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Performance Evaluation for VOIP over IP and MPLS

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Abstract— Corporates and multisite organizations are now applying VOIP usage all over their branches, this made offices with no boundaries and reduced a huge amount of cost for their infrastructure; facilitated exchanging for voice, video and Data .Growing demand for such usage has pushed the wheel for improving and applying more techniques to make this service more reliable, efficient and scalable. In this paper a simulation were performed and compared for a multisite office network for G.723 VOIP communication traffic applied on two network infrastructure models: one for IP and the other for MPLS, the results came encouraging for the MPLS model.

Keywords- component; MPLS; VOIP; CODECS; Multisite Offices.

I. Introduction

Voice over IP is also known as IP telephony or broadband telephony. It routes voice conversations over IP-based networks including the Internet. VoIP has made it possible for businesses to realize cost savings by utilizing their existing IP network to carry voice, video and data; especially where businesses have underutilized network capacity that can carry VoIP at no additional cost on their Local Area Networks.

WAN bandwidth is probably the most expensive and important component of an enterprise network, Network administrators must know how to calculate the total bandwidth that is required for Voice traffic and how to reduce overall utilization, a description in detail for coder-decoders (Codecs), codec complexity and the bandwidth requirements for VoIP calls.

A codec is a device or program capable of performing encoding and decoding on a signal or digital data stream. Many types of codecs are used to encode-decode or compress - decompress various types of data that would otherwise use a large amount of bandwidth on WAN links. Codecs are especially important on low-speed serial links where every bit of bandwidth is needed and utilized to ensure network reliability.

There are a lot of types of codec and each type of them has its pros and cons, for this simulation G.723 was chosen G.723: Describes a dual-rate speech coder for multimedia communications. This Compression technique can be used for compressing speech or audio signal components at a very low bit rate as part of the H.324 family of standards. This codec has two bit rates associated with it [1]:

— r63: 6.3 kb/s; using 24 byte frames and the Multipulse Linear predictive coding with Maximum Likelihood Quantization (MPC-MLQ) algorithm.

— r53: 5.3 kb/s; using 20 byte frames and the (Algebraic code-excited linear prediction) ACELP algorithm.

MPLS is a switching mechanism that assigns labels (numbers) to packets, and then forward packets based on labels. The labels are assigned at the edge of the MPLS network, and forwarding inside the MPLS network is done solely based on labels. Labels usually correspond to a path to Layer 3 destination addresses; similar to IP destination- based routing. Labels can also correspond to Layer 3 VPN destinations (MPLS VPN) or non-IP parameters, such as a Layer 2 circuit or outgoing interface on the egress router. That means that it acts like glue between layer 3 and 2 to make forwarding decision based on who is available, such as Any Transport over MPLS (AToM), quality of service (QoS), or source address.

Multiprotocol Label Switching (MPLS) is a tunneling technology used in many service provider networks [2], MPLS domain has two main types of switches: MPLS core switch which consists of Label Switch Routers (LSRs) and the other is MPLS edge which consists of Label Edge Routers (LERs), the main components of MPLS technology are explained as follows:

•Label Switch Router (LSR) – A router which is located in the MPLS domain and forwards the packets based on label switching is called LSR and usually this type is located the provider cloud; as soon as LSR receives a packet it checks the look-up table and determines the next hop, then before forwarding the packet to next hop it removes the old label from the header and attaches new label.

•Label Edge Router (LER) – LER handles L3 lookups and is component that is responsible for adding or removing the labels from the packets when they enter or leave the MPLS domain. Whenever a packet is entering or leaving MPLS domain it has to pass through LER router, when a packet enters into MPLS domain through LER which is called “Ingress router”, or when a packet leaves the MPLS domain through LER which is called Egress router.

•Label Distribution Protocol (LDP) - the protocol where the label mapping information is exchanged between LSRs .It is responsible in establishing and maintaining labels between switches and routers.

•Forward Equivalence Class (FEC) – set of packets where they have related characteristics which are forwarded with the same priority to the same path. This set of packets is has the same MPLS label. Each packet in MPLS network is assigned with FEC only once at the Ingress router.

•Label Switched path (LSP) – the path set by signaling protocols in MPLS domain. In MPLS domain there are number of LSPs that are originated at Ingress router and traverses one or more core LSRs and terminates at Egress router.

In general, MPLS has two planes:

Control Plane: - The Control Plane is responsible for the routing information exchange and label distribution between adjacent devices.

Data Plane: - The Data Plane is responsible for forwarding packets according to the destination IP address or label using LFIB managed by the control plane.

In MPLS routers control plane and data plane are separated entities. This separation allows the deployment of a single algorithm that is used for multiple services and traffic types [3].

There are two ways to create LSPs in the MPLS network, one is control driven LSP and the other is explicitly routed LSP. Control driven LSP are also called as hop-by-hop LSP which are set using LDP protocol. Explicitly routed LSPs are also called as constraint based routed LSPs (CR-LSPs), and that’s will be used in our simulation to define the path.

II. RELATED WORKS

Fine-tuning, analyzing and optimizing voice traffic over data networks have been a major challenge to researchers and developers ,many techniques have been proposed based on analyses from real word and simulated traffic.

Authors in [3] have made a comparative analysis of MPLS over Non-MPLS networks and showed that MPLS have a better performance over IP networks, through this paper a comparison study has been made on MPLS signaling protocols (CR-LDP, RSVP and RSVP-TE) with Traffic Engineering by explaining their functionality and classification. The Simulation of MPLS and Non-MPLS network is done; performance is compared by with consideration of the constraints such as packet loss, throughput and end-to-end delay on the network traffic.

Authors in [4] analyzed three commonly used codecs using peer-to-peer network scenario. The paper presents OPNET simulator and they were considered only in Latency, Jitter and Packet loss. They were able to present from the results that G.711 is an ideal solution for PSTN networks with PCM scheme. G.723 is used for voice and video conferencing however provides lower voice quality. Music or tones such as DTMF cannot be transmitted reliably with G.723 codec. G.729 is mostly used in VoIP applications for its low bandwidth requirement that’s why this type is mostly common on the WAN connections and to transport voice calls between multisite branches.

Authors in [5] they calculated the minimum number of VoIP calls that can be created in an enterprise IP network. The paper presents OPNET simulator designing of the real-world network model. The model is designed with respect to the engineering factors needed to be reflected when implementing VoIP application in the IP network. Simulation is done based on IP network model to calculate the number of calls that can be conserved.

Authors [6] in their Simulation experiments, they observed that SIP module provides and reduced congestion over access networks. The proposed architecture efficiently utilized the available bandwidth on various links, they also proposed that SIP based QoS Module can be further improved by the use of integration of Integrated Services, Multi-Protocol Label Switching, Resource Reservation Protocol and Differentiated Services mechanisms.

III. Aim of Work and Assumptions Used

The main goal of this research is to analyze the voice and data traffic that could run over MPLS network to be within an acceptable range to communicate voice compared to an IP traditional network, this will insight network managers , researchers and designers to determine quickly and easily how well VoIP will perform on a network prior to deployment, prior to the purchase and deploy for VoIP equipment, it is imaginable to predict the number of VoIP calls that can be maintained by the network while satisfying VOIP requirements , therefor simulation is divided in two scenarios:

Scenario 1: In this part of the simulation the VoIP traffic is send from source (VoIP_A) to destination (VoIP_B) in the two networks using G.723 encoding; MPLS Network as shown in Figure 1 and Traditional IP networks as shown in figure 2 .The main goal is to compare the performance of VoIP traffic in the both networks by using performance metrics, i.e., voice jitter, packet End-to-End delay, packet loss and throughput. The simulation results obtained are analyzed to determine the effective technology used for transmitting VoIP traffic.

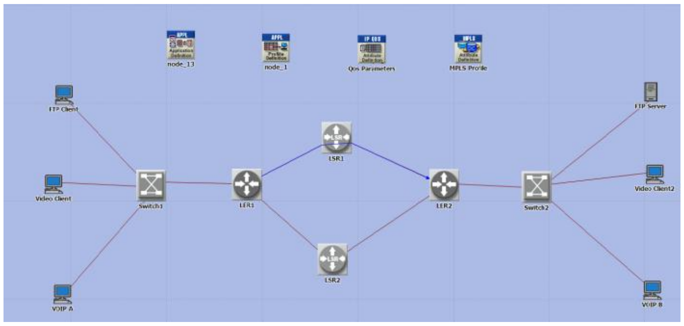

Figure 1. VOIP over MPLS Network Model

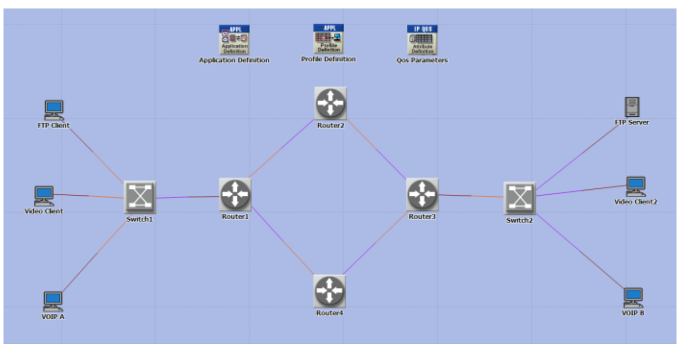

Figure 2. VOIP over IP Network Model

Scenario 2: In this part of the simulation, an approach is made to estimate the approximate minimum number of calls that can be maintained in the both networks considering all the factors to have a successful call. This approach can be used to estimate the number of calls, in a real network. This is done by designing the real network in the OPNET. We use the End-to-End delay performance metric obtained from the simulation to estimate the approximate minimum number of calls maintained in both networks.

We will consider the background traffic excluding the VoIP traffic to be as 50% of link capacity, as explained in (Fortz, Rexford & Thorup 2002) 60% link capacity is the max-utilization allowed of a link to protect it from bursts, and to make close to real-world environment we will generate FTP traffic and video streaming from two work-station beside VOIP traffic, in both models all links will connecting routers, switches, workstations will be at 100 Mbps links.

IV. PROPOSED WORK

In MPLS model Traffic Engineering is implemented by using CR-LDP signaling protocol, which is configured in by defining FECs in MPLS definition attribute and setting LDP parameters in the routers. The CR-LSP is established can be visible in the Fig.1 as a blue colored link, this will guarantee that the traffic is evenly distributed in the MPLS network.

In IP model all routers will be replaced with normal IP routers, Open Shortest Path Find First OSPF protocol will be used to route all IP traffic that including voice that is transmitted from VOIP A to VOIP B.

G7.23 Codec was chosen in this simulations because beside it's low bandwidth requirements it has mean of opinion score of: 3.62 which is very close to G.729 and has the following characteristics: Optimized for high performance on leading edge DSP architectures, Multichannel implementation, Multi-tasking environment, compatible Common compressed speech frame stream interface to support systems with multiple speech coders, Dynamic speech coders selection if multiple speech codecs available.

The acceptable total End to End delay for VOIP traffic is 150-200ms and preferred at 150ms which includes: Coder (Processing) Delay, Queuing/Buffering Delay, Packetization Delay, Serialization Delay, Network Switching Delay. The Scheme encoder for G.723 would be 5.3 kb/s with total algorithmic delay of 37.5ms and 30ms sample blocks [7]. The minimum delay that was gained in this simulation was 100ms which is acceptable and we will look for any anomaly above 150ms total delay to calculate the acceptable total number of calls.

The first VoIP call is created at the 100th second of the simulation as this time will be used to train the network for the current environment then a call will be created for every 2 seconds in simultaneous fashion and this process will be repeated to the End of the simulation till it reaches the threshold where end to end delay is not accepted. VoIP calls are added continuously in a static interval to both network models, in this way it will be possible to determine the number of calls that can be retained in the given network.

Maintained number of calls = (100- threshold time)/2

V. SIMULATION RESULTS

The simulation duration were made in a way to search for the anomaly and looking for an End to End delay above 150ms as this the threshold that we are looking for as accepted quality voice call along with other factors like Jitter and delay variation, so for IP model the simulation duration were about 500ms and for the MPLS model the simulation duration lasted for 1500ms to reach that satisfying point. It is clearly observed from the graphs that MPLS model have a better performance than IP model for voice traffic and have a better measurement factors and consequently higher number of marinated calls.

In both scenarios Voice traffic starts at 100s, IP model duration is 500ms and MPLS model duration is 1500ms, both received number of packets is compared to send packets, it is shown that MPLS model lasts longer before dropping voice packets. IP Model starts dropping packets at time 441s of the duration while the MPLS models starts dropping voice packets at 1200s as shown in figure 3.

Figure3. Send and received voice packets

It is also shown that at this time frame that IP model at 441s Jitter starts to increase and on MPLS model Jitter starts to increase time 1200s as shown in figure 4.

Figure 4.Voice Jitter for both models

Delay variation is noticed for IP model to be at time 441 as shown in figure 5.

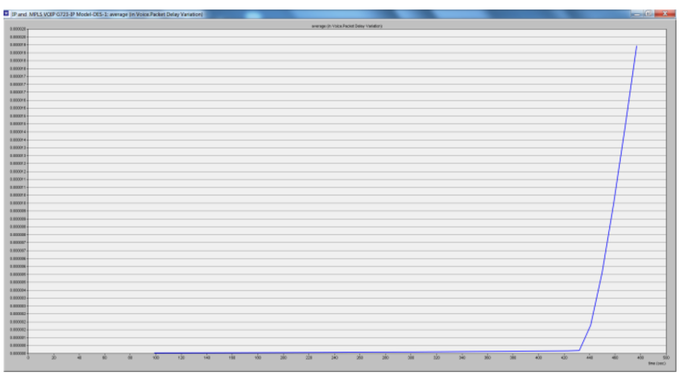

Figure 5. IP delay variation.

And for MPLS model it is noticed that that delay variation starts at time 1200s as show in figure 6

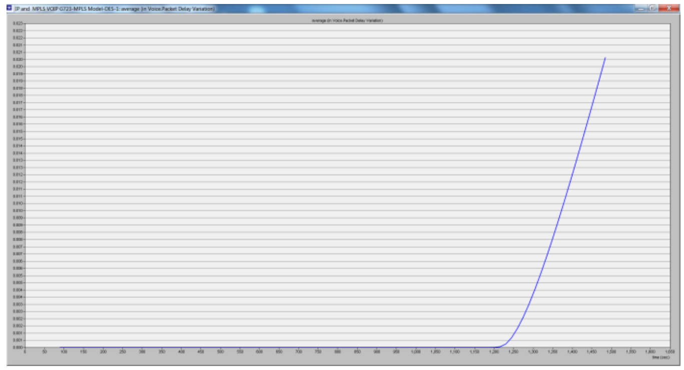

Figure 6. MPLS delay variation.

The most important factor that we will consider is end to end delay which was found above the acceptable quality voice call for IP model at time 441s as shown in figure 7.

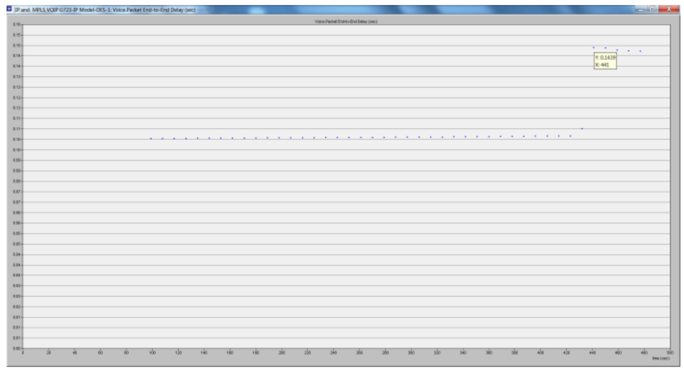

Figure 7. IP End To End delay

And for MPLS model end to end delay was found above acceptable quality voice call at time 1220s as shown in figure 8.

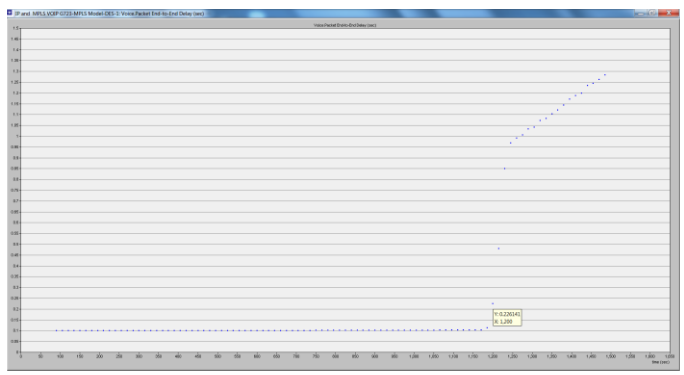

Figure 8. MPLS End To End delay

As we explained before that voice calls are added every 2 seconds and keeping the calls active in simultaneous fashion, so now in this part we can easily calculate the number of maintained calls during the simulation staring from time 100s as this is the time for the first calls.

IP model number of calls =

(441-100)/2= 170 calls with acceptable quality.

MPLS model number of calls =

(1200-100)/2= 550 Calls with acceptable quality

VI. CONCLUSION

Fine-tuning, analyzing and optimizing voice traffic over data networks have been a major challenge to researchers and developers ,many techniques have been proposed based on analyses from real word and simulated traffic

It was clear from the simulation that MPLS model had a better overall performance for voice traffic transmission and would have better utilization.

when we analyzed conventional IP and MPLS network from VOIP factors prospective, performance analysis were made focusing on voice metrics such as Voice End-to-End delay, Voice jitter, Voice packet delay variation, Voice packet send and received which gave the possibility to calculate number of calls in a timely manner to help designers and network managers to have a better understanding for the options that they could have for implementing such challenging tasks like VOIP deployment over LAN and WAN usage.

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