# Voice over Internet Protocol with IP Phones

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***Abstract*–In future the Internet becomes the critical information infrastructure for both personal and business applications for fast and reliable. A voice over internet protocol is IP based packet-switched networks (i.e. Internet) with public packet switched networks (i.e. EIGRP). The key building block of a VoIP gateway is an IP Phone system that interfaces with the EIGRP, RIP, OSPF and converts analog signals from these protocols to digital signals that can be manipulated by a computer and vice versa. Today's exterior routing protocol, BGP and PSTN is known to be packet loss in reacting and recovering from network failures. Many techniques have been focused on the performance and security and reducing packet loss. However, those approaches require modifying BGP, SIP or PSTN which makes them impractical in the Internet. So in this paper, we proposed a simple and practical approach to strengthen the working of packet tracer ,no packet loss and security to reach a message and voice call by using interior routing protocols RIP, EIGRP, OSPF. Our experimental result reveals that it can reduce packet loss, cost and security during link failures significantly by using Cisco Packet tracer 5.3.3.**
***Keywords:* VOIP, RIP, OSPF, EIGRP, Security, no Packet loss.**

## 1. Introduction

In voice over internet protocol maintenance is very important activity according to performance and cost, which also include enhancement to improve the performance. In voice over internet protocol many exterior routing protocols BGP, SIP or PSTN have been applied to improve the performance and reliability but those have not given the good quality performance for customer's expectations to small and medium business applications [5].

Voice over IP (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 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]. Signaling protocols are used to set up and tear down calls, carry information required to locate users and negotiate capabilities. One of the main motivations for Internet telephony is the very low cost involved. Some other motivations are:

- Demand for multimedia communication.
- Demand for integration of voice and data networks.

This paper concentrates on implementation and, performance, issues, cost savings to voice calls with Cisco Packet tracer 5.3.3 and requirements for service providers, especially for corporations with large data networks, Small-Medium Business (SMB) customers. It works with the traditional telephone networks via an IP gateway and serves as a business infrastructure for existing and future services like web, email, video and streaming media. The high cost of long-distance and international voice calls is the crux of the issue.

It is from regulatory taxes imposed on long-distance voice calls. Such surcharges are not applicable to long-distance circuits carrying data traffic [1]; thus, for a given bandwidth, making a data call is much less expensive than making a voice call. In addition to that we use the IP Telephones .Recently, several approaches have been extended to distribute multiple paths to provide the information to networks so here we used RIP, EIGRP, OSPF routing protocols and implemented for shortest path to transmitting data with packet lossless and reliability for calls by using Cisco Packet Tracer 5.3.3.

An example of emerging value-added services is Hosted VoIP, which provides voice service with full functionality and it gives security also [3].

### 1.1 Requirements:

VoIP requires a connection to the Internet through an ISP, a VoIP service to extend the reach to traditional landlines, and VoIP software to actually place calls. Plain Old Telephone Service (POTS) requires none of these prerequisites. It is important to note that Digital Subscriber Line (DSL) internet service uses traditional phone lines for your internet connection; in this case, you already have telephone service to begin with [4] ,[7]. You may wish to weigh the expected benefits of VoIP against these costs given your current operating environment or operating system.

VoIP communication nearly always requires a high-speed (broadband) internet connection for reliable functionality. Even given typical broadband connection speeds, though, service interruptions or degradation of quality is possible due to high internet traffic [1], [4]. For example, if you are trying to place a VoIP call while other people are using a lot of bandwidth on the same internet connection, the sound quality of your VoIP call or general VoIP availability may be affected.

### 1.2 Services Offered:

VoIP has enabled many new services, the most basic being Internet telephony service to users over an IP network. This may involve PC-to-PC or PC-to-phone, Phone-to-phone connectivity [4]. The services can be categorized as follows:

(1). *Residential VoIP*. Voice over broadband services sold to residential and home office customers.

(2). *Hosted VoIP for businesses.* Voice service with full PBX like functionality for SMB (small & medium business)

(3). *Long-distance bypass*. IP trunking service offered by carriers to enable long drag voice providers to bypass long distance toll networks.
(4). *IP trunking services*. Connected islands of PSTN networks using private IP networks
(5). *VoIP peering*. Allows direct peering of VoIP networks to completely bypass PSTN networks wherever possible it.
(6). *Voice-capable* routers and switches, toll bypass, over WANs to transport packetized voice traffic such as telephone calls over IP data intranetworks or internetworks.

**1.3 Main issues:**

For VOIP to become popular, some key issues need to be resolved. Some of these issues stem from the fact that IP was designed for transporting data while some issues have arisen. [5] Because the vendors are not conforming to the standards and protocols.

The key issues are:
(1). Quality of voice

For voice communications over IP to become acceptable to the users, the delay needs to be less than a threshold value and the IETF (Internet Engineering Task Force) are working on this aspect [5].

(2). Interoperability

In a public network environment, products from different vendors need to operate with each other if voice over IP is to become common among users. To achieve interoperability, standards are being devised [5].

(3). Security

This problem exists because in the Internet, anyone can capture the packets meant for someone else. Some security can be provided by using encryption and tunneling [5].

(4). Scalability

As researchers are working to provide the same quality over IP as normal telephone calls but at a much lower cost, so there is a great potential for high growth rates in VOIP systems [5].

**1.4 Contributions:**

We should follow when we are doing any research on some particular topic [6].

*(1) Increasing the VoIP capacity:* Our objective is to increase the number of supported VoIP calls with the given quality measure over internet.

(2) *Maintaining the VoIP quality:* Supporting delay sensitive real time application such as VoIP over mesh network is a challenge.

**1.5 Advantages and Disadvantages:**

There are many advantages and disadvantages also.

(1). Short for Voice over Internet Protocol, VOIP technology can transmit voice, video and data across wide geographical areas; it is possible to hold teleconferences with offices or clients in virtually any part of the globe [8].

(2).With a one-time expense of setting up a VoIP system, business can achieve significant savings on long distance phone services and traveling costs.

(3). Cisco is one of the pioneers of this area. It offers various packages of VoIP services that are suited to many different business sizes [10].

(4). Routing protocols are very useful to improve the software and hardware reliability.

(5).It Supports emergency calls at all times and allow prioritization also for performance and reliability.

## 2. Related work

**Security:**

At this point while we are transport a message with the intention of communication motivation initially it goes to the control first and after that starting switch it will exceed to the additional devices those which are connected to the switch. So now we comprise to talk about how the significance or message is disappearing to other devices with no packet failure or crash [1].

We send a message starting source in the direction of destination that we decide any network as a source and any network as a destination. When we are sending a message to some particular destination from source then, the packet ought to not share with the other devices. If the packet shared with other devices that means the packet has some information used on that device. So as to means the packet information failure to source.

When we are using BGP or PSTN protocols it is some complex to provide security and no packet loss is moreover not feasible that much therefore for security and no packet loss we use EIGRP, OSPF, RIP protocols. These protocols provide fast performance, full of security and no packet loss from source to destination [11]

**Logical workspaces:**

The logical workspace allows people to get a global view of the network through real-time or simulation mode. The devices are added, linked, and configured on this workspace by using some commands on this tool. It has more options to do the configuration when we are doing our research on this tool.

This is the logical workspace; whatever we connected those devices will shows like the above diagram [14]

**Physical workspaces**

The physical workspace gives a physical dimension to the logical network topology. This workspace is particularly important for wireless labs, where the distance parameter is one of the factors that determine if a device is able to connect or not connect to another device. The physical workspace is divided into four layers to reflect the physical scale of real life environments: Intercity, City, Building, Wiring Closet [14].

**OSPF: Open shortest path first**

The Open Shortest Path First (OSPF) protocol is an intra domain routing protocol based on link state routing. Its domain is also an autonomous system.

Areas, Metric, Types of Links, Graphical Representation, OSPF Packets, Link State Update Packet, Other Packets, and Encapsulation. These are the topics in OSPF [9].

*Area*: OSPF divides an autonomous system into areas. An area is a collection of networks, hosts and routers.

*Metric:* OSPF allows assign a cost, called the metric, to each route. The metric can be based on a type of service (minimum delay, maximum throughput, and so on).and we have some examples on link vector routing protocols [9].

It has different types of links like stub network, transient, point to point, realistic networks [8].
(i). It uses IP, has a value in the IP Header (8 bit protocol field).
(ii). It is Interior routing protocol; its domain is also an autonomous system.
(iii). Special routers or backbone routers responsible to dissipate information about other AS into the current system.
(iv). Divides an AS (autonomous systems) into areas. Metric based on type of service. Minimum delay (rtt), maximum throughput, reliability, etc.

**Shortest Path Calculation:**
Here we take one network and calculate which the shortest path in that network for sending is and receiving the messages or making the voice calls.
*Example:[4] ,[8]*
And this example will continued in the paper presentation of open shortest path first.it has open shortest path first packet and header format also [8], and routing table as shown.
After stabilized the routing tables for autonomous systems are

## Results

Here I observed the performance when I sent a message from one network to another network. It shows the time event and also shows whether it is success or not. It shows each and every step from where to where the message and voice calls are going to the destination.

Here first we have do RIP, EIGRP and OSPF configuration. This configuration is not sufficient for dialing a call to other network, we should do dial peer configuration also for all networks.
Now we are sending a packet from IP phone 16 to IP phone 13. Now we will see that how it is going from IP phone 16 to IP phone 13 and how packet is going with security and without packet loss it will reach the destination.

| Vis | Time(sec) | Last Device | At Device | Type | Infr |
|---|---|---|---|---|---|
| | 0.000 | IP Phone16 | Switch0 | ICMP | R |
| | 0.001 | Switch0 | Router0 | ICMP | R |
| | 0.002 | Router0 | Router1 | ICMP | R |
| | 0.003 | Router1 | Switch1 | STP | R |
| | 0.003 | Switch1 | IP Phone13 | ICMP | R |
| | 0.003 | Switch1 | Router1 | STP | R |
| | 0.004 | Router1 | Router0 | STP | R |
| | 0.004 | Router0 | Switch0 | STP | R |
| | 0.004 | Switch0 | IPhone17 | ICMP | X |
| | 0.004 | Switch0 | IPhone18 | STP | X |
| | 0.005 | Swicth0 | IP Phone19 | STP | X |
| | 0.005 | Switch0 | IP Phone16 | STP | R |
| | 0.005 | Switch0 | Server0 | STP | X |
| | 0.006 | Switch0 | Laptop0 | STP | X |

This is the way how the message is going from source to destination and here we can see the time for reaching a message and it takes only 0.016 sec compared to BGP which takes 3 sec time to reach a message. R-Right .It means that the message is going with secure and no packet loss at that moment. X-it means that the message with secure, share the information but no packet loss.

| Last status | Source | Destination | Type | Time | Periodic | Nu |
|---|---|---|---|---|---|---|
| Successful | IP Phone16 | IP Phone13 | ICMP | 0.000 | N | R |

This experiments result shows the performance of VoIP. When we compared to other protocols this EIGRP, OSPF, RIP protocols gives very fast performance.

## III. Conclusion

In this report we present a summary of the implementation of an integrated VOIP system with redistribution of routing protocols. Our major conclusions are as follows. In terms of functionality and services that can be supported for EIGRP, OSPF and RIP V2 protocols and are very similar. Fewer interoperability issues are expected among its implementations. We use the OSPF for the purpose of shortest path for going the message from one end to another end. The two protocols are comparable in their QoS support (similar call setup delays, no support for resource reservation or class of service (QoS) setting) [2]. Finally, we implemented a VOIP system using Packet tracer5.3.3 tool and analyzed the traffic characteristics. For the simulation we have considered a number of issues that may arise during the implementation phase of the network.

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