SIMULATION STUDY ON DEPLOYMENT OF VoIP ON WIRELINE NETWORK

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ABSTRACT

Voice over Internet Protocol is a revolutionary technology which has become a key topic in both the growing Internet industry and established telecommunications industry. VoIP has become a potential alternative to the Public switching telecommunication network due to its reduced cost. However, despite its reduced cost it has so many challenges which affect its successful deployment. This paper presents a detailed simulation approach for deploying VoIP successfully using the OPNET network simulator. The approach and work presented in this paper predict, prior to the deployment of VoIP equipment the number of VoIP calls that can be sustained by an existing network while satisfying quality of service requirements of all network services and leaving adequate capacity for future growth. This is achieved by carrying out a number of simulations on a segmented network, which provided graphical results used in the study of VoIP traffic and other major devices. These results will benefit network planners in the successful implementation of VOIP services.

KEYWORDS: Network Management, VoIP, Simulation, OPNET, QoS.

INTRODUCTION

The long-hyped integration of voice and data on a single network is finally becoming reality. Major industry analysts are now discussing the explosion of VoIP deployments and corporate rollouts have commenced in order to take advantage of the operational efficiencies and competitive advantages that are facilitated through VoIP's advanced communication services. Almost universally, industry analysts talk about incomplete and delayed VoIP implementations caused by poor pre-deployment analysis, planning and lack of well integrated management tools that address both networked voice and data applications. The risks are not only the investments made in the VoIP equipment and upgraded infrastructure, but also in the potential impact on organizational productivity. The reputation of the IT organization can hinge on the success or failure of a VoIP implementation. The solution to mitigate these risks is the main approach taken in this paper by knowing the exact number of calls that is needed on a given network before deployment.

This research provides a simulation approach that involved the use of OPNET as a model to depict the real performance of a given network under VoIP and data load traffic. OPNET contains a vast amount of models of commercial available network elements and has various real-life network configuration capabilities. This makes the simulation of real-life network environment close to reality. Other features of Opnet include GUI (graphical user interface) and comprehensive library of network protocol and models.

VoIP DEPLOYMENT

Flowchart below gives an 8-step methodology for a successful VoIP deployment [19]. The first four steps can be performed in parallel. Before embarking on the simulation in step 7, step 5 must be carried out which requires upfront modification to the existing network. Both step 6 and 7 can be done sequentially. The final step is the pilot deployment.

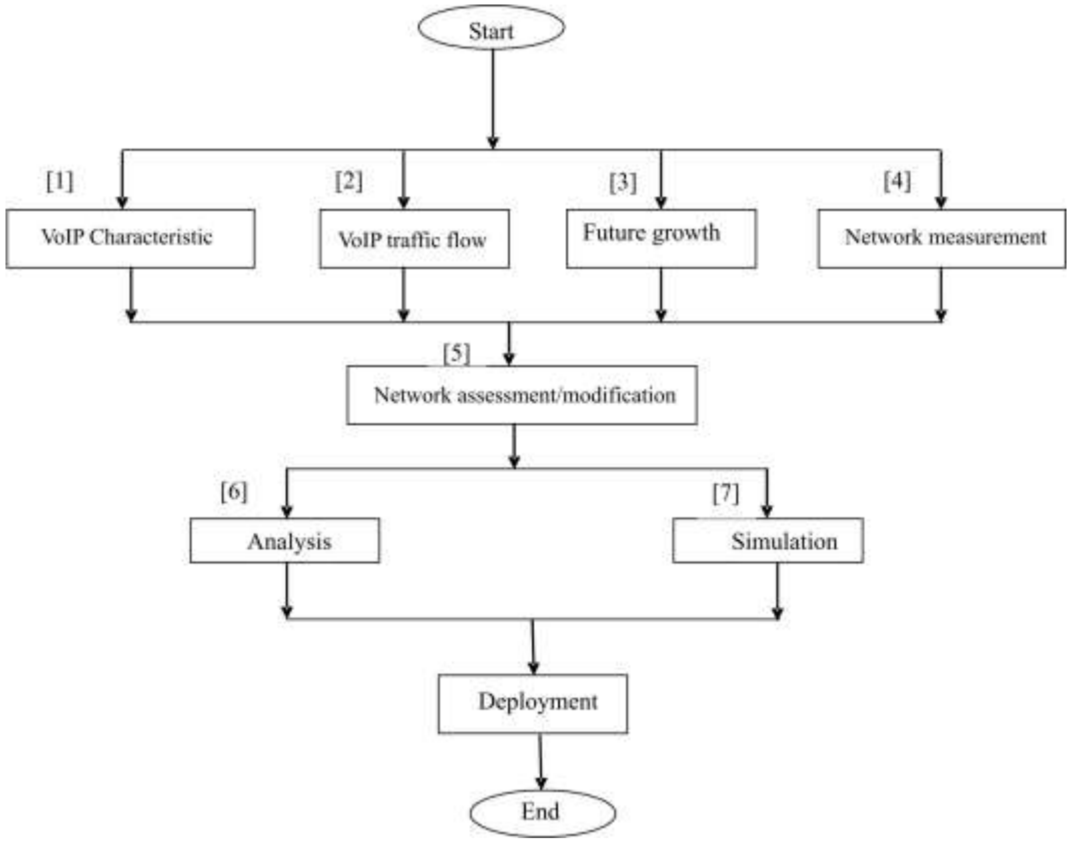

Figure 1: Flowchart Illustrating VoIP Deployment

VoIP TRAFFIC CHARACTERISTICS, REQUIREMENTS AND ASSUMPTIONS

When introducing a new network service such as VoIP, one has to characterize first the nature of its traffic, QoS requirements and any additional components or devices. For simplicity, in this paper an assumption of point-to-point conversation for all VoIP calls with no call conferencing is considered. For deploying VoIP, G.714 imposes a maximum total one-way packet delay of 150ms end-to-end for VoIP applications [3, 4, 5]. A delay of up to 200ms was considered to be acceptable. This delay can be broken down into at least three different contributing components, which are as follows: (i) encoding, compression and packetization delay at the sender (ii) propagation, transmission and queuing delay in the network and (iii) buffering, decompression, depacketization, decoding, and playback delay at the receiver [17].The required bandwidth for a single VoIP call in one direction is 64 kbps. G.711 codec samples 20ms of voice per packet. Therefore, 50 such packets are transmitted per second. Each packet contains 160 voice samples in order to give 8000 samples per second. Each packet is sent in one Ethernet frame. With every packet of size 160 bytes, headers of additional protocol layers are added. These headers include RTP + UDP + IP + Ethernet. In this paper, the G.711u codec is adopted as standard for the required delay and bandwidth. However to obtained best possible quality of service, it is possible to implement different ITU-T codecs that yield much less required bandwidth per call and relatively a bit higher, but acceptable end-to-end delay. Below are common ITU codecs' and their defaults.

Table 1: Common ITU-T codecs and their defaults

Codec	Data Rate (kbps)	Datagram size (ms)	A/D Conversion Delay (ms)	Combined Bandwidth(kbps)
G.711u	64.0	20	1.0	180.0
G.711a	64.0	20	1.0	180.0
G.729	8 .0	20	25.0	180.0
G.723.1(MPMLQ)	6.3	30	67.5	180.0
G.723.1(ACELP)	5.3	30	67.5	180.0

Throughout this paper, point-to-point voice calls is assumed and no voice conferencing is implemented. Also, any signaling traffic generated by the gatekeeper when a call is on-going is also not considered. This research and design work is based on the worst-case scenario for VoIP call traffic. The signaling traffic involving the gatekeeper is only generated prior to the establishment of the voice call and when the call is finished. This traffic is relatively limited and small compared to the actual voice call traffic. In general, the gatekeeper generates no signaling traffic throughout the duration of the VoIP call for an already established on-going call.

VoIP TRAFFIC

One way to model the VoIP traffic in OPNET is to use the predefined voice application. Basically, an application in OPNET is a collection of tasks of which each task is defined as a set of phases. Each phase takes place between two endpoints and has a configurable traffic behavior. The time on which each task starts and the duration that it takes can be configured when defining the application. Applications can be defined and configured using the Application Definition object while the Task Definition object is used to define and configure tasks. In the case of predefined applications, the underlying tasks are already defined but usually some flexibility is given to the user to configure their attributes.

GROWTH CAPACITY

The projected growth expected of the network has to be all taken into consideration to extrapolate the required growth capacity or the future growth factor. To achieve the desired output, 20% of the available network capacity is reserved for future growth and expansion. For simplicity, network resources of the router, switches, and switched-Ethernet links are evenly distributed. However, this percentage in practice can be variable for each network resource and may depend on the current utilization and the required growth capacity. In this simulation approach, the reservation of this utilization of network resources is done upfront before deploying the new service and only the left-over capacity is used for investigating the network support of the new service to be deployed.

Network Measurements

Measurements were performed to determine the existing traffic load. This is a vital step as it can potentially affect results to be used in the simulation. There are a number of tools available commercially and non-commercially to perform network measurements. Popular open-source measurement tools include MRTG, STG, SNMPUtil and GetIF. A few examples of popular commercially measurement tools include HP Open View, Cisco Netflow, Lucent VitalSuite, Patrol DashBoard, Omegon NetAlly, Avaya ExamiNet and NetIQ Vivinet Assessor [3]. To obtain adequate assessment, network measurements were taken over a long period of time. Sometimes it is desirable to take measurements over several days or a week. One has to consider the worst-case scenario for network load or utilization in order to ensure good QoS at all times including peak hours. The peak hour is different from one network to another and it depends totally on the nature of business and the services provided by the network. Table below shows a summary of peak-hour utilization for traffic of links in both directions connected to the router and the two switches of the network topology of Figure 2.

Table2: Peak Rate Measurement

Link	Bit rate(Mbps)	Packet rate(pps)	Utilization
Router⇔ switch1	9.44	812	9.44%
Router ⇔ switch2	9.99	869	9.99%
Switch1 ⇔ floor1	3.05	283	6.1%
Switch1 ⇔ floor2	3.19	268	6.38%
Switch 1 ⇔ fileserver	1.89	153	1.89%
Switch1⇔Database	2.19	172	2.19%
Switch2 ⇔ Floor3	3.73	312	7.46%
Switch2 ⇔ Email server	2.12	191	2.12%
Switch2 ⇔ HTTP server	1.86	161	1.86%
Switch2 ⇔ firewall	2.11	180	2.11%
Switch 2 ⇔ Proxy	1.97	176	1.97%

NETWORK ASSESSMENT /MODIFICATION

In this step the existing network based on the existing traffic load and the requirements of the new service to be deployed is assessed. Immediate modifications to the network may include adding and placing new servers or devices, upgrading PCs, and redimensioning heavily utilized links. As a good upgrade rule, a topology change is kept to minimum and should not be made unless it is necessary and justifiable.

ANALYSIS

VoIP is bounded by two important metrics. First is the available bandwidth, second is the end-to-end delay [11].The actual number of VoIP calls that the network can sustain and support is bounded by these two metrics. Depending on the network under study, either the available bandwidth or delay can be the key dominant factor in determining the number of calls that can be supported.

Bandwidth Bottleneck Analysis

Bandwidth bottleneck analysis is an important step to identify the network element, whether it is a node or link that puts a limit on how many VoIP calls can be supported by the existing network. For any path that has N network nodes and link, the bottleneck network element is the node or link that has the minimum available bandwidth.

Delay Analysis

For the existing network, the maximum tolerable end-to-end delay for VoIP packet is 150ms. The maximum number of VoIP calls that can be sustained is bounded by this delay. It should be noted that the major goal is to determine the network capacity of the VoIP, i.e the maximum number of calls that existing network can support while maintaining VoIP QoS. This can be done by adding calls incrementally to the network while monitoring the threshold or bound for VoIP delay. When the end-to-end delay, including network delay becomes larger than 150ms, the maximum number of calls can then be known.

SIMULATION

The object of the simulation is to verify analysis results of supporting VoIP calls. OPNET Modeler contains a vast amount of models of commercially available network elements, and has various real-life network configuration capabilities. This makes the simulation of real-life network environment close to reality. Other features of OPNET include GUI interface, comprehensive library of network protocols and models, graphical results and statistics.

MODELING THE NETWORK

VoIP gateway is modeled as an Ethernet workstation. The enterprise servers are modeled as Ethernet servers. All network elements have been connected using a 100Base-T links. Figure 2 shows described the network topology.

SIMULATION PROCEDURE

The simulation model for the existing network under study is illustrated in Figure 2. The simulation model of the provided organization network is exact replica of the real network. However the specific device that is needed, i.e. the Cisco 3600 router and catalyst 2950 Ethernet switch were not available. Hence, generic router and Ethernet switch model is used to represent the router and the switches in our network. VoIP gateway is modeled as an Ethernet workstation since collecting statistics inside the enterprise network is the only interest. The enterprise servers are modeled as Ethernet servers. All network elements are connected using a 100Base-T links.

Floor LANs are modeled as subnets that enclose an Ethernet switch and three designated Ethernet workstations used to model the activities of the LAN users. One of these workstations generates the background traffic of the floor while the other two act as parties in VoIP sessions. For example, the Ethernet workstations for floor 1 are labeled as A1_C2, F1_C2 and A1_C3. A1_C1 is a source for sending VoIP calls. A1_C2 is a sink for receiving VoIP calls. A1_C3 is a sink and source of background traffic. It is also observed that the floor LANs does not represent exactly the floor multimedia PCs or IP phones. However, building a model with such exact floor network configurations will make simulation almost manual. This is because it requires for each time a new VoIP call (or a group of calls) is added to perform two tasks: first adding individual PCs with individual profiles and settings and then running the simulation. This has to be repeated manually and results have to be examined after each simulation run. The simulation approach is an automated one, as the simulation is configured to automatically keep generating three calls every three seconds. The approach and model has no impact on the performance of internal nodes and links inside the core network.

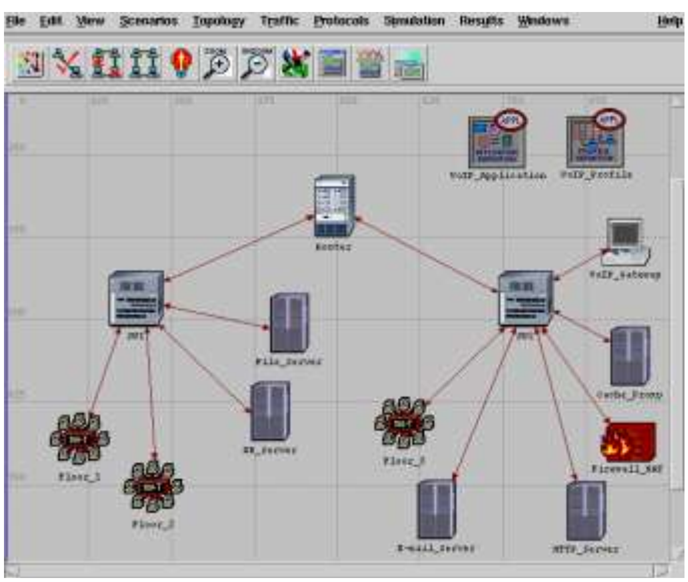

Figure 2: Network Topology of an Enterprise

PILOT DEPLOYMENT

Before embarking on changing any of the network equipment, it is always recommended to conduct a pilot deployment of VoIP in a test lab to ensure smooth upgrade and transition with minimum disruption of network services. A pilot deployment comes after training of IT staff. A pilot deployment is the place for the network engineers, support and maintenance team to get firsthand experience with VoIP systems and their behaviour. During the pilot deployment, the new VoIP devices and equipment are evaluated, configured, tuned, tested, managed and monitored.

SIMULATION RESULTS

Before the simulation started, OPNET was configured to obtain graphed results for numerous network components which include VoIP traffic, router, switches, and links. Graphed results for some of the most important components were generated. The OPNET simulation was configured for 8 minutes. The generation of background traffic by default in OPNET started at 40 seconds from the start time of the simulation run. The VoIP traffic started at 70 seconds at which a total of 3 VoIP bi-directional calls were initially added. Then, every 2 seconds 3 VoIP calls are added. The Simulation stops at 8 minutes in which a total of 615 calls got generated. Since the simulation stops at 8 minutes, the last 3 calls to be added were at 7 minutes and 58 seconds. Figure 3 shows the VoIP traffic and the corresponding end-to-end delay as VoIP calls are added every two seconds. Figure 3a shows the total VoIP traffic that was sent, received, and dropped. Figure 3b is a zoom-in version of Figure 3a focusing on the mismatch region between traffic sent and received. From Figure 3a, it is clear that the total VoIP traffic generated by the end of simulation run is very close 61,500 pps.

One can determine the total number of calls that the network can sustain by examining network bandwidth or delay bounds. Firstly, the bandwidth bound was investigated. Figure 3a and Figure 3b show clearly that not all of VoIP packets being sent get received. That is, there is a mismatch between traffic sent and received. Figure 3b captures clearly the addition of the three calls every 2 seconds and how this addition is repeated in gradual steps of 300 pps. The number of calls supported by the network can be determine by examining the X and Y axes. Examining the X axis of the simulation run time, it is clear that the last successful addition of three calls was at exactly 4 minutes and 48 seconds as seen clearly in Figure 4b. The next addition was at 4 minutes and 50 seconds which resulted in a mismatch. For the last successful addition of voice calls, which occurred at 4 minutes and 48 seconds, traffic volume was obtained (see Y axis) of exactly 33,000 pps or 330 VoIP calls.

Figure 4a shows the corresponding VoIP end-to-end delay. This delay does not exceed 80ms earlier. The delay stays less than 80ms until a simulation time of 4 minutes and 54 seconds at which the delay increases sharply. One can then find out the number of VoIP calls that the network can support to satisfy the 80ms time constraint. Therefore, one can conclude that, based on these simulation results, the number of voice calls to be supported by the network is bounded more by the network bandwidth than the delay. Hence, the number of the VoIP calls that the network can support based on simulation is 330 calls.

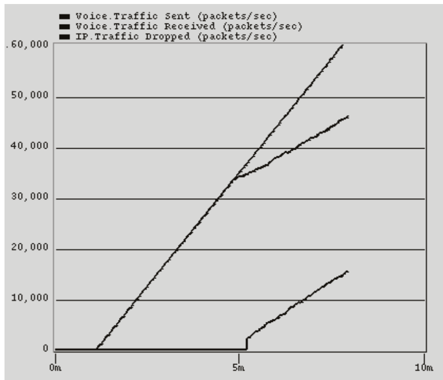

Figure 3a

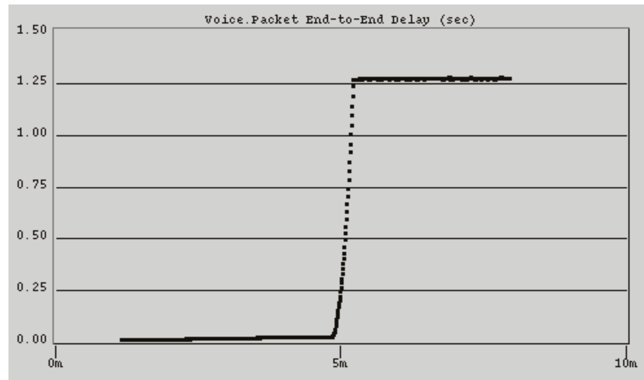

Figure 4a

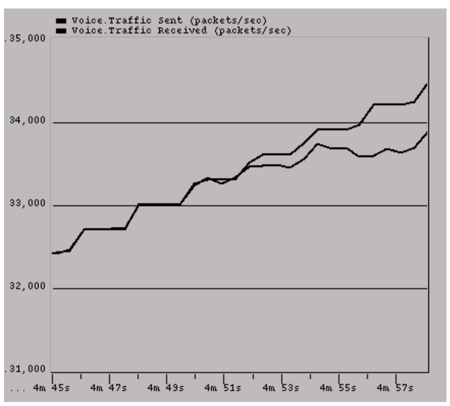

Figure 3b

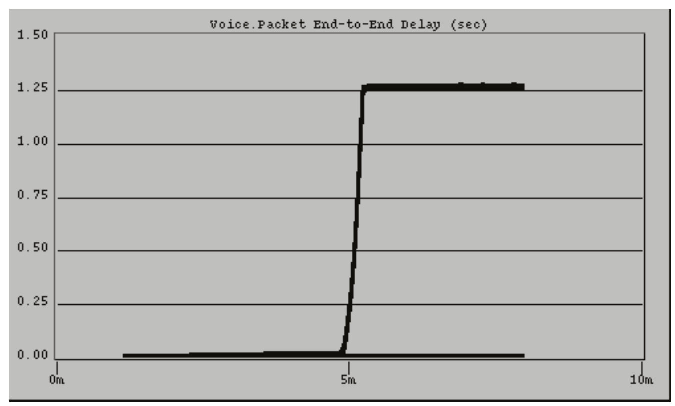

Figure 4b

FINAL SIMULATION RUN

Based on the simulation results, the existing network can support 330 VoIP calls while satisfying the VoIP QoS of throughput, latency and packet loss. Attention was focused on finding out the number of voice calls that the network can sustain. As a final check to ensure a healthy network and a normal behaviour for all network elements, a final simulation run was carried out in which 330 voice calls was added all at once at the start of the simulation, after 70 seconds. The simulation run execute for a prolong amount of time, for good 5 minutes at a steady state in other to examine the health of each network element. The simulation run showed a mismatch between traffic sent and received and a delay of more than 80ms. However, a successful simulation run of 306 voice calls showed normal and healthy results with no packet loss, end-to-end delay of 2.25ms and adequate utilization of router and switch CPUs and links. To account for the extra queuing delay produced by the edge links due to generating VoIP traffic from the same workstation, as opposed to separate PCs, a delay of 0.10ms has to be subtracted from 2.25ms. Therefore the effective end-to-end network delay would be 2.15ms.

Replacing Generic Router with Layer 3 Switch (DES-3326)

For best network practices and justification of results, the generic router was removed and replaced with a layer three switch (DES-3326) after reviewing the condition stated in step 5 of Section 2, a total of 313 calls was obtained as against 306 calls recorded earlier. This showed clearly that the router is a bottle neck on the network, since it has to undergo routing of inter-floor calls from floor1 to floor2. This result also can help network planners to make alternative use of network devices that would provide more calls capability and meet standard requirement of QoS. The figure below depicts graphically the voice/IP traffic after the replacement of the generic router with DES-3326 Ethernet switch. The layer 3 switch was used because of its high performance value in networking routing. However, a layer 3 device can support the same networking routing as network router do. Both inspect incoming packets and make dynamic routing decisions based on the source and destination addresses inside.

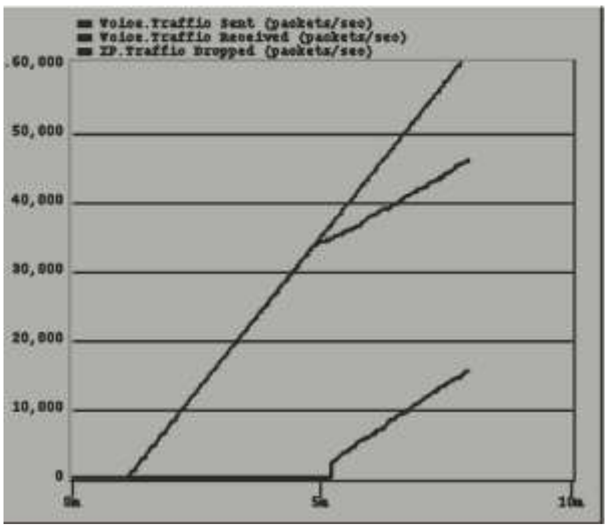

Figure 5: Graphical Result Using Layer three switch

IMPLICATIONS OF RESULTS

Based on the simulation results and observations of network under study, the following network decision can be justified:

- The existing network, with a reserved growth factor of 20%, can safely support up to 306 calls while meeting the VoIP QoS requirements and having no negative impact on the performance of existing network services or applications.
- A safety growth factor of 20% is maintained across all network resources.
- The replacements of layer 3 switch after removing the router shows high discrepancy and require thorough engineering analysis before deciding on which layer three devices is best for any given network topology.
- The network capability to support VoIP is bounded more by the network throughput than the delay. This is due to the fact that the existing network under study is small and does not have a large number of intermediate nodes. The network delay bound can become dominant if it was a large-scale LAN or WAN.

CONCLUSION

This paper presented a step by step methodology on how VoIP can be deployed successfully. The methodology can help network planners to decide quickly and easily how well VoIP will perform on a network prior to deployment by making alternative use of network devices. Only peer–to-peer voice calls have been considered.

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