Analysis of the Routing Protocols in Real Time Transmission: A Comparative Study

IKram Ud Din[1] Saeed Mahfooz[2]

Muhammad Adnan[3]

GJCST Computing Classification C.2.2

***Abstract*-During routing, different routing protocols are used at the routers to route real time data (voice and video) to its destination. These protocols perform well under different circumstances. This paper is about to evaluate the performance of RIP, OSPF, IGRP, and EIGRP for the parameters: packets dropping, traffic received, End-to-End delay, and variation in delay (jitter). Simulations have been done in OPNET for evaluating these routing protocols against each parameter. The results have been shown in the graphs which show that IGRP performs the best in packets dropping, traffic received, and End-to-End delay as compared to its other companions (RIP, OSPF, and EIGRP), while in case of jitter, RIP performs well comparatively.**

Keywords-Routing, Protocol, Delay, Packet Loss, Jitter

I. Introduction

A protocol is a set of rules that reveals how computer systems communicate with each other across networks. A protocol also functions as the common medium by which different hosts, applications, or systems communicate. The data messages are exchanged when computers communicate with one another. Examples of messages are sending or receiving e-mail, establishing a connection to a remote machine, and transferring files and data. There are two classes of protocols at the network layer, i.e., routed and routing protocols. The transportation of data across a network is the responsibility of the routed protocols, and routing protocols permit routers to appropriately direct data from one place to another. In other words, protocols that transfer data packets from one host to another across router(s) are routed protocols, and to exchange routing information, routers use routing protocols. IP is considered as a routed protocol while routing protocols are: i). Routing Information Protocol (RIP), ii). Interior Gateway Routing Protocol (IGRP), iii). Open Shortest Path First (OSPF), and iv). Enhanced Interior Gateway Routing Protocol (EIGRP), etc. To forward data packets, the Internet Protocol (IP) uses routing table. RIP uses hop count to determine the path and distance to any link in the internetwork. In case of multiple paths to a destination, RIP selects the path that has fewest hops. The only routing metric RIP uses is hop count; therefore, it does not necessarily opt for the fastest path to a destination [1]. IGRP is developed to address the problems associated with routing in large networks that are beyond the scope of RIP.

IGRP can select the fastest path based on the bandwidth, delay, reliability and load. By default, it uses only bandwidth and delay metrics. To allow the network to scale, IGRP also has a much higher maximum hop-count limit than RIP. OSPF was developed by the Internet Engineering Task Force (IETF) in 1988. OSPF shares routing information between routers belonging to the same autonomous system. It was developed to address the needs of scalable, large internetworks that RIP could not. EIGRP is an advanced version of IGRP that provides superior operating efficiency such as lower overhead bandwidth and faster convergence [1].

As we are examining the video and voice packets during video conferencing and voice packet transmission in this paper, therefore a short introduction of those protocols must also be inevitable that are used for the transmission of these packets. In video conferencing, Real Time Transport Protocol (RTP) is used for carrying out video packets, and for session establishment between the two systems, either H.323 or SIP is used. RTP provides end-to-end network transport functions premeditated for real time applications such as video and voice. Those functions comprise payload-type identification, time stamping, delivery monitoring and sequence numbering [2].

Voice over Internet Protocol (VoIP) is a means of compressing voice using a standardized codec, then encapsulating the results within IP for transport over data networks. For establishing and transporting VoIP traffic, H.323 is a standard protocol [3].

The H.323 standard has been developed by the ITU-T for vendors and equipment manufacturers who provide VoIP service. It was originally developed for multimedia conferencing on LANs, but was later extended to VoIP. The 1st and 2nd versions of H.323 were released in 1996 and 1998, respectively. Currently, its version 4 is under consideration.Session Initiation Protocol (SIP) is the Internet Engineering Task Force (IETF) standard for multimedia or voice session establishment over the Internet. It was proposed as a standard in February 1999. SIP: a detailed protocol that stipulates the commands and responses to set up and tear-down calls. It also details features such as proxy, security, and transport (TCP or UDP) services. SIP describes end-to-end call signaling between devices. SIP defines, as the name implies, how the session is established between two IP nodes with or without media [2].

About-[1,2,3] Department of Computer Science, University of Peshawar, Pakistan
(e-mail[1];ikramuddin205@yahoo.com)
(e-mail; saeedmahfooz@yahoo.com)
(e-mail; adnan5283@yahoo.com)

The goal of this study is to measure the performance of throughput, packet loss, jitter, and delay in real time transmission. The simulations have been done in OPNET, because OPNET has originally been developed for network simulation, and it is fully usable as an ample simulation tool with higher investment. OPNET provides a complete development environment for the specification, simulation and performance analysis of communication networks [4], [5], [6]. OPNET must be able to simulate different network devices and various kinds of transmission lines, and display such information as packet end-to-end delay, delay variation (jitter), and packet loss in the network. The main purpose is to analyze how the network having speech activity. The voice quality can be characterized by two measurements: i) delay of the signal, and ii) distortion of the signal. The delay disturbs the interactivity, while distortion reduces the legibility [7]. Many factors such as a heavy load in the network that creates higher traffic, may contribute to the congestion of network interface [8]. Therefore, this research is important to be managed in order to measure and predict data transfers in real time applications. The remaining paper is structured as: Section 2 describes the work done in the evaluation of routing protocols. Section 3 illustrates the working environment for the implementation of these protocols. Section 4 explains the OPNET simulations of the mentioned protocols. Section 5 concludes our work, and references are given in section 6.

II. RELATED WORK

Privacy and security become necessary requirements for Voice over IP (VoIP) communications that need security services such as integrity, confidentiality, non-replay, non-repudiation, and authentication. Quality of Service (QoS) of the voice is affected by jitter, delay, and packet loss [9].

Normally, telecommunication network consists of routers which optimize the packets' transmission. Practically, a packet is transmitted through a number of paths from one router to another. The selection of path is based on routing tables' information usually received according to routing protocol. A routing protocol is one that provides techniques facilitating a router to build a routing table. It also shares routing information with other neighboring routers.

When a router is switched off, the packets passing through that router is passed to another router. This operation is known as "routing protocol convergence". Packets are possibly to be lost during a routing protocol convergence [10].

Networks like the Internet are renowned today. Such networks consist of routers, switches and hubs, communication media, and firewalls. Servers and clients are usually interconnected by networks. During communication through the Internet, there may be many possible routing paths and many routers between a source and destination. When packets arrive at a router, the router decides as to the next hop in a path to the destination. For making this decision, many algorithms are used, such as RIP, OSPF, IGRP, and EIGRP, etc. The RIP and OSPF try to route the packets to a destination via the path consisting fewest number of nodes (routers). The IGRP and EIGRP attempt to route the packets based on shortest path, shortest delays, and greatest bandwidth factors.

The invention of Curtis et al [11] makes routing decisions. In their invention, a best path is determined according to an IGRP, EIGRP, OSPF, BGP or other routing task that can provide multiple routing paths. A first variety of routers in the best routing path is determined.

Their invention also makes decision for routing a received packet. If the first variety of routers had a noise level, the packet is forwarded to a next router in the best routing path. If not, then according to said IGRP, EIGRP, OSPF, BGP, or the other routing function in a second routing path is determined [11].

A network facilitates the delivery of packets from a source to destination. This delivery is possible through routers. Packets have destination addresses that let routers to determine how to route the data packets. A router has a routing table which stores network-topology information. With the help of network-topology information, the router forwards packets to the destination. A routing protocol consists of methods to select the best path and exchange topology information. There are two main classes of routing protocols: distance vector routing protocols, e.g. RIP and IGRP, and link-state routing protocols, e.g. OSPF. For enterprise networks, OSPF is often preferred [12], [13].

To exchange service availability and network reachability information, router implements one or more routing protocols. In a specific implementation, the border router implements RIP, OSPF, IGRP, EIGRP, or BGP [14].

Routing protocols accept network state information and then on the basis of such accepted information, update network topology information. Routing protocols also distribute the network state information. Path generation and forwarding information generation are also duties of the routing protocols [15], [16].

III. WORKING ENVIRONMENT

When a node wants to transmit real time applications (video or voice) over IP then it must have to pass through a router. For transmission of real time applications, real time transport protocol (RTP) is used and the session is established between two remote stations through session initiation protocol (SIP) or H.323. Except, these real time transmission protocols, some routing protocols are also used which route the real time applications to its destination. These are: RIP, OSPF, IGRP and EIGRP.

Consider the following scenario having two servers i.e. VoIP and video, and two clients which are: VoIP and video client. The distribution of the servers and clients are at two different location, i.e., servers are located at site Lahore (in this case) and the clients at the other site (say Karachi).

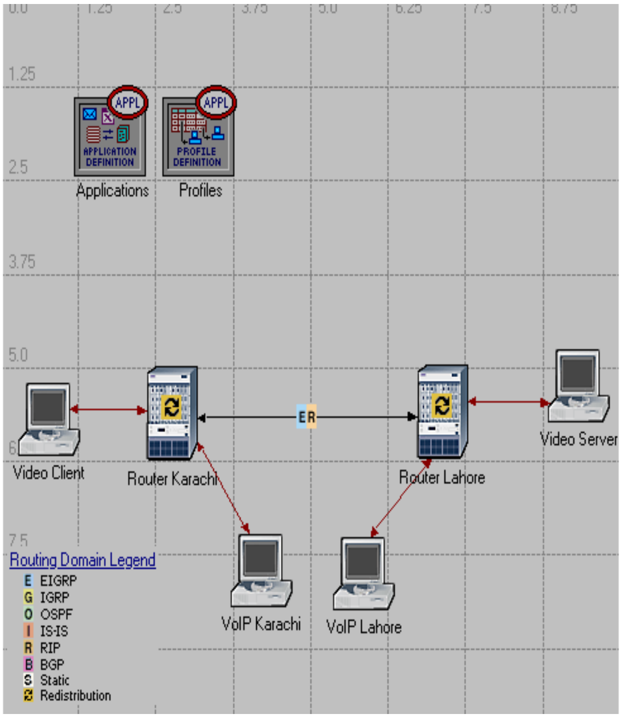

Fig. 1: structure of the network

A. IP Packet/Traffic Dropping

When a router or switch is unable to receive incoming data packets at a given time, is called Packet loss/drop. The real time applications (video or voice) are drastically degraded by packet loss [17].

B. Video/Voice Traffic Receiving

Video/voice traffic is the total number of audio and video packets received during video conferencing or other type of real time communication (e.g., IP telephony).

C. End-to-End delay

End-to-end delay depends on the end-to-end data paths/signal paths, the payload size of the packets, and the CODEC. Delay is the latency; one-way or round-trip, encounter when data packets are transmitted from one place to another. In order to maintain the expected voice quality for Voice over IP (VoIP), the roundtrip delay must remain within almost 120 milliseconds. [17].

D. Variation in Delay (Jitter)

In computer networks, the term jitter means variations in delay of packets received. Jitter is an essential quality of service (QoS) factor in evaluation of network performance. It is one of the significant issues in packet based network for real time applications [18]. The variation of interpacket delay or jitter is one of the principal factors that disturbs voice quality [19]. Jitter plays a vital role for the measurement of the Quality of Service (QoS) of real time applications. The effect of end-to-end delay, packet loss, and jitter can be heard as: The calling party says, "Hello Sir, how are you?" With end-to-end delay, the called party hears,…...Hello Sir, how are you? With packet loss, the called party hears, He.lo….r, w are you? With jitter, the called party hears, Hello…Sir, how....are… you? [2].

IV. Simulation Results

In this section, a scenario was tested in which the delay, packet loss, and jitter were examined.

Figure 2 shows the number of IP packets dropped per second. Figure 3 illustrates the traffic received during video conferencing. The voice traffic received is shown in figure 4. The end-to-end delay in voice packets is given in figure 5, while variation in delay or jitter is clear from figure 6.

A. Performance Evaluation

The number of packets dropped is given in figure 2; in which the less number of packets is lost when IGRP is implemented at the routers. While a huge amount of packets is dropped if OSPF works as a routing protocol. IGRP also works well in case of receiving video and voice packets, given in figure 3 and 4, respectively. The end-to-end delay and variation in delay (jitter) in voice traffic is shown in figure 5 and 6, respectively, in which IGRP is also the best protocol. In the given figures, the X-axis shows the amount of time and the Y-axis shows the number of packets in figure 2, 3, and 4, and in figure 5 and 6, it shows the value of jitter and delay.

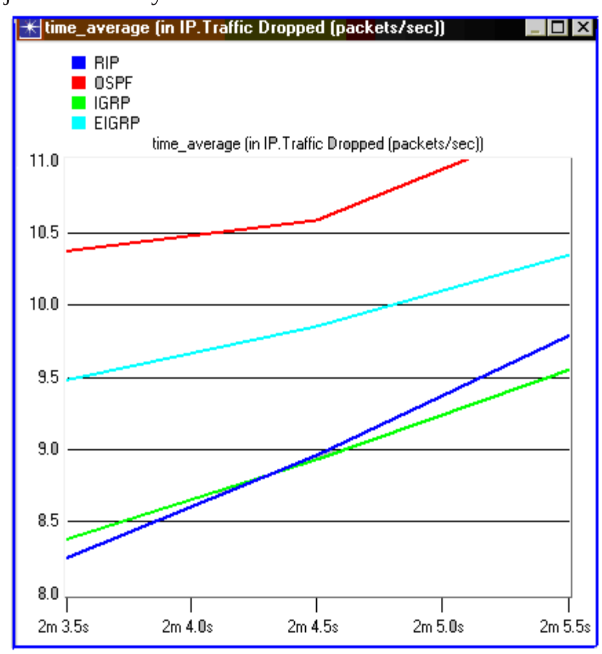

Fig. 2: Number of packets dropped per second

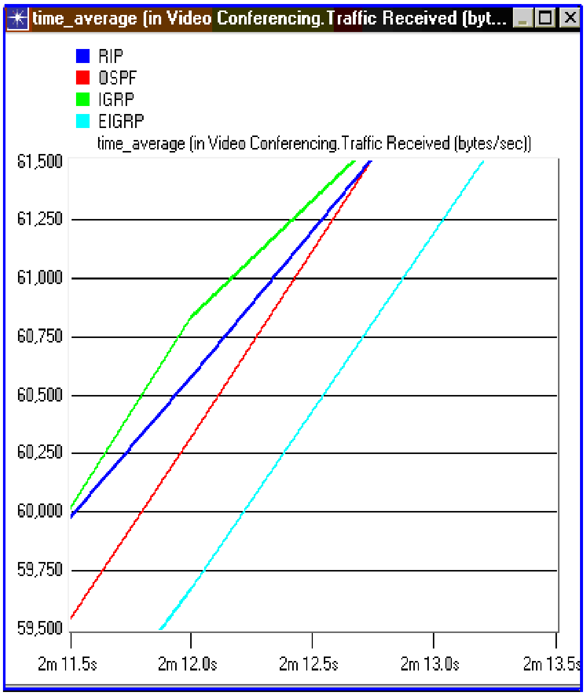

Fig. 3: video traffic received per second

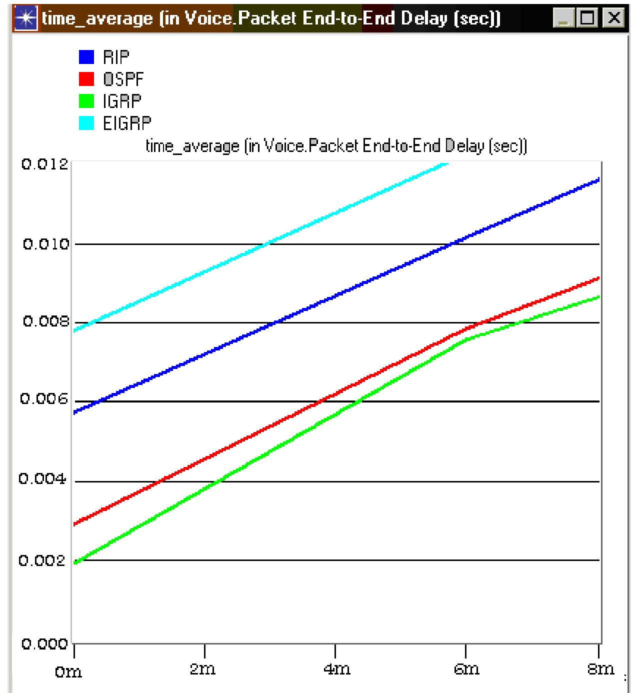

Fig. 5: End-to-End Delay in voice Packets

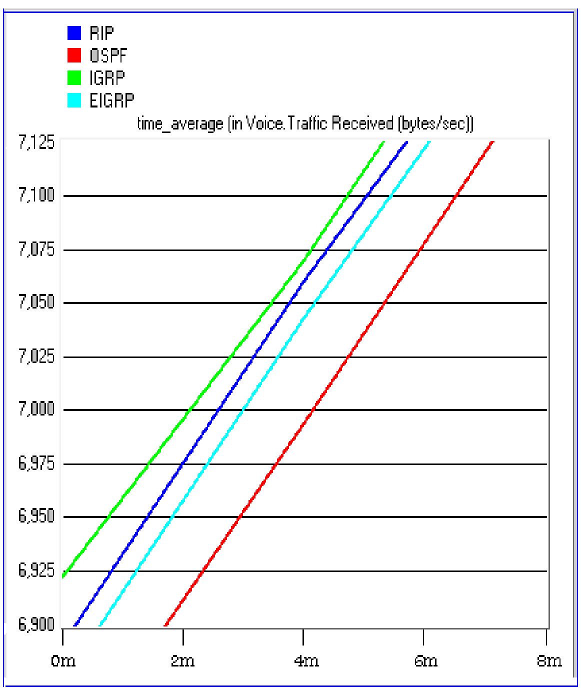

Figure 4: voice traffic received per second

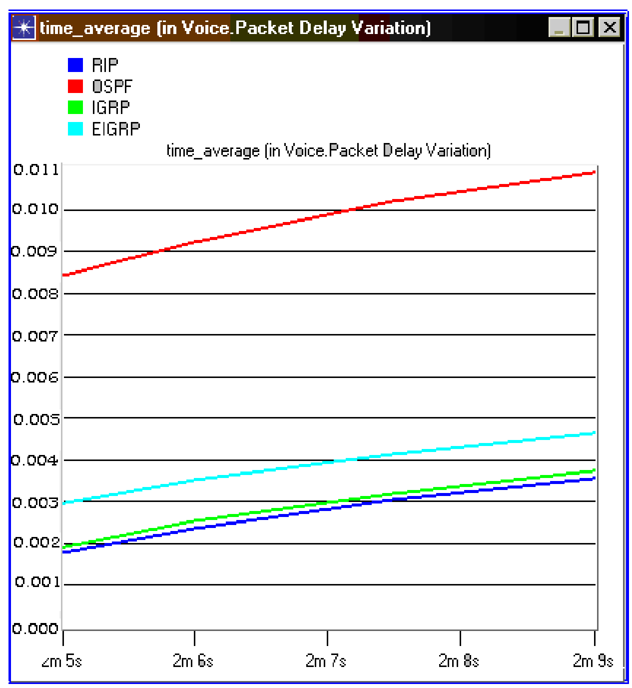

Fig. 6: Jitter in Voice Packets

V. Conclusion

The size of today's networks has been growing quickly and support complicated applications, e.g., video conferencing and voice messages. Quality transmission is demand of the time. This needs some good results producing routing protocols at the routers. The work done in this paper analyzes the available routing protocols: RIP, OSPF, IGRP and EIGRP for packets dropping, traffic received, End-to-End delay, and variation in delay (jitter). Our work is based on OPNET simulation for each of these parameters. The study presents a comprehensive result for each protocol against the parameters: packets dropping, traffic received, End-to-End delay, and variation in delay (jitter) one by one. IGRP performs well in packets dropping, traffic received, and End-to-End delay as compared to its other companions (RIP, OSPF, and EIGRP), while in case of jitter; RIP performs a bit well than IGRP.

VI. References

1) Cisco Systems, I., Cisco Networking Academy Program CCNA 1 and 2 Companion Guide Third Edition. 2003.
2) Cisco Systems, I., Cisco Voice Over IP. Student Guide, ed. V. 4.2. 2004.
3) Shufang Wu, M.R., Riadul Mannan, and Ljiljana Trajkovic, OPNET Implementation of Megaco/H.248 Protocol. 2003.
4) Mohd Ismail Nazri and A.M. Zin, Emulation Network Analyzer Development for Campus Environmetn and Comparison between OPNET Application and Hardware Network Analyzer. European Journal of Scientific Research, 2008. 24(2): p. 270-291.
5) Mohd Ismail Nazri and A.M. Zin, Evaluation of Software Network Analyzer Prototyping Using Qualitative Approach. European Journal of Scientific Research, 2009. 26(3): p. 170-182.
6) Sood, A., Network Design by using OPNET™ IT GURU Academic Edition Software. Rivier Academic Journal, 2007. 3(1).
7) Sjögren, H.R.C., Voice over IP, simulated IP-network, in School of Mathematics and Systems Engineering. 2008, Växjö University.
8) Chang, W.K. and H. S, Evaluating the performance of a web site via queuing theory, in software quality-ECSQ. 2002, springer-Verlag, Berlin, Heideberg: Helsinki, Finland. p. 63-72.
9) Mohd Nazri Ismail and M.T. Ismail, Analyzing of Virtual Private Network over Open Source Application and Hardware Device Performance. European Journal of Scientific Research, 2009. 28(2): p. 215-226.
10) Nicolas Dubois and B. Fondeviole, Method for Control Management Based on a Routing Protocol. US Patent, 2008. No: US 2008/0025333 Al.
11) Richard Scott Curtis and J.D. Forrester, System, Method and Program for Network Routing. US Patent, 2008. No: US 2008/0317056 Al.
12) Thomas P. Chu, R.N. and Y.-T. Wang, Automatically Configuring Mesh Groups in Data Networks. US Patent, 2010. No: US 2010/0020726 Al.
13) Xiaode Xu, M.S. and D. Shah, Routing Protocol with Packet Network Attributes for Improved Route Selection. US Patent, 2009. No: US 2009/0059908 Al
14) Rosenberg, J., Peer-to-Peer Network including Routing Protocol Enhancement. US Patent, 2009. No.: US 2009/0122724 Al.
15) Bruce COLE and A.J. Li, Routing Protocols for Accommodating nodes with Redundant Routing Facilities. US Patent, 2009. No: US 2009/0219804 Al.
16) Russell I. White, S.E.M., James L. Ng, and Alvaro Enrique Retana, Determining an Optimal Route Advertisement in a Reactive Routing Environment. US Patent, 2009. No.: US 2009/0141651 Al.
17) Paul J. Fong, E.K., David Gray, et.al, Configuring Cisco Voice Over IP, ed. S. Edition.
18) C. Demichelis and P. Chimento, IP Packet Delay Variation Metric for IP Performance Metrics (IPPM). Request for Comments: 3393, 2002.
19) Pedrasa, J.R.I. and C.A.M. Festin, An Enhanced Framing Strategy for Jitter Management, in TENCON 2005 IEEE Region 10. 2005: Melbourne, Qld. p. 1-6.