Deploying VoIP over a Small Enterprise Network

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Abstract

The possibility of voice communications traveling over the Internet, rather than the PSTN, first became a reality in February 1995 when Vocaltec, Inc. introduced its Internet Phone software. This technology has advanced rapidly. This paper is aimed at making suggestions on things to consider when deploying VoIP onto a small enterprise network. Though the suggestions to be presented are centered on small network, some of the issues are applicable to a bigger network. These suggestions are based on some theoretical analysis, which are supported with some experimental work. An experiment was conducted for about 24 hours in order to capture the network measurement of a small LAN. The results obtained from the measurement were used in analyzing the whole network based on queuing theory; this provides ground for making judicious modifications to the network in order to support the deployment of Voice over IP.

I. INTRODUCTION

The VoIP concept is simply the ability to send voice messages over IP-based data networks with a suitable quality of service (QoS) and superior cost/benefit. What this means is that, we can incorporate voice traffic in an IP datagram, this way we can use the same medium for exchanging both data and voice messages, which will definitely cut-down the cost of telephony.

Although IP is a network layer protocol for TCP/IP protocol suite, it is not directly used for voice transmission. To implement voice over IP, some modifications are required in TCP/IP. The TCP/IP protocol layers and VoIP protocol layers are shown in Figure 1.

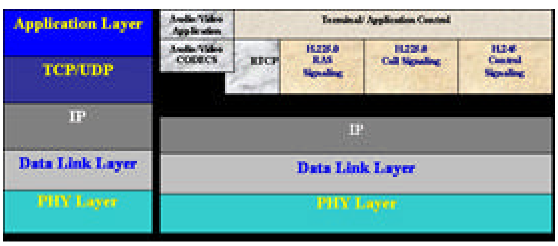

Figure 1: TCP/IP protocol layers and VoIP protocol layers

Like all other voice communications VoIP needs two types of protocols:

1. Protocol for sending the conversation data in the IP medium and
2. Protocol for the signaling

In order to send the conversation data in the IP medium RTP/RTCP (Real Time Protocol/Real Time Control Protocol) protocol is used over UDP. RTP is responsible for controlling the voice packet and voice quality. RTCP, on the other hand, is used for exchanging messages between session users regarding the quality of session like lost RTP packets, delay etc. Signaling protocol is needed for call setup, monitoring call progress and call release. The protocols available for this purpose are:

1. IETF (Internet Eng. Task Force): SIP and S/MGCP
2. ITU-T(International Telecom. Union): H.323
3. MEGACO/H.248 has developed jointly by IETF and ITU

II. DESCRIPTION OF THE NETWORK SETUP

In this section we discuss the setup of the network used for conducting the experiment. A small enterprise network consisting of a router, two switches; a couple of servers and three switched LAN of workstations and IP telephones (distributed over three different floors of a building) were used for the experiment. The first two floors are connected to the backbone router via a 3COM 3300 superstack switch, which is serving as an interface between the router and two other servers; File Server and Database Server.

The third floor is connected to the second switch, which has the same specification as the first switch; this switch is also connected to the backbone router. Other devices connected to this switch include Mail Server, Company Web Server, Web Cache Proxy and the link to the Internet, with a Firewall on this link, serving as shield from intruders on the network. In order to isolate broadcast and multicast traffic, VLANs are employed. All VLANs are port based. There are 3 VLANs on switch 1. VLAN1 includes port P1, P2 and P12.

VLAN2 includes P3 and P12. VLAN3 includes P4 and P12. Switch 2 has two VLANs. VLAN1 includes P7 and P11. VLAN2 includes P5, P6, P23, P24 and P11. Fig 2 shows the setup of the network used.

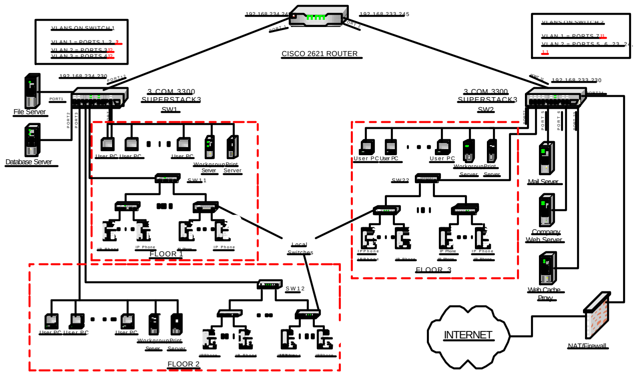

Figure 2: Logical diagram of the existing network

III. REQUIREMENTS AND ASSUMPTION

The most important requirement is to ensure QoS. In order to achieve this, the maximum acceptable end to end delay in packet delivery for optimal voice quality is 150ms (according to ITU G. 114 recommendation) [3].

Assumptions

The following are some of the assumptions we made in connection to the experiment.

a) All the voice session will be point to point. In other words, we considered only unicast traffic.
b) G. 711 CODEC is used for voice coding and decoding. It is chosen as it supports [3][4]:
   i) high speed and high bandwidth (64kbps)
   ii) the best voice quality, since it does no compression, introduces the least delay
   iii) less sensitive than other CODECs to packet loss.
c) Echo cancellers are built-in to CODECs.
d) There is little or no packet loss in the network.
e) The delay jitter and other factors for delay are considered in the Delay Analysis.

IV. NETWORK MEASUREMENT

Two tools were used for taking the traffic measurements of the network for a period of 24 hours, the behavior of the traffic is captured after every 10 minutes. The tools used are: Getif and SNMP Traffic Grapher (STG). In this section, we give a summary of our observations.

- Peak hour: 11.10am (29^{th} March) to 12:10 pm (29^{th} March) to 12.10 pm approx. (One hour)
- Over utilized interfaces: the port1&2 of router, Port 12 of switch 1 and port 11 of switch2
- Under utilized interfaces: All other ports have very less utilization

V. NETWORK DELAYS

End-to-end delay consists of the delay incurred by the voice signal from the instant it is produced by the speaker until the listener at the destination hears it. We must identify delays since there is a constraint of 150ms of end-to-end delay for optimal voice quality. The end-to-end delay is made up of the following components [14]:

1. Propagation Delay, T_{prop}
2. Queuing Delay, T_q
3. Fixed Component Delay, T_{fix}
4. Serialization Delay, T_{ser}
5. CODEC Delay, T_{CODEC}

Among these delays, only queuing delay is a variable, others are almost fixed for any network. We can equate total delay as

$$T_{total}= T_{tq}+ T_{prop}+T_{CODEC}+ T_{fix} + \mathbf{T_{ser}} = T_{tq}+ \text{Fixed delays} \quad (1)$$

The following assumptions were made in the delay calculation

1. Serialization delay was ignored, we only considered the other delays (i.e. propagation delay, fixed component, packetization and jitter buffer), the sum of these delays is equal to 85ms. Therefore, for maintaining the QoS, our tolerable variable (queuing) delay is 65 ms.
2. We considered 80% utilization of the devices and extra 20% of the capacity is left for the future growth of the network. According to this assumption service rate of the router, switches and interfaces will be 20kbps, 1.04Mpps and 80Mbps respectively.
3. All the queues (of the interface, switches and router) were considered as M/M/1 queues assuming packets arrive according to a Poisson process, buffer sizes are infinite and the different queues are independent. M/M/1 is preferred since it gives worst case i.e. an analysis based on this assumption gives conservative results. This is nice because tables are available for the M/M/1 case and values can be looked up quickly.

VI. TRAFFIC TRAVERSAL PATHS

There are three different scenarios of transmitting voice from one part of the network to another: (i) intra floor calls (i.e. calls within a floor), (ii) calls from floor 1 to floor 2 or vice versa and (iii) calls from floor 1 or 2 to floor3 and vice versa. In order to calculate the delay, the worst scenario was considered where by a user generates voice traffic from floor 1 to floor 2 or vice versa. As we can see from the logical diagram of the network in Figure 2, for a traffic to traverse from floor 1 to floor 2, it will have to pass through the router because floor1 and floor2 are on two different VLANs. This makes the interface between the router and switch 1 to be utilized twice for the same traffic. But before the queuing analysis let's start by succinctly discussing all the possible scenarios for traffic traversal in the network.

1. Intra-floor traversal (Floor1 to Floor1 or Floor2 to Floor2 or Floor3 to Floor3): this scenario is a situation whereby voice traffic is generated within a floor and the traffic does not travel outside the limits of that particular floor.
2. Inter-floor traversal (i.e Floor1 to Floor3/Floor2 to Floor3/Floor1 to Floor2): this scenario is a situation whereby traffic traverse from one floor to another, in this case it goes beyond the boundaries of the floor where the traffic is generated. Based on the setup of the current network, see Figure 2, traffic can travel from one floor to another in two ways. The first way is from floor 1 to floor 2 (or vice vera) where only switch 1 and the router are involved in the process. While the other way is from floor 1 or floor 2 to floor 3 (or vice versa), in this case both two switches in addition to the router are involved in the process.

One factor that could affect the number of sessions obtainable for the voice service is the way calls are distributed across the floors of the network. For the modeling, the calls were distributed in the following way:

(F1-F1): (F2-F2): (F3-F3): (F1-F2): (F1-F3 or F2-F3) = 4: 4: 4: 2: 1

This means that, for the delay calculation seven calls were initially considered: 4 of the calls within a floor, 2 between floor 1 and 2 and only one call between either floor 1 or floor 2 and floor 3. This way seven calls were added for each delay calculation to see how it affects the traffic. Call distribution in percentage is shown in Table 1.

Table 1: Calls Distribution

	F1	F2	F3
F1	26.66	13.33	6.66
F2		26.66	
F3			26.66

VII. QUEUING ANALYSIS

Queues play significant role in determining traffic delays in a network. Hence queuing analysis is very important in determining the network performance. In this section an analytical model of the network expressed as a set of equations is presented.

The analytical model for the worst case scenario is first of all developed then it used for generalizing a model for the whole network. In section VI the traffic traversal paths are discussed. The worst case scenario is the situation whereby traffic is generated from floor1 to floor2 or vice versa. To establish a call between floor1 and floor2 traffic will pass through switch11, switch1, router, switch1 and swich12. The link between the switches and router are all full duplex. When a link goes down, traffic may follow a different path because of automatic routing. Calculation will only be made for the usual case.

For simplicity all the queues are modeled as M/M/1 queues and equations are derived using the following basic formula of the M/M/1 system:

Delay in the system,

$$T = 1/(m - l) \qquad (2)$$

Where, µ = The service rate and ? = arrival rate

There is traffic from different interfaces in a single queue. And queuing theory allows mixing traffic with following formula:

$$l = \Sigma\, l_i \text{ , i=1 to n} \qquad (3)$$

Now considering a single call from floor1 to floor2 and background traffic between components we can derive equation (4) to (8) expressing delays in various components.

$$T_{sw11} = \frac{1}{m_{ps} - 2l_p} + \frac{1}{m_0 - l} \qquad (4)$$

$$T_{sw1} = \frac{1}{m_{ps} - (2l_p + \Sigma\, l_{sw1bgi})} + \frac{1}{m_0 - (l + \Sigma\, l_{sw1bgo})} \qquad (5)$$

$$T_{router} = \frac{1}{m_{pr} - (2l_p + \Sigma\, l_{rtbgi})} + \frac{1}{m_0 - (l + \Sigma\, l_{sw1bgo})} \qquad (6)$$

$$T_{sw1} = \frac{1}{m_{ps} - (2l_p + \Sigma\, l_{sw1bgi})} + \frac{1}{m_0 - l} \qquad (7)$$

$$T_{sw12} = \frac{1}{m_{ps} - 2l_p} + \frac{1}{m_0 - l} \qquad (8)$$

Where,

m_0 = Service rate of the interfaces in bps

m_{ps} = Service rate of the switch in pps

m_{pr} = Service rate of the router in pps

l = Arrival rate of voice traffic in bps

l_P = Arrival rate of voice traffic in pps

$\Sigma\, l_{sw1bgi}$ = incoming background traffic to switch1 in bps

$\Sigma\, l_{sw1bgo}$ = Total outgoing background traffic from switch1 to the router in bps

$\Sigma\, l_{rtbgi}$ = Total incoming background traffic to router in pps

$\Sigma\, l_{rtbgo}$ = Total outgoing background traffic from router to switch1 in pps

A simple program using C Programming Language was developed to solve the above equations in order to find the maximum number of sessions. In the program, sessions of calls (as assumed in call distribution) were added until one of the following two conditions was satisfied: (i) the delay becomes more than 65 ms (ii) the traffic coming in to the router exceeds the service rate of the router. The output of the program shows that the total number of sessions that can be supported by the network is given below:

Intra floor = 244
Floor 1 to Floor 2 = 122
Floor 1 to Floor 3 = 61

The interesting point here is that the delay of the voice traffic from floor 1 to floor 2 is 8.7 ms. This proves that the bottleneck of the network is the router as it has an effective service rate of only 20 Kpps (Kilo packets per seconds), whereas the links and switches in the network are good enough to support excellent quality of voice over the network. From Figure 3, it can be observed that the delay shoots out after 122 sessions. This implies that that the maximum number of simultaneous sessions that can be supported while maintaining high QoS is 122.

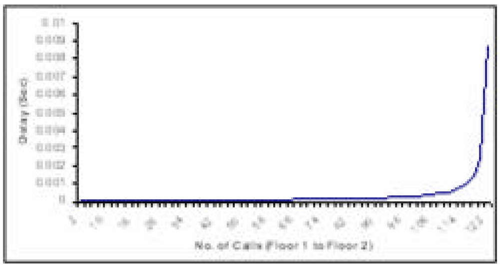

Figure 3: Graph showing increase in delay with an increase in the number of sessions

Increasing the number of sessions requires adding more IP telephone sets to the network. Whenever the ports of the switches at a particular level are filled, another level of local switches can be added to the model.

VIII. CONCLUSION

1. Queuing analysis is a good approach in determining the bottleneck of a network by considering the worst case scenario of the network traffic.
2. Most often, one of the nodes in a network may be a bottleneck; if such node is identified then replacing it may be the right decision to take if doing so is cost-effective.

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