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Enhancing the connectivity of VOIP based Local Area Network (LAN) Using Packet Tracer

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Abstract: This paper focuses on enhancing the connectivity of VOIP based Local Area Network (LAN) network using Packet Tracer. The simulation was carried out on the CISCO packet tracer by creating a LAN using a Switch and then enabling VoIP on the router by configuring the Dynamic Host Control Protocol (DHCP) server as well as the IP phones on the local area network. The result showed that the phone with IP number 201 has a good connectivity with phone IP 211.On picking up, the receiver of the dialed IP phone 201 became connected with the dialing phone, and this was seen on both phone interfaces.

Keywords: *CISCO, Voice over Internet Protocol (VOIP), Packet Tracer, DHCP*

I. INTRODUCTION

Voice over Internet Protocol (VoIP) which is also referred to as internet telephony is a technology that transmits voice signal in real time using the Internet Protocol (IP) over a public internet or private data network. In a simpler term, it converts voice signal which is analog to a digital signal in your telephone before compressing and encoding it into long strings of IP packets for onward transmission over the underlying IP network to the receiver. At the receiving end, the received IP packets are reassembled in order before decompressing and processing through the use of a Digital to Analogue Converter (DAC) to generate the initial signal transmitted. Its existence is basically based on two fundamental technologies, the telephone and the internet [1].

Voice over Internet Protocol (VOIP) uses the Internet Protocol (IP) to transmit voice as packets over an IP network. So VoIP can be achieved on any data network that uses IP, like the Internet, Intranets and Local Area Networks (LAN). Here the voice signal is digitized, compressed and converted to IP packets and then transmitted over the IP network [2].

The VoIP systems can come in different forms. Its basic structure is functionally similar to that of PSTN that allows it to communicate with the second party at the other terminal of the connection which is either a VoIP system or traditional analog telephone [3]. Figure 1 shows a Workflow of internet telephony.

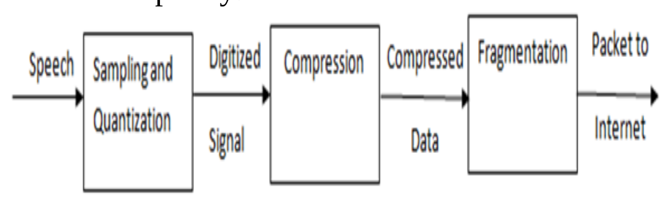

Figure 1: Workflow of Internet Telephony

II. MATERIALS AND METHODS

A. Materials

The materials used n this experiment include the following: Cisco packet tracer, Router, switch, HP laptop, 3 IP phone (IP phone 0, IP phone 1, and IP phone 2). These materials were in the VoIP LAN network as shown in Figure 2.

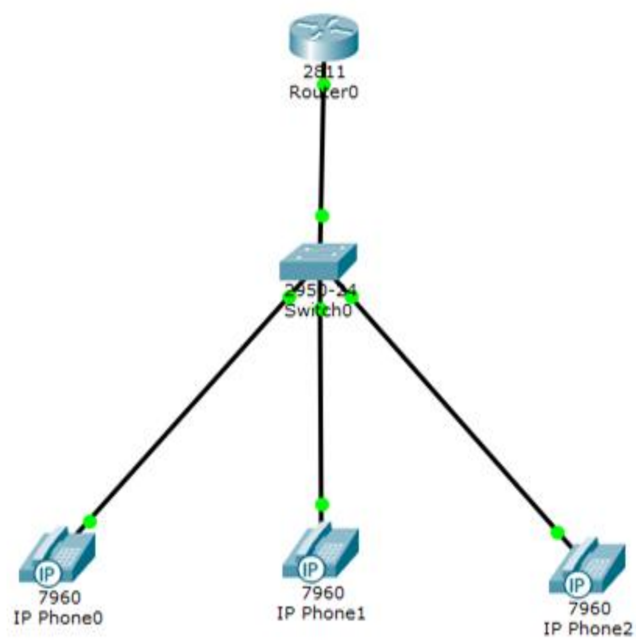

Figure 2: VoIP based LAN Network with Packet Tracer

B. Method

In order to achieve an efficient VoIP based LAN network with packet tracer, the Cisco packet tracer was installed on window laptop 32 operating system. Each of the materials used in the experiment was given an IP address and configured appropriately as follows:

1) Router Setup

In order to be able to make VoIP calls, the router has to be set up to enable VoIP.

First is to configure router interface: from the CLI (Command Line Interface) as follows:

%move to enable mode%
%to global configuration%
%name the interface%
%IP address and subnet mask%
%no shutdown%
%exit%

Configure DHCP server: this is it enable IP addresses to be automatically assigned to all the devices.

%ipdhcp pool voice%
%default-router IP address%
#option 150 IP address%
%network IP address subnet mask%
%exit%

Enable Call Manager Express: Call Manager Express (CME) is the name for the telephony service that will run on the Cisco routers.

The first line *%telephony-service%* initiates the service on the router; The next two lines set how many phones you want on the network.

2) Switch Configuration

The following configurations are made on the switch to make the interfaces support VoIP.

- Setting host name for switch
- Adding descriptions for significant interfaces
- Assigning VoIP phone ports to voice VLAN

These changes are made by implementing the commands below on the CLI. Figure3 shows a switch configuration.

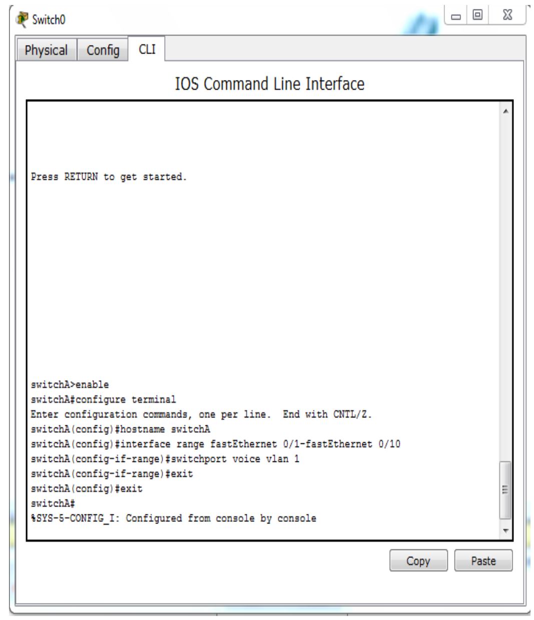

Figure 3: Switch Configuration

3) IP Phone Configurations

The DHCP server automatically assigns IP addresses to the IP phones, in order to allow the phones make calls, it is necessary to assign phone numbers to it, this is done on the CLI of the router as shown in the figure below:
Phone numbers are: 201, 211 and 221 for IP phone 1, 2 and 3 respectively.
Figure 4 shows how to assign numbers to IP phones.

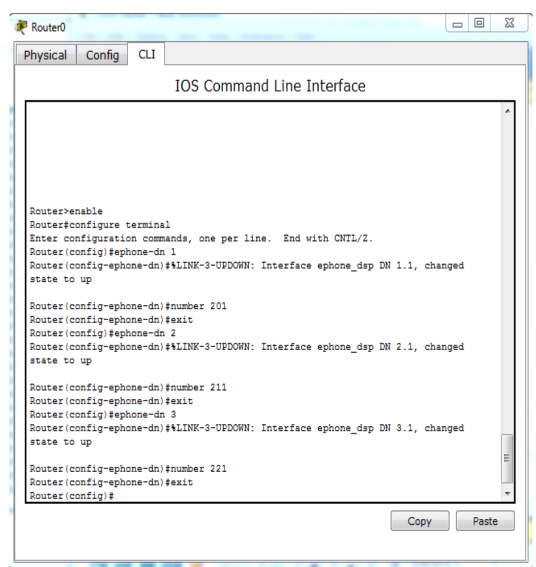

Figure 4: Assigning numbers to IP phones

III. RESULT AND DISCUSSION

Having configured the all the interfaces on the router and the switch to support VoIP, and configuring the IP phones on the router, call can now be made from the GUI interfaces of the VoIP phones on the simulated VoIP network on the packet tracer. To make call from any of the phones, the receiver is picked up and then the phone number dialed. The phone being dialed rings out and the light on the phone continues blinking until the receiver is picked up. On picking up the receiver of the dialed phone, it becomes connected with the dialing phone, and this can be seen on both phone interfaces. Figure 5 shows the Configured IP phones.

Figure 5: Configured IP Phones

CONCLUSION

IP telephony is really a great innovation, in terms of both economic and infrastructure requirements. VoIP calls can be made from anywhere within the same local area network with no additional cost for the voice signal, thereby reducing cost. From this research, it is evident that VoIP can be easily deployed in an already existing local area network which makes for its flexibility. Adopting the use of VoIP for calls will really go a long way in reducing the cost being spent on voice calls

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