Comparative Analysis of Best-Effort (non- Quality of Service) networks and Quality of Service enabled networks in Voice over IP (VoIP) Enterprise Local Area Network

Sahidul Islam
Faculty of Information ,Media & Electrical Engineering
Th Köln
Cologne,Germany
sahidul.islam@smail.th-koeln.de

Prof. Dr.-Ing. Andreas Grebe
Faculty of Information ,Media & Electrical Engineering
Th Köln
Cologne,Germany
andreas.grebe@th-koeln.de

***Abstract*—— Voice over IP (VoIP) telephony is nowadays a common component of multimedia converged networks and it has revolutionized the world of telecommunication. Due to its interactive nature it becomes a very attractive service. To be utilized properly Voice over IP requires to maintain a precise level of quality. Quality of Service (QoS) is determined by factors like jitter, traffic sent, traffic received and end-to-end delay. Local Area Network is designed, simulated and analyzed to investigate the effect of Quality of Service (QoS) on the performance of the network. This paper investigates the behavior of VoIP and Best -Effort traffic without and with QoS implementation This experiment examines the performance of non-QoS and QoS scenario. Sending and receiving voice traffic and sending Iperf load are used as performance parameters. The result demonstrates a significant difference between the performance of both scenarios. Therefore, the Quality of Service (QoS) has improved the overall performance of the network significantly.**

Keywords— QoS, VoIP, LAN, Codec, Bandwidth, Best-Effort, Quality of Service, Wireshark, Asterisk

I. INTRODUCTION

Voice over IP (VoIP) is a common and vital component of multimedia coverage networks .Using different social media applications a huge amounts of voice traffic are transferred between millions of people across the world. VoIP service requires a specific and precise level of quality to be ensured. The end user perception of the quality is determined by subjective testing as a function of the network impairments such as delay, jitter, packet loss, and blocking probability. The amount of impairment introduced by a packet network depends on the particular QoS mechanism implemented [1] Factors like the delay the packet delay variation (jitter), and the data loss rate determine the Quality of Service (QoS) of the network [2]. Main challenge in multimedia services over IP is that real-time traffic must reach its destination within a predefined time interval (delay) and with some acceptance of the delay variation (jitter). UDP/IP operates on a best-effort basis and also allow permits dropping of packets on the way to a destination that's why it is difficult[3].

Quality of Service (QoS) indicates the statistical performance guarantee of a network . Parameters like average packet loss, average delay, average jitter and average throughput define Quality of Service of a network. In real-time transmission Quality of Service mechanism is deployed to give same sort of priority to the voice and video streams, to insure they will be delivered to the desired destination correctly. Quality of service has been used in many ways with a view to provide best service to network traffic in a network environment, such as priority queuing, application specific routing, bandwidth management, traffic shaping and many other ways. The QoS implementation divided into Application Layer QoS and Network Layer Qos.

1) Application layer QoS – Application layer QoS are implemented at the end system. The value of jitter is mainly controlled at end system in such scenario.

2) Network layer QoS – Network Layer Quality of Service is controlled. at the router and switch (at the network layer). Usually the bandwidth compliance and delay is controlled in the intermediate system in such cases Quality of Service (QoS) is a feature of routers and switches which prioritizes traffic so that more important traffic can pass first. As a result, the performance is improved critical network traffic. LANs with high volumes of local traffic or VoIP phones are benefited by Quality of Service tools[4].

In this experiment network level QoS has been implemented.

The purpose of this paper to investigate Comparative Analysis of Best-Effort networks and Quality of Service enabled networks for Voice over IP (VoIP) Enterprise Local Area Network. At first we will implement the Best-Effort (without Quality of service) to the network and then we will implement Quality of service to the same network. Finally, this experiment will give a comparative view of both scenarios.

II. NETWORK TOPOLOGY

Our network topology consist of Lan A, Lan B and Lan C. Network topology is shown in following figure 1.Lan A consist of Iperf ,VoIP phone , Switch S1 and Router R1.Lan B is the intermediate connecting path between Router R1 and Router R2. We applied DiffServ in this area. Serial Link speed between R1 and R2 is 125kbps. Lan C consist of Router R2, Switch S2 , VoIP phone. Lan C is the destination of Iperf Load.

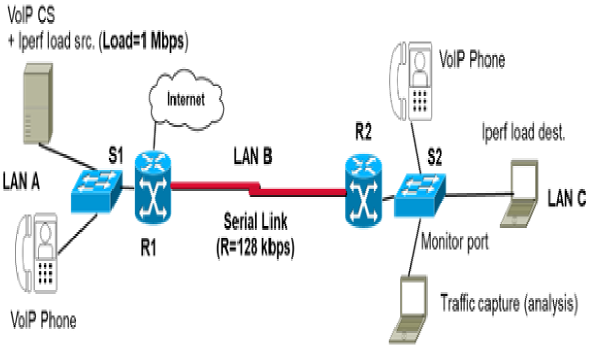

Figure-1: Network Topology

A. *LAN A*

Lan A consist of Switch S1, Router R1, Asterisk server , VoIP phone etc. LAN A and its subnet addressing is shown in following figure 2 and table 1 respectively.

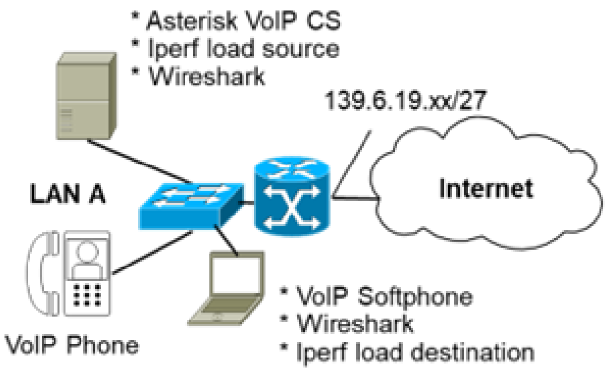

Figure-2: LAN A

Table -1: LAN A Subnet Addressing

LAN A	Subnet Address	Subnet Mask
Default Gateway	10.6.0.1	255.255.255.0
Asterisk Server	10.6.0.2	255.255.255.0
VoIP Phone	10.6.0.3	255.255.255.0
PC Softphone	10.6.0.4	255.255.255.0
Optional	10.6.0.5	255.255.255.0

Router R1 is connected to internet with subnet address 139.6.19.154 and subnet mask 255.255.255.224

B. *LAN B*

LAN B subnetting address is shown in following table 2. The clock master is generated in the data communication equipment interface of Router R1. The clock rate is 125kbps.

Table 2: LAN B Subnet Addressing

LAN B	IP Address	Subnet Mask
R1 Serial Interface	10.6.1.1	255.255.255.0
R2 Serial interface	10.6.1.2	255.255.255.0

C. *LAN C*

LAN C subnet address is shown in following table 3. We applied default router in router R2 and static router in router R1.

Table 3: LAN C Subnet Addressing

LAN C	Subnet Address	Subnet Mask
Default Gateway	10.6.2.1	255.255.255.0
Asterisk Server	10.6.2.2	255.255.255.0
VoIP Phone	10.6.2.3	255.255.255.0
PC Softphone	10.6.2.4	255.255.255.0
Optional	10.6.2.5	255.255.255.0

D. *Asterisk Server*

Asterisk is open-source free software which uses VoIP implementation of Private branch exchange (PBX). Asterisk is similar to hardware PBX, It allows user to make calls to one another using softphones. Also, it is possible to connect to other telephone services including Public Switch Telephone using Asterisk [5,9].

Asterisk provides not only One-To One call but also it provide different services including voice email, meeting call, email notification of voicemail, text messaging, text to voice, call hold, call parking and forwarding calls, music on hold, auto attendance. Asterisk server is enough intelligent to make decisions, like what next to do.

E. *Monitor Port on Switch S2*

DNS Lookup is disabled to prevent the router from attempting to translate incorrectly entered commands. Also, Configured a Monitor PC. Monitor PC is not used for general communications. The purpose of monitor PC is capturing monitored IP flows for traffic analysis. A monitor port in a Switch is used to copy all Ethernet frames, send and received, from selected switch ports to a separate switch port, which is connected with a monitoring device.

III. ANALYZE BEST-EFFORT BEHAVIOR FOR VOIP AND IP LOAD TRAFFIC

A. *Codec*

A CODEC is an algorithm which is used to encode and decode the voice ssignal. Analog voice stream cannot transmit over internet, so it needs to be digitized to transmitte over the Internet. When transmitted signal reach the other end it also needs to be decoded to restore the analog stream. Encoding and decoding can be done in variety of different ways. Among all the codecs few popular CODECs are used and discussed in this paper. Every Codec uses different methods to compress and decompress the voice stream and each CODEC has aprocessing delay to the overall end-to-end delay [10-11]

In this experiment we examined PCMU, iLBC, G 729 A/B codecs, which are supported by our VoIP phones. Application layer throughput of codecs are shown in following table 4

Table -4: Parameters of Codec

Codec	Sampling Rate (Hz)	Application Layer Throughput (kbps)
PCMU	8000	64
iLBC	8000	13,33/15,2
G729 A/B	8000	8

For each codec, phone call from LAN A to LAN C is established and data is captured in Wireshark. RTP IP flow is filtered in Wireshark to select the VoIP user data. RTP provides end-to-end service for transmitting data in real-time. RTP can be used for unicast and multicast services. Separate copy of data are sent from source to destination for unicast service on the other hand a source sends only one copy of data for multicast.

Network is responsible for transmitting data to multiple locations. RTP enables to identify the type of data being transmitted, what the order of the packet, synchronies media streams. However, RTP does not give any guarantee of the arrival of the packets. It is the responsibility of receiver to reconstruct and detect loss packets from the information provided by packet header. RTP does not provide any guarantee for timely delivery or any other Quality of Service guarantee.

Data link layer throughput of selected codec are shown in following table 5

Table 5: Data Link Layer Throughput of Codec

Codec	Data Link Layer Throughput (kbps)
PCMU	85,6
iLBC	27,4
G729 A/B	29,6

Due to overhead encapsulation data link layer throughput higher than application layer throughput. Also, Wireshark does not capture frame check sequence.

B. QoE of VoIP Codec

Codec Selection:
PCMA, iLBC codecs are selected to examine Quality -of-Experience. Sampling rate and Application layer throughput are shown in following table 6.

Table-6: Application Layer Throughput of Codec

Codec	Sampling Rate (Hz)	Application Layer Throughput (kbps)
PCMA	8000	64
iLBC	8000	15.2

C. QoE with different Link Bandwidth

Result of perceptional quality also known as Quality-of-Experience (QoE) is shown in following table -7. For each link bandwidth and codec a phone call is established and experience of voice and speech quality is shown in following table 7.
Table -7: Perceptional Impression and its rating

Codec	Link Bandwidth (kbps)	QoE Description	Rate (scale of 5)	Rank.
PCMA	32	Very high Latency. Take more time to establish call. Call quality is not good.	1.9	6
	64	Medium latency and no no call interruption. Clear voice and the call quality is acceptable.	3.7	4
	128	No latency and no call interruption. Clear voice and the call quality is good.	4.7	2
iLBC	32	Medium latency and no call interruption. A bit noisy voice and the call quality is acceptable	3.5	5
	64	Very low latency and no call interruption. Voice is clear. Voice and call quality is good.	4.4	3
	125	No latency and no call interruption. Voice is clear and call quality is excellent.	4.9	1

IV. BEST- EFFORT SCENARIO

A. Traffic Load Generation

To check the bandwidth sharing behavior in a best effort (BE) scenario udp traffic is generated in parallel. At first call is not generated only load of 1Mbps UDP traffic for duration of 10 seconds are send from source to destination pc and Application Layer Throughput and Wireshark Data Link Layer Throughput (Mbps) shown in following table 8.

Table 8: Application Layer Throughput and Wireshark Data Link Layer Throughput (Mbps

Load Generation	Iperf Application Layer Throughput (Mbps)	Wireshark Data Link Layer Throughput (Mbps)
1 Mbps	.122	.126

B. Analyze Bandwidth Sharing

Now, A phone call is established and parallel to 1Mbps UDP traffic for duration of 10 seconds are send from source to destination pc. Observation is shown in following figure 3

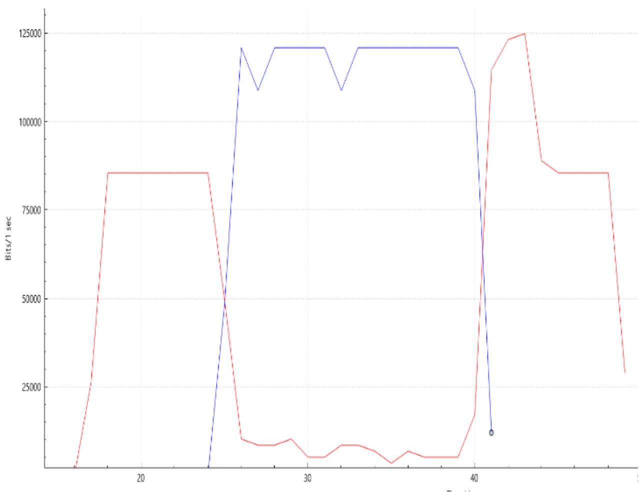

Figure -3: Bandwidth Sharing in BE Scenario

Best effort is shown in above figure 3. From the figure it is clear that while 1Mbps UDP traffic (blue color curve) for duration of 10 seconds are sending from source to destination UDP traffic is sharing almost all of the bandwidth and at this time call quality is not accepted. Serial link bandwidth 125kbps is shared by both network traffic flows. Data link layer throughput of both traffic flow in this scenario is shown in following table 9.

Table 9: Data Link Layer Throughput

Link Bandwidth	Load Traffic Data Link Layer Throughput (kbps)	VoIP Traffic Data Link Layer Throughput (Mbps)
125 kbps	121	.856

V . IMPLEMENT DIFFSERV DOMAIN

Real-time multimedia applications require a certain level of quality of service (QoS) from the network. The Differentiated Services (DiffServ) model provide the building blocks for a scalable IP QoS solution. It defines a set of service classes with respective forwarding rules. depending upon ToS content a packet coming to the router with a Type of Service (ToS) field get better services than other. Diffserv approach are good match with internet architecture and it can be initially deployed with a minimalist approach [12].

.

There are different forwarding classes as follows.

Expedited Forwarding (EF) – EF is intended to provide a building block for low jitter, low delay, and low loss services by ensuring that the traffic is served at a certain configured rate over a suitably defined interval. EF is independent of the offered load of non-EF traffic to that interface.

Assured Forwarding (AF) – AF Per Hop Behavior (PHB) group provides delivery of IP packets in four independently forwarded AF classes. Within each AF class, an IP packet can be assigned one of three different levels of drop precedence, low, medium and high.

Our network run two network services. One is VoIP and other is Best-Effort service. VoIP includes signaling traffic and media data traffic. The VoIP service use a PCM codec for sampling and SIP Signaling. Best-Effort service for any other traffic. Services and their class,dscp class and value are shown in the following table.

Table 10: Class and DSCP Value

Service	Class	DSCP Class	DSCP Value
VoIP	Gold	EF	46
Iperf	Premium	AF11	14
Web Traffic	Best-Effort	BE	0

A. Traffic Matching by ACL

Router R1 and R2 are ingress routers in our Diffserv Domain. Monitoring is configured in LAN C. SIP signaling and RTP media data is send and received from the Asterisk server.Iperf load is used to check the QoS behavior and traffic prioritization in the DiffServ domain. Following IP and port numbers are filtered to match traffic flow.

Table -11: IP address and port number

IP flow	Source Address	Source Port	Destination Address	Destination Port
SIP	10.6.0.2	5060	10.6.2.3	5060
RTP	10.6.0.2	16390	10.6.2.3	5004
Iperf	10.6.0.2	58193	10.6.2.2	5001

Now , ACL is created to select VoIP traffic and Iperf to match on both routers. Class-maps for R1 and R2 are created to to implement a match ip flows to DiffServ classes. Also, policy-maps are created on both routers to set DSCPs.

B. QoS Schedling

VoIP service guarantee link layer throughput of one VoIP call plus additional bandwidth for signaling. We estimate 10% of link layer bandwidth for the signaling part. Required bandwidth for the VoIP service shown in table 12. Also, policy-maps on both routers are created for traffic forwarding in the DiffServ domain.

Table -12: Total required bandwidth for VoIP Service

Service	Bandwidth
VoIP RTP Bandwidth	90
VoIP Signaling Bandwidth	10
Total required Bandwidth	100

VI. Result

A Analyse QoS Services Implementation

Quality of Service is implemented in our networks. Traffic load generation and VoIP call with PCMA are established in parallel. The QoS implementation provides priority for VoIP traffic. Wireshark I/O-Graph for both network traffic flows are shown in figure 4. Red color curve is for VoIP traffic. From the graph VoIP traffic is prioritized when both traffic sharing the link bandwidth

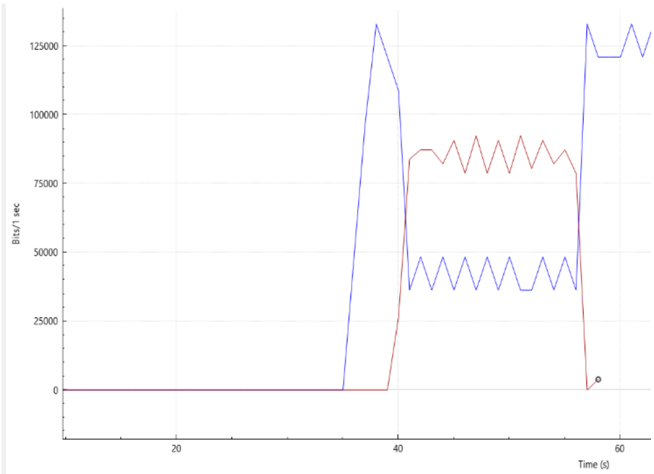

Figure 4: Quality of Service implemented networks.

Voip Traffic Data Link Layer Throughput (kbps) and Load Traffic Data Link Layer Throughput (kbps) are shown in following table 13.

Table 13: Throughput table

Link Bandwidth (Kbps)	Voip Traffic Data Link Layer Throughput (kbps)	Load Traffic Data Link Layer Throughput (kbps)
125	83.6	41.3

Discussion

Comparative scenarios of non -Qos and QoS enabled networks are shown in this paper. Section three and four describe BE scenario and section four describe QoS scenario. Figure 3 illustrates the Best-Effort and figure 4 deficits the Quality-of-Service . From figure 4 it is clear that Quality of Service is implemented in our network successfully and overall performance of networks is improved.

References

[1] Kun I. Park, Ph.D. The MITRE Corporation USA, "QOS IN PACKET NETWORKS", Springer 2005

[2] B. Goode, "Voice over Internet protocol (VoIP)," in Proceedings of the IEEE, vol. 90, no. 9, pp. 1495-1517, Sept. 2002, doi: 10.1109/JPROC.2002.802005.

[3] BUR GOODE, IMPLEMENTING VOICE OVER IP, 2003 by John Wiley & Sons, Inc

[4] https://www.researchgate.net/publication/224959510_Design_and_implementation_of_a_differentiated_service_based_qos_modelfor_real-time_interactive_traffic_on_constrained_bandwidth_ip_networks

[5] J. Meggelen, L. Madsen and J. Smith, Asterisk, 1st ed. Sebastopol,Calif.: O'Reilly, 2007

[6] Datarkar, Trupti, N. P. Bobade, and M. A. Gaikwad. "Voice overInternet Protocol (VOIP) Based On Asterisk." IJAR 1.8 (2015):499-502.

[7] "VoIP Solutions for Unified Communications & Collaboration(UC) | CounterPath", Counterpath.com, 2017. [Online]. Available:http://www.counterpath.com/index.php?menu=products&smenu=xlite. [Accessed: 16- Jun- 2016].

[8] Penton, J., and A. Terzoli. "Asterisk: A converged tdm and packetbased communications system." Proceedings of SATNAC 2003-Next Generation Networks (2003).

[9] Iseki, Fumikazu, Yuki Sato, and Moo Wan Kim. "VoIP systembased on Asterisk for enterprise network." Proc. IEEEInternational Conference on Advanced CommunicationTechnology (ICACT). 2011.

[10] https://www.researchgate.net/publication/261191950_QoS_analysis_of_VoIP_traffic_for_different_codecs_and_frame_counts_per_packet_in_multimedia_environment_using_OPNET

[11] M. Aamir and S. M. A. Zaidi, "QoS analysis of VoIP traffic for different codecs and frame counts per packet in multimedia environment using OPNET," 2012 15th International Multitopic Conference (INMIC), Islamabad, Pakistan, 2012, pp. 275-281, doi: 10.1109/INMIC.2012.6511508.

[12] https://www.researchgate.net/publication/3350357_Traffic_engineering_enhancement_to_QoS-OSPF_in_DiffServ_and_MPLS_networks