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| Image result for cisco backgroun |
| VoIP with CMEDerek Liu |
| Periods 0,3,4  Mr. Mason & Mr. Hansen  CCNP Lab 5 |

**VoIP and FXO Configuration with CME**

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Purpose

Other than data, networks also can also route voice. This is possible through voice over IP or VoIP. Voice samples can be transmitted over the internet rather than standard copper phone lines. However, rather than just configuring VoIP, we also explored possible ways for VoIP phones to access analog lines and make calls through there.

Background Information

Voice over IP or VoIP is a technology that allows phone calls to be made over the internet. VoIP converts voice into digital signals and sends them over the internet. These signals are then interpreted and converted back to voice. This is a very call-effective method of calling as it doesn’t require the setup of copper phone lines since the internet is used as the medium for transferring the data. While we were only using VoIP in a LAN environment, calls can be made across a WAN through h.323 standards. H.323 defines the signaling and control messages and is commonly used in conjunction with SIP or SCCP and MGCP.

In this lab, we used SCCP as seen through the firmware being pushed to the phones through TFTP (firmware starts with P00). SCCP stands for skinny call control protocol and is Cisco proprietary. It is used for signaling and control of voice and video calls between Cisco IP phones. IT is often used in conjunction with with RTP and RTCP to transmit both audio and voice over IP networks. SCCP can be used with either CUCM or CUCME. The main difference between SIP and SCCP is that SCCP is Cisco proprietary.

In this lab, VoIP was made possible through Cisco unified communication manager express or CUCME. The CUCME is a software-based IP private branch exchange (PBX) system that runs on Cisco routers. This is convenient for small to medium sized businesses as they won’t need to have a dedicated server in order to run CUCM.

The phones in this lab didn’t need to be configured beyond just plugging it into a PoE switch. This is due to TFTP which allows for configuration files to be sent through UDP to the phones. TFTP is often used in network environments where it is necessary to transfer files to or from devices that do not have the resources or capabilities to support more complex file transfer protocols. It is also commonly used to transfer configuration files and software images to and from Cisco devices.

VoIP was originally developed around 1995 by a company called VocalTec. This voice technology is relatively new and is slowly being incorporated into our networks as an alternative to traditional phone services. Currently, it is being used in a lot of popular services such as Skype, WhatsApp, Google Hangouts, and Discord.

Lab Summary

In this lab, we established a LAN connection between two Cisco VoIP phones, meaning they were able to call each other with VoIP. One of the 7940 model and the other of the 7960 model. This was done by using Cisco unified communication manager express or CUCME/CME. One Cisco 2811 router with FXO voice cards was used to run CME and transfer calls from VoIP to analog. A PoE switch was used to provide power to the VoIP phones and create routes to the internet. Two computers were connected to the phones and configured with DHCP and were able to access the internet. This lab was considered successful when the computers were able to access the internet, the Cisco phones were able to contact each other through VoIP, and the Cisco phone was able to make a call to a cell phone. This last requirement was tested by making a call to a local pizza delivery place and ordering a pizza.

Lab Commands

**(Bolded comments are not part of the commands used and are just comments)**

**Switch Commands**

vlan 20

name VOICE **name vlan. We have a vlan 10 for data but realized it wasn’t necessary.**

interface FastEthernet0/1

switchport trunk encapsulation dot1q **trunk connection for router on a stick**

switchport trunk native vlan 50 **assigns native vlan as 50**

switchport mode trunk

interface FastEthernet0/2

switchport mode access

switchport voice vlan 20 **assign voice vlan**

spanning-tree portfast **not necessary for this lab**

**Router Commands**

ip dhcp excluded-address 192.168.20.1 192.168.20.5 **excludes dhcp addresses that can be assigned**

ip dhcp pool VOICE20 **name dhcp pool**

network 192.168.20.0 255.255.255.0

default-router 192.168.20.1

option 150 ip 192.168.20.1 **Cisco proprietary command for Cisco IP Phones**

interface FastEthernet0/0

ip address dhcp **interface requests dhcp address**

interface FastEthernet0/0.20 **activates sub interface**

encapsulation dot1Q 20

ip address 192.168.20.1 255.255.255.0

tftp-server P00308000500.sbn **sends files that the Cisco phones require**

tftp-server P00308000500.loads

tftp-server flash:P00308000500.bin alias P00308000500

tftp-server flash:P00308000500.sb2

voice-port 1/0/0 **the port number for our FXO port**

ring number 3 **number of rings before FXO port answers call**

connection plar opx 1010

caller-id enable **transfers the caller’s telephone number to the called**

dial-peer voice 83 pots **creates a dial-peer. The number doesn’t matter**

destination-pattern 91[2-9]..[2-9]...... **destination pattern to forward calls. 9 to call out. 1 for US country code. [2-9] to prevent accidental dialing of 911. Dots represent digits of the 10 digit phone number.**

port 1/0/0 **fxo port number**

forward-digits all **forwards digits to the called**

telephony-service **enables telephony service**

max-ephones 2 **max of 2 ephones in our network**

max-dn 2 **max of 2 directory numbers in our network**

ip source-address 192.167.20.1 port 2000 **source address for SCCP messages**

auto assign 1 to 2 **automatically assign dn numbers to the two phones**

system message ad astra per aspera

ephone-dn 1

number 1010 **assign a number to ephone**

ephone 1

mac-address 001D.A219.FA62 **configure mac-address for ephone**

type 7940 **model of the phone**

button 1:2

Network Diagram

Diagram

Description automatically generated

Process

Start by creating a network like the one shown in the Network Diagram section. In this case we used a Cisco 2811 Router to run CUCME and PoE switch in order to provide power to the IP phones.

The computers behind the IP phones should be able to receive an IP address at this point when set as a DHCP client. First, ensure that the router has CUCME in its flash. This can be confirmed with a show flash: command. In our case, we didn’t have CME files in our router. After establishing a connection between the computer, installing a TFTP server, we copied a CME tar file into the router with the command copy tftp://10.0.0.2/ flash: (file-name). The tar file was then extracted with archive tar /extract (file name) flash: and CME files could be found the flash.

Something that could not be accurately modeled in the Network Diagram section was the FXO port connection. The in the 2811 router, there were FXO port voice interface cards. These were connected with an RJ11 jack to an analog line.

Configurations

***Router Configuration:***

hostname Router

boot-start-marker

boot-end-marker

logging message-counter syslog

no aaa new-model

memory-size iomem 10

no network-clock-participate slot 1

dot11 syslog

ip source-route

no ip cef

ip dhcp excluded-address 192.168.20.1 192.168.20.5

ip dhcp pool VOICE20

network 192.168.20.0 255.255.255.0

default-router 192.168.20.1

option 150 ip 192.168.20.1

no ipv6 cef

multilink bundle-name authenticated

voice-card 0

no dspfarm

voice-card 1

no dspfarm

vtp domain cisco

vtp mode transparent

archive

log config

hidekeys

vlan 20

interface FastEthernet0/0

ip address dhcp

duplex auto

speed auto

no shutdown

interface FastEthernet0/0.20

encapsulation dot1Q 20

ip address 192.168.20.1 255.255.255.0

interface FastEthernet0/0.50

encapsulation dot1Q 50 native

interface FastEthernet0/1

no ip address

shutdown

duplex auto

speed auto

interface FastEthernet0/0/0

interface FastEthernet0/0/1

interface FastEthernet0/0/2

interface FastEthernet0/0/3

interface Serial0/1/0

no ip address

shutdown

interface Serial0/2/0

no ip address

shutdown

clock rate 2000000

interface Serial0/2/1

no ip address

shutdown

clock rate 2000000

interface Serial0/3/0

no ip address

shutdown

clock rate 2000000

interface Serial0/3/1

no ip address

shutdown

clock rate 2000000

interface Vlan1

no ip address

shutdown

ip forward-protocol nd

no ip http server

no ip http secure-server

ip flow-export version 9

tftp-server P00308000500.sbn

tftp-server P00308000500.loads

tftp-server flash:P00308000500.bin alias P00308000500

tftp-server flash:P00308000500.sb2

control-plane

voice-port 1/0/0

ring number 3

connection plar opx 1010

caller-id enable

voice-port 1/0/1

voice-port 1/0/2

voice-port 1/0/3

voice-port 1/1/0

voice-port 1/1/1

dial-peer voice 82 pots

destination-pattern 9[2-9]..[2-9]......

port 1/0/0

forward-digits 10

dial-peer voice 83 pots

destination-pattern 91[2-9]..[2-9]......

port 1/0/0

forward-digits all

dial-peer voice 81 pots

destination-pattern 9[469]11

port 1/0/0

forward-digits 3

telephony-service

max-ephones 2

max-dn 2

ip source-address 192.167.20.1 port 2000

auto assign 1 to 2

system message ad astra per aspera

max-conferences 8 gain -6

transfer-system full-consult

create cnf-files version-stamp Jan 01 2002 00:00:00

ephone-dn 1

number 1010

ephone-dn 2

number 1020

ephone 1

device-security-mode none

mac-address 001D.A219.FA62

type 7940

button 1:2

ephone 2

device-security-mode none

mac-address 0015.2B47.6685

type 7960

button 1:1

line con 0

line aux 0

line vty 0 4

login

scheduler allocate 20000 1000

end

***Switch Configuration:***

hostname Switch

boot-start-marker

boot-end-marker

no aaa new-model

system mtu routing 1500

vtp domain CCNP

vtp mode transparent

authentication mac-move permit

ip subnet-zero

spanning-tree mode pvst

spanning-tree etherchannel guard misconfig

spanning-tree extend system-id

vlan internal allocation policy ascending

vlan 10

name DATA

vlan 20

name VOICE

vlan 50

name NATIVE

vlan 99

name MANAGEMENT

interface FastEthernet0/1

switchport trunk encapsulation dot1q

switchport trunk native vlan 50

switchport mode trunk

interface FastEthernet0/2

switchport mode access

switchport voice vlan 20

spanning-tree portfast

interface FastEthernet0/3

switchport mode access

switchport voice vlan 20

spanning-tree portfast

interface FastEthernet0/4

interface FastEthernet0/5

interface FastEthernet0/6

interface FastEthernet0/7

interface FastEthernet0/8

interface FastEthernet0/9

interface FastEthernet0/10

interface FastEthernet0/11

interface FastEthernet0/12

interface FastEthernet0/13

interface FastEthernet0/14

interface FastEthernet0/15

interface FastEthernet0/16

interface FastEthernet0/17

interface FastEthernet0/18

interface FastEthernet0/19

interface FastEthernet0/20

interface FastEthernet0/21

interface FastEthernet0/22

interface FastEthernet0/23

interface FastEthernet0/24

interface GigabitEthernet0/1

interface GigabitEthernet0/2

interface Vlan1

no ip address

shutdown

ip classless

ip http server

ip sla enable reaction-alerts

line con 0

line vty 0 4

login

line vty 5 15

login

end

Problems

Going into the lab with more ideas of what we wanted rather than how to do it, we ran into a multitude of problems along the way. The first problem we ran into was we were unsure of where to start or what to configure. Our initial guess was to edit settings directly on the phones. This was proven to be ineffective as there was no way to save the changes made on the phones. We turned to configuring the router and the switches instead. The topology of the network was figured out through research online and finding information on how to establish a connection between the IP phones in a LAN with DHCP and router on a stick. The IP phones were able to receive a number and the computers were able to access the internet. While the phones were able to call each other, they displayed an error of “file not found.” After configuring the LAN, we believed that we were close to finishing the lab as the only thing left to configure was a connection to the analog line. However, that is where most of the problems arose.

We had a working analog line which we had tested with an analog phone. We also knew that there were voice ports – both FXS and FXO -- on the router which could be seen on running configuration. After identifying that we needed to connect the analog line to the FXO port, the original approach was the directly try to configure FXO voice port as if it were an ethernet port. This did not work.

After more research, it was realized that our flash didn’t contain appropriate files for CME even though telephony services were allowed to be set up on the router. We obtained a tar file containing the respective files and extracted them. The necessary files were then pushed to the phones through downloading a TFTP service on the computer.

After CME was successfully installed on the router, the phones were still unable to call through the analog line, each time displaying a message of “unknown call.” Due to the lack of guides on the specific goal we were pursuing, we spent a lot of time trying commands. Progress was made when configuring a dial-peer with a destination pattern of 9……….. and assigning it to the FXO voice port as it not longer showed an error of “unknown call.” We were then able to use a forward-digits 7 command and a call was established and connected but no voice could be heard (only beeps). Trying different numbers actually established calls with people who answered, however their number wasn’t the number we had typed into the phone and dialed (e.g. dial in number for pizza hut and we are connected with an art teacher). At this point we continued to work the with forward-digits command. The forward-digits all command was then configured, and it allowed a connection with the correct number.

Conclusion

While there were a lot of problems, we encountered in the lab especially when configuring a connection between VoIP and analog, we were successfully able to create a LAN VoIP connection, connect the computers to the internet, and make a call with a VoIP phone through an analog line. This was made possible through an FXO voice port and CUCME. We learned not only how to use a telephone, but also about the growing influence of VoIP and its usage in many of the voice services we engage with today.

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