Performance with Voice over Internet Protocol

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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. So we need these routing protocols to implement and maintain the performance of those applications in the presence of failures. Today's exterior routing protocol, BGP and PSTN is known to be very slow in reacting and recovering from network failures. Many techniques have been focused on the performance and reliability of routing. 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 reliability and performance 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 improve the performance during link failures significantly by using Cisco Packet tracer 5.3.3.**
***Keywords:* VOIP, RIP, OSPF, EIGRP, Performance.**

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

Here we have to improve the performance of voice over internet protocol. For that I used here some routing protocols like EIGRP, OSPF, and RIP. We can use any routing protocols but I used to choose these above routing protocols because for fast performance.

Objective:

(1). Distinguish between intra and inter domain routing
(2). Understand distance vector routing and RIP
(3). Understand link state routing and OSPF
(4). Understand about EIGRP

(1). Intra and Interdomain routing:

Routing inside an autonomous system is referred to as intradomain routing. Routing between autonomous systems is referred to as interdomain routing.

(2). RIP (Routing information protocol):

RIP is a routing protocol for exchanging routing table information between routers. Routing updates must be passed between routers so that they can make the proper choice on how to route a packet. It is one of the popular interior and dynamic protocols used to sharing neighborhood the information by using some different metrics. Example distance vector routing protocol [10] like each router sends all or part of its routing table in routing.

The Routing Information Protocol (RIP) is an intra domain routing. Protocol used inside an autonomous system. It is a very simple protocol based on distance vector routing.

The topics discussed in this section include:

RIP Message Format, Requests and Responses, Timers in RIP RIP Version 2, Encapsulation and examples on distance vector routing protocol [9].

RIP is one hop count how many networks a packet crosses and networks are treated as equally. RIP version 1 does not recognize subnets. This feature was added in RIP version 2.Because RIP only uses hop count as a metric, packets may be forced to take a slower route with less hops over a faster route with more hops with limit as [10]. Here in routing information protocol some timers also available for prevent collisions, deleting the route from the table also.

Characteristics of RIP:

i. Distance vector routing protocol.
ii. Uses hop count as a path selection metric.
iii. Three types of timers.
iv. Multiple stability features.

Each router sends all or part of its routing table in routing updates. However, the updates are only sent to neighboring routers.

- Hop Count **--** Fifteen Hop Limit.

Hop count is the sum of all the legs in a route. After 15 hops, the packet is discarded.

RIP Packet:

It contains some portions and it has some multiple stability features for send and receives updates about the information and hold-down mechanisms to put hold-down the data [10].

1	1	2	2	2	4	4	4	4
A	B	C	D	C	E	C	C	F

A-Command - request or response ,B-Version # - specifies version of RIP ,C-Zero- Not used ,D-Address Family Identifier (AFI) - Indicates type of address being specified (IP, IPX, etc) ,E-Address-Specifies IP address of entry's ,F-Metric of hops traversed from source to destination .It will initialize and update the routing table [10].

(3). 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 [10]. The instability of distance vector routing and creation of loops can be avoided in path vector routing.

Note: *OSPF packets are encapsulated in IP datagrams.*

(4). EIGRP: Enhanced Interior Gateway Routing Protocol

EIGRP is an enhanced version of IGRP. The same distance vector technology found in IGRP is also used in EIGRP, and the underlying distance information remains unchanged. The convergence properties and the operating efficiency of this protocol have improved significantly. This allows for an improved architecture while retaining existing investment in IGRP [11].

The convergence technology is based on research conducted at SRI International. The Diffusing Update Algorithm (DUAL) is the algorithm used to obtain loop−freedom at every instant throughout a route computation. This allows all routers involved in a topology change to synchronize at the same time [11].

How Does EIGRP Work?

EIGRP has four basic components:

(i) Neighbor Discovery/Recovery
(ii) Reliable Transport Protocol
(iii) DUAL Finite State Machine
(iv) Protocol Dependent Modules

(i). Neighbor Discovery/Recovery:

It is the process that routers use to dynamically learn of other routers on their directly attached networks. Routers must also discover when their neighbors become unreachable or inoperative. This process is achieved with low overhead by periodically sending small hello packets [11]. As long as hello packets are received, a router can determine that a neighbor is alive and functioning.

(ii).The reliable transport:

It is responsible for guaranteed, ordered delivery of EIGRP packets to all neighbors. For efficiency, reliability is provided only when necessary. For example, on a multi−access network that has multicast capabilities, such as Ethernet, it is not necessary to send hellos reliably to all neighbors individually [11]. So EIGRP sends a single multicast hello with an indication in the packet informing the receivers that the packet need not be acknowledged.

EIGRP uses five packet types:

· Hello/Acks, Updates, Queries, Replies, Requests [11].

Neighbor Table: It stores the data about the neighboring routers, i.e. those directly accessible through directly connected interfaces like fastethernet and serial interfaces should be up then it will correct the configuration[9].

Topology Table: It aggregates the routing tables gathered from all directly connected neighbors, not store the complete network topology and contains a list of destination networks in the EIGRP-routed network with their respective metrics for every destination [13]. Every destination in the topology table can be marked either as "Passive", which is the state when the routing has stabilized , knows the route to the destination, or "Active" when the topology has changed and the router is in the process of updating its route to that destination [12].

Routing table: Stores the actual routes to all destinations; the routing table is populated from the topology table with every destination network that has its successors and has an optionally feasible to the successor identified, and so the successors and feasible successors are serve as the next hop routers for these destinations to the networks.

EIGRP associates six (6) different vector metrics with each route and considers only four (4) of the vector metrics in computing the Composite metric [11]:

(i). Bandwidth:

Required minimum Bandwidth (in kilobits per second) along the path from router to destination network.

(ii). Delay:

Total Delay (in 10s of microseconds) along the path from router to destination network.

(iii). Hop Count:

Number of routers a packet passes through when routing to a remote network, used to limit the EIGRP AS.

Results

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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.

Vis	Time(sec)	Last Device	At Device	Type	Inforn
	0.000	--	IP Phone16	ICMP	--
	0.001	IP Phone16	Switch0	ICMP	--
	0.002	Switch0	Router0	ICMP	--
	0.003	--	IP Phone20	STP	--
	0.003	Router0	Router1	ICMP	--
	0.003	--	IP Phone20	STP	--
	0.004	IP Phone20	Switch3	STP	--
	0.004	IP Phone20	PC0	STP	--
	0.004	Router1	Switch1	ICMP	--
	0.004	--	IP Phone20	STP	--
	0.005	--	Switch1	STP	--
	0.005	--	Switch1	STP	--
	0.005	--	Switch1	STP	con
	0.006	Switch1	Router1	STP	--
	0.006	Swithch1	IP Phone3	STP	--
	0.006	Swithch1	IP Phone4	STP	--
	0.006	Switch1	IP Phone5	STP	con
	0.007	Switch1	Router0	ICMP	--
	0.008	Router1	Router0	ICMP	con
	0.010	Swithch0	IP Phone16	ICMP	--
	0.016	--	IP Phone14	STP	--

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.

Last status	Source	Destination	Type	Time	Periodic	Nu
Successful	IP Phone16	Phone3	ICMP	0.000	N	0

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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