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Assessing and Redesigning Enterprise Networks through NS-2 to Support VoIP

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Abstract

In the recent past, Voice over IP (VoIP) deployments over data networks are gaining popularity due to the massive growth in the broadband internet access. Successful deployment of these applications depends directly on the performance of the underlying data network. Based on this and the fact that today's data networks are operated to perform many significant applications, network administrators seek out a way to measure the impact of these applications on the existing network performance before deploying them. Occasionally, network redesign is necessary; considering redesign's alterations should preserve most of the existing network characteristics to reduce overall impact of deploying new applications in the network performance. In this paper, we evaluated readiness of the existing enterprise network through NS-2 to support VoIP and based on findings a solution for redesigning the enterprise network is proposed to enhance the new network performance and leaving sufficient capacity for future growth. We applied our approach on a medium size enterprise network as a case study, the results prove improvements in network performance after redesigning the existing enterprise network.

Keywords: Redesign, Network Assessment, VoIP, Simulation, NS-2, Data Analysis;

1. Introduction

Deploying multimedia applications such as Voice over IP (VoIP) over data networks are gaining an ever increasing attractiveness in the Internet community, due to the massive growth in the broadband internet access. Most of the network administrators find it extremely advantageous and cost effective to deploy VoIP over their existing data networks. However, one has to keep in mind that, VoIP constitutes challenges for data networks since it merge two networks with different QoS standards, multimedia and data. While multimedia applications require timely packet delivery and very sensitive to low delay, jitter and packet loss, traditional data networks are characterized by their burst traffic and high bandwidth demand at burst time but they are more tolerant with the respect to these performance metrics. That is, optimizing a high performance network for data traffic may adversely affect successful VoIP deployment. To overcome this obstacle, existing data network performance must be enhanced with a mechanism that ensures the required QoS to carry real time traffic and sustain the QoS of the existing services as well.

However, when deploying a new network service such as VoIP over existing network, network administrators are

faced with numerous strategically and challenging questions. Questions similar to: Is the existing infrastructure ready to handle this added burden? What is the impact of the new VoIP load on the QoS for currently running network services and applications? Is it essential to redesign the existing network to grantee successful VoIP deployment without interrupting the obtainable services? If so, what is the estimated cost of the network redesigning and enhancement changes? Else, how many VoIP calls can the network support before upgrading prematurely any part of the existing network hardware? These interrelated and endless questions make the network assessment and redesigning problem a computationally difficult to solve.

In previously related work [1], utilizing NS-2, a topology redesign solution is proposed to ensure network reliability considering retaining or enhancing the network performance in term of bandwidth utilization and packet loss rate. As an expansion to it, this paper focuses on redesigning the existing network to guaranty successfully deployment of VoIP application. Two new terms are added to the performance measurements in this work, latency and number of sustained VoIP calls. Additionally, extensive complex simulations are conducted to assess the network performance in close proximity to communication network real life scopes. The contributions of this paper are threefold: First, we propose a framework for assessing network compliance to support VoIP. Second, we systematically analyzed the simulation results concentrating on the proposed performance measurements. Third, based on findings, we propose a redesign solution to the existing network to improve network performance, satisfying QoS requirements of all existing and added networked services and leaving sufficient capacity for future growth. Yet again, a successful deployment of the VoIP application depends closely on the performance of the underlying data network. As a case study, we illustrate how to apply our approach to a typical medium size network.

2. Related Works

The past decades have witnessed an increasing in the number of VoIP applications deployments over networks, many researchers [2-5] have addressed the performance assessing and analyzing issues for supporting VoIP over networks. In addition, the emerging multimedia market has led to the development of new tools [6-8] for testing the performance of multimedia applications. Other efforts [9-12] have studied the implementation of a simulation tool in assessing network to support multimedia applications. For the most part, none of the previous efforts [2-12] have tackled network topology redesigning problem to enhance the existing network performance, which is our major contribution and the primary objective of this study.

More than a few approaches [13-21] have tackled the network topology designing and redesigning problems. The major differences between our work and these attempts, that we assess network performance using simulation tool and based on findings we redesign the existing network to guarantee successful deployment of a new service and provide more capacity for future expansion.

3. VoIP Service in Data Networks

The long hyped convergence of voice and data onto a single network is finally becoming reality. Obscurity from merging voice and data networks lies in the fact that, voice applications require the network to provide some features that are not very important to data applications. File downloads and database programs, for example, require every byte to be delivered correctly, but they are flexible with regard to time it takes to get the bytes from one location to another. Voice, on the other hand, requires the bytes to arrive in a very timely manner, although it is more flexible about losing a few bytes here and there. Because of this, there is concern about relying on a delay-tolerant data infrastructure to support delay-sensitively voice services.

For employing VoIP over data networks, one has to characterize first the nature of its traffic, QoS requirements, and any additional components or devices. For deploying VoIP, a Call Manager's server must be installed to the existing network. The server handles establishing, terminating, and authorizing connections of all VoIP calls. As a network administrator, placement of this server in the network is another major problem in designing the network. Other hardware requirement includes VoIP client terminals, which can be a separate VoIP device, has to be added to the network client nodes. The quality of a VoIP call depends on low latency, jitter and packet loss. Based on [22-23], QoS requirements for VoIP are constrained as follow: 1) Latency should be below 150 ms, 2) Jitter should be less than 40 ms, and 3) packet loss should be below 3%. Additionally, the required bandwidth for one direction VoIP call is 90.4 kbps with packet overhead [24].

4. Assessment and Redesigning Framework

Assess and redesign problem is appraising the readiness of existing enterprise network to support VoIP through NS-2 and base on findings we determine a set of modifications to redesign the network to enhance its performance and leaving sufficient capacity for future growth with minimum cost.

4.1. Performance Model

We present network performance in terms of latency, bandwidth utilization, packet loss rate and number of sustained voice calls statistics. Throughout our assessment, we consider the number of calls supported by existing network part of the network measurements instead of jitter. Due to the fact that, higher levels of jitter are more likely to occur on either slow or heavily congested links and since we are using dual-duplex links of 100Mbps and 1Gbps, then we could ignore jitter metric.

Latency is the time it takes for a packet initiated from source node to reach the destination node. Latency (Γ) between node i to node j can be estimated according to Equation (1), where ε is packet arrival rate and μ is service rate.

$$\Gamma_{ij} = (t_j - t_i) + 1/(\mu - \varepsilon) \quad (1)$$

Bandwidth utilization is varying depending on network topology and on how network is configured and used. In most networks, bandwidth usage gradually increases as users increase using more network resources. We calculated the bandwidth utilization using Equation (2). Given capacity γ_{ij} and traffic η_{ij} through the link from i to j , bandwidth utilization of a link β_{ij} is calculated as the ratio of the traffic to capacity.

$$\beta_{ij} = \eta_{ij} / \gamma_{ij} \quad (2)$$

Packet loss rate, which is number of requests successfully fulfilled during the lifetime of the simulation, as stated in Equation (3), where the packet loss rate Ω_i of a node i is the ratio of dropped packet δ_i to the total packets v_i entered the node queue, as stated in Equation (3).

$$\Omega_i = \delta_i / v_i \quad (3)$$

Number of VoIP calls, which is the total number of packets successfully traveled from VoIP source node to VoIP destination node. A call from node i to node j is considered to be successful voice call, which means the total number of calls will be incremented, if and only if: latency of voice packet send from node i to node j is less than 150 ms; available bandwidth on node i , node j and every intermediate nodes and links are greater than 90.4 kbps; and loss rate of a voice packet should be less than 3%.

4.2. Redesign Outlines

Based on simulation and analytical findings, recommendations can be made to redesign the existing network by determining a good arrangement of reconfigurable portions of the existing network infrastructure. It is inherently a dynamic problem; because of fact that networking environment, requirements and priorities, and available resources (services, devices, links and nodes) are all change with time. For simplicity reasons, we restrict topology network redesigning in additions, replacing and upgrading of switches or routers devices, reconfiguring switches or routers devices and swapping location between any nodes. Thus, network redesign's goal is to preserve most features of the existing network by requiring as little changes as possible. In our work, voice packet latency threshold is set to 150ms, bandwidth utilization threshold is set to 65%, packet loss rate threshold for data and voice packet is set to 10% and 3% respectively.

In our methodology, we assume that all the network device nodes (routers and switches) are identical in nature. The possible alteration may include: (1) adding network devices or links and define its throughput and capacity respectively; (2) geographical relocating nodes and links; and (3) modifying network configuration (such as QoS standards, routing protocols, etc.).

4.3. Case Study

We have selected a medium size network to study the efficiency of our redesign methodology. The network consists of two-hundred clients, nine servers, nine Layer-2 switches, three Layer-3 switches and a router. The clients and servers operate as a client-server pair. The clients and servers are provided with full duplex links of 100 Mbps capacity and queuing delay of 10 ms to the switches. Links of 100 Mbps capacity were employed for Layer-2 switches connected to Layer-3 switches. The links between Layer-3 switches and the router are configured with 1 Gbps. Open Short Path First (OSPF) routing protocol is configured in the network. The network is configured with a total of eleven VLANs to segregate broadcast and multicast traffic. Based on the hardware requirements for deploying VoIP of section 3, a Call Manager must be added to the existing network this is considered as a network design issue. So, since the local servers are connected directly to SW-3 and since most of the clients are also connected to SW-0, connecting the Call Manager to SW-3 is practical in order to provide accessibility and security to the server. It is proper to include the Call Manager to be a member of SW-3 VLAN so it can access locally the database and file servers to record and log phone calls. Fig. 1 shows the existing enterprise network topology before and after the incorporation of necessary VoIP components.

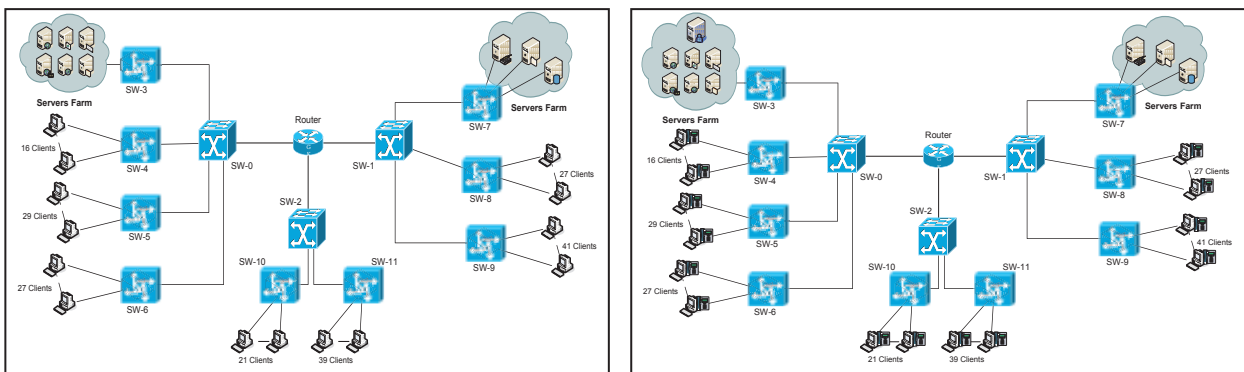


Fig. 1. (a) Enterprise network topology under study; (b) Enterprise network topology with VoIP equipments.

5. Simulation Results

We have utilized UDP and TCP with Drop tail queuing, which is First-In First-Out (FIFO). The exponential traffic generator modes are applied to every client and a CBR traffic stream is applied to every server resource. In all NS-2 simulation scenarios, the simulation time was set to 360 seconds and the interval time between two packets for CBR traffic stream is 0.05 milliseconds. The default packet size is 500 bytes and the maximum is set to 1024 bytes. To reduce high variations in random traffic model, we have repeated the simulation runs three times and used the average as our results. We assume that for every 25 simulated seconds 50% of the clients send and receive local voice packets and the generation of voice traffic is stated at 20 simulated seconds. Furthermore, we assume that 45% of traffic takes place between neighbor's resources and 55% traffic is uniformly distributed among the rest.

Our first NS-2 simulation run is aimed to study the influence of deploying VoIP on existing network performance (results are shown in sub-section 5.1). Keep in mind that, redesign modifications are not yet applied to the existing network. Based on findings, necessary network alterations are proposed to enhance the network performance to guarantee successful VoIP deployment with capabilities for future growth. In sub-section 5.2, we studied the effectiveness of the proposed network redesign solution.

5.1. Simulation Results for VoIP Deployment in Existing Network

Fig. 2 depict performance of the existing network during simulation runtime. In these figures, we show performance results for router, some of switches and some of links. As for the other network devices and elements, they showed healthy behavior and were under the performance thresholds. Fig. 2(a) illustrates the loss rate (Ω) of voice and data packets for the first simulation run. From the graph, it can be noticed that voice Ω remains lower than

3%, until 215 simulated seconds, while data Ω already exceeds 10%, this is justified by the priority of voice packet over data packet in the QoS configuration. The data Ω increased sharply at 155 seconds, after that Ω of both traffic type increased rapidly. The reason for that is that, at this simulated time, the bandwidth utilization (β) of the network devices and interconnected links exceed 65%, as shown Fig. 2(b). It confirms that β has reached 100% short time before the end of the simulation time. As described, at beginning of the simulation run, the network β (less than 30%) and then swiftly get higher at 20 simulated time as the voice traffic is generated. As clients generate voice call requests with a rate of 100 requests every 15 seconds, β keeps gradually increasing.

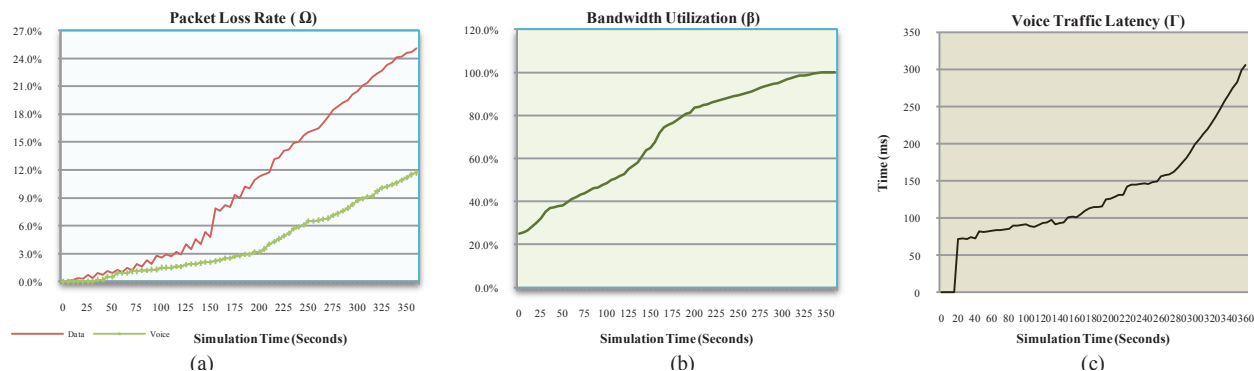


Fig. 2. (a) Packet Loss Rate; (b) Bandwidth Utilization; (c) Voice Packet Latency,

The results of the latency (Γ) of the voice traffic shown in Fig.2(c) show serious traffic congestion. The voice traffic Γ (306 ms) at the end of the simulation run is far worse than the required Γ (150 ms). It goes beyond the specified threshold at 257 simulated seconds, this is because β of links interconnecting network devices approached 85%. Additional cause for this incident is that the voice Ω reached performance threshold at 219 simulated seconds.

From the above findings, it is confirmed that last successful voice call requests sent and received were at 195 simulated seconds, at which voice Ω exceeds 3%. So, number of voice calls sustained by the existing network is $100 \times (195 - 20) / 25 = 700$ calls. Also, we could conclude that the existing network had three bottlenecks (Router, SW-0 and SW-1). Additionally, links connecting SW-0 with Router, SW-1 with Router, SW-2 with Router and SW-3 with SW-0 failed to sustain network reliability. That is, the existing network performance affected directly by these three devices and four links.

5.2. Simulation Results for VoIP Deployment in Proposed Network Redesign

Normally, network administrators recommend replacing primary bottlenecks devices of the existing network; especially if it expected network growth in near future, but the drawback of doing so is the cost. Our main objective is to redesign the existing network to support deploying VoIP and enhancing network performance with minimum cost. Toward achieving this, we proposed, applying load balancing using existing network devices, adding three 1Gbps links and reconfigure existing network to solve bottleneck and availability problems. In our proposal, we created a new VLAN for VoIP equipments and apply the changes to all related switches. Additionally, we enabled Layer-3 switching for local VLAN and OSPF routing for other routing services in switches SW-0, SW-1 and SW-3. As a result, the network consists of four routers, i.e. reducing the routing overhead on the Router device for local traffic. Fig. 3(a) illustrates redesign solution proposed in this paper to the network under study.

Data Ω varies significantly depending upon the voice traffic generation in existing network (Fig. 2(a)), whereas the effects of voice traffic generation on data Ω is less obvious in proposed network as shown in Fig. 3(b). We attribute this result to the VLAN created to isolate voice traffic from data traffic. What's more, voice Ω exceeds performance threshold at 259 simulated seconds. It shows that data Ω of proposed network hardly reached the performance threshold at 324 simulated seconds, whereas in existing network it surpass threshold at 182 simulated seconds. Additionally, 11.3% is the maximum Ω witnessed at the end of the simulation, at the same time it is 25.1% in existing network. Moreover, enhancement in voice Ω in proposed network by 6.8%. Also, at 261 simulated seconds voice Ω go beyond 3% in proposed network. Thus 66 seconds more than the existing network of voice calls

have been sustained. Next figures demonstrate performance results of existing network before and after applying suggested network modifications.

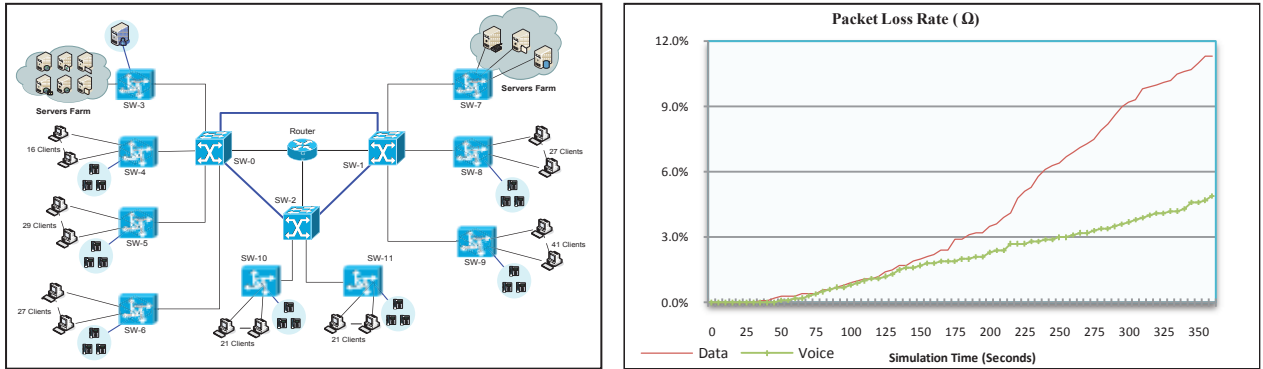


Fig. 3. (a) Proposed redesign network with VoIP equipments; (b) Packet Loss Rate,

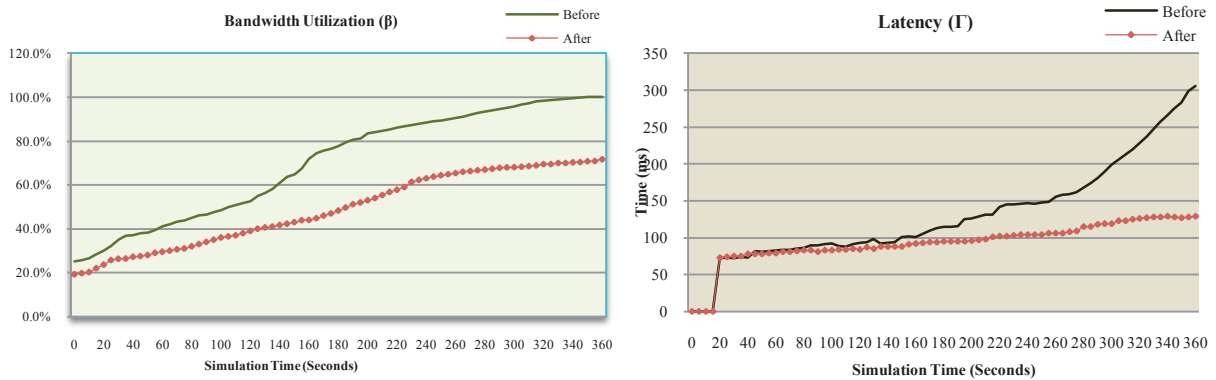


Fig. 4. (a) Bandwidth Utilization before and after redesign existing network; (b) Latency before and after redesign existing network,

In Fig.4(a), we noticed that β increasing patterns of proposed network almost identical to those found in existing network with enhancement ranged from 5.9% to 29.5%. The cause of this is identified in fact that the traffic generation model was not changed but a new bandwidth capacity was added in the proposed network. Simulation results for Γ are astonishing; Γ of voice traffic was under performance threshold for the whole simulation time, with a maximum of 129 milliseconds compared to 306 milliseconds recorded in existing network, as shown in Fig.4(b). The cause of this significant result is justified by the creation of VoIP VLAN and applying load balancing and redundancy to the existing network. Finally, by analyzing the second simulation results, we noticed that the last successful voice packet sent and received satisfied VoIP quality of service was at 255 simulated seconds. The voice traffic generation started at 20 simulated seconds. With a rate of 100 calls every 25 seconds, the number of voice call sustained by proposed network is 940 ($100 \times (255 - 20) / 25$), which is 240 voice calls more than existing network.

From the above simulations we can conclude that redesign the existing network has a great impact on existing network performance and number of calls supported.

6. Conclusions and Future Works

We have introduced a new methodology to assess the readiness of a network in-operation to support VoIP. Additionally, we proposed a new approach of network redesign modifications through NS-2 simulation results. The efficiency of this methodology has been tested over an existing network of 200 clients, 9 servers, 12 switches and a router. The simulation results prove improvements in bandwidth utilization, data loss rate, voice loss rate, latency and number of VoIP calls by 29.5%, 13.8%, 6.8%, 57.8% and 25.5% respectively after redesigning the existing enterprise network. In the future, more extensive complex simulations will be carried-out to model a systematic method to redesign network under growth.

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