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Deployment of VoIP Communications in B&A Spy Agency: Design and Implementation

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ABSTRACT

VoIP, Voice over Internet Protocol is one of the emerging and bleeding edge technologies that have in most cases sorted out the budget problems for the transmission and reception of voice communications. Indeed it is to use the Data Network infrastructure (Packet Switching Network) for the transmission and reception of voice calls. Though, people do prefer to use circuit switching network as it gives a dedicated path between end users that results in a crystal clear voice quality. Also the Digital communications “Modulation for the Transmitter end” and “Demodulation for the Receiving end” is involved (Digital Signal Processing to provide phenomenal voice quality) but no one can regret a fact that is; occasionally it becomes too expensive for the organization to provide dedicated paths to all employees. A promising solution is yes of course VoIP but the deployment of this technology introduces issues like Voice Quality, Jitter/ Delay, Quality of Service, Denial of Service etc. which is a challenge. In order to deploy VoIP communications in an existing Network it is very crucial and important to do proper planning and to analyze the network critically and justify whether it the Network is ready to cope with the VoIP services, as it can cause serious effects to the Network in case it is over burden. After doing that to assess and then evaluate by testing the network. This research is a case study based on B&A, A spy agency that has faced problems in the future regarding to call management and expansion. This research will cover the design and implementation of VoIP simulated in OPNET.

Keywords: *VoIP, DoS, VoIP Communications, Qos, B&A Spy Agency.*

1 AN OVERVIEW OF B&A SPY AGENCY

As proposed before that B&A is a spy agency. To provide not just the voice services but with reliability and security are the two main key aspects of the medium enterprise organization. Reliability means that whenever a user dials a number it don’t wait for ages also a satisfaction from the employees are required that they are getting clear voice quality without any problems like listening back to their own voices or delay etc. As B&A is a spy agency so the share of information should be secure.

Table 1: Users and the Applications Used

Building No.	Total no. of floors	Applications
1	3	http, email, database, printing
2	3	Same as above
3	3	Same as above
4	3	Same as above
5	4	Same as above
6	4	Same as above
7	4	Same as above including the CAD applications

As discussed before that this arbitrary organization was on the way to expansion and it has been expanded now. Presently there are a total number of seven buildings. A total number of users have increased to 1200 from 600 (Almost doubled). The table below is to highlight what is included now;

- Total no of users in each building Less than or equal to 50
- Building number 1 to 4 is physically located close to each other
- Building 5 and 6 is located close to each other
- Building 07 is on a separate location apart from the above building number 1 to 6
- Server room and core switch room are in building 07 along with a dedicated server for software applications
- All the buildings to have their respective access and distributed switches on their respective floor 1.
- The used topology is star
- The floors are connected to access switch by 10 Base T link. Access switches are connected to Distributed switches by 100Base T and then to distributed switches by 1000 base X. Servers are connected by 1000 Base X link

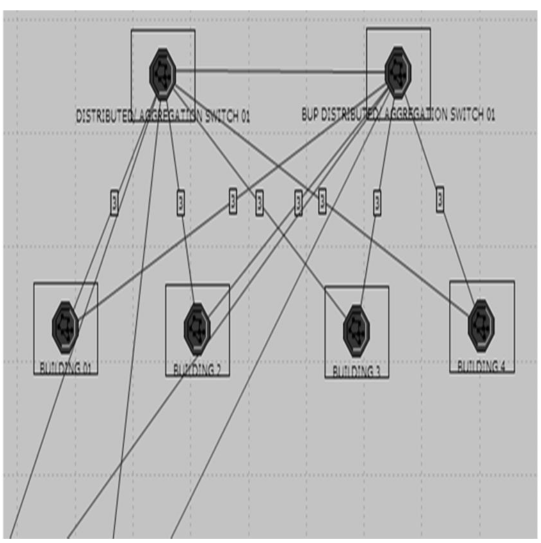

Fig. 1. Building 1 to Building 04 on the same geographic location

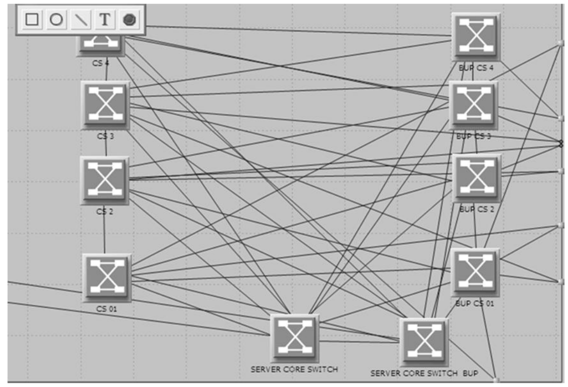

Fig. 2. Core switch room at Building 07

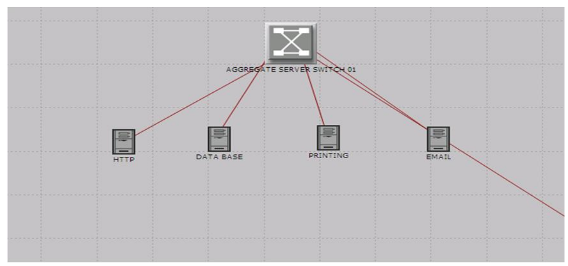

Fig. 3. Server room at Building 07

The users are almost doubled now causing difficulty in managing the PBX (A Local-Exchange). Though the infrastructure did provide the network users of that time with a separate cabling connection for the voice communications, but now it is a very difficult and hectic thing to re infrastructure [1].

As the employees, most of the time communicates each other by doing emailing so, an approximation to answer those questions are;

- Total number of simultaneously calls during the peak hours = (30 to 50)
- Duration of each call during the peak hours = (3 to 5 minutes)
- Total number of users with dedicated numbers = 750
- Total number of simultaneously calls during off peak hours = (30-50)
- Duration of each call during off peak hours = (15-20) minutes
- Requirement to call outside the organization (internationally) including peak and off peak hours= (10-20)
- Duration of each call outside the organization (internationally) including peak and off peak hours = (30-45) minutes

- No of users allowed to call internationally = 200

It was reported before that with the PBX it was difficult to maintain the infrastructure as the employees were facing a high waiting time during the peak hours and they were facing the blocking probability which was very high. The organization was very concerned as the optimized bill was out of the budget.

2 THE VOICE OVER IP COMMUNICATIONS: AN INTRODUCTION:

Voice over IP, VoIP is defined as to use the Internet Protocol to transmit and receive voice communications [8]. However, the main purpose of the internet protocol is to transmit and receive data over different applications like email, http, ftp, www, database, printing and many more. In order to implement VoIP communications on IP based network.

VoIP, Voice over Internet Protocol is one of the most promising and bleeding edge technologies in the field of modern Telecommunications and Information and Technology industry [1]. No one can regret that the deployment of VoIP reduces the cost as compared to the old fashion circuit switch telephony but the challenges are considerably significant and it also introduces too many risks. As compared to the trendy circuit switching network the packet switching, VoIP does not show immunity towards various security threats also there are various challenges about the Voice Quality, Security, Packet Loss, Jitter and so many.

The positive aspects of the deployment of VoIP are however; it is less expensive, flexible, scalable, reconfigurable, manageable etc. There are also so many question marks when deploying the VoIP Communications in an existing network.

In order to deploy VoIP communications the following questions have to be considered;

1. What is the meaning of Security and Reliability for B&A Spy Agency?
2. How can the network be assessed or what are the assessment tools and the parameters that need to be measure or calculated?
3. How the signalling has to be done? What is the VoIP architecture and is the Bandwidth sufficient to support Voice calls?
4. How to simulate and test the network and why? Which tool to use in this case OPNET but then the question whether or not OPNET will give me the desired results or not?
5. What is the required Quality of Service threshold? Provision of an assurance to meet it. Is Quality of service provision is sufficient to provide reliability and security or something else has to be done? Identification of security risks and the provision of an effective solution to avoid the security risks and failure.
6. How can be the modification of the Network done? What are the devices that the organization needs to consider for example a soft phone installed in the Computer with the connected handset or has to be a telephonic handset?

3 WHAT IS SECURITY AND RELIABILITY FOR B&A SPY AGENCY?

It is very critical to define a threshold and explain what actually reliability and security means for the organization. It is quite obvious that people do trust public switched telephonic network because of the dedicated switched network.

At the time of designing the network infrastructure the Engineers thought that circuit switching telephony is the best solution for this small enterprise organization and they gave dedicated paths to all users. As far as security for the same organization is concerned it is to provide a guaranteed telephony service between two end users without any problems technically speaking delay, jitter, noise, echo etc. Also the waiting time to establish the connection should be as little as it can. In reality the users demands as soon as he/ she dials up a number the connection gets established without any waiting time.

Apart from the security perspective; the Network Engineer perhaps didn’t realize at that time that the same enterprise will be demanding an expansion and then how they will manage and provide more connections. Of course there is just a single PBX for call routing and management and if every user wants to make a call how many dedicated paths they will provide.

A calculation was made for the previous telephonic network;

Availability of the medium = Failure Time / Total Time

Unavailability = 1- Availability

After assuming the calculations the availability calculated was 55% and 45% unavailability during the busy hours which was a bit concerned.

The reason why the users were facing the system unavailability was only the delay in the process as there was not much link capacity before for the PSTN. It was also causing delays as the dedicated circuitry had to be generated for the calling end parties.

As now the organization has been expanded and the geographical map does not allow to re infrastructure the cabling so it is a huge problem that has to be tackled. The prominent solution is yes the Ethernet packed based network can be used for voice communications. Deploying VoIP seems to be a good choice as the network provider has assured that the link and carrier will provide its best service but it is again opening different obstacles as it has to be made clear that how many errors the system will introduce, if adding soft-phones what will be the cost and error rate, what will be issues originated by compatibility, is the network capable of handling calls, how will it affect the band width and system capacity.

"Security" for B&A means that the data cannot be intercepted and securely communicated. The engineers at that time planned as because of circuit switching there will be dedicated path and the connection will be physical so there will be no security threats and the users will be fully satisfied.

Switching to VoIP communication there can be three different types of security issues;

3.1 Denial of Service

A sort of attack that keeps the user away to use the services. As the VoIP communications have to be deployed on a packet based network so if the scenario like this occurs will be a great disappointment. One can consider many examples when different sites based on internet get attacked. An example of Denial of service can be when a user is using the MAC address of the router and sending de authenticated messages to the network. In this scenario all the users will be unavailable to authenticate with the network. Probably it may occur in the wireless networks but as far as B&A spy agency is concerned none of the out siding users can use the network or login to it.

3.2 Service Stealing

This relates to if someone is making use of the services that he/ she is not paying for. As far as B&A Spy agency is concerned all employees have to follow the strict rules and regulations.

3.3 Confidentiality and Integrity

Eaves dropping, one of the examples when a fake user wants to listen. In order to stop or prevent proper encryption can be used to make the data preventable from being listened.

4 THE NETWORK ASSESSMENT TOOLS

As the voice communications has to be deployed on a network that was originally designed for the data communications and service like http, email, www. , data base and file printing so it is very critical and necessary to assess network whether it will or it will not support the VoIP deployment.

As far as Data Network is concerned the parameters are most likely bit error rate, round trip delay, CPU utilization time, Servers down time and load parameters, packet loss and delay etc. but for the VoIP communication the parameters are of course that the end users are able to listen and speak without any disturbance produced by factors like noise, jitter, delay and packet loss.

Wallingford (2005) discussed that the most important parameters to assess the network are [7];

1. Bandwidth
2. Packet Loss
3. Delay/ Jitter
4. Network Utilization

In order to calculate the above factors which are not the only once there are different tools that can be able to monitor and give results. In this very case the assessment tools that can be utilized are "Observer" and "Wireshark". Yes some online bandwidth calculation tools can also be availed if required. In order to simulate and test OPNET will be used.

According to Wireshark; "It is a free and open source packet analysing tool. Before, in 2006 it was known as ethereal. The plus point is it is free and the network can be assessed and monitored".

According to Observer "It is a complete solution not only for monitoring the network problems but also the provision of back time analyses is assured. Other prominent features are reporting, monitoring for the routes, alarms, application tools, trouble shooting and many more".

5 SIMULATION OF THE NETWORK USING OPNET

In order to simulate the network availability and reliability different statistics are simulated by using OPNET to come with a concrete solution. The reason why to use OPNET is because of the software packed with features. Voice over IP communications can be deployed. Because the Ethernet Packet based network has to be used for the voice over IP communications so it is of great interest that for both the data and voice the calculations has to be made. What if the voice is deployed on a data network and it is not providing the best service for data. Different statistics can be obtained to discuss the factors that cause severe problems in a network. One of the statistics is given below; rest of them will be discussed later in the report.

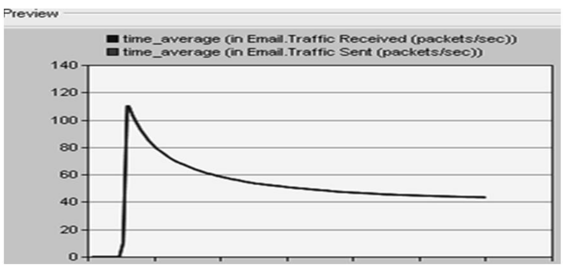

Fig. 4. Simulation result for the Email

Application Traffic sent and receive
Figure above shows the simulation result for the email traffic. After simulating the Network with the VoIP deployment the simulation justified that as much traffic was send was received with no delay or error. That reflects that the network is capable of handling traffic both data/ voice efficiently. What the simulation justifies is that the network is capable to handle load and there is not much link failure issues and downtime report.
It is also important to analyze the voice results because as a user we only can say about the voice quality, availability etc. but has no command what actually is running behind in the background as it is hidden from the user.

6 DELAY

If defined for the data communication, delay referred to the time for a successful communication that take place between the transmitter and the receiver. It is also referred to latency. If defined for the voice communication the concept remains the same the only different is that the time has been meant for the voice communication between the transmitter and receiver. Hens and caballero (2008) explained that because of the one way end to end delay increases in the voice communication/ telephonic conversation routing becomes more and more complex [4]. Hence it is very difficult to tackle with this characteristic.

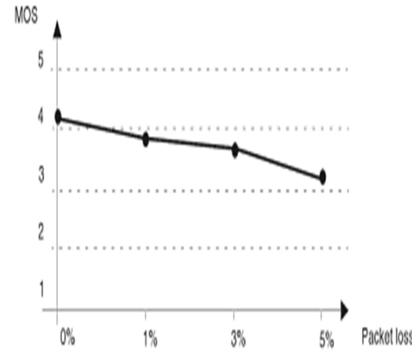

Fig. 5. Performance Analysis for the G. 711 PCM quality by Hens and caballero (2008)[4]

Another challenge is that as compared to the old fashion circuit switched telephone network the delay and latency is significantly high and there are a number of reasons including no dedicated circuit and increase number of equipment and communication.

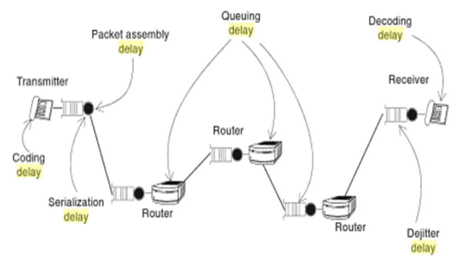

Fig. 6. Different equipment that causes delay in VoIP by Hens and Caballero (2008) [4]

Different types of delays explained by the same authors in the same literature explained above are Coding/ Decoding Delay, Packet Assembly Delay, Serialization Delay, Queuing Delay, De-jittering Delay,

As the codec used for this deployment is G.711, it is already explained that it generates 8 bits after 125 microseconds. As compared to other coding schemes the PACKETIZATION Delay for G.711 is less and the only reason is the effective and efficient compression takes place is less time.

After simulating the VoIP deployment in OPNET the following graph was obtained;

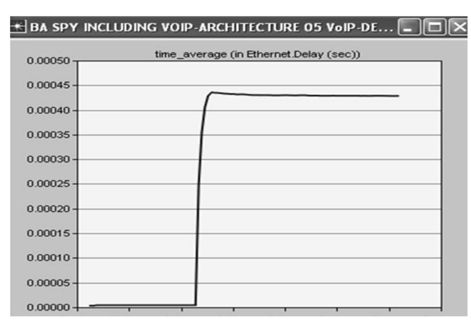

Fig. 7. Delay for the overall network simulation graph

Ethernet delay after the deployment shows that the total delay calculated is less than 0.45ms that is considerably acceptable as compare to 150ms which is too much. The most important statistic to obtain here is MOS.

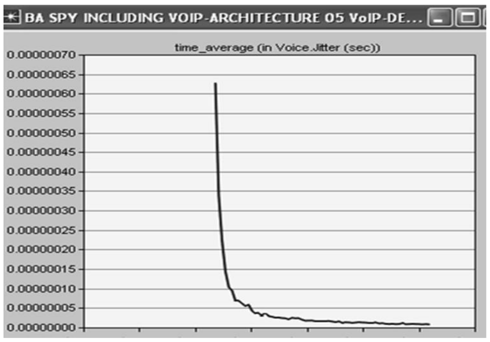

Fig. 8. Jitter value simulation graph for VoIP communications

After simulating the desired network evaluation for the VoIP configurations, the most promising result obtained was of course jitter. It seems that the jitter is signficantly low in the beginning and it almost reaches to a threshold of 0 which is ideal. Of course it is not true as for the voice communications using the network it is true that one has to face jitter values but might be because of the provision of relative nodes for the backbone and efficient transmission media the results are very satisfactory. Also real time traffic will definitely introuduce jitter but the positive thing is that the network is capable to introduce a less jitter call quality.

7 MOS

One of the most serious issues in any data network is that it uses UDP, User Data Gram Protocol which is a connection less oriented protocol. Though Voice over IP, VoIP communications uses RTP, Real time transmission protocol but it is connection less. It assures its best effort but there is no guarantee as compared to TCP, Transmission control protocol that provides reliability and security in terms of effectively handling the data. In case if the data loss occurs that will have an adverse effect on the voice communications. If an email is considered it would not matter if it reaches with a little delay still understandable but in case of voice if a packet is loss means distorted voice quality.

In order to measure data loss it has to be calculated when two parties are busy in voice communication. For that MOS score is calculated.

MOS is an abbreviation to Mean Opinion Score. To calculate MOS, Hersent, Petit and Gurle (2005) explained that for the telephone coders that do operate between the frequency ranges of 4kbp to 32kbps, the most commonly used methodology is ACR, Absolute category rating [5]. In order to achieve MOS value this test is being done. It is also explained in the same literature that it is not the only way to assess the quality of voice. Other methods are DCR, Degradation category rating and Comparison category rating, CCR. Wallingford (2005) explained that it is not entirely necessary to determine the MOS value but it is a positive aspect to consider if the number of users is satisfactory [7].

The reason why MOS calculation is used in this project is because the project is simulated in OPNET and it can produce MOS graphs. In order to justify the speech quality is according to standards it is a good way to compare and justify.

The authors in the same literature also explained that this calculation is based on the call quality between two effective call parties. The rating is made on the basis of call qualities. Though the value is absolute but there has to be some sort of reference nodes available in order to justify the MOS value/ score. To do that, some sort of radio references are inserted. Also Raake (2006) explained and defined MOS, Mean Opinion Score. It is indeed a standard specified by ITU-T.

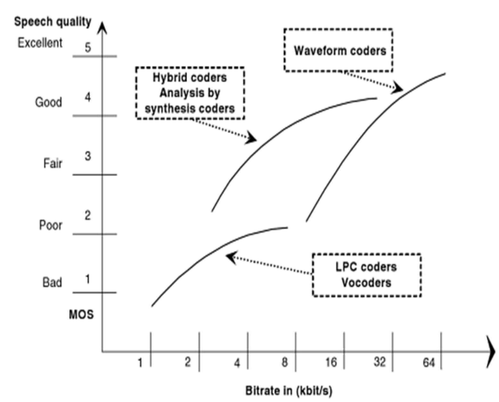

Fig. 9. MOS, as a function of BR, Bit rate and the coder used, by Hersent, Petit and Gurle (2005) [5]

Table 2: ITU Coders MOS recommended values [2]

Standard	G.711	G.726 or G.721	G.728	G.729	G.723.1
Date of approbation	1972	1990 (1984)	1992	1995	1995
Bitrate (kbit/s)	64	16/24/32/40	16	8	6.3–5.3
Type of coder	Waveform: PCM	Waveform: ADPCM	ABS: LD-CELP	ABS: CS-ACELP	ABS: MP-MLQ, CS-ACELP
Speech quality	4.2	2/3.2/4/4.2	4.0	4.0	3.9/3.7

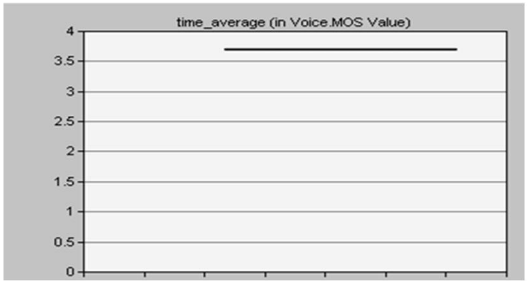

Fig. 10. MOS value for the VoIP deployment [2]

Figure shows that the MOS value after simulation is approximately 3.7 which are satisfied according to the literature.

8 HOW MORE SERVICES CAN BE PROVIDED TO THE NETWORK: A CONCLUSION?

Though discussed earlier about the QoS, Quality of service provision but the question here is whether assurance to provide quality of service will guarantee the best voice communication experience with reliability and security? The answer is might be yes but in most cases no. we have to think beyond. On a packet based network the main targets are;

1. Monitoring the Network (Identification)
2. Controlling (Precautionary measurements)
3. Management (Trouble shooting and allocation of resources)

By doing so one will be able to minimize the packet loss and delay but it cannot be reached to zero. So still quality of service I must say that do not provide a complete solution. Something has to be done beyond the expectations. As far as Ethernet is concerned most of the time the dominant quality of service used is class of service represented by CoS. The argument to justify is whether prioritizing is sufficient. Durkin and James (2003) explained that although MPLS is a prominent solution as it assigns a virtual label of 32bit long header and the promising features are segmentation, prioritization and expedition of the traffic yet again so many other measurements have to be done like [3];

The far most challenge in packet based network is to minimize the packet loss but in order to utilize the bandwidth effectively and efficiently one has to use the selection of appropriate codec and in this case it is G.711. Now in order to get higher value of MOS one has to use coding that introduce low bit error rate but again one has to tolerate the packet loss in order to utilize the bandwidth.

Another challenge is the number of users is not finite. Sometimes the network will be congested but sometimes a very few users will be accessing the network. Again a challenge!
There can be different impairments present in the network. Generalizing a network it comprises of:

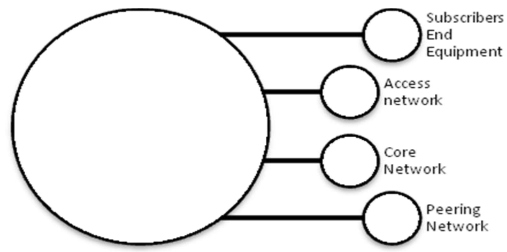

Fig. 11. Layered approach for the Network

The subscribers do use different types of devices so the problems that can be caused from the subscriber side using the equipment can be;

1. An introduction to the acoustic echo because of the improper isolation of the headphones and speakers could destruct the voice.
2. Different end users do use different equipment. Might be one is using the VoIP phone set or software to access a subscriber who is connected to the PSTN so different audio levels and different CODECS selection for two different users on a same call.
3. As the coaxial cable is being used so to convert two to four wires might be a mismatch in impedance, that will result in an echo termed as hybrid.

Different techniques can be used like voice signal processing where different CODECS can be tuned according to the requirements [7][8]. Also there are some echo cancellation devices and adoptive noise filters that can stop the echo problems. One can also use matched filters and calculation on SNR and BER as different modulation schemes bring their own BER. Overall it seems a promising solution to provide B&A Spy Agency with the VoIP technology.

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