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Design and Simulation of VoIP System for Campus usage: A Case Study at PTU

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ABSTRACT

Voice communication is one of the most important parts of human development. People need to communicate at great distances have influenced their development to all levels and at all times. Voice over Internet Protocol (VoIP) is one of the most widespread used and common technologies that are related to voice communications. VoIP uses internet protocol data packets to transfer voice, fax, and other data over the shared network. Several tools have emerged around this protocol. Systems and software those are capable of managing communications with telephone calls or faxes and other advanced functionalities. In this paper, VoIP network architecture design is simulated using Cisco Packet Tracer. RIP routing protocol, VoIP technology, Network Topology design and Wireless technology play important roles to achieve the main goal.

KEYWORDS: *VoIP, Wireless Access Point, Network Topology, Cisco Packet Tracer*

I. INTRODUCTION

Technology has always tried to facilitate needs for remote communication from the beginning of evolution. In VoIP process, signals from spoke are going through the telegraph to the current communications through telephony mobile. Internet telephony refers to communications services such as voice, fax, SMS, and/or voice-messaging applications. That are transported via the internet, rather than the public switched telephone network (PSTN). The voice information travels to its destination in countless individual network packets across the Internet. Thereby eliminating toll charges, which is why they are cheaper than calls over PSTN.

In VoIP network, voice is packetized, compressed and then transmitted, in a form of data packets, over IP-based networks. Thus, voice quality of a call depends largely on the quality of the network, which varies due to numerous factors such as the hardware, the provider, the current internet connection status, the available bandwidth, etc. [1].

The steps involved in originating a VoIP telephone call are as follows:

- Signaling and media channel setup
- Digitization of the analog voice signal
- Encoding
- Packetization
- Transmission as Internet Protocol (IP) packets over a packet-switched network

On the receiving end, the above steps take place in the reverse order:

- Receiving the IP packets
- Decoding of the packets
- Digital-to-analog conversion which reproduces the original voice stream.

Fig. 1 shows VoIP Network Infrastructure for university campus. This network design consists of 6 departments. The main target is to communicate within departments using IP phones with networking facilities. Thus, configurations of router for accessing internet, obtaining telephony service, interfacing multiple networks and creating wireless network, are deployed in this design.

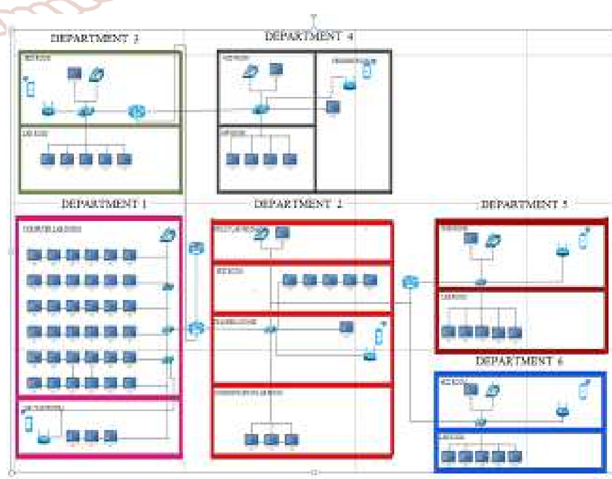

Figure1. VoIP Network Infrastructure Design

II. VOIP PROTOCOL

A protocol is a set of rules or procedures that needs to be followed to allow an orderly communication. There are a number of protocols that are employed in order to provide for VoIP communication services. They can be implemented in using both the proprietary and open protocols and standards. In order to be able to communicate using a VoIP system, there are two types of protocol that must be used [1]. They are:

A. Signaling Protocol

- controls and manages the call
- includes elements such as call set up, clear down, and call forwarding
- examples of such protocol are H.323 - ITU standard, Session Initial Protocol (SIP) - IETF standard, and Skinny Client Control Protocol(SCCP) - Cisco proprietary

B. Data Exchange Protocol

- manages the data exchange for the VoIP traffic
- handles both audio and video (e.g. Real-Time Transport Protocol (RTP) [2]

III. SOFTWARE IMPLEMENTATION

Fig. 2 shows the simulation of VoIP network architecture design for University campus. In this network design, Cisco Packet Tracer Software is used as a powerful simulation tool. In this design, four (2811) routers, five (WRT300N) wireless routers, six (2960-24TT) switches and six (7960) IP phones are used as main networking components. RIP routing protocol is used to interface the multiple network in each department and ISP router use default route. Each department uses IP phone to communicate over voice among departments. This network is designed to provide internet access on wireless devices. Switches in department 2 are connected using the mesh network topology.

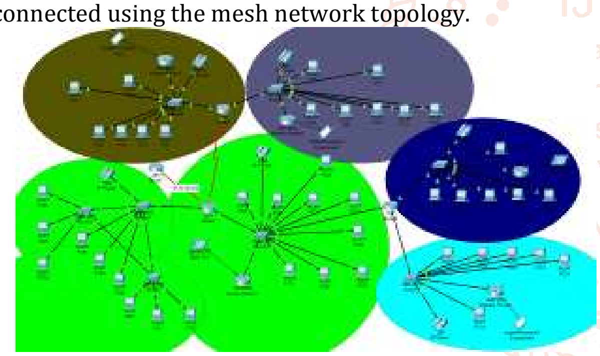

Figure2. Network Simulation Design

After the design implementation, it is needed to assign the default gateway's IP address, Wireless Router's IP address, Access Point name and VoIP directory number of each department. Table 1 shows the assignment of the whole network.

Table1. Assigning the Whole Network

No.	Department	GateWay	APName	Router IP	VoIP No.
1	Dep. 1	192.168.2.1	ECWifi	192.168.102.1	1008
2	Dep. 2	192.168.1.1	ECWifi	192.168.102.1	1007
3	Dep. 3	192.168.10.1	CWifi	192.168.110.1	2006
4	Dep. 4	192.168.20.1	MCWifi	192.168.120.1	2005
5	Dep. 5	192.168.100.1	MPWifi	192.168.210.1	1006
6	Dep. 6	192.168.200.1	EPWifi	192.168.220.1	1005

IV. ROUTER CONFIGURATION

In this section, configurations of router in desired network are described. There are three main configurations such as RIP routing configuration, Telephony Service configuration and ISP router configuration for internet access network. One of these routers is used as ISP router and the other is used as interfacing router to route the multiple network for departments and for VoIP network. Fig. 3 shows the configuration of ISP router to obtain internet access for each department.

```
Router>en
Router#conf t
Enter configuration commands, one per line.  End with CNTL/Z.
Router(config)#hostname ISP
ISP(config)#int s0/1/1
ISP(config-if)#ip add 10.10.10.1 255.255.255.0
ISP(config-if)#no shut

%LINK-5-CHANGED: Interface Serial0/1/1, changed state to down
ISP(config-if)#exit
ISP(config)#ip route 0.0.0.0 0.0.0.0 10.10.10.2
ISP(config)#exit
ISP#
%SYS-5-CONFIG_I: Configured from console by console

ISP#
```

Figure3. ISP Router Configuration

IP phones are used to communicate the 5 departments to each other. So, the router is needed to conFigurefor obtaining the telephony service in these departments. Before configuration the routers for telephony-service, it is needed to choose the correct router that supports the telephony. Fig. 4 shows the configuration of router for VoIP network in department 2. Fig. 5 shows the configuration status of DHCP server in running configuration mode

```
ECR>en
ECR#conf t
Enter configuration commands, one per line.  End with CNTL/Z.
ECR(config)#int f0/0
ECR(config-if)#ip add 192.168.1.1 255.255.255.0
ECR(config-if)#no shut
ECR(config-if)#ip dhcp pool voip
ECR(dhcp-config)#network 192.168.1.0 255.255.255.0
ECR(dhcp-config)#default
ECR(dhcp-config)#default-router 192.168.1.1
ECR(dhcp-config)#option 150 ip 192.168.1.1
ECR(dhcp-config)#exit
ECR(config)#tele
ECR(config)#telephony-service
ECR(config-telephony)#max-ephones 5
ECR(config-telephony)#max-dn 5
ECR(config-telephony)#ip so
ECR(config-telephony)#ip source-address 192.168.1.1 port 2001
ECR(config-telephony)#auto assign 1 to 5
ECR(config-telephony)#exit
ECR(config)#exit
```

Figure4.Telephony Service and DHCP Server Configuration

```
ip dhcp pool voip
 network 192.168.1.0 255.255.255.0
 default-router 192.168.1.1
 option 150 ip 192.168.1.1
ip dhcp pool voice
 network 192.168.2.0 255.255.255.0
 default-router 192.168.2.1
 option 150 ip 192.168.2.1
!
```

Figure5. DHCP Server Configuration Status

To achieve the main goal of this VoIP network configuration, it is also needed to configure the switch in each department. Thus, the configuration of switch for VoIP network in Department 2 is shown in Fig. 6 as sample configuration. Similarity, the switches are configured for voice access of each department.

```
Switch>en
Switch#conf t
Enter configuration commands, one per line.  End with CNTL/Z.
Switch(config)#int range f0/4-8
Switch(config-if-range)#swit
Switch(config-if-range)#switchport mode access
Switch(config-if-range)#swi
Switch(config-if-range)#switchport voice vlan 1
Switch(config-if-range)#exit
Switch(config)#exit
Switch#
```

Figure6. Configuration of Voice Access in Switch

The last configuration of this section is for RIP routing protocol that is used to route the different network of 5 departments in University campus. Fig. 7 shows the configuration of router in department 1 using GUI. This configuration provides to route the networks. Fig. 8 shows how to specify the IP address of interface for router in department 1.

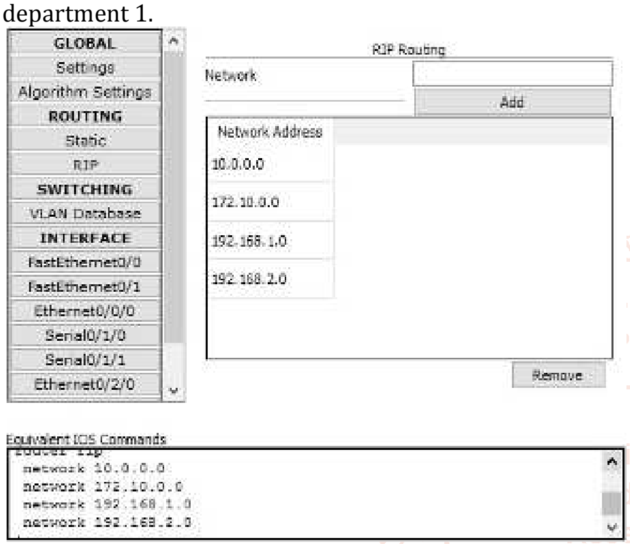

Figure7. RIP Routing Configuration in Department 1

```
!
interface FastEthernet0/0
 ip address 192.168.1.1 255.255.255.0
 duplex auto
 speed auto
!
interface FastEthernet0/1
 ip address 192.168.2.1 255.255.255.0
 duplex auto
 speed auto
!
interface Ethernet0/0/0
 no ip address
 duplex auto
 speed auto
!
interface Serial0/1/0
 ip address 172.10.10.1 255.255.255.0
 clock rate 2000000
!
interface Serial0/1/1
 ip address 10.10.10.2 255.255.255.0
 clock rate 2000000
!
```

Figure8. Assigning the IP Address of Fast Ethernet 0/0 Interface

V. WIRELESS NETWORK CONFIGURATION

Cisco wireless routers are deployed as access points in each department to build the wireless network. The configuration of wireless access point in Department 4 is shown in Fig. 9.

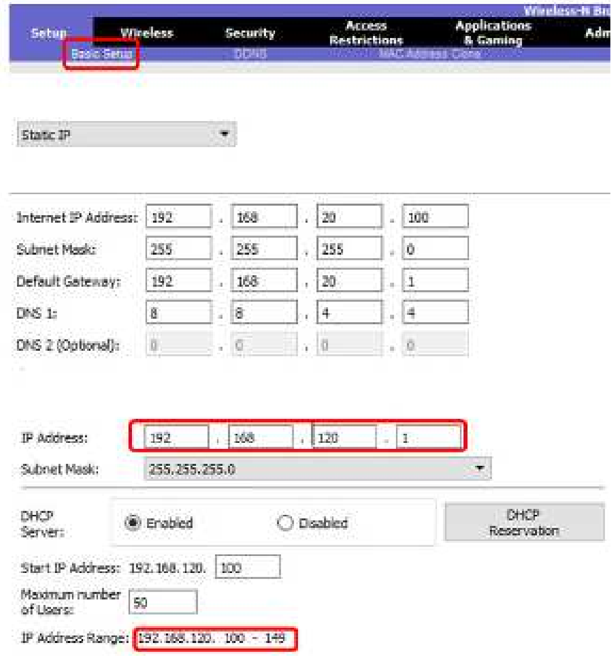

Figure9. Wireless AP Configuration in Department 4

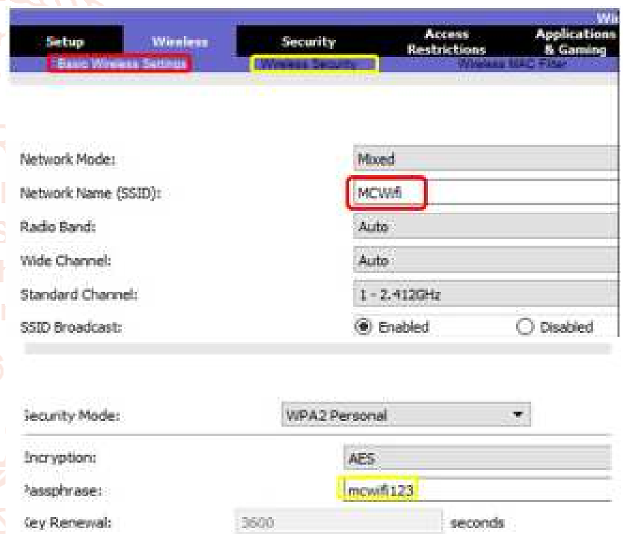

Figure10. SSID Name and Security Setting

VI. END DEVICES CONFIGURATION

To connect the wire and wireless network from end devices such as PCs, mobile phone and tablet pc, it is needed to conFigurethese devices. To connect the wire and wireless network from end devices such as PCs, mobile phone and tablet pc, it is needed to conFigurethese devices. Fig. 11 shows how to conFigurethe PCs in Department 5 and Fig. 12 is configuration of mobile phone to connect the MPWifi.

Figure11. PC Configuration in Department 5

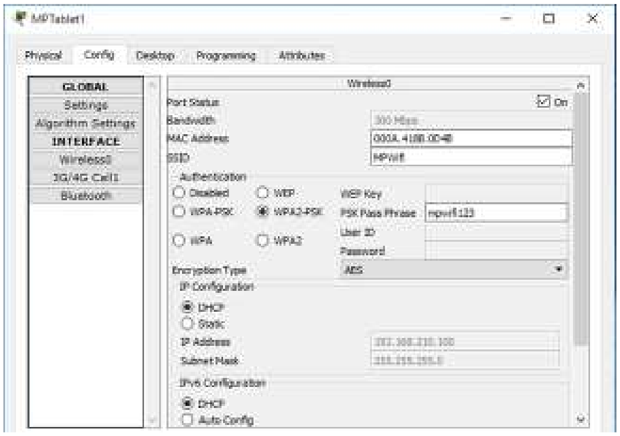

Figure12. Configuration of Mobile Phone in Department 5

VII. TESTING RESULT USING PING COMMAND

Fig. 13 is the testing result or internet connection in Head of Department room.

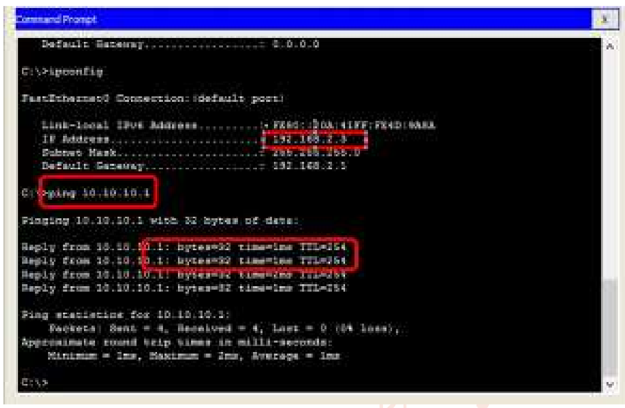

Figure13. Testing Results for Internet Connection

Local Area Network (LAN) connection testing is one of the vital approaches to accomplish the main target. Thus, Fig. 14 shows the testing result of LAN network connection.

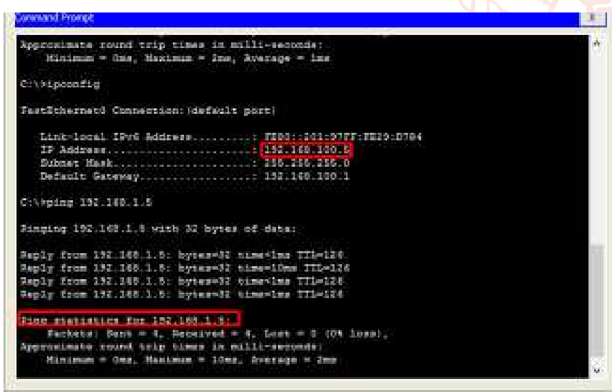

Figure14. Testing Result for Connections from Department 5 to Department 1

The wireless network connection testing that is needed to test the connection with each other via mobile phone. Thus, the wireless connection testing result for connection between Department 3 and Department 6 is shown in Fig. 15.

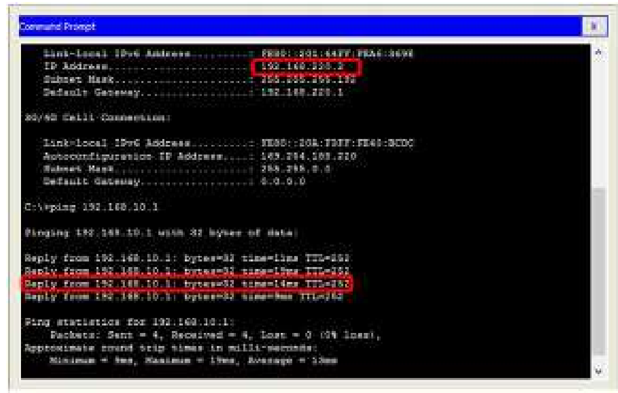

Figure15. Testing Result for Connection from Department 6 to Department 3

VIII. TESTING RESULT FOR VOIP NETWORK

VoIP technology is used to communicate within departments in University campus. Fig. 16 is testing result for IP phones connection between Department 1 and department of Department 3.

Figure16. Ring Out from Department 1

Figure17. Phone Ringing in Department 3

Fig. 18 is the testing result for VoIP connection from Department 4 to Department 2 when the connection is available between Department 1 and Department 3. Fig. 19 shows the VoIP connection from Department 5 to Department 4 that is ringing out to Department 1.

Figure18. VoIP Busy Connection from Department 4 to Department 2

Figure19. VoIP Busy Connection from Department 5 to Department 4

IX. CONCLUSION

Voice over IP (VoIP) protocol is used to carry voice signal or the IP network. This allows user to use IP Telephone instead of the dedicated voice transmission telephone lines. This technology offers cost savings by making more efficient use of the existing network.

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