

DAYANANDA SAGAR COLLEGE OF ENGINEERING

DEPARTMENT OF COMPUTER SCIENCE & ENGINEERING

(Autonomous Institution Affiliated to VTU, Belgaum)

ShavigeMalleshwara Hills, Kumara swamy Layout, Bangalore -560078



MINI PROJECT REPORT (CN CS-52)

“VOICE OVER IP NETWORK IN CISCO PACKET TRACER ”

Submitted in partial fulfillment for award of AAT Marks in subject Computer Networks (5th Semester) Bachelor of Engineering.

Submitted By,

SL.NO	USN	NAME
1.	IDS16CS713	GAJULA.MEGHANA
2.	IDS16CS725	P.VAMSHIDHAR
3.	IDS17CS409	HARISH.P
4.	IDS17CS419	NAVYA.R

**Signature of Faculty in
charge,**

Table of contents

1. Abstract
2. Introduction
3. Problem statement
4. Design of topology
5. Simulation phase
6. Screenshots
7. Conclusion

ABSTRACT:

Voice over IP (VOIP) uses the Internet Protocol(IP) to broadcast voice as packets over an IP network. Therefore, VOIP can be accomplished on any data network that uses IP address, such as Internet, Intranets and Local Area Networks (LAN).The overview of the technology and how this technology can be applied for the integration of voice and data networks. It is also focusing on how VoIP configure on LAN and how it works.

INTRODUCTION:

Packet Tracer is a cross-platform visual simulation tool designed by Cisco Systems that allows users to create network topologies and imitate modern computer networks. The software allows users to simulate the configuration of Cisco routers and switches using a simulated command line interface. Packet Tracer makes use of a drag and drop user interface, allowing users to add and remove simulated network devices as they see fit. The software is mainly focused towards Certified Cisco Network Associate Academy students as an educational tool for helping them learn fundamental CCNA concepts. Previously students enrolled in a CCNA Academy program could freely download and use the tool free of charge for educational use. Cisco systems released different versions of the packet tracer where we have done this project in cisco packet tracer 7.0.

Voice over Internet Protocol (also voice over IP, VoIP or IP telephony) is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. The terms Internet telephony, broadband telephony, and broadband phone service specifically refer to the provisioning of communications services (voice, fax, SMS, voice-messaging) over the public Internet, rather than via the public switched telephone network (PSTN).

The steps and principles involved in originating VoIP telephone calls are similar to traditional digital telephony and involve signaling, channel setup, digitization of the analog voice signals, and encoding. Instead of being transmitted over a circuit-switched network, the digital information is packetized, and transmission occurs as IP packets over a packet-switched network.

Early providers of voice-over-IP services offered business models and technical solutions that mirrored the architecture of the legacy telephone network. Second-generation providers, such as Skype, built closed networks for private user bases, offering the benefit of free calls and convenience while potentially charging for access to other communication networks, such as the PSTN. This limited the freedom of users to mix-and-match third-party hardware and software. Third-generation providers, such as Google Talk, adopted the concept of federated VoIP—which is a departure from the architecture of the legacy networks. These solutions typically allow dynamic interconnection between users on any two domains on the Internet when a user wishes to place a call.

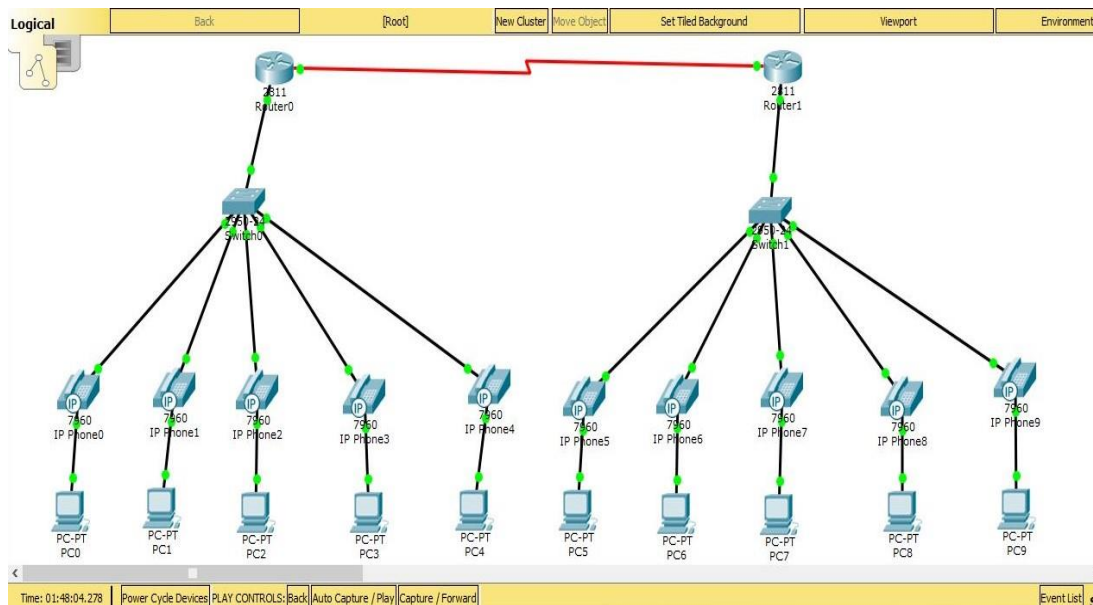
In addition to VoIP phones, VoIP is also available on many personal computers and other Internet access devices. Calls and SMS text messages may be sent over mobile data or Wi-Fi. VoIP allows modern communications technologies (including telephones, smart phones, voice and video conferencing, email, and presence detection) to be consolidated using a single unified communications system.

PROBLEM STATEMENT:

VoIP is another method of making phone Calls that can be very economical or totally free. The 'phone' components are not always present anymore, as people can communicate without a telephone set. VoIP is the transmission of voice and audio, video content over Internet Protocol (IP) networks. VoIP is enabled by a group of technologies and methodologies used to deliver voice communications over the internet, enterprise local area networks or wide area networks.

DESIGN:

To implement voice over ip using cisco packet tracer we have routers, switches and ip phones and the end devices will be pc's. The following figure represent the complete topology that are connected to each other. Here in the topology we have two routers which are of 2811 router, two switches of type 2950 switch, 10 ip phones where in cisco packet tracer we have 7960 ip phones which is used by default and the end devices will be pc where we have 10pc's in the below given topology. Here we have connections of type copper straight except between two routers which has serial connection between each of them.



SIMULATION PHASE:

Here in the simulation phase we are going to start with the configuring the switches and routers in command line interface(CLI) where we are supposed to enter the commands to configure each of them.

Start configuring router:

```
Router>enable
```

```
Router#configure terminal
```

Enter configuration commands, one per line. End with CNTL/Z.

```
Router(config)#host r1
```

```
r1(config)#int fa0/0
```

```
r1(config-if)#ip add 1.0.0.1 255.0.0.0
```

```
r1(config-if)#no shut
```

```
r1(config-if)#
```

%LINK-5-CHANGED: Interface FastEthernet0/0, changed state to up

%LINEPROTO-5-UPDOWN: Line protocol on Interface FastEthernet0/0, changed state to up

```
r1(config-if)#exit
```

```
r1(config)#ip dhcp pool voip
```

```
r1(dhcp-config)#network 1.0.0.0 255.0.0.0
```

```
r1(dhcp-config)#default-router 1.0.0.1
```

```
r1(dhcp-config)#option 150 ip 1.0.0.1
```

```
r1(dhcp-config)#exit
```

Now configure router to interact with telephones.

```
r1(config)#telephony-service
```

```
r1(config-telephony)#max-ephones 5
```

```
r1(config-telephony)#max-dn 5
```

```
r1(config-telephony)#ip source-address 1.0.0.1 port 2001
```

```
r1(config-telephony)#auto assign 4 to 6
```

```
r1(config-telephony)#auto assign 1 to 5
```

```
r1(config-telephony)#exit
```

Now its time to assign phone numbers to ip phones:

```
r1>enable
```

```
r1#configure terminal
```

Enter configuration commands, one per line. End with CNTL/Z.

```
r1(config)#telephony-service
```

```
-dn)#ephone-dn 2
```

```
r1(config-ephone-dn)%%LINK-3-UPDOWN: Interface ephone_dsp DN 2.1, changed  
state to up
```

```
number 1000
```

```
r1(config-ephone-dn)#exit
```

```
r1(config)#telephony-service
```

```
r1(config-telephony)#ephone-dn 1
```

```
r1(config-ephone-dn)#number 1001
```

```
r1(config-ephone-dn)#
```

```
r1(config)dial-peer voice 20 voip
```

```
r1(config-dial-peer)destination-pattern 1...
```

```
r1(config-dial-peer)session target ipv4:192.168.1.0
```

Now configure switch:

```
Switch>enable
```

```
Switch#configure terminal
```

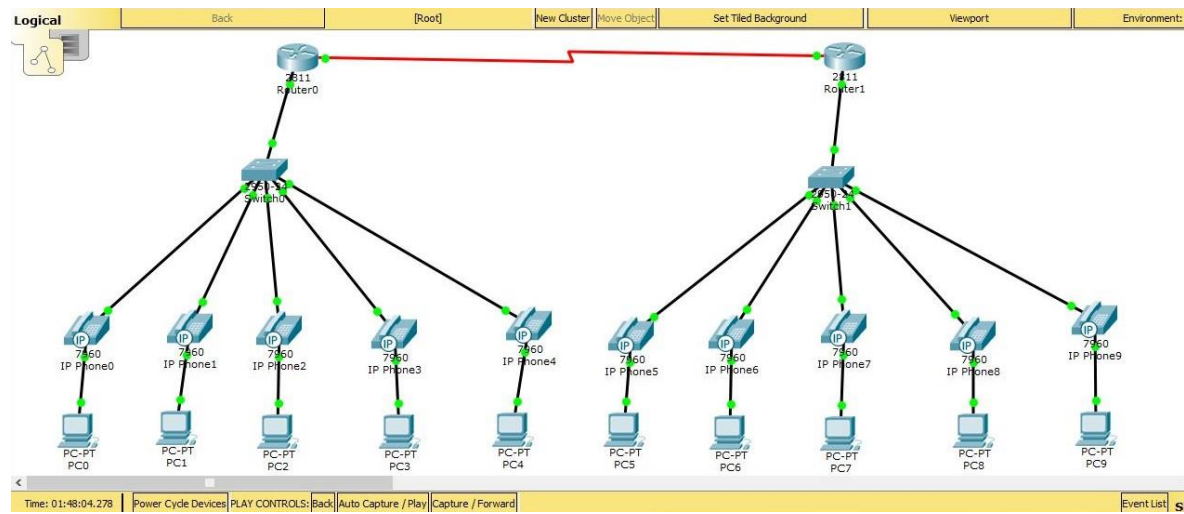
Enter configuration commands, one per line. End with CNTL/Z.

```
Switch(config)#int range fa0/1-5
```

```
Switch(config-if-range)#switchport mode access
```

```
Switch(config-if-range)#switchport voice vlan 1
```

SCREENSHOTS:



As told in the design phase, this is the entire topology of our project which is voice over ip network(voip, IP telephony).



You can clearly see in above diagram that the ip phone number is 1004 and from there i am dialing ip phone which number is 1003. You can see in output ring out option which means phone is ringing. Now lets go to other phone and receive the call and see what happens after receiving this dial number.



You can clearly see when i received the call coming from ip phone 1004, my phone showing Connected option on screen when we pick the call. Here whenever we get a call from other ip phone the light starts blinking on the phone for which we have dialed.

CONCLUSION:

VoIP stands for Voice over Internet Protocol. It is also referred to as IP Telephony, Internet Telephony and Internet Calling. It is another method of making phone Calls that can be very economical or totally free. The 'phone' components are not always present anymore, as people can communicate without a telephone set. VoIP is the transmission of voice and audio, video content over Internet Protocol (IP) networks. VoIP is enabled by a group of technologies and methodologies used to deliver voice communications over the internet, enterprise local area networks or wide area networks, Therefore, it is suggested that in this way people can send their voice from one location to another location with the data by their existing broadband internet without any additional charges. In this way they can save their money It is found that IP phone automatically receives the IP address and phone number via router When we call to phone number 1001(caller) by the phone 1004(caller) then it is found that caller phone has received the call successful from caller phone and a link was found in between caller and caller. When caller put the receiver on phone then caller shows a message "The line is disconnected".