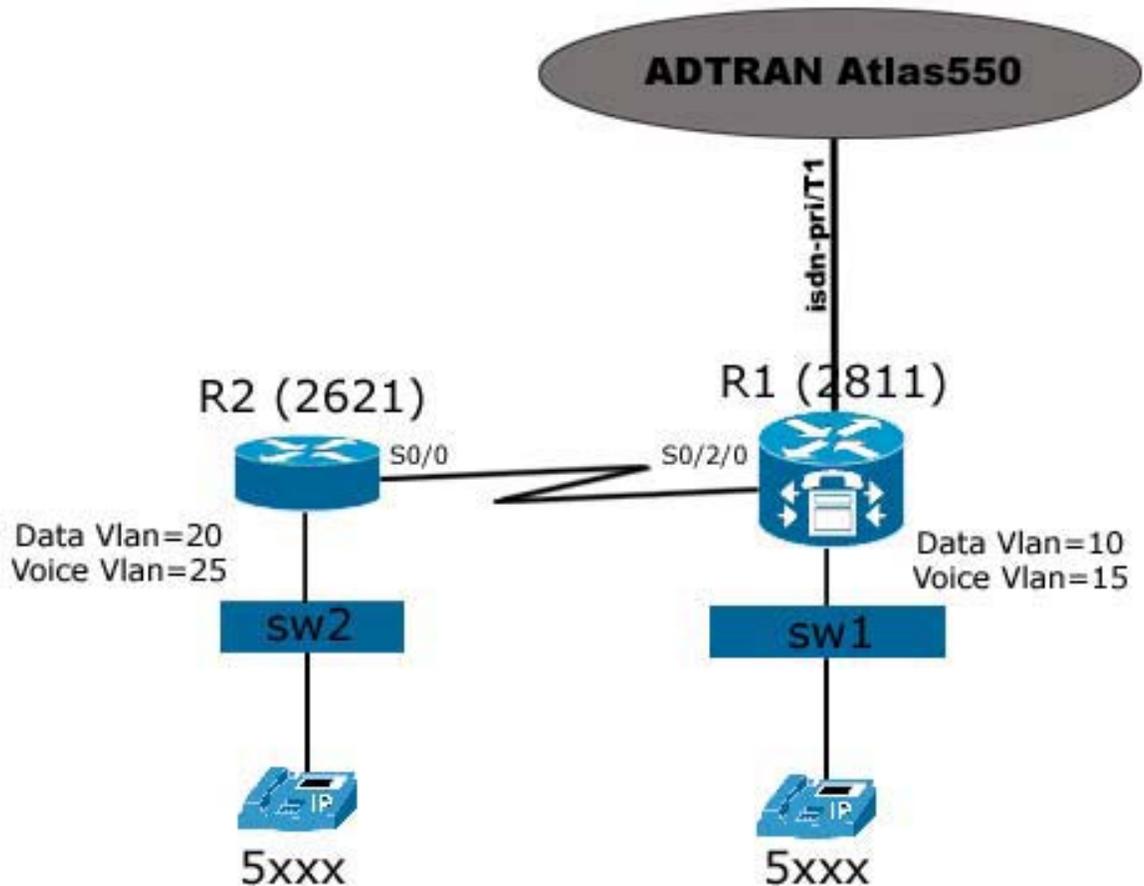## Case Study 1: Registering IP Phones with a remote Call

Manager

Objectives

- Place calls from IP Phones under R1 to IP Phones under R2

- Place calls from any IP Phone (under R1 and under R2) to the regular PSTN/POTS Phone

- Place calls from the regular PSTN/POTS phone to IP Phones.

Scenario

The ACME Inc. has a new office at another city. In order to save money, a VoIP environment will be implemented by you. The CME Router will be located at the company's main office. At the remote office, the remote router (R2) is not able to run the CME software. The IP Phones under R2 must register at main office's CME Router (R1).

Equipment Requirements

2 2811/2621XM;
2 Cisco IP Phones;
1 Regular Phone;
1 ADTRAN Atlas550 with a PSTN and PRI/T1 card;
2 Cisco Vlan Capable Switches;

1 1MFT-T1 card
1 NM-HDV Network Module;
Appropriated cables and power supplies,

Notes:

- If using switch modules connected to the routers instead external 3550/3560 switches, check the "IPTX Appendix A - Using Switch Modules on the CME Routers" document.

- If using a FXO card on R1 to connect the PSTN instead a ISDN-PRI/T1 or a you are planning to use a FXS card to connect regular phones to the routers, check the "IPTX Appendix B - Using a FXO card to reach the PSTN and FXS card to reach regular internal phones" document.

Step 1 Basic Switch Configuration

a. Based on the figure, connect the cables but do NOT connect the IP Phones yet.

b. Connect to the console port of the Switches and power them on. If the switches have a configuration already on them, erase it and reload them.

Note: The vlan information is NOT stored on NVRAM on the newer switches. There is a file, stored on the flash memory, named **vlan.dat**. This file must be erased too.

```
Switch1# delete flash:vlan.dat
Switch1# erase startup-config
Switch1# reload
```

c. Two VLANs must be created and configured on the switch and will be used to separate the voice and data packets. Create the VLANs on the switches (commands to the switch2 are listed after):

```
Switch1# vlan database
Switch1(vlan)# vlan 10 name DATA state active
Switch1(vlan)# vlan 15 name VOICE state active
Switch1(vlan)# exit
```

d. Configure the switch ports where the IP Phones will be connected. Despite the fact that the IP Phones do work with access mode ports, Cisco recommends that you configure the switch ports, connected to the IP Phones, as trunk ports. This way, it is possible to connect a PC on the IP Phone's PC Port (this port is present in most of the Cisco IP Phones). In order to accomplish this task, it is necessary to tell the Switch which of the created VLANs is the Voice Vlan (**switchport voice vlan** command) and to specify the native VLAN on the link between the Switch and the IP Phone. Usually the Native VLAN on this link will be the DATA VLAN (**switchport native vlan** command). The idea is tag the traffic to the IP Phone (IP Phones do support 802.1q) and the traffic

headed to the PC will be be untagged (PC's do not support tagged packets). Finally, configure the port connected to the Router as a trunk port.

```
Switch1(config)# interface range fast0/5 – 10
Switch1(config-if-range)# switchport trunk encapsulation dot1q
Switch1(config-if-range)# switchport trunk native vlan 10
Switch1(config-if-range)# switchport mode trunk
Switch1(config-if-range)# switchport voice vlan 15
Switch1(config-if-range)# no shut


Switch1(config)# interface fast0/1
Switch1(config-if-range)# switchport trunk encapsulation dot1q
Switch1(config-if-range)# switchport mode trunk
Switch1(config-if-range)# no shut
```

e. Configure the switch2 (connected to R2) in a similar way.

```
Switch2# delete flash:vlan.dat
Switch2# erase startup-config
Switch2# reload

Switch2# vlan database
Switch2(vlan)# vlan 20 name DATA state active
Switch2(vlan)# vlan 25 name VOICE state active
Switch2(vlan)# exit

Switch2(config)# interface range fast0/5 – 10
Switch2(config-if-range)# switchport trunk encapsulation dot1q
Switch2(config-if-range)# switchport trunk native vlan 20
Switch2(config-if-range)# switchport mode trunk
Switch2(config-if-range)# switchport voice vlan 25
Switch2(config-if-range)# no shut

Switch2(config)# interface fast0/1
Switch2(config-if-range)# switchport trunk encapsulation dot1q
Switch2(config-if-range)# switchport mode trunk
Switch2(config-if-range)# no shut
```

Step 2: Basic CME Router Configuration

l.   In this lab, you will be working with two routers. Connect to their console port and power them on. If the routers have a configuration already on it, erase the router and reload it.

m. Enter privilege exec mode.

n. Configure the hostname, passwords, interfaces, console and telnet access, eigrp routing protocol like the figure and make sure that both routers can reach each other. Use the **ping** command to troubleshoot, if necessary. The R1 and R2 configuration files are listed end of this document. Your files should be similar to the listed here.

Step 3 Voice CME Router Configuration

3.1 Preparing the Routers to handle the VLANs

Create and configure two sub-interfaces on the fastEthernet 0/0 of R1 and two sub-interfaces on the fastEthernet 0/0 of R2. They will be used by the VLANs configured on the switch.

On R1:
R1(config)# int fa0/0
R1(config-if)# no ip address
R1(config-if)# no shutdown

R1(config)# int fa0/0.10
R1(config)# encapsulation dot1q 10
R1(config)# ip address 10.10.0.1 255.255.255.0
R1(config)# no shut

R1(config)# int fa0/0.15
R1(config)# encapsulation dot1q 15
R1(config)# ip address 10.15.0.1 255.255.255.0
R1(config)# no shut

On R2:
R2(config)# int fa0/0
R2(config-if)# no ip address
R2(config-if)# no shutdown

R2(config)# int fa0/0.20
R2(config-subif)# encapsulation dot1q 20
R2(config-subif)# ip address 10.20.0.1 255.255.255.0
R2(config-subif)# no shut

R2(config-subif)# int fa0/0.25
R2(config-subif)# encapsulation dot1q 25
R2(config-subif)# ip address 10.25.0.1 255.255.255.0
R2(config-subif)# no shut

3.2 Configuring the DHCP Server on the Routers

The PCs and IP Phones will need IP Addresses. Configure the DHCP Pools on both R1 and R2 routers so the routers will be able to teach IP Addresses to PCs and IP Phones.

On R1:
R1(config)# ip dhcp pool DATA
R1(dhcp-config)# network 10.10.0.0 255.255.255.0
R1(dhcp-config)# default-router 10.10.0.1

R1(config)# ip dhcp pool VOICE
R1(dhcp-config)# network 10.15.0.0 255.255.255.0
R1(dhcp-config)# default-router 10.15.0.1
R1(dhcp-config)# option 150 ip 10.15.0.1

Note: The **option ip 150** command is used to provide the ip address of the TFTP server.

On R2:

```
R2(config)# ip dhcp pool DATA
R2(dhcp-config)# network 10.20.0.0 255.255.255.0
R2(dhcp-config)# default-router 10.20.0.1

R2(config)# ip dhcp pool VOICE
R2(dhcp-config)# network 10.25.0.0 255.255.255.0
R2(dhcp-config)# default-router 10.25.0.1
R2(dhcp-config)# option 150 ip 10.25.0.1
```

To exclude the routers ip address from the pool:

On R1:

```
R1(config)# ip dhcp excluded-address 10.10.0.1
R1(config)# ip dhcp excluded-address 10.15.0.1
```

On R2:

```
R2(config)# ip dhcp excluded-address 10.20.0.1
R2(config)# ip dhcp excluded-address 10.25.0.1
```

Add the networks in the routing protocol.

On R1:

```
R1(config)# router eigrp 1
R1(config-router)# network 10.10.0.0 255.255.255.0
R1(config-router)# network 10.15.0.0 255.255.255.0
R1(config-router)# no auto-summary
```

On R2:

```
R2(config)# router eigrp 1
R2(config-router)# network 10.20.0.0 255.255.255.0
R2(config-router)# network 10.25.0.0 255.255.255.0
R2(config-router)# no auto-summary
```

3.3 Configuring the Telephony Service on the CME Router

On this scenario, R1 will be running the Cisco CME Software and R2 is a router which is not able to run CME. All the IP Phones connected to R2 must register on R1.
The first step it to enable the CME software on R1. Use the telephony-service to enable the CME and enter the telephony configure mode.
Note: You can use the telephony-service setup command to start a "configuration script" but it will not be used here.

On R1:
Enter the telephony-service configuration mode by issuing the telephony-service command on the global configuration mode.
Use the max-ephones and max-dn to specify the max number of ephones and the numbers of ephones-dn, respectively.

The `ip source-address` command specifies the address and port used by the CME software.
Create the phones configuration files on the flash memory by issuing the `create cnf-files`.
Specify the format of the dial plan to be used on internal IP Phones using the `dialplan-pattern` command.
Finally, the `secondary-dialtone 9` command sets the 9 digit as the indicator for another dial-tone. It is used to make an off network call(get an outside line)..

```
R1(config)# telephony-service
R1(config-telephony)# max-ephones 4
R1(config-telephony)# max-dn 20
R1(config-telephony)# ip source-address 10.15.0.1 port 2000
R1(config-telephony)# auto assign 1 to 4
R1(config-telephony)# create cnf-files version-stamp Jan 01 2002 00:00:00
R1(config-telephony)# dialplan-pattern 1 5... extension-length 4
R1(config-telephony)# secondary-dialtone 9
R1(config-telephony)# end
```

3.4 Configuring the ephones and ephone-dn's

You will need to setup the ephone-dn's and ephones on R1 in order for the voice service to work.
From the configuration mode, create the ephone-dn's. The `ephone-dn` command is used to create the ephone-dn. `number`, `description` and `name` will be use to specify the phone number of this ephone-dn, set a description to this ephone-dn and a name, respectively.

```
R1(config)# ephone-dn  1  dual-line
R1(config-ephone-dn)# number 5011
R1(config-ephone-dn)# description Fred's Phone
R1(config-ephone-dn)# name Fred

R1(config)# ephone-dn  2  dual-line
R1(config-ephone-dn)# number 5012
R1(config-ephone-dn)# description Barney's Phone
R1(config-ephone-dn)# name Barney
```

Once you have the ephone-dn's created, you can assign them to the e-phones. Use the `ephone` command and sub-commands to setup the e-phones.
The `mac-address` command is used to tell the router the mac address of your IP Phone. You can find this information on a stamp under the IP Phone. Copy the mac address of your IP Phones and replace the "x's" on the command line by it. Use the format xxxx.xxxx.xxxx (4 strings separated by dots). Be careful! If the mac address is wrong, registration on CME will be denied.
Use the `type` command to specify the model of your IP Phone. In the example bellow we are using the Cisco IP Phone 7960. Fell free to change it to your IP Phone Model.
The button command has the function of assign an ephone-dn to a button on your IP Phone. Different models of IP Phone have different numbers of buttons on its panel. In this example we are assigning the ephone-dn 1 to a 7960 phone (mac address xxxx.xxxx.xxxx) panel's button 1 and ephone-dn 2 to the 7940 phone (mac address yyyy.yyyy.yyyy) panel's button 1.

```
R1(config)# ephone  1
R1(config-ephone)# mac-address xxxx.xxxx.xxxx
R1(config-ephone)# type 7960
R1(config-ephone)# button  1:1

R1(config)# ephone  2
R1(config-ephone)# mac-address yyyy.yyyy.yyyy
R1(config-ephone)# type 7940
R1(config-ephone)# button  1:2
```

Note: If you have more IP Phones, just add more ephone and ephone-dn block of commands.

3.5 Configuring the PSTN/POTS interface on the CME Router

The PSTN/POTS cloud is being simulated by the ADTRAN Atlas550. This cloud provides us a PRI/T1 connection so it is necessary to specify the switch type. The isdn switch-type primary-ni command specifies primary-ni as the switch type according with the cloud's switch type. The framing esf and linecode b8zs commands are used to configure the T1 controller according the Atlas550 (the cloud). The pri-group timeslots 1-24 command tells the router which timeslots will be used for data.

```
R1(config)# isdn switch-type primary-ni
R1(config)# controller T1 1/0/0
R1(config-controller)# framing esf
R1(config-controller)# linecode b8zs
R1(config-controller)# pri-group timeslots 1-24
```

Note: If you get an error message like "no resources available" after you issue the pri-group timeslots 1-24 command, check to see if there is enough memory on the NM-HDV module.
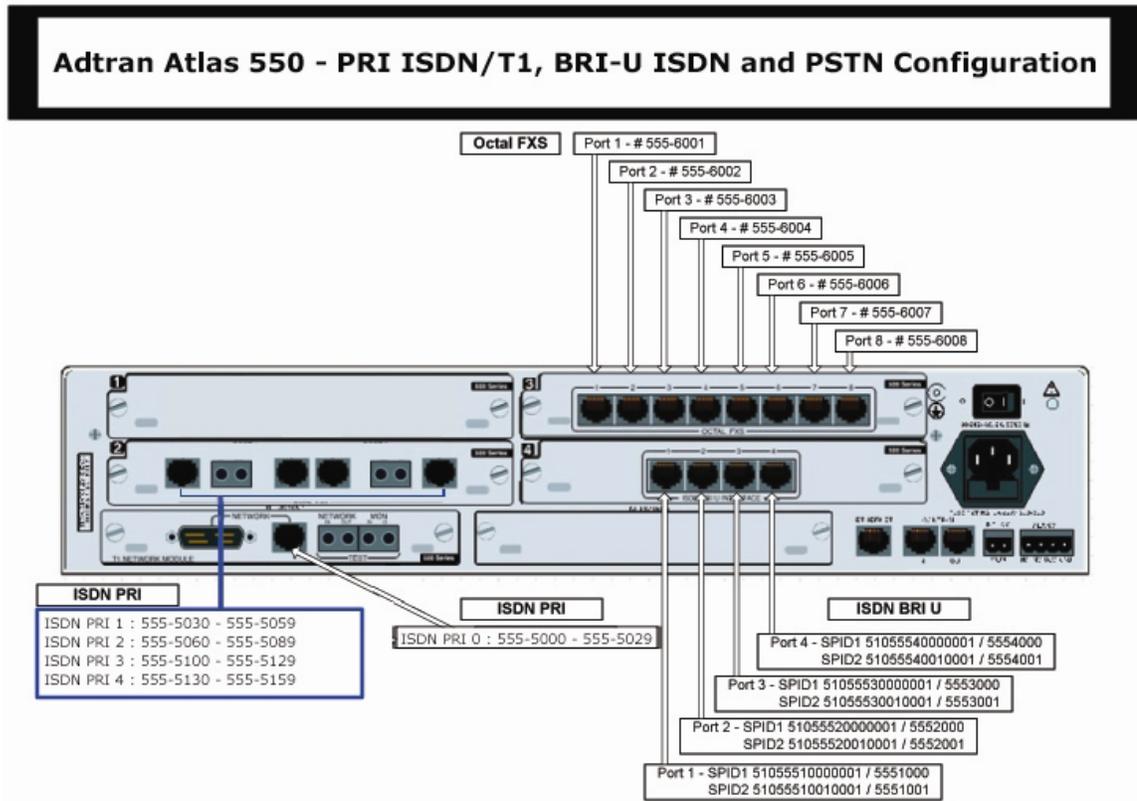
If using the NM-HDV2-1T1/E1 module it is necessary issue the card type t1 <slot> 1 command BEFORE issue the commands above, where <slot> is the slot in use by the Network Module and T1/E1 card. When finished the card type and isdn commands, you should be able to see a voice-port entry on the show running-config command output. This entry is important because will be used inside the dial-peer.

3.6 Configuring Dial Peers

R1 will receive calls from the POTS system so it is necessary to make it handle inward calls. In the real world the dial plan used is E.64. We have created our own dial plan and configured it on the Atlas550. The Atlas550 is configured to forward calls with 555-5xxx numbering destination to the PRI ports.
Use the voice translation-rule, voice translation-profile, translate called and rule commands to setup a translation. In this scenario, the inside IP Phones and regular phones have 5xxx numbering. The idea is split the 555-5xxx number, discard the first 3 digits and keep the 4 last remaining digits.

Note: The ADTRAN Atlas550 must be configured to forward calls to the PRI ports every time a 5555xxx number is dialed. Also, make sure you connected the CME Router's PRI port to the right ADTRAN Atlas550 PRI port

Adtran Atlas 550 - PRI ISDN/T1, BRI-U ISDN and PSTN Configuration

Octal FXS
Port 1 - # 555-6001
Port 2 - # 555-6002
Port 3 - # 555-6003
Port 4 - # 555-6004
Port 5 - # 555-6005
Port 6 - # 555-6006
Port 7 - # 555-6007
Port 8 - # 555-6008

ISDN PRI
ISDN PRI 1 : 555-5030 - 555-5059
ISDN PRI 2 : 555-5060 - 555-5089
ISDN PRI 3 : 555-5100 - 555-5129
ISDN PRI 4 : 555-5130 - 555-5159

ISDN PRI
ISDN PRI 0 : 555-5000 - 555-5029

ISDN BRI U
Port 4 - SPID1 51055540000001 / 5554000
SPID2 51055540010001 / 5554001
Port 3 - SPID1 51055530000001 / 5553000
SPID2 51055530010001 / 5553001
Port 2 - SPID1 51055520000001 / 5552000
SPID2 51055520010001 / 5552001
Port 1 - SPID1 51055510000001 / 5551000
SPID2 51055510010001 / 5551001

```
R1(config)# voice translation-rule 1
R1(cfg-translation-rule)#  rule 1 /^555\(.+\)/ /\1/

R1(config)#voice translation-profile DID
R1(cfg-translation-profile)# translate called 1
```

R2 doesn't have PRI/T1 connection to the POTS/PSTN. There is no need to issue commands on R2.

At this point R1 is almost ready but we still need to setup the dial-peers on it. The dial-peer will handle both the outgoing and incoming calls. R1 must to be able to forward/receive calls to/from the POTS system and forward/receive call to/from other IP Phones or internal regular phones as well (if using an FXS card on R1). To accomplish this, dial-peers will be used.

The translation-profile incoming DID command specifies the translation-profile to be used by the dial-peer.

The destination-pattern 9[2-9]...... command specifies the match string. If the dialed number matches this string, the dial-peer will be used.

This pattern is useful for local calls. If you want long distance or international calls, more patterns must be added here.

The port command specifies which voice port will be used by the router to forward the calls to PSTN.

Use the forward-digits command to specify how many digits, of the dialed number, will be forwarded, from left to the right. In this case, we are forwarding just the PSTN 7 digits, excluding the 9 used to gain the secondary dial tone.

The port command specifies the voice-port used to forward/receive calls. In order to issue this command, you must choose the right voice-port to use. Use the show running-config command and look for a voice-port entry. If your hardware is not working (like cards, network modules, etc) you will not be able to see the voice-port entry on your running-config file.

The show diag command might be useful while troubleshooting hardware issues.

```
R1(config)# dial-peer voice 100 pots
R1(config-dial-peer)# translation-profile incoming DID
R1(config-dial-peer)# destination-pattern 9[2-9]......
R1(config-dial-peer)# incoming called-number .
R1(config-dial-peer)# direct-inward-dial
R1(config-dial-peer)# forward-digits 7
R1(config-dial-peer)# port 1/0/0:23
```

Step 4: Connecting IP Phones, Regular Phones, PCs and placing calls.

Now, everything is setup the IP Phones must be connected on the switches and the user's PC to the IP Phone.

Connect one of the IP Phones to one of the VOICE VLAN's port on SW1 (port 5 to port 10) and the other to one of the VOICE VLAN's port on SW2 (port 5 to port 10 too) on SW2.

Since R2 is not running the CME software, both IP Phones will connect to 10.15.0.1 (R1 ip address) and download the configuration files.

Note: It is common for the IP Phones to reboot or take a little time to register with the CME but if the IP Phones are spending more than 2 minutes trying to contact the TFTP server or it is trying to contact other TFTP server but R1, the IP Phone probably has old configurations. On the IP Phone keypad, use the following key sequence in order to erase any eventual configuration on the IP Phone and reboot it: **#**
use the debug ephone register command to monitor the register process.
(works for Cisco IP Phones 79xx)

After registered check the IP Phones display. You should be able to see the extension number and a name as well (5011/5012 and Fred/Barney). Troubleshoot if necessary.

Configure the network card on the user's PC to learn the network configuration automatically via DHCP and plug the user PC on the IP Phone's PC port. The PC should learn an IP address from the DATA DHCP pool at the router and be able to access the network. Troubleshoot if necessary.

Finally, connect a regular phone to the port 8 of the PSTN/POTS Module on ADTRAN Atlas550. The Atlas550 is configured to use the 555-600[1-8] numbers on the ports 1-8 of the PSTN/ POTS module. Connecting a regular phone to the port 8 of the PSTN/ POTS on Atlas550 makes this phone reachable by the number 555-6008. This regular phone is used to simulate a regular phone located at somewhere else in world.

Step 5: Testing and Troubleshooting

This scenario is simulating a real world situation. R2 is a router located at a remote site and is not able to run the CME software (low memory, old IOS version, etc).

R1 is a CME capable router and is located at the main office. Since it has the CME software running on it, the IP Phones under R2 must register at R1.
Playing the role of main site, R1 has a PSTN connection as well. This connection is achieved through a PRI/T1 link and all the calls headed to the PSTN will be forwarded to this link. Incoming PSTN calls will arrive over this link too and R1 is able to forward them correctly.
The IP Phones are located at the user's office where there is a PC. This PC may be using the IP Phone as the access switch and thus, to access the data network.

Try to place calls from one IP Phone to another using the extension numbers showed on the IP Phone's display. The calls should be successful completed.

Try to place calls from the IP Phones to the regular phone (dial 95556008 from the IP Phones) and from the regular phone to both IP Phones (dial 5555011 and 5555012 from the regular phone). All calls should be working fine. Troubleshoot if necessary.

When working with the Telephony Company Cloud and something doesn't work, it is useful make sure the router is receiving the calls, that the Cloud is forwarding the calls well. For example: If you can see the call process on the router but the IP Phone is not ringing, it means the cloud is doing its job and the problem is probably the way the router is handling the incoming calls.
On this scenario, the connection with the company is simulated using an ISDN/T1 link which is very similar to a real world situation. A very useful command when working with an ISDN and PSTN is the `debug isdn q931` command.

Issue the `debug isdn q931` command on R1, place a call from the PSTN to one of the internal IP Phones and watch the messages on the R1 console while the call is completed. You will be able to see all the ISDN signalization (if the cloud is working fine and forwarding call to the router).
Now disconnect the T1 cable and place the call again. You will not see the ISDN messages because the call can't reach the router since the cable is disconnected.

The `show isdn status` command can be used to make sure your ISDN connection is established. A common output of this command at R1 when the PRI/T1 ISDN link is established and up is similar to the following:

```
r1#sh isdn status
Global ISDN Switchtype = primary-ni
ISDN Serial1/0/0:23 interface
     dsl 0, interface ISDN Switchtype = primary-ni
   Layer 1 Status:
     ACTIVE
   Layer 2 Status:
     TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
   Layer 3 Status:
     0 Active Layer 3 Call(s)
   Active dsl 0 CCBs = 0
   The Free Channel Mask:  0x807FFFFF
   Number of L2 Discards = 0, L2 Session ID = 5
   Total Allocated ISDN CCBs = 0
```

Another useful command is the debug ephone register. This command allows you to track the e-phones register process.
Disconnect on of the IP Phones. Issue the debug ephone register and then connect the IP Phone again. Take a look on the debug messages on the R1 console.

The show running-config, show ip interface brief, show ip route, show ip protocols are very useful and powerful troubleshoot commands as well.

Below are the basic configuration files for R1 and R2. If you are planning to start from these files be careful: on the Cisco Router 2800 series the interface naming convention is changed.

Basic R1 configuration file:

```
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname r1
!
boot-start-marker
boot-end-marker
!
enable secret 5 $1$E70n$H.Rezw/Yhb4EAJVbIrmHa1
!
no aaa new-model
!
resource policy
!
no network-clock-participate slot 1
voice-card 1
 dspfarm
!
ip subnet-zero
!
!
ip cef
no ip dhcp use vrf connected
!
no ip domain lookup
```

```
no ip ips deny-action ips-interface
!
no ftp-server write-enable
!
interface FastEthernet0/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial0/2/0
 ip address 192.168.0.1 255.255.255.0
 clockrate 56000
 no shutdown
!
interface Serial0/2/1
 no ip address
 shutdown
 clockrate 2000000
!
router eigrp 1
  network 192.168.0.0
 no auto-summary
!
ip classless
!
ip http server
no ip http secure-server
!
control-plane
!
!
line con 0
 logging synchronous
line aux 0
line vty 0 4
 password cisco
 login
!
scheduler allocate 20000 1000
!
end
```

Basic R2 configuration file:

```
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname r2
!
```

```
boot-start-marker
boot-end-marker
!
enable secret 5 $1$E70n$H.Rezw/Yhb4EAJVbIrmHa1
!
no aaa new-model
!
resource policy
!
no network-clock-participate slot 1
voice-card 1
 dspfarm
!
ip subnet-zero
!
!
ip cef
no ip dhcp use vrf connected
!
no ip domain lookup
no ip ips deny-action ips-interface
!
no ftp-server write-enable
!
interface FastEthernet0/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial0/2/0
 ip address 192.168.0.2 255.255.255.0
 clockrate 56000
 no shutdown
!
interface Serial0/2/1
 no ip address
 shutdown
 clockrate 2000000
!
router eigrp 1
 network 192.168.0.0
 no auto-summary
!
ip classless
!
ip http server
no ip http secure-server
!
control-plane
!
line con 0
 logging synchronous
line aux 0
```

```
line vty 0 4
 password cisco
 login
!
scheduler allocate 20000 1000
!
end
```

Case Study 2: Registering IP Phones with a local CME and forwarding calls to a remote Call Manager
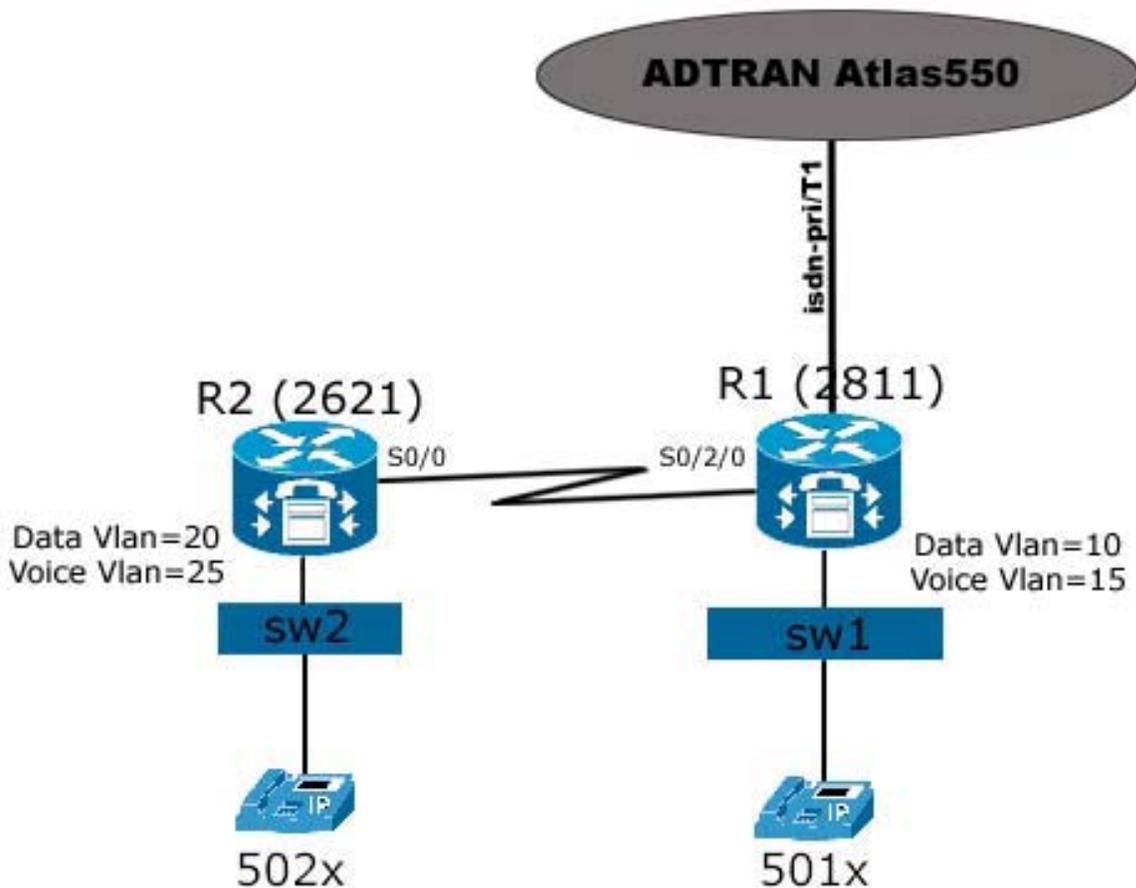
Objectives

- Place calls from IP Phones under R1 to IP Phones under R2

- Place calls from any IP Phone (under R1 and under R2) to the regular PSTN/POTS Phone

- Place calls from the regular PSTN/POTS phone to IP Phones.

Scenario

The ACME Inc. has a new office in another city. In order to save money, a VoIP environment will be implemented by you. One of the CME Routers will be located at the company's main office and the other at the company's remote office. R1 is the Router with PSTN/POTS connection and for this reason it must be configured to forward the POTS/PSTN calls to R2's IP Phones.

R1 has IP Phones under it to and must be able to forward the calls from these IP Phones to R2's IP Phones.

R2 must be able to forward calls to POTS/PSTN and calls to R1's IP Phones as well.

Equipment Requirements

2 2811/2621XM;
2 Cisco IP Phones;
1 Regular Phone;
1 ADTRAN Atlas550 with a PSTN and PRI/T1 card;
2 Cisco Vlan Capable Switches;
1 1MFT-T1 card
1 NM-HDV Network Module;
Appropriated cables and power supplies,

Notes:

- If using switch modules connected to the routers instead external 3550/3560 switches, check the "IPTX Appendix A - Using Switch Modules on the CME Routers" document.

- If using a FXO card on R1 to connect the PSTN instead a ISDN-PRI/T1 or a you are planning to use a FXS card to connect regular phones to the routers, check the "IPTX Appendix B - Using a FXO card to reach the PSTN and FXS card to reach regular internal phones" document.

Step 1 Basic Switch Configuration (Note: If you went through the CS1 lab and still have the switch's configuration on it, you may skip this step)

f. Based on the figure, connect the cables but do NOT connect the IP Phones yet.

g. Connect to the console port of the Switches and power them on. If the switches have a configuration already on them, erase it and reload them.

Note: The vlan information is NOT stored on NVRAM on the newer switches. There is a file, stored on the flash memory, named **vlan.dat**. This file must be erased too.

```
Switch1# delete flash:vlan.dat
Switch1# erase startup-config
Switch1# reload
```

h. Two VLANs must be created and configured on the switch and will be used to separate the voice and data packets. Create the VLANs on the switches (commands to the switch2 are listed after):

```
Switch1# vlan database
```

```
Switch1(vlan)# vlan 10 name DATA state active
Switch1(vlan)# vlan 15 name VOICE state active
Switch1(vlan)# exit
```

i. Configure the switch ports where the IP Phones will be connected. Despite the fact that the IP Phones do work with access mode ports, Cisco recommends that you configure the switch ports, connected to the IP Phones, as trunk ports. This way, it is possible to connect a PC on the IP Phone's PC Port (this port is present in most of the Cisco IP Phones). In order to accomplish this task, it is necessary to tell the Switch which of the created VLANs is the Voice Vlan (**switchport voice vlan** command) and to specify the native VLAN on the link between the Switch and the IP Phone. Usually the Native VLAN on this link will be the DATA VLAN (**switchport native vlan** command). The idea is tag the traffic to the IP Phone (IP Phones do support 802.1q) and the traffic headed to the PC will be be untagged (PC's do not support tagged packets). Finally, configure the port connected to the Router as a trunk port.

```
Switch1(config)# interface range fast0/5 – 10
Switch1(config-if-range)# switchport trunk encapsulation dot1q
Switch1(config-if-range)# switchport trunk native vlan 10
Switch1(config-if-range)# switchport mode trunk
Switch1(config-if-range)# switchport voice vlan 15
Switch1(config-if-range)# no shut


Switch1(config)# interface fast0/1
Switch1(config-if-range)# switchport trunk encapsulation dot1q
Switch1(config-if-range)# switchport mode trunk
Switch1(config-if-range)# no shut
```

j. Configure the switch2 (connected to R2) in a similar way.

```
Switch2# delete flash:vlan.dat
Switch2# erase startup-config
Switch2# reload

Switch2# vlan database
Switch2(vlan)# vlan 20 name DATA state active
Switch2(vlan)# vlan 25 name VOICE state active
Switch2(vlan)# exit

Switch2(config)# interface range fast0/5 – 10
Switch2(config-if-range)# switchport trunk encapsulation dot1q
Switch2(config-if-range)# switchport trunk native vlan 20
Switch2(config-if-range)# switchport mode trunk
Switch2(config-if-range)# switchport voice vlan 25
Switch2(config-if-range)# no shut

Switch2(config)# interface fast0/1
Switch2(config-if-range)# switchport trunk encapsulation dot1q
Switch2(config-if-range)# switchport mode trunk
Switch2(config-if-range)# no shut
```

Step 2 Basic CME Router Configuration

    o. In this lab, you will be working with two routers. Connect to their console port and power them on. If the routers have a configuration already on it, erase the router and reload it.

    p. Enter privilege exec mode.

    q. Configure the hostname, passwords, interfaces, console and telnet access, eigrp routing protocol like the figure and make sure that both routers can reach each other. Use the **ping** command to troubleshoot, if necessary. The R1 and R2 configuration files are listed end of this document and your files should be similar to the listed here.

       Note: Be careful. The 2800 series router has changed the interface naming convention.

Step 3 Voice CME Router Configuration

3.1 Preparing the Routers to handle the VLANs

Create and configure two sub-interfaces on the fastEthernet 0/0 of R1 and two sub-interfaces on the fastEthernet 0/0 of R2. They will be used by the VLANs configured on the switch.

On R1:
```
R1(config)# int fa0/0
R1(config-if)# no ip address
R1(config-if)# no shutdown

R1(config)# int fa0/0.10
R1(config)# encapsulation dot1q 10
R1(config)# ip address 10.10.0.1 255.255.255.0
R1(config)# no shut

R1(config)# int fa0/0.15
R1(config)# encapsulation dot1q 15
R1(config)# ip address 10.15.0.1 255.255.255.0
R1(config)# no shut
```

On R2:
```
R2(config)# int fa0/0
R2(config-if)# no ip address
R2(config-if)# no shutdown

R2(config)# int fa0/0.20
R2(config-subif)# encapsulation dot1q 20
R2(config-subif)# ip address 10.20.0.1 255.255.255.0
R2(config-subif)# no shut

R2(config-subif)# int fa0/0.25
R2(config-subif)# encapsulation dot1q 25
R2(config-subif)# ip address 10.25.0.1 255.255.255.0
R2(config-subif)# no shut
```

3.2 Configuring the DHCP Server on the Routers

The PCs and IP Phones will need IP Addresses. Configure the DHCP Pools on both R1 and R2 routers so the routers will be able to teach IP information to PCs and IP Phones.

On R1:
```
R1(config)# ip dhcp pool DATA
R1(dhcp-config)# network 10.10.0.0 255.255.255.0
R1(dhcp-config)# default-router 10.10.0.1

R1(config)# ip dhcp pool VOICE
R1(dhcp-config)# network 10.15.0.0 255.255.255.0
R1(dhcp-config)# default-router 10.15.0.1
R1(dhcp-config)# option 150 ip 10.15.0.1
```

Note: The **option ip 150** command is used to provide the ip address of the TFTP server.

On R2:
```
R2(config)# ip dhcp pool DATA
R2(dhcp-config)# network 10.20.0.0 255.255.255.0
R2(dhcp-config)# default-router 10.20.0.1

R2(config)# ip dhcp pool VOICE
R2(dhcp-config)# network 10.25.0.0 255.255.255.0
R2(dhcp-config)# default-router 10.25.0.1
R2(dhcp-config)# option 150 ip 10.25.0.1
```

To exclude the routers ip address from the pool:

On R1:
```
R1(config)# ip dhcp excluded-address 10.10.0.1
R1(config)# ip dhcp excluded-address 10.15.0.1
```

On R2:
```
R2(config)# ip dhcp excluded-address 10.20.0.1
R2(config)# ip dhcp excluded-address 10.25.0.1
```

Add the networks in the routing protocol.

On R1:
```
R1(config)# router eigrp 1
R1(config-router)# network 10.10.0.0 255.255.255.0
R1(config-router)# network 10.15.0.0 255.255.255.0
R1(config-router)# no auto-summary
```

On R2:
```
R2(config)# router eigrp 1
R2(config-router)# network 10.20.0.0 255.255.255.0
R2(config-router)# network 10.25.0.0 255.255.255.0
R2(config-router)# no auto-summary
```

3.3 Configuring the Telephony Service on the CME Router

On this scenario, R1 and R2 will be running the Cisco CME Software. The IP Phones connected at R1 will register on R1 and the IP Phones connected to R2 will register on R2. The first step it to enable the CME software on R1/R2. Use the `telephony-service` to enable the CME and enter the telephony configure mode.
Note: You can use the `telephony-service setup` command to start a "configuration script" but it will not be used here.

On R1:
Enter the telephony-service configuration mode by issuing the `telephony-service` command on the global configuration mode.
Use the `max-ephones` and `max-dn` to specify the max number of ephones and the numbers of ephones-dn, respectively.
The `ip source-address` command specifies the address and port used by the CME software.
Create the phones configuration files on the flash memory by issuing the `create cnf-files`. Specify the format of the dial plan to be used on internal IP Phones using the `dialplan-pattern` command.
Finally, the `secondary-dialtone 9` command sets the 9 digit as the indicator for another dial-tone. It is used to make an off network call (get an outside line)..

```
R1(config)# telephony-service
R1(config-telephony)# max-ephones 4
R1(config-telephony)# max-dn 20
R1(config-telephony)# ip source-address 10.15.0.1 port 2000
R1(config-telephony)# auto assign 1 to 4
R1(config-telephony)# create cnf-files version-stamp Jan 01 2002 00:00:00
R1(config-telephony)# dialplan-pattern 1 5... extension-length 4
R1(config-telephony)# secondary-dialtone 9
R1(config-telephony)# end
```

On R2:
R2 needs similar configurations but as long R2 don't have a connection to PSTN/POTS some commands will not be necessary.
Issue the following command on R2:

```
R2(config)# telephony-service
R2(config-telephony)# max-ephones 4
R2(config-telephony)# max-dn 20
R2(config-telephony)# ip source-address 10.25.0.1 port 2000
R2(config-telephony)# auto assign 1 to 4
R2(config-telephony)# create cnf-files version-stamp Jan 01 2002 00:00:00
R2(config-telephony)# dialplan-pattern 1 5... extension-length 4
R2(config-telephony)# end
```

3.4 Configuring the ephones and ephone-dn's

You will need to setup the ephone-dn's and ephones on R1 in order for the voice service to work.
From the configuration mode, create the ephone-dn's. The `ephone-dn` command is used to create the ephone-dn. `number`, `description` and `name` will be use to specify the phone number of this ephone-dn, set a description to this ephone-dn and a name, respectively.

On R1:
```
R1(config)# ephone-dn  1  dual-line
R1(config-ephone-dn)# number 5011
R1(config-ephone-dn)# description Fred's Phone
R1(config-ephone-dn)# name Fred

R1(config)# ephone-dn  2  dual-line
R1(config-ephone-dn)# number 5012
R1(config-ephone-dn)# description Barney's Phone
R1(config-ephone-dn)# name Barney
```

Once you have the ephone-dn's created, you can assign them to the e-phones. Use the `ephone` command and sub-commands to setup the e-phones.
The `mac-address` command is used to tell the router the mac address of your IP Phone. You can find this information on a stamp under the IP Phone. Copy the mac address of your IP Phones and replace the "x" character on the command line by it. Use the format xxxx.xxxx.xxxx (4 strings separated by dots). Be careful! If the mac address is wrong, registration on CME will be denied.
Use the `type` command to specify the model of your IP Phone. In the example bellow we are using the Cisco IP Phone 7960. Fell free to change it according to your IP Phone Model.
The button command has the function of assign an ephone-dn to a button on your IP Phone. Different models of IP Phone have different numbers of buttons on its panel. In this example we are assigning the ephone-dn 1 to the 7960 phone (mac address xxxx.xxxx.xxxx) panel's button 1 and ephone-dn 2 to the 7940 phone (mac address yyyy.yyyy.yyyy) panel's button 1.

```
R1(config)# ephone  1
R1(config-ephone)# mac-address xxxx.xxxx.xxxx
R1(config-ephone)# type 7960
R1(config-ephone)# button  1:1

R1(config)# ephone  2
R1(config-ephone)# mac-address yyyy.yyyy.yyyy
R1(config-ephone)# type 7940
R1(config-ephone)# button  1:2
```

On R2:
Again, R2 needs some similar commands and you can change the phone type and MAC addresses to fit your equipment.
Note we are changing the numbers. R2's IP Phones have 502x dial plan while R1's IP Phones have 501x dial plan. Issue the following on R2:

```
R2(config)# ephone-dn  1  dual-line
R2(config-ephone-dn)# number 5021
R2(config-ephone-dn)# description George's Phone
```

```
R2(config-ephone-dn)# name George

R2(config)# ephone-dn  2  dual-line
R2(config-ephone-dn)# number 5022
R2(config-ephone-dn)# description MrSpacely's Phone
R2(config-ephone-dn)# name MrSpacely

R2(config)# ephone  1
R2(config-ephone)# mac-address uuuu.uuuu.uuuu
R2(config-ephone)# type 7960
R2(config-ephone)# button  1:1

R2(config)# ephone  2
R2(config-ephone)# mac-address vvvv.vvvv.vvvv
R2(config-ephone)# type 7940
R2(config-ephone)# button  1:2
```

Note: Despite two IP Phones were configured on R1 and on R2, this lab just requires one IP Phone under R1 and one IP Phone under R2.

3.5 Configuring the PSTN/POTS interface on the CME Router

The PSTN/POTS cloud is being simulated by the ADTRAN Atlas550. This cloud provides us a PRI/T1 connection so it is necessary to specify the switch type. The isdn switch-type primary-ni command specifies primary-ni as the switch type according with the cloud's switch type. The framing esf and linecode b8zs commands are used to configure the T1 controller according the Atlas550 (the cloud). The pri-group timeslots 1-24 command tells the router which timeslots will be used for data.

```
R1(config)# isdn switch-type primary-ni
R1(config)# controller T1 1/0/0
R1(config-controller)# framing esf
R1(config-controller)# linecode b8zs
R1(config-controller)# pri-group timeslots 1-24
```

Note: If you get an error message like "no resources available" after you issue the pri-group timeslots 1-24 command, check to see if there is enough memory on the NM-HDV module.

If using the NM-HDV2-1T1/E1 module it is necessary issue the card type t1 <slot> 1 command before issue the commands above, where <slot> is the slot in use by the Network Module and T1/E1 card. When finished the card-type and isdn commands, you should be able to see a voice-port entry on the show running-config command output. This entry is important because will be used inside the dial-peer.
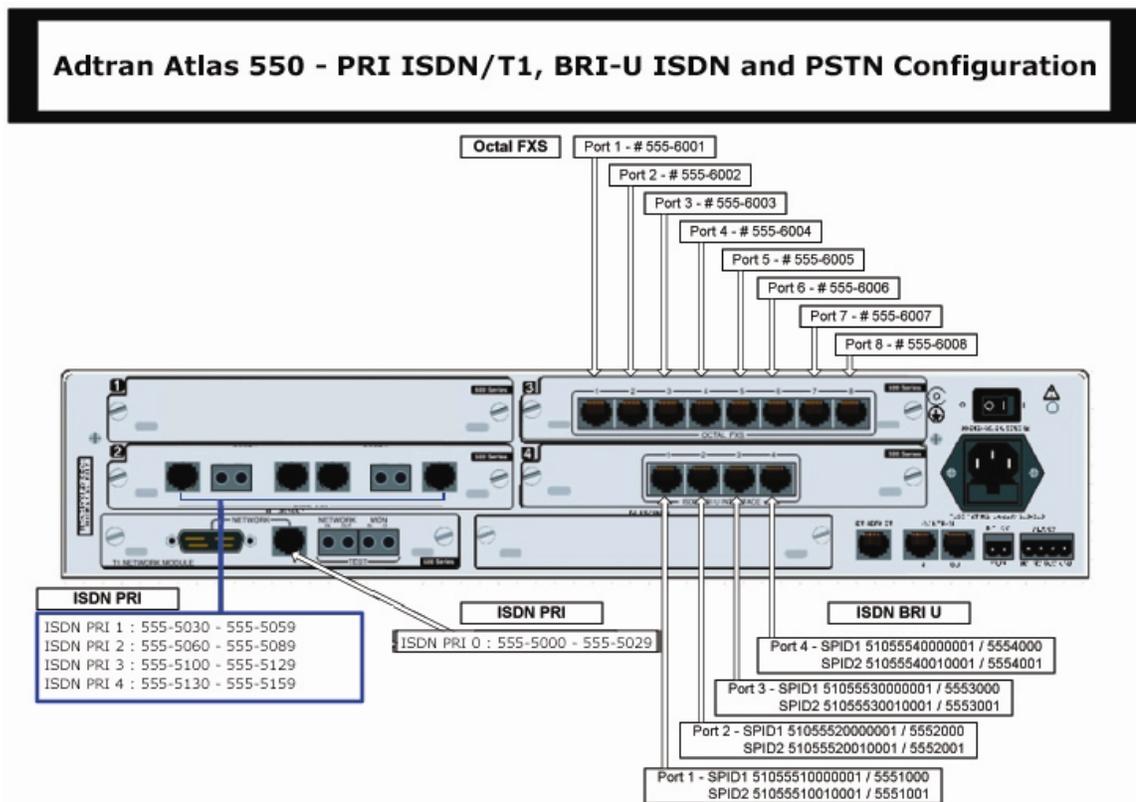
Since R2 doesn't have POTS/PSTN connection, it doesn't need any commands at this point.

3.6 Configuring Dial Peers

R1 will receive calls from the POTS system, so it is necessary to make it handle these calls. In the real world the dial plan used is E.64 but here we have created our own dial plan and configured it on the Atlas550. The Atlas550 is configured to forward calls with 555-5xxx numbering destination to the PRI ports.

Use the voice translation-rule, voice translation-profile, translate called and rule commands to setup a translation. In this scenario, the inside IP Phones and regular phones have 5xxx numbering. The idea is split the 555-5xxx number, discard the first 3 digits and keep the 4 last remaining digits.

Note: The ADTRAN Atlas550 must be configured to forward calls to the PRI ports every time a 5555xxx number is dialed. Also, make sure you connected the CME Router's PRI port to the right ADTRAN Atlas550 PRI port.

Adtran Atlas 550 - PRI ISDN/T1, BRI-U ISDN and PSTN Configuration

```
R1(config)# voice translation-rule 1
R1(cfg-translation-rule)#  rule 1 /^555\(.+\)/ /\1/

R1(config)#voice translation-profile DID
R1(cfg-translation-profile)# translate called 1
```

R2 doesn't have PRI/T1 connection to the POTS/PSTN. There is no need to issue commands on R2 here.

At this point R1 and R2 are almost ready but we still need to setup the dial-peers on them. The dial-peer will handle both the outgoing and incoming calls.
Since R1 will receive call from the POTS system and from the CME software at R2, it is necessary to make it handle both kinds of calls. In order to accomplish this task, R1 will

need 2 dial-peers: one pots dial-peer type (will handle POTS/PSTN calls) and one voip dial-peer type (will handle the calls between R1 and R2).

R1 must to be able to forward/receive calls to/from the POTS system and forward/receive calls to/from other IP Phones (registered at R1 and registered at R2) or internal regular phones as well (if using an FXS card on R1/R2, see IPTX Case Study Appendix-B). R2 must be able to forward calls from R2 IP Phones to R1's IP Phones, calls from R2's IP Phones to PSTN regular phones and to handle incoming calls to its IP Phones. To accomplish these tasks, dial-peers will be used.

The pots dial-peer at R1 will be responsible for the PSTN/POTS calls. Here are the commands:

```
R1(config)# dial-peer voice 100 pots
R1(config-dial-peer)# translation-profile incoming DID
R1(config-dial-peer)# destination-pattern 9[2-9]......
R1(config-dial-peer)# incoming called-number .
R1(config-dial-peer)# direct-inward-dial
R1(config-dial-peer)# forward-digits 7
R1(config-dial-peer)# port 1/0/0:23

R1(config)# dial-peer voice 200 voip
R1(config-dial-peer)# destination-pattern 502.
R1(config-dial-peer)# session target ipv4:192.168.0.2
```

The translation-profile incoming DID command specifies the translation-profile to be used by the dial-peer and only R1 needs it. (R2 doesn't have POTS/PSTN connection)
The destination-pattern 9[2-9]...... command specifies the match string. If the dialed number matches this string, the dial-peer will be used.
This pattern is useful for local calls. If you want long distance or international calls, more patterns must be added here.
The port command specifies which voice port will be used by the router to forward the calls to PSTN.
Use the forward-digits command to specify how many digits, of the dialed number, will be forwarded. When forwarding calls to PSTN/POTS, R1 is forwarding just the PSTN 7 digits, excluding the 9 used to gain the secondary dial tone.
The port command specifies the voice-port used to forward/receive calls. In order to issue this command, you must choose the right voice-port to use. Use the show running-config command and look for a voice-port entry. If your hardware is not working (like cards, network modules, etc) you will not be able to see the voice-port entry on your running-config file.
The show diag command might be useful while troubleshooting hardware issues.

The voip dial-peer at R1 will handle the calls between R1 and R2.
The destination-pattern 502. says to the CME software to forward any call starting by 502 to the address specified on the session target command (in this case, the address of the R2 serial port connected to R1).

On R2:

Some small but important changes must be made on R2 commands. It is necessary to change the dial-peer type. On R1 the pots dial-peer type was used but since R2 has no direct connection to the PSTN/POTS, all calls from R2, heading phones outside from R2 have to pass through R1, only a voip dial-peer type will be necessary. Create 2 voip dial-peers type: one to handle calls to PSTN/POTS and another to handle calls to R1's IP Phones.

```
R2(config)# dial-peer voice 100 voip
R2(config-dial-peer)# destination-pattern 501.
R2(config-dial-peer)# session target ipv4:192.168.0.1
R2(config-dial-peer)# incoming called-number .

R2(config)# dial-peer voice 101 voip
R2(config-dial-peer)# destination-pattern 9[2-9]......
R2(config-dial-peer)# session target ipv4:192.168.0.1
R2(config-dial-peer)# incoming called-number .
```

Step 4: Connecting IP Phones, Regular Phones, PCs and placing calls.

Now, everything is setup the IP Phones must be connected on the switches and the user's PC to the IP Phone.

Connect one of the IP Phones to one of the VOICE VLAN's port on SW1 (port 5 to port 10) and the other to one of the VOICE VLAN's port on SW2 (port 5 to port 10 too) on SW2.

One of the phones will register on R1 and the other on R2 and download the configuration files.

Note: It is common for the IP Phones to reboot or take a little time to register with the CME but if the IP Phones are spending more than 2 minutes trying to contact the TFTP server or it is trying to contact other TFTP server but R1/R2, the IP Phone probably has old configurations. On the IP Phone keypad, use the following key sequence in order to erase any eventual configuration on the IP Phone and reboot it: * * # * *
(works for Cisco IP Phone 79xx)

After register, check the IP Phones display. You should be able to see the extension number and a name as well (5011/5021 and Fred/Barney/George/MrSpacely). Troubleshoot if necessary.

Note the debug ephone register command can be very useful when troubleshooting register problems.

Configure the network card on the user's PC to learn the network configuration automatically via DHCP and plug the user PC on the IP Phone's PC port. The PC should learn an IP address from the DATA pool at the router and be able to access the network. Troubleshoot if necessary.

Finally, connect a regular phone to the port 8 of the PSTN/POTS Module on ADTRAN Atlas550. The Atlas550 is configured to use the 555-600[1-8] numbers on the ports 1-8 of

the PSTN/ POTS module. Connecting a regular phone to the port 8 of the PSTN/ POTS on Atlas550 makes this phone reachable by the number 555-6008. This regular phone is used to simulate a regular phone located at somewhere else in world.

Step 5: Testing and Troubleshooting

R1 and R2 are CME capable routers. R1 is located at the main office and R2 is located at the remote office. The phones under R1 will register at R1 and the phones under R2 must register at R2.

Playing the role of main site, R1 has a PSTN connection as well. This connection is achieved through a PRI/T1 link and all the calls headed to the PSTN will be forwarded to this link. Incoming PSTN calls will arrive over this link too and R1 is able to forward them correctly.
The IP Phones are located at the user's office where there is a PC. This PC may be using the IP Phone as the access switch and thus, to access the data network.

Try to place calls from one IP Phone to another, use the extension numbers showed on the IP Phone's display. The calls should be successful completed.

Try to place calls from the IP Phones to the regular phone (dial 95556008 from the IP Phones) and from the regular phone to both IP Phones (dial 5555011 and 5555021 from the regular phone). All calls should be working fine. Troubleshoot if necessary.

When working with the Telephony Company Cloud and something doesn't work, it is useful make sure the router is receiving the calls, that the Cloud is forwarding the calls well. If you can see the calls process on the router, it means the clouds is doing its job and the problem is probably the way the router is handling the incoming calls.
On this scenario, the connection with the company is simulated using an ISDN/T1 link which is very similar to a real world situation. A very useful command when working with an ISDN and PSTN is the debug isdn q931 command.

The show isdn status command can be used to make sure your ISDN connection is established. A common output of this command at R1 when the PRI/T1 ISDN link is established and up is similar to the following:

```
r1#sh isdn status
Global ISDN Switchtype = primary-ni
ISDN Serial1/0/0:23 interface
     dsl 0, interface ISDN Switchtype = primary-ni
   Layer 1 Status:
     ACTIVE
   Layer 2 Status:
     TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
   Layer 3 Status:
     0 Active Layer 3 Call(s)
   Active dsl 0 CCBs = 0
   The Free Channel Mask:  0x807FFFFF
   Number of L2 Discards = 0, L2 Session ID = 5
```

Total Allocated ISDN CCBs = 0

Another useful command is the debug ephone register. This command allows you to track the e-phones register process.
Disconnect on of the IP Phones. Issue the debug ephone register and then connect the IP Phone again. Take a look on the debug messages on the R1 console.

The show running-config, show ip interface brief, show ip route, show ip protocols are very useful and powerful troubleshoot commands as well.

Below are the basic configuration files for R1 and R2. If you are planning to start from these files be careful: on the Cisco Router 2800 series the interface naming convention is changed.

Basic R1 configuration file:

```
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname r1
!
boot-start-marker
boot-end-marker
!
enable secret 5 $1$E70n$H.Rezw/Yhb4EAJVbIrmHa1
!
no aaa new-model
!
resource policy
!
no network-clock-participate slot 1
voice-card 1
 dspfarm
!
ip subnet-zero
!
!
ip cef
no ip dhcp use vrf connected
!
no ip domain lookup
no ip ips deny-action ips-interface
!
no ftp-server write-enable
!
interface FastEthernet0/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
```

```
!
interface Serial0/2/0
 ip address 192.168.0.1 255.255.255.0
 clockrate 56000
 no shutdown
!
interface Serial0/2/1
 no ip address
 shutdown
 clockrate 2000000
!
router eigrp 1
  network 192.168.0.0
 no auto-summary
!
ip classless
!
ip http server
no ip http secure-server
!
control-plane
!
!
line con 0
 logging synchronous
line aux 0
line vty 0 4
 password cisco
 login
!
scheduler allocate 20000 1000
!
end
```

Basic R2 configuration file:

```
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname r2
!
boot-start-marker
boot-end-marker
!
enable secret 5 $1$E70n$H.Rezw/Yhb4EAJVbIrmHa1
!
no aaa new-model
!
resource policy
no network-clock-participate slot 1
voice-card 1
 dspfarm
!
ip subnet-zero
!
ip cef
```

```
no ip dhcp use vrf connected
!
no ip domain lookup
no ip ips deny-action ips-interface
!
no ftp-server write-enable
!
interface FastEthernet0/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial0/2/0
 ip address 192.168.0.2 255.255.255.0
 clockrate 56000
 no shutdown
!
interface Serial0/2/1
 no ip address
 shutdown
 clockrate 2000000
!
router eigrp 1
 network 192.168.0.0
 no auto-summary
!
ip classless
!
ip http server
no ip http secure-server
!
control-plane
!
line con 0
 logging synchronous
line aux 0
line vty 0 4
 password cisco
 login
!
scheduler allocate 20000 1000
!
end
```

Note: Be careful. The 2800 series router has changed the interface naming convention.