoing: Default, Calling Name Presentation - Outgoing: Default.
 - Outbound Call Routing:** Digit Discard Instructions: Set Manually.
 - Inbound Call Routing:** Remove all but the last 4 digits of the incoming number.

Note: Outbound Call Routing: In order to allow the Cisco BE3000 to pass the “8” outside dial access code to CUBE, you will need to choose the “Set Manually” option and then add/edit the Call Number Pattern – see example below.

The screenshot shows the "Edit Digit Discard Instructions" dialog box. It contains the following fields and controls:

- Copy Manual Settings from Connection (Optional): [Dropdown] [Load] [Reset]
- Called Number Pattern: 81XXXXXXXXXX
- Remove First Digits: None
- Prefix With Digits: [Empty]
- Final Number: 81XXXXXXXXXX
- Number Type: Subscriber
- Buttons: [Minus] [Plus]



5.6 Configuring Cisco BE3000 Devices (SPA8800 IP telephony gateway)

The screenshot displays the Cisco Business Edition 3000 Administrative Interface. The main content area is titled "Edit SPAB8621F883490" and contains the following configuration fields:

- Device Type: SPA8800
- MAC Address: B8621F883490
- IP Address: 192.168.1.4
- Name: SPAB8621F883490
- Description: SPA8800 Fax Testing
- Site Association: CentralSite

Below these fields is a section for "Advanced Settings" which is currently collapsed. The settings visible are:

- Fax Mode: G.711 Passthrough
- FXS Port Input Gain: -3
- FXS Port Output Gain: -3
- DTMF Playback Level: -6
- DTMF Playback Length: 1
- FXS Port Impedance: 600

At the bottom of the configuration area are "Save" and "Reset" buttons. The interface also shows a navigation menu on the left with options like Monitoring, Users/Phones, Connections, Network, PSTN Connections, Connection Groups, Devices, and Sites. The footer of the interface includes the text "©2013, Cisco Systems, Inc. All rights reserved." and a taskbar at the bottom with "Done" and "Internet" icons.

Note: When configuring the SPA8800 to support fax equipment, ensure that the Fax Mode is set to G.711 Pass-through as it is the most reliable fax relay method offered by the SPA8800 gateway.



5.7 Configuring SPA8800 Advanced Voice Settings

The screenshot displays the SPA8800 Configuration Utility web interface. The top navigation bar includes 'Network' and 'Voice' tabs. Below this, there are tabs for 'Info', 'System', 'SIP', 'Provision', and 'Regional'. The 'Provision' tab is active, and the 'Line 1' sub-tab is selected. The main configuration area is divided into three sections: 'Configuration Profile', 'Firmware Upgrade', and 'General Purpose Parameters'. The 'Configuration Profile' section includes fields for 'Provision Enable' (set to 'yes'), 'Resync Random Delay' (2), 'Resync Error Retry Delay' (3600), 'Resync From SIP' (yes), 'Resync On Reset' (yes), 'Resync Periodic' (3600), 'Resync Trigger 1' and '2' (empty), 'Resync After Upgrade Attempt' (yes), 'Resync Fails On FNF' (yes), and 'Profile Rule' (ftp://192.168.1.250/spa\$MA.cnf.xml). The 'Firmware Upgrade' section includes 'Upgrade Enable' (yes), 'Downgrade Rev Limit' (empty), 'Upgrade Error Retry Delay' (3600), 'Upgrade Rule' (empty), and various log messages. The 'General Purpose Parameters' section includes 'GPP A' (empty). At the bottom, there are 'Undo All Changes' and 'Submit All Changes' buttons. The footer shows '© 2009 Cisco Systems, Inc. All Rights Reserved.' and 'SPA8800 IP Telephony Gateway'.

Note: During the first time installation of the SPA8800, configure the Profile Rule as **“ftp://BE3000 IP address/spa\$MA.cnf.xml”**. In the above example, the IP address assigned to the BE3000 is 192.168.1.250.



5.8 Configuring Cisco BE3000 Usage Profile

The screenshot displays the Cisco Business Edition 3000 Administrative Interface. The main content area is titled "Edit Usage Profile - IDT Testing Usage Profile". The interface is divided into several sections:

- Profile Information:** Name: IDT Testing Usage Profile, Description: IDT Testing Usage Profile.
- Allowed Calls:** Highest Level of Calls Allowed: International Calls, Emergency Calls: Allow.
- Call Features:** Call Barge: Allow user to barge in on calls; Call Park: Allow user to park call and pick call up from another phone; Call Pickup Groups: Allow user to pick up calls of another user (5 Users assigned to this usage profile are not in a pickup group -> Show All); Reach Me Anywhere: Allow user to be reached on multiple phones at the same time (Warning: Reach Me Anywhere is supported only for PRI gateways and SIP trunks); Extension Mobility: Allow Cisco Extension Mobility to be used on phone of user; Allow user to use Cisco Extension Mobility service.
- Voicemail:** Allow user to use Voicemail service; Allow user to divert an incoming call to voicemail; Allow user to be notified of new voicemails via email.
- Forwarding:** Forward Busy Calls To: Voicemail; Forward No Answer Calls To: Voicemail.
- Audio For Hold:** Sample Audio Source.

Buttons for "Save" and "Reset" are located at the bottom of the configuration area. The footer of the interface shows "©2013, Cisco Systems, Inc. All rights reserved." and a taskbar with "Done", "Internet", and "100%" zoom level.

Note: On the Usage Profile page, you can set the highest level of calls allowed to be assigned to users. You can also setup forwarding of busy / no answer calls.



5.9 Configuring Cisco BE3000 Users

The screenshot shows the Cisco Business Edition 3000 Administrative Interface. The left sidebar contains navigation options: Monitoring, Users/Phones, Phones, Departments, Usage Profiles, Hunt Lists, Call Pickup Groups, Attendant Group, and Phone Applications. The main content area is titled "Users" and displays a table of user information. The table has columns for Last Name, First Name, User ID, Usage Profile, Line Numbers, and Actions. The data rows are:

Last Name	First Name	User ID	Usage Profile	Line Numbers	Actions
Albert	Eric	eaibert	IDT Testing Usage Profile	7201	Edit Delete...
Benson	Paul	pbenson	IDT Testing Usage Profile	7202	Edit Delete...
Fax	Eric	efax1	IDT Testing Usage Profile	7205	Edit Delete...
Fax	Eric	efax3	IDT Testing Usage Profile	7207	Edit Delete...
Fax	Eric	efax2	IDT Testing Usage Profile	7206	Edit Delete...

Below the table are buttons for "Add User..." and "Import Users/Phones...". The interface also shows a filter field for "Filter/Last Name" and a "Clear Filter" button. The bottom of the page includes a copyright notice: "©2013, Cisco Systems, Inc. All rights reserved."

The screenshot shows the "Edit User - eaibert" form in the Cisco Business Edition 3000 Administrative Interface. The form is divided into several sections:

- General**: Includes tabs for "General", "Speed Dials", and "Calling Features".
- User Information**: Fields for First Name (Eric), Last Name (Albert), E-mail Address (eaibert@dt.net), and Usage Profile (IDT Testing Usage Profile).
- System and Device Access**: Fields for User ID (eaibert), Password, Confirm Password, Phone PIN, and Confirm Phone PIN. There is a checkbox for "User must change password at next login" and a checked checkbox for "Enable Administrator Access".
- Line Numbers**: A table with columns for Line Number, External Caller ID, and Call Forward All. The first row shows Line Number 7201, External Caller ID 19545567202, and a "Phone Number" dropdown.
- Cisco Mobile Solutions**: A checkbox for "Enable Cisco Mobile Client Support" and a checkbox for "Configure Cisco Mobile Client Number" with a "Phone Number" dropdown. There are also fields for "Phone Type" and "Device ID".

The bottom of the page includes a copyright notice: "©2013, Cisco Systems, Inc. All rights reserved."



5.10 Configuring Cisco BE3000 Phones

The screenshot shows the Cisco Business Edition 3000 Administrative Interface. The left sidebar contains navigation options: Monitoring, Users/Phones (selected), Users, Phones, Departments, Usage Profiles, Hunt Lists, Call Pickup Groups, Attendant Group, and Phone Applications. The main content area is titled "Phones" and displays a table of phone configurations. The table has columns for Name, Owner, Extension, Description, Model, and Actions. Below the table are buttons for "Add Phone..." and "Import Users/Phones...".

Name	Owner	Extension	Description	Model	Actions
SEPB4E9B08DA705	ealbert	7201	Eric Albert Test Phone	Cisco 3905	Edit Delete
SEPF02929E237C4	pbenson	7202	Paul Benson Test phone	Cisco 3905	Edit Delete
SPA621F88349001	efax1	7205	SPA800 Fax device port 1	Analog Phone (SPA800)	Edit Delete
SPA621F88349003	efax2	7206	SPA800 port 2	Analog Phone (SPA800)	Edit Delete
SPA621F88349005	efax3	7207	SPA800 port 3	Analog Phone (SPA800)	Edit Delete

The screenshot shows the "Edit Phone" configuration page for the phone with extension 7201. The page includes fields for Registration, Phone Type (Cisco 3905), MAC Address (B4E9B08DA705), Device Name (SEPB4E9B08DA705), and Description (Eric Albert Test Phone). There is a checkbox for "Do Not Disturb". The "Extensions" section contains a table for configuring multiple extensions.

Extension	Owner
1 7201	ealbert
2	
3	
4	
5	
6	



5.11 Configuring CISCO CUBE

Show version

```
vpcube#sh version
Cisco IOS Software, C880 Software (C880VOICE-UNIVERSALK9-M), Version 15.1(4)M4, RELEASE
SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2012 by Cisco Systems, Inc.
Compiled Wed 21-Mar-12 01:24 by prod_rel_team
```

```
ROM: System Bootstrap, Version 12.4(22r)YB5, RELEASE SOFTWARE (fc1)
```

```
vpcube uptime is 3 weeks, 1 day, 23 hours, 36 minutes
System returned to ROM by reload at 17:44:06 UTC Tue May 7 2013
System restarted at 17:44:51 UTC Tue May 7 2013
System image file is "flash:c880voice-universalk9-mz.151-4.M4.bin"
Last reload type: Normal Reload
Last reload reason: Reload Command
```

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
<http://www.cisco.com/wwl/export/crypto/tool/stqrg.html>

If you require further assistance please contact us by sending email to export@cisco.com.

Cisco 881 (MPC8300) processor (revision 1.0) with 498688K/25600K bytes of memory.
Processor board ID FTX1707811A

5 FastEthernet interfaces
1 Virtual Private Network (VPN) Module
256K bytes of non-volatile configuration memory.
125440K bytes of ATA CompactFlash (Read/Write)



License Info:

License UDI:

```
-----  
Device#  PID      SN  
-----  
*0      C881-CUBE-K9    FTX1707811A
```

License Information for 'c880-iad'
License Level: advipservices Type: Permanent
Next reboot license Level: advipservices

Configuration register is 0x2102

vpcube#

Show running-configuration

vpcube#sh run
Building configuration...

```
Current configuration : 9149 bytes  
!  
! Last configuration change at 19:34:37 UTC Tue May 14 2013 by vpo  
! NVRAM config last updated at 15:48:33 UTC Wed May 15 2013 by vpo  
! NVRAM config last updated at 15:48:33 UTC Wed May 15 2013 by vpo  
version 15.1  
no service pad  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname vpcube  
!  
boot-start-marker  
boot-end-marker  
!  
!  
logging buffered 51200 warnings  
!
```



```
no aaa new-model
!
crypto pki token default removal timeout 0
!
crypto pki trustpoint TP-self-signed-1254493017
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-1254493017
  revocation-check none
  rsakeypair TP-self-signed-1254493017
!
!
crypto pki certificate chain TP-self-signed-1254493017
  certificate self-signed 01
    3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
    31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
    69666963 6174652D 31323534 34393330 3137301E 170D3133 30333034 31363035
    35335A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
    4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 32353434
    39333031 3730819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
    8100DAD4 B1D52AA8 9D80D332 64F3271F 4B0A8901 FA989871 32819139 4D33C06B
    E7CF9A2E 010CD052 974E977F 970023AB 8D1E3359 0C153159 2A1A6415 A07AEDF6
    9A28CF52 B8778FA1 1438A4BE B084CE08 8CC9DB1B 395A0AB9 BF965982 A3D6159E
    A50D5BDE AB2C2003 0A4EFFD5 217DBFBB 839D59B8 8145BDBA FCA954E4 F9039E6F
    08390203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
    551D2304 18301680 144FE13B 03859D11 30C74F73 36BD0653 320D6966 1B301D06
    03551D0E 04160414 4FE13B03 859D1130 C74F7336 BD065332 0D69661B 300D0609
    2A864886 F70D0101 05050003 8181004B A00CE8A0 FED299B3 F1B67CF4 7E983039
    A8ED5605 B5A9D7D8 EBE6709D 6CE328D8 397D0A8F B4B5B0E7 EFEC552B EB5FB2CE
    EEB72BAF A1A0675A B8E78EA8 9E8D5BD8 75D7CB6F 3DCD0D41 CB2B1FEC 3C652E55
    B82E84A0 7ED09F8C 4B3CAF4F F7F2620E 0CE14115 918B635F 1E96C9E9 BF72BFAC
    5D172FE8 581363E2 D401B113 A2F0DC
  quit
ip source-route
!
!
!
ip dhcp excluded-address 10.10.10.1
!
ip dhcp pool ccp-pool
  import all
  network 10.10.10.0 255.255.255.248
  default-router 10.10.10.1
  lease 0 2
!
!
ip cef
```



```
no ip domain lookup
ip domain name yourdomain.com
no ipv6 cef
!
!
!
!
!
multilink bundle-name authenticated
!
!
!
voice service voip
ip address trusted list
  ipv4 169.xxx.xxx.xxx
  ipv4 66.xxx.xxx.xxx
  ipv4 192.xxx.xxx.xxx
  ipv4 67.xxx.xxx.xxx
no ip address trusted authenticate
address-hiding
allow-connections sip to sip
redirect ip2ip
fax protocol pass-through g711ulaw1
sip
  header-passing
  error-passthru
asserted-id pai2
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
  g729 annexb-all
  sip-profiles 100
  no call service stop
!
voice class codec 13
  codec preference 1 g729r8
  codec preference 2 g711ulaw
!
voice class sip-profiles 1004
```

¹ This command enables the router to perform G.711 pass-through for faxing.

² This command enables the router to send P-Asserted Identity header on the outbound calls.

³ This command sets the codec preference to be assigned to dial-peers. IDT recommends that G.729 codec be set as the preferred codec on calls.

⁴ This SIP Profile is recommended when setting up a SIP Trunk with IDT / Net2phone. IDT recommends including the /siptrunking in the SIP contact header of the REGISTER as well as in the INVITE.



```
request REGISTER sip-header Contact modify "<sip:(.*)@(.*):5060>" "<sip:siptrunking@\2>"
request INVITE sip-header Contact modify "<sip:(.*)@(.*):5060>" "<sip:siptrunking@\2>"
!
!
!
!
voice translation-rule 15
rule 1 /81/ /1\1/
!
!
voice translation-profile outbound_fax_g711
translate called 1
!
!
license udi pid C881-CUBE-K9 sn FTX1707811A
!
!
username vpo privilege 15 secret 4 Ai2HyYxj4jk0kY.qvF6sy3b8QsoqM1PHF/3rXWebe.6
!
!
!
!
!
!
!
!
!
!
!
!
interface FastEthernet0
switchport access vlan 110
no ip address
!
interface FastEthernet1
no ip address
!
interface FastEthernet2
no ip address
!
interface FastEthernet3
no ip address
!
interface FastEthernet4
```

⁵ The command allows CUBE to string off the leading "8" from the dialed string using for outbound G.711 fax calls.



```
description "Uplink to IDT ISP"
ip address 67.133.xxx.xxx 255.255.255.0
duplex auto
speed auto
!
interface Vlan1
description $ETH_LAN$
ip address 10.10.10.1 255.255.255.248
ip tcp adjust-mss 1452
!
interface Vlan110
description LAN segment
ip address 192.168.1.5 255.255.255.0
!
ip default-gateway 67.133.xxx.xxx
ip forward-protocol nd
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
!
ip route 0.0.0.0 0.0.0.0 67.133.xxx.xxx
ip route 67.133.xxx.xxx 255.255.255.255 FastEthernet4
ip route 192.168.0.0 255.255.255.0 192.168.1.1
!
access-list 23 permit 10.10.10.0 0.0.0.7
!
!
!
!
!
control-plane
!
!
!
!
mgcp profile default
!
!
dial-peer voice 2051 voip
description "Int'l calls to IDT - IDT facing side"
destination-pattern 011T
session protocol sipv2
session target sip-server
```



```
voice-class codec 1
voice-class sip profiles 100
dtmf-relay rtp-nte6
no vad
!
dial-peer voice 2050 voip
description "Int'l calls to IDT - IP-PBX facing side"
session protocol sipv2
incoming called-number 011T
voice-class codec 1
voice-class sip profiles 100
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2041 voip
description "N11 Calls to IDT - IDT facing side"
destination-pattern .11
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip profiles 100
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2040 voip
description "N11 Calls to IDT - IP-PBX facing side"
session protocol sipv2
session target sip-server
incoming called-number .11
voice-class codec 1
voice-class sip profiles 100
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2000 voip
description "Outgoing calls to IDT - IP-PBX facing side"
session protocol sipv2
session target sip-server
incoming called-number .T
voice-class codec 1
voice-class sip profiles 100
dtmf-relay rtp-nte
no vad
!
```

⁶ This command enabled DTMF digit passing using RFC2833



```
dial-peer voice 2001 voip
description "Outgoing to IDT - IDT facing side"
destination-pattern 1T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip profiles 100
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2011 voip
description "Incoming calls to IP-PBX - IP-PBX facing side"
destination-pattern 195455672027
session protocol sipv2
session target ipv4:192.168.1.2508
voice-class sip profiles 100
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2010 voip
description "Incoming calls to IP-PBX - IDT Facing side"
session protocol sipv2
incoming called-number 19545567202
voice-class sip profiles 100
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2100 voip
description "Outgoing G.711 fax calls to IDT - IDT facing side"
translation-profile outgoing outbound_fax_g711
destination-pattern 8T
session protocol sipv2
session target sip-server
voice-class sip early-offer forced
voice-class sip profiles 100
dtmf-relay rtp-nte
codec g711ulaw
fax-relay sg3-to-g3
fax rate 14400
fax protocol pass-through g711ulaw
!
dial-peer voice 2101 voip
description "Incoming G.711 fax calls - IP-PBX facing side"
```

⁷ This is the DID assigned by IDT / Net2phone to you.

⁸ This is the IP address assigned to the BE3000 in this example.



destination-pattern 19545567205⁹

```

session protocol sipv2
session target ipv4:192.168.1.250
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 100
dtmf-relay rtp-nte
codec g711ulaw
fax-relay sg3-to-g3
fax rate 14400
fax protocol pass-through g711ulaw
!
dial-peer voice 2102 voip
description "Incoming G.711 fax calls - IDT facing side"
session protocol sipv2
session target sip-server
incoming called-number 19545567205
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 100
dtmf-relay rtp-nte
codec g711ulaw
fax-relay sg3-to-g3
fax rate 14400
fax protocol pass-through g711ulaw
!
!
num-exp 5567202 19545567202
sip-ua
credentials username 263xxxxxxx password 7 075B701A1B realm net2phone10
authentication username 263xxxxxxx password 7 03500A5D53
retry invite 2
retry register 10
timers connect 100
registrar ipv4:169.132.xxx.xxx expires 360011
sip-server ipv4:169.132.xxx.xxx
host-registrar
!
banner login ^C^C
banner motd ^CC@
#####

```

⁹ This is the separate number used for G.711 inbound faxing

¹⁰ Used by CUBE for registration purposes. The actual SIP registration Username and Password will be provided to you by your IDT Account Manager and must be kept confidential.

¹¹ The SIP proxy server IP address will be provided to you by your IDT Account Manager as well.



```
# This system is the property of Net2Phone and is      #
# specifically for the use of authorized users for legitimate #
# business purposes only.                             #
#                                                     #
# Individuals using this computer system without authority, or in #
# excess of their authority, are subject to having all of their #
# activities on this system monitored and recorded by system #
# personnel.                                          #
#                                                     #
# In the course of monitoring individuals improperly using this #
# system, or in the course of system maintenance, the activities #
# of authorized users may also be monitored.        #
#                                                     #
# Anyone using this system expressly consents to such monitoring #
# and is advised that if such monitoring reveals possible #
# evidence of criminal activity, system personnel may provide the #
# evidence of such monitoring to law enforcement officials. #
#####
^C
!
line con 0
login local
line aux 0
line vty 0 4
privilege level 15
login local
transport input telnet ssh
!
end

vpocube#
```