Configure A VoIP Network

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Abstract— Voice over internet protocol (VoIP) is a protocol of voice transmission over the internet and other networks based on packet switching. The VoIP came as an alternative for the public switch telephone network (PSTN) for voice transmission. A VoIP enabled speech server can include a speech application which can be configured to communicate with a VoIP telephony gateway server over a VoIP communications route. The VoIP enabled speech server include a VoIP call control interface to the VOIP telephony gate server, the VoIP call control interface establishing the VoIP communications route. In operation, the speech application can receive VoIP packets from the VoIP telephony gateway server over the VoIP communications path.

Keywords- *VoIP, IP addressing, DHCP, IP telephony, VLAN, Routing Algorithams, Server Pool, Gateway*

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I. INTRODUCTION

Voice over Internet Protocol (VOIP) is a technology that allows you to make voice / telephone calls using a broadband Internet connection instead of a regular phone line, which has been the method used for the last hundred years or so.

VOIP services vary. Some services using VOIP may only allow you to call other people using the same service, but others may allow you to call anyone who has a telephone number anywhere in the world. While some services only work via your computer or a special VOIP phone, other services allow you to use a traditional phone with an adaptor.

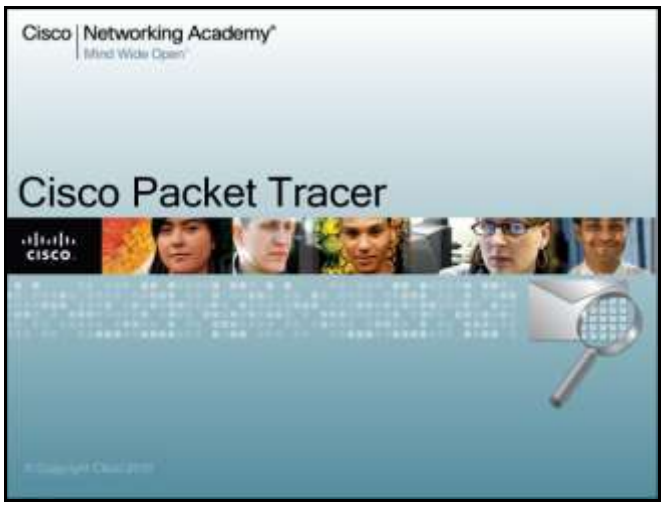
The Voice / Signal that comes through VOIP is digitized in order to be routed across the Internet or Intranet. There are two digital coding standards that are most commonly used for VOIP – H.323 and Session Initiation Protocol (SIP).

H.323 was developed many years ago mainly for video conferencing over ISDN telephone lines. Because H.323 was built long ago, its technology is considered outdated and comes with quite unnecessary overhead costs, lacking the features and flexibilities that SIP has to offer.

SIP is an end-to-end client / server session signaling protocol. It is much less complicated when it comes to implementation and programming from a developer's standpoint. At the same time, it provides all of the features needed to scale and provide reliable voice communication [7].

II. OVERVIEW OF SOFTWARE

A. *Cisco Packet Tracer Student Version*



Version: 6.1.0.0120

Figure 1. Version of cisco packet tracer

B. *Operating Modes*

- Realtime Mode:

  In Realtime Mode, your network runs in a model of real time, within the limits of the protocol models used. The network responds to your actions immediately as they would in a real device. For example, as soon as you make an Ethernet connection, the link lights for that connection will appear, showing the connection state (see the "Connections/Links" page for details). Whenever you type a command in the CLI (such as ping or show), the result or response is generated in real time and you see it as such [9].

- Simulation Mode:

 ISSN: 2321-8169
2115 - 2120

In Simulation Mode, you can "freeze" time -- you have direct control over time related to the flow of PDUs. You can see the network run step by step, or event by event, however quickly or slowly you like. You can set up scenarios, such as sending a ping packet from one device to another. However, nothing "runs" until you capture it (the first time through, as with a protocol sniffer) or play it (re-playing the captured events as an animation). When you capture or play the simulation, you will see graphical representations of packets traveling from one device to another. You can pause the simulation, or step forward or backward in time, investigating many types of information on specific PDUs and devices at specific times [9].

III. IP Addressing And Subnetting

A. *IP Addressing*

Network layer addressing lies at Layer 3 of the OSI model. This enables a group of computers to be given similar logical addresses. Logical addressing is similar to determining a person's address by looking at his or her country, state, ZIP code, city, and street address. Routers forward traffic based on the Layer 3 or network layer address. IP addressing supports five network classes. The bits at the far left indicate the network class, as follows:

- Class A networks are intended mainly for use with a few large networks because they provide only seven bits for the network address field.
- Class B networks allocate 14 bits for the network address field and 16 bits for the host address field. This address class offers a good compromise between network and host address space.
- Class C networks allocate 21 bits for the network address field. They provide only 8 bits for the host field, however, so the number of hosts per network can be a limiting factor.
- Class D addresses are reserved for multicast groups, as described formally in RFC 1112. In class D addresses, the four highest-order bits are set to 1, 1, 1, and 0 [2].

B. *Subnetting*

For the subnet address scheme to work, every machine on the network must know which part of the host address will be used as the subnet address. This is accomplished by assigning a subnet mask to each machine. A subnet mask is a 32-bit value that allows the recipient of IP packets to distinguish the network ID portion of the IP address from the host ID portion of the IP address. The network administrator creates a 32-bit subnet mask composed of 1s and 0s. The 1s in the subnet mask represent the positions that refer to the network or subnet addresses.

Not all networks need subnets, meaning they use the default subnet mask. This is basically the same as saying that a network doesn't have a subnet address. Figure 1 shows the default subnet masks for Classes A, B, and C. These default masks cannot change. In other words, you can't make a Class B subnet mask read 255.0.0.0. If you try, the host will read that address as invalid and usually won't even let you type it in. For a Class A network, you can't change the first byte in a subnet mask; it must read 255.0.0.0 at a minimum. Similarly, you cannot assign 255.255.255.255, as this is all 1s—a broadcast address. A Class B address must start with 255.255.0.0, and a Class C has to start with 255.255.255.0 [2].

Class	Format	Default Subnet Mask
A	network.node.node.node	255.0.0.0
B	network.network.node.node	255.255.0.0
C	network.network.network.node	255.255.255.0

Figure 1. Default Subnet Mask

IV. VoIP Network Configuration

Based on the Cisco's Hierarchical Network Design Model, this simulated network has VOIP call routing features enabled between subnets with core services centralized in the datacenter.

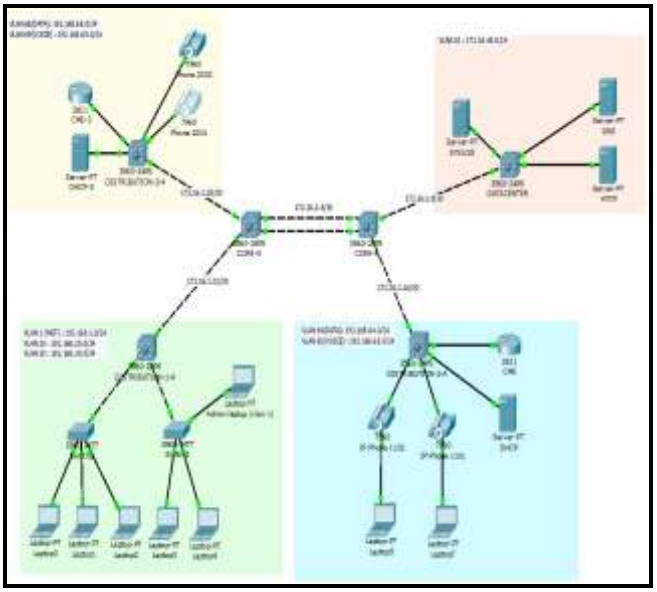


Figure 2. VoIP Network

ISSN: 2321-8169
2115 - 2120

A. Routing Configuration

- VLAN Database Configuration:

The Cisco 2811 routers support VLAN configuration. We can manage the VLANs on the router from the VLAN Database panel. We can add VLANs by entering a name and a VLAN number and pressing the Add button [9].

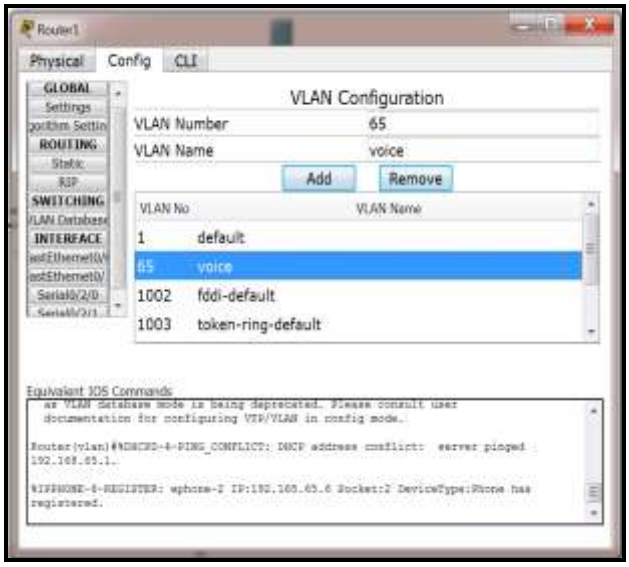


Figure 3. VLAN Configuration Window

- Interface Configuration:

A router can support a wide range of interfaces like serial, modem, copper Ethernet, and fiber Ethernet. Each interface type may have different configuration options, we can set the Port Status (on or off), IP Address, Subnet Mask, and Tx Ring Limit [9].

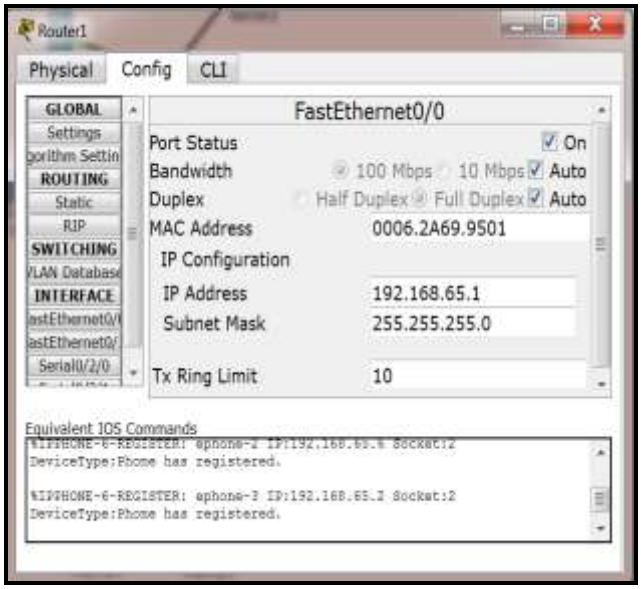


Figure 4. Interface Configuration Window

B. Server Configuration

- DHCP Service Configuration:

In the DHCP service configuration, we can set up a DHCP server with many different IP address pools. To add a DHCP pool, enter the Pool Name, Default gateway, DNS Server address, Starting IP address , Subnet Mask, and the Maximum number of Users, then click add [9].

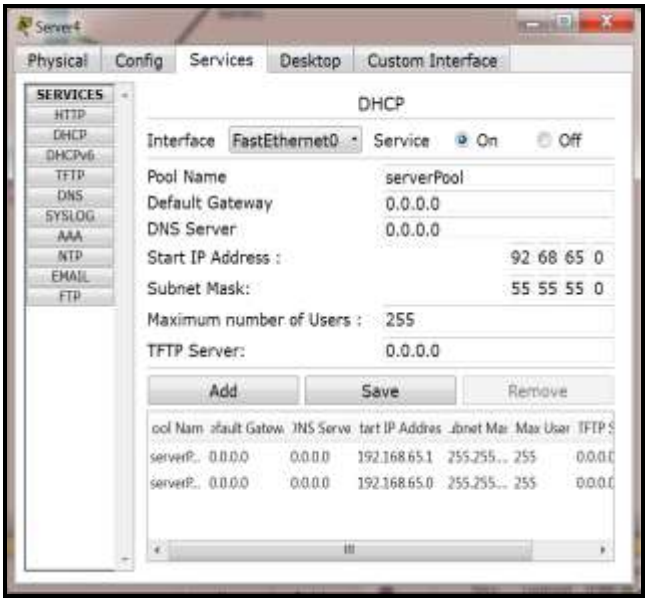


Figure 5. DHCP Server Configuration Window

C. IP Phone-GUI

Figure 6. IP Phone-GUI

 ISSN: 2321-8169
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D. Laptop Configuration

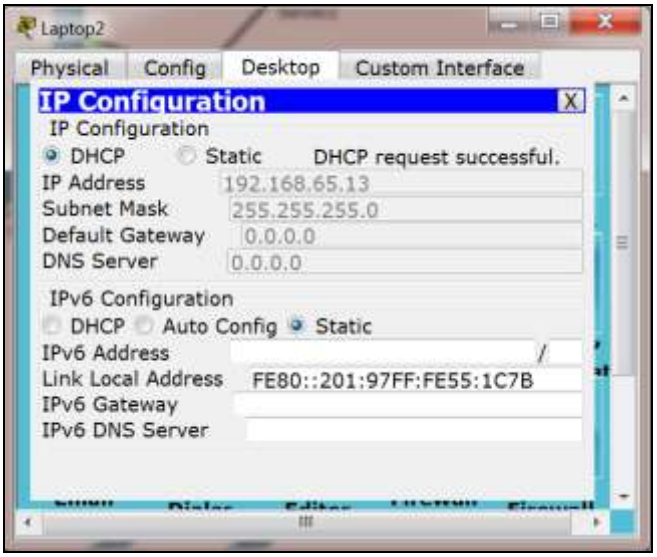

Figure 7. Laptop Configuration

We can configure the global settings and interface settings on the laptop with the Configure tab. Additionally, the Desktop tab provides tools to configure IP settings, configure dial-up settings, use a terminal window, open a host command line interface,configure Linksys wireless settings, generate PDUs, and issue SNMP requests [9].

V. Simulation And Result

A. Simulation and Panel

Packet Tracer simulations not run on a linear time scale. Time is determined by the events that occur. The Event List keeps track of all such PDU instances and lists their information in various fields:

- Visible:

  An "eye" icon in the field means that an event is happening at the current simulation time.
- Time:

  This field indicates the time (in seconds) at which the event occurred, relative to the last time the simulation scenario restarted.
- Last Device:

  This field indicates the previous location of the packet.
- At Device:

  This field indicates the current location of the packet.
- Type:

  This field indicates the packet type (ACL Filter, ARP, BGP, CDP, DHCP, DNS, DTP, EIGRP, FTP, H.323, HTTP, HTTPS, ICMP, ICMPv6, IPSec, RIP, RTP, SMTP, SNMP,TCP, TFTP, Telnet, UDP, and VTP).

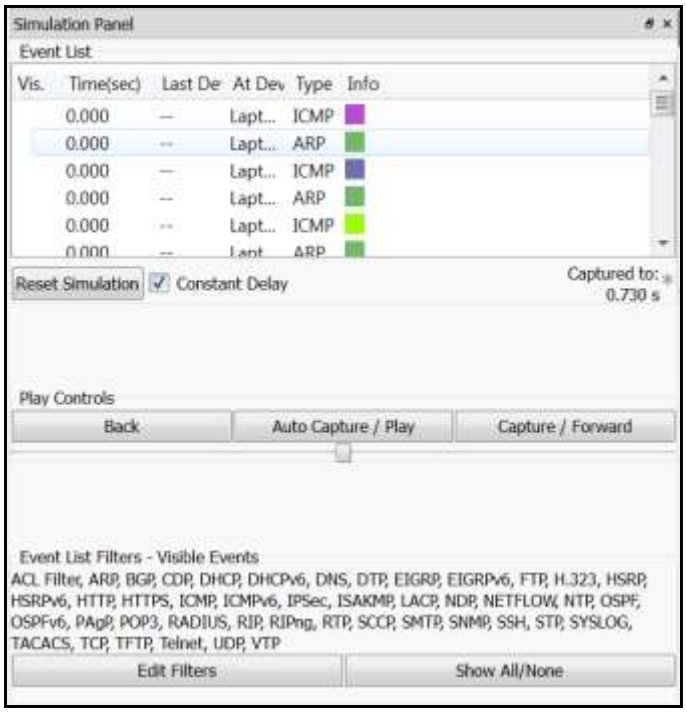

Figure 8. Event List Window

- Info:

  This field shows detailed information about the packet instance, broken up into each layer of the OSI model. Learn more about this field in the "PDU info" page [9].

B. PDU List Window

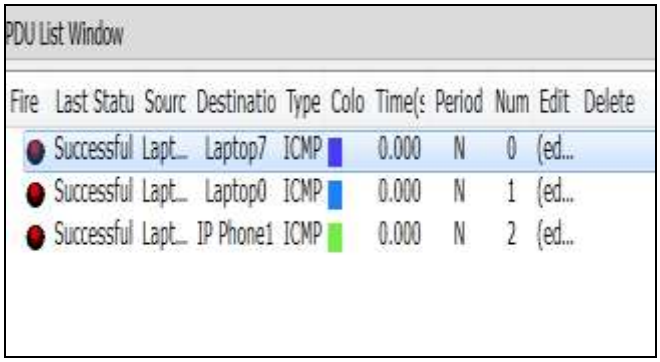

Figure 9. PDU List Window

C. Results

- IP Phone 2002 To IP Phone 1102:

ISSN: 2321-8169
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Figure 10. IP Phone No. 2002

Figure 11. IP Phone No. 1102

- Laptop 0 To Laptop 7:

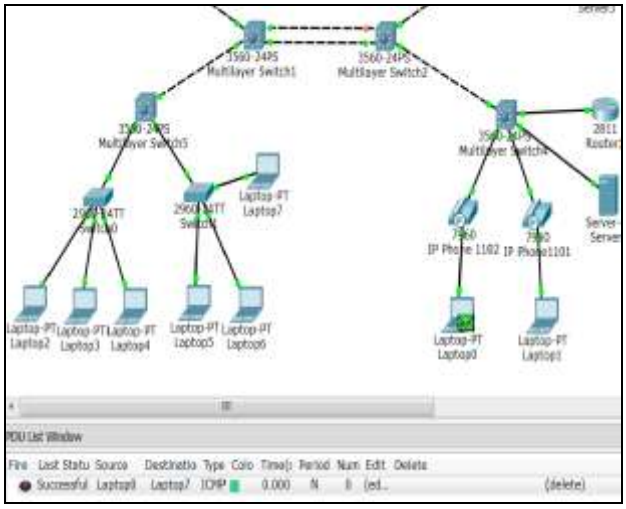

Figure 12. Result 1

- Server 1 to Router 1:

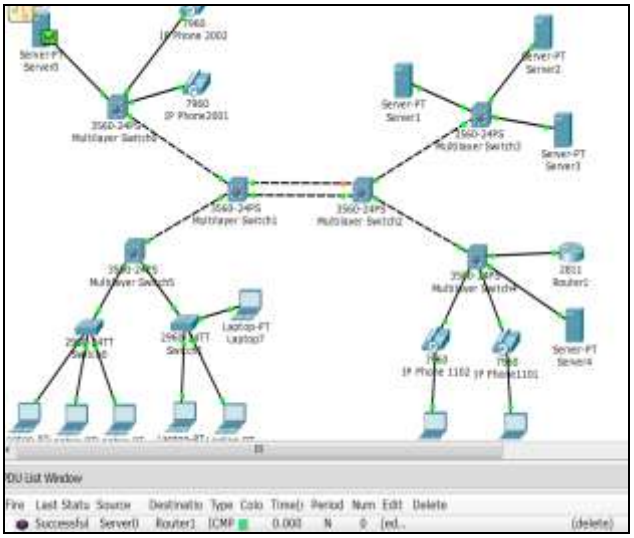

Figure 13. Result 2

- Multiple simulation:

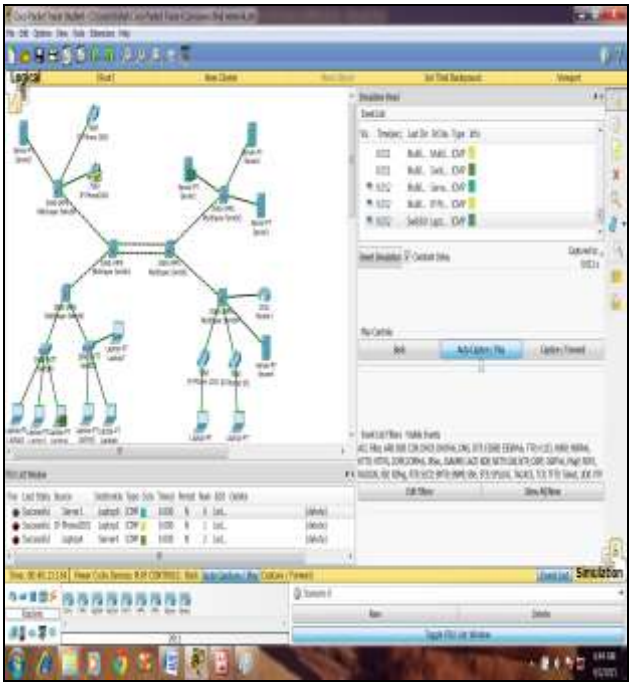

Figure 14. Result 4

VI. CONCLUSION

VoIP is an evolutional step in voice communication that makes use of the widely spread and well established internet backbone. VoIP has manage to provide a much cheaper means of voice communication but still it is not wholly embraced by all this might because of its trade off low cost for poor QoS.

The nature of transmitting voice data over internet will always result to packet loss. The techniques used to counter to packet loss need to be closely monitored as most of them trade off packet loss with delay.

Benefits of Voice over IP (VoIP), include cost savings, single infrastructure savings, and new Applications. DHCP enables a *device* (a PC or an IP phone) to dynamically receive

an IP address (that is, the IP address does not need to be statically configured into the device). So, for instance, if you have an IP phone configured with DHCP, you can move the phone wherever you need and still keep the same phone number. This is similar to moving your laptop from office to office and still being able to log in to the same network server.

ACKNOWLEDGMENT

A debt of gratitude to Mr. A. I. Darvadiya, our internal guide , for their guidance and suggestion at all stages of our project , right from the selection of the project , constant encouragement , practical guidance.

To Mr. D. N. Khandhar our H.O.D for being kind and helpful to us , and stand by us at every stage of our final year project.

To Mr. K. H. Wandra, Principle of C.U .Shah college of Engineering and Technology to give us opportunity to work on this very demanding and rewarding project.

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