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The Evaluation of Voice-over Internet Protocol (VoIP) by means of Trixbox

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Abstract

In this research paper, we evaluate Voice-over Internet Protocol (VoIP) technology, implement and integrate VoIP with IP Private Branch Exchange (IP PBX), track the behavior of data and voice quality using different voice codec and provide the appropriate solutions to improve the quality and reduce the cost of the customer's calls. The main issue is to identify the problematic areas when network is under attack. We introduce VoIP BOX which optimize the bandwidth, improve the reliability, scalability and security of customer's calls. We also examine the attacks on VoIP PBX network and provide the appropriate solutions how to stop or minimize the internal and external attacks launched by Attacker. The introduced VoIP BOX which is integrated with VoIP PBX, achieves quality of service on low bandwidth, almost zero jitter and high security.

Keywords: VoIP; Trixbox, PBX; QoS; DoS; SME

INTRODUCTION

VoIP is the latest technology used to transmit conversations digitally over the Internet. In other words, VoIP uses the internet protocol (IP) as a carrier to transfer the voice over the network [1]. VoIP is rapidly growing in the communication industry and globally changing the life style of businesses and residential consumers. Dropping price and hundreds of versatile additional features also attracts the public and private telephone consumers towards VoIP. Cost is one of the driving factors for switching towards the VoIP. The implementation of PBX with VoIP is called IP PBXs or VoIP PBX, which can be Hosted PBX (hosted with the provider) or Non-Hosted PBX (hosted at Data Center or 2 customer's premises). It started with the idea of trying to use IP to transport PBX traffic. According to Infonetics Latest Research Reports Show the VoIP Market is Rapidly Shifting toward Small & Medium Businesses Business: VoIP Service is growing rapidly because of its cost savings, efficiency and benefits. Frost & Sullivan reports Hosted Business IP Telephony is growing at over 28% annually in North America and rest of the world. In fact, business VoIP services will make up almost one third of all VoIP service revenue by 2010 [2].

VoIP was well initiated in the marketplace by local and long distance carriers in 2004. The Private Brach Exchange (PBX) was established in 1990 and was amalgamated with VoIP at the end of 2004 and now is known as VoIP PBX or IP PBX. IP PBX serves to small, medium and large enterprises to transmit their voice and video over the data network and make well-matched with Public Switched Telephone Network (PSTN). Now a days, most of the institutions and companies are going towards VoIP and IP PBX to lesser their cost over 50% with high trustworthiness, scalability and protection.

Since the last decade the Small and Medium sized Enterprises (SME) are taking up modern technologies including VoIP. VoIP is the technology of this era that permits the different services to travel on a single channel or network, also known as Quad-play technology i.e. video streaming, voice, data and surveillance cameras. The basic idea of VoIP is that it provides the transportation of voice through Internet protocol but in actual it accomplish the level of triple play, It means that the voice , video and data can travel over a single wire e.g. PTCL in Pakistan, Virgin Media in UK. Skype is a marvelous invention of VoIP which plays a vital role to make the technology compatible with additional features such as video, voice, live chat, audio and video call conferencing and call forwarding [3]. There are many benefits of VoIP technology with versatile features for data and voice consumers but still many companies are using old method of telecommunication system, is providing by Public Switched Telephone Network (PSTN) providers [4]. In other words, VoIP system gives opportunity to its users to consign phone calls that translate signal from analog to digital and sustain full duplex conversation. A Private Branch Exchange (PBX) is the advanced telephone system which supports to the small, medium and large organizations [5]. It provides internal dial tone to the private organizations, as opposed to the telephone company which supports many businesses or the general public. It provides the connectivity between the internal and external lines with the outside network lines. The latest advancement in the VoIP PBX is to use Internet Protocol to carry calls along with data and video. All modern PBXs support VoIP [6]. The Voice over IP (VoIP) supports its domestic and corporate users to operate the available broadband link for data and voice communications. VoIP technology is not dependent on the particular broadband provider and geographical position, patrons can be anywhere of the world [28-30].

Even though VoIP technology has various advantages but VoIP introduced different level of security threats such as VoIP Packet Injection, VLAN modification, VoIP social engineering, VoIP toll fraud, VoIP voice mail hacks, address spoofing, VoIP call eavesdropping, VoIP packet modification, VoIP phishing or Vishing, Denial of Services (DoS) and many more [7]. In 2008, VoIP networks received the highest level of threats from Denial of Services (DoS) and Distributed Denial of Services (DDoS) attacks. According to Weinschenk, VoIP technology can be secured by using some protocols [8]. To avoid the network from attackers, network administrator needs to implement proper encryption policies and firewalls but still there is some chance that the network will be affected. Therefore, VoIP and Network engineers regularly update the VoIP network with latest security tools to make sure that VoIP network is under control [4]. The rest of the research report is structured as follows: Section 2 is about Overview of VoIP and PSTN, Section 3 is the Methodology of the paper, Section 4 is about the Evaluation of VoIP and DoS and finally conclusion is drawn in section 5.

OVERVIEW of VOIP and PSTN

In this section we discusses the general idea of Public Switched Telephone Network (PSTN) and Voice over Internet Protocol (VoIP) technologies and a comparison of these two technologies is also drawn. Numerous aspects are considered for example, reasonably priced, portability, less maintenance requirement and free of charges national and international calls are being considered during the comparison of these two technologies.

PSTN Technology

PSTN has been developed since Alexander Graham Bell made the first voice transmission over wire in 1976 [4]. After the invention of Alexander Graham Bell, one-way calling system was established but afterward bi-directional or duplex communication was established, due to which both consumers were able to communicate with each other simultaneously. In this period in place of electronic switches telephone operator was working, the operator asked consumer/caller where they wanted to dial and then they manually link the two voice paths. Later on, the human switches were swapped by electronic switch known as modern PSTN. Day by day technology is changing and it requires multimedia discussion on regular basis that established Digital Subscriber Line (DSL) and cable providers. The most recent technology offers the variety to consumers to select the provider as per their requirements. These substitutes comprise packet based networks for voice and data integration with Least Cost Routing (LCR) and trunking throughout the busy hours.

The common channel signaling standard is Signaling System 7 (SS7) that supports PSTN by means of managing call establishment, billing, trade of information, routing, operations and also support of intelligent services of network. The SS7 protocol plays vital role for Voice over IP (VoIP) and it also inter works with PSTN.

VoIP Technology

The main theme of Voice over IP (VoIP) was pioneered by the Institute of Electrical and Electronic Engineers (IEEE) in 1974, but formally introduced to the market in 2004. At present, Voice over Internet Protocol is the component of telecommunication and data communication and is extensively accessible in the market and facilitating their users such as residential, small and medium sized businesses, institutions and large organizations. The consumers of VoIP uses broadband and local area network services as a carrier and transmit the digital signals that carries the packets of voice from one point to another. The transfer of voice packets should be in real time environment and quality of voice should not be affected by losses such as packet loss, delay, jitter and many more [9]. VoIP covers other technologies such as telephone, internet, fax machine, email, and the web, etc. VoIP supports call centers and prepaid/postpaid calling card service provider and many more [10].

VoIP protocol provides other communication media like text messaging and video conferencing. VoIP is used by mobile operators for their back-haul (base station-to-backbone) network connections. Wireless technology has a goal to migrate to the fourth generation (4G), such as LTE (e.g., Verizon) and WiMax (e.g., ClearWire/Sprint) is to transport Voice over IP, instead of using dedicated, circuit switched circuits. However, many Internet-based services happen to carry voice and video, Examples include chat in multiplayer online games, instant messaging with voice and video, voice-enabled, Web-based customer service, and so on.

In many large businesses/organizations, consumer telephone connects directly or indirectly with switch that can be installed on computer (software PBX) or can be available with built in hardware, and at times that can also be connected through the telephone company`s line known as Private Branch Exchange (PBX). The PBX is not operated by Telephone Company but by enterprise. On large scale manufactures of PBX are Nokia, Siemens, GTE, Cisco, Lucent, Avaya, Nortel, Trixbox, Lucent, etc. The problems between the traditional PBXs and private telephone extensions are approximately reduced due to VoIP technology. It is easier for central management to maintain and control VOIP service rather than traditional PBX.

As per the new In-Stat report, it took only three years (2007-2010) that VoIP penetration has become doubled in numbers. In 2013, it is expected to grow up to 79 percent penetration. At present around 33 percent of enterprises are using VoIP technology. VoIP adopters have a good understanding of the cost savings associated with VoIP, and have oriented their limited budgets to optimizing efficiency and savings by replacing legacy TDM voice solutions [14].

The growth of VoIP is demonstrated in Figure 1.

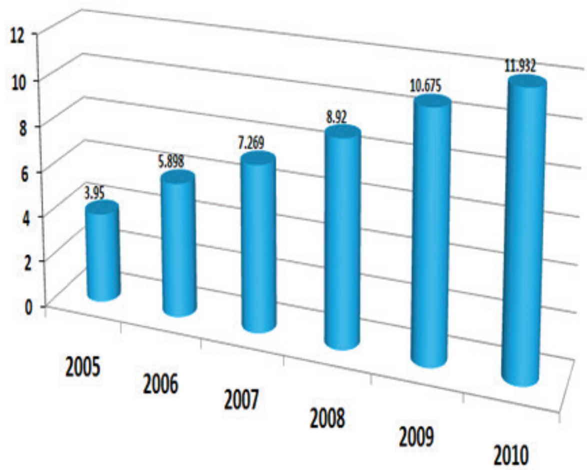

Figure 1. Growth of VoIP

The consumers need high-quality speed internet for example wireless router, DSL, ADSL, WiMax and IP Phone with the appropriate configuration to consign a telephone call. The process of calling and confirming that the call forwarded to proper recipient is handled by VoIP software. There are a number of ways to use the VoIP service, usually there are three ways which are being adopted by the VoIP consumers such as: Analog Telephone Adopters (ATA), IP devices, and Soft Phone to Soft Phone means PC to PC [11]. Analog Telephone Adopter (ATA): Analog Telephone Adaptor is the device to connect with the regular PSTN phone and ATA coverts analog signals into digital using TDM algorithm [12].

IP Devices: IP phone or soft phone (built in software) can be IP device that connects to Internet. The IP phone uses RJ-45 (network) connector instead of the regular RJ-11 connectors but is analogous to normal phone. RJ-45 can be connected with the hub, router, switch, etc. Most up-to-date IP phones are with built in wireless features that allow the consumer to download the soft phone and also put the calls by using Wi-Fi facility under the IEEE 802.11 protocol but the security is the main concern during the use of wireless internet from open access public hot spots. Computer-to-Computer Communication: The most easy and reasonably priced way is to operate the VoIP features from PC to PC by the use soft phone. Computer to computer conversation is totally free of cost by using VoIP technology.

Reliable and free of charge communication requires some peripheral devices such as speakers, microphone, sound card along with software and internet connections. The most common VoIP providers in the world are given below in Table 1.

Table 1. VoIP Providers

VoIP Providers (US)	Call Vantage, Vonage, VoIP.com, BroadVoice, Lingo, AT&T, Packet8, Via Talk etc. (Aboutcoip, 2010)
VoIP Providers (UK):	Choice VOIP, BMyWeb-Calls, Globe 7, Tpad, etc (MyVoipProvider, 2010)

Different devices required for the functioning of VoIP are shown in Figs. 2 and 3.

Figure 2. Soft Phone

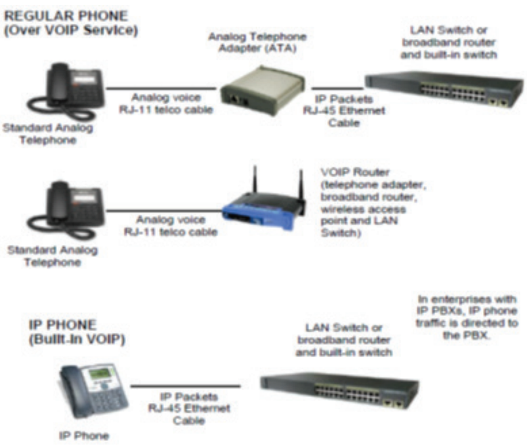

Figure 3. The Devices Used for VoIP

Comparision of VoIP and PSTN

There are many reasons to switch from PSTN service to VoIP. The most important reasons are given in Table 2:

Table 2. Comparison of VoIP and PSTN

Features	VoIP	PSTN
Economical	YES - VoIP service is extensively available in the market and also more economical rather than the PSTN phones. VoIP services saves more than 70% cost as compare to regular phone bills due to its additional enhanced feature.	NO – PSTN is broadly available but not as much economical as compare to VOIP.
	its additional enhanced feature.	
Additional Features	YES - The additional features of VoIP for the residential users are caller identification (CID), Call forwarding to another number and email address, conference calls and call hunting, etc. due to its distinct features difference is created between regular phone service and VoIP service. Hence, VoIP has preference over regular phone service.	NO – Mostly features are not available in the PSTN Services. Only Some are available in the PSTN but to utilize its additional services customer has to pay additional amount.
Free International (long distance) Calls	YES – This is the most attractive feature for the people who are staying abroad as well as franchise businesses because most of the VoIP service providers are providing free unlimited local and long distance calls, such as Lingo, Vonage, etc [13].	
NO – In PSTN these features are unavailable.		
Utilize of the Existing Services	YES –The basic requirement for the VOIP service is high speed internet i.e. broadband, DSL, etc, as when the users are connected through Wide Area Networks (WAN).	

Generally every single consumer at home is paying much money for using the internet even though when internet is not in use. Consequently, when VoIP is used through the existing internet service, money will be saved.	NO – in PSTN network there is not any perception of Internet or broadband. Hence, PSTN consumers are unable to utilize such features.	
Portability	YES - VoIP has made life easier and mobile. VoIP consumer can carry their IP Phone or device provided from their carriers and is able to work anyplace or anywhere not bound of geographical location. Only availability of internet service is required.	NO –For the PSTN consumer Portability service is not available.
Low Maintenance Requires and 24X7 Customer Support	YES - VoIP consumers do not have to create any agreement with any phone repair companies.24X7 cost free service is provided to residential users by VoIP. The disconnection of VoIP service is usually due to stoppage of internet connections or deletion of IP settings provided by the provider. Mostly consumers just require rebooting the IP phone or ATA adopters, it will bring back the link with the provider. As a result, it is not necessary to keep the contract with the regular phone companies.	NO – PSTN consumers make an agreement with their provider for fear that of line failure and customer service is not available 24X7.
Personal Number Portability	YES – VOIP service allows its user to keep existing number If any user has phone number and want to switch to VoIP service.	YES – PSTN users are able to keep their existing number at the same time as migrating to other carrier.
Automatic Billing	YES – Some of the VOIP providers offer post paid billing and some pre-paid billing procedure as well but both are offering automatic billing features. Consequently, it's not necessary for the user to go online or their physical locations to charge their account they just go for the automatic billing options and automatically bill will be direct debt from the account.	NO – Automatic billing feature is not available in PSTN Operators.

VoIP/IP PBX is a grand low priced service and it has leaded the users towards most up-to-date technology. For that reason, VoIP technology is the most advantageous for the existing and prospective consumers to cut their cost and make use of the marvelous features.

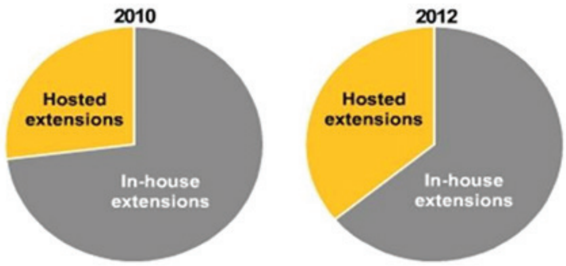

Figure 4. Comparison of Hosted Extension

MATERIAL and METHODS

The basic idea is to assess the IP PBX (Trixbox), execute the IP PBX and incorporate the VoIP with VoIP BOX and give the answer how to get better the number of calls and quality and ensure the behavior of voice and data traffic at low bandwidth, proposed solutions of how to get better quality of service and presentation at low bandwidth, converse about the intimidation and safety measure problems to VoIP technology with suggestions and recommend the improved thoughts and procedures to solve the problems. This research presents the advantage to individual users, small and medium sized enterprises, large companies, institutions and service providers to cut cost with the enhanced attributes over and above to make it more protected, consistent, flexible and full-bodied.

The procedures adopted in this research paper are: achieve a comprehensive understanding of different vendors of VoIP PBX and its working, figure out VoIP parameters and tools with Linux and Windows Operating Systems, amalgamate Voice over IP PBX with VoIP BOX, set up Modulation of Management (HUD-LITE) tools for security, organize the trunks for inbound and outbound call routes, arrange IVR (Interactive Voice Response) setup for incoming calls, configure and handle messages forward to the e-mail address, converse hazards, threats and attacks of VoIP technology face and recognize the countermeasure, explore several procedures and method that are well-suited to use as a security feature of IP PBX, talk about critical analysis of this research, authenticate that the new development will offer more safety and dependability in the other's people study and situation i.e. cost effective solution, scrutinize the evaluation survey report of PSTN and VoIP, that will properly clarify to save more than 50% cost with sophisticated, simple, safe and more well-organized telephone exchange using voice over IP (VoIP) particularly with low bandwidth.

This research is execution of the actual VoIP scenarios and gives the selection of procedures of VoIP and IP PBX technology along with their benefits and recognizes performance issues of VoIP. The evaluation and optimization of the voice and data traffic by using VoIP BOX are also discussed in this paper. We simulate Virtual Machine (VMware), Switches, VoIP BOX, IP phones and other technologies available in the topology.

Trixbox provides the practical execution of IP PBX functionality and incorporation with VoIP BOX, which includes Call Manager Express, PSTN and modern techniques of securities. Flooding attack is used to examine the security of VoIP and recommendation is provided to be secured from these types of attack. The Research report on VoIP/IP PBX and its performance is based on actual environment; therefore, the results and user guide manuals can be utilized as guideline for The IUB as well as any other small and medium sized organizations.

Evaluation of VoIP and DoS

The evaluation about the key factors of Voice over IP PBX and Denial of Service (DoS) are discussed in this section.

Evaluation of VoIP

At present, VoIP users are using normal IP PBX which is insignificant as in contrast to the bandwidth. The bandwidth issues are still faced by more than 50% countries and usually have usual architecture of VoIP. Hence, where ever the bandwidth is concerned VoIP Box plays a vital role. Figure 5 given below demonstrates the usage of call on different codec with and without VoIP Box.

Codec	G.729	G.711	G.722	GSM
Without VOIPBOX each call uses	42.4kps	106kps	63.6kps	42.4kps
With VOIPBOX each call uses	8.1kps	64.1kps	32.1kps	3.1kps
Calls per 256kps	28	3	7	17
Dramatically reduce jitter	+700%	+50%	+233%	+340%
Run voice and data on a single link without compromising data transfer rates				

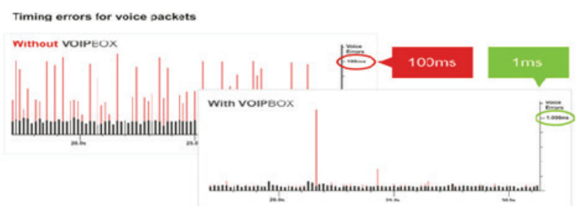

Figure 5. Comparison Chart and Graph with and without VoIP BOX

Numerous calls are progressed due to VoIP Box by utilizing similar bandwidth and without compressing of the audio stream and by reducing jitter. VoIP Box is also helpful to make use of accessible capacity of bandwidth and progress the capability around 70% to 100%. SIP helps to produce multiple calls with the help of VoIP BOX all together towards VoIP PBX, with and without VoIP BOX and VQ Manager monitor the quality of the calls, Jitter, delay and many more. Figure 6 shows the users call chart with multiple codec:

Codec	With VOIPBOX each call uses	Without VOIPBOX each call uses	No. of calls per 256kps of bandwidth with VOIPBOX
G.729	8.1kps	42.4kps	28 (+700%)
G.711	64.1kps	106kps	3 (+50%)
G.722	32.1kps	63.6kps	7 (+233)
GSM	13.1kps	42.4kps	17 (+340)

Figure 6. Call Support with / without VoIP BOX [21]

Scenario

The first foremost task of EGS is to modify the existing network by getting better quality of service which is provided to existing employees and make possible trouble-free communication among other EGS employees all around. EGS maintains flexible current network communication due to scalable network design. The second task of EGS for improved network is the accessibility to all employees within company`s intranet and also providing accessibility throughout different access levels to suppliers, external users and vendors. The availability of need is important for company user. EGS makes the network available on demand and also elimination of denial of service (DoS) to authorized users. Moreover, EGS needs improved telephone devices for the VoIP service, server consolidation and new security mechanisms must be implemented so as to reduce maintenance cost.

In addition, by modifying the network, recovery and backup should be implemented to avoid failure or downtime of any of the devices either critical or non-critical. All level of staff should be given proper training of new technology. The education plan should be designed for EGS to all staff regarding network. Additionally, "VOIP Box will be provided to the EGS which will increase the bandwidth utilization from 70% to 100%" [21].

EGS expects the network modification with VoIP to be completed within the next few weeks of deployment within a specific timeframe. As the company is taking numerous delivery orders from all around the world due to this it is very important. Hence, the budget of company proves that EGS saves around 50% cost with more enhanced features.

a) Analysis of EGS Existing Infrastructure & Proposed Solution

EGS IT team has provided information which is foundation of comparison report. The projections will clear the dissimilarity among the services and most important cost effective solution.

Table 3. Price Comparisons of PSTN & VoIP Technology Report

Existing Infrastructure	Cost / Features	Proposed Solution	Cost / Features
PSTN Business Incoming lines for each location (total 3 major locations)	£600.00 (Limited Use) / Month for only 3 incoming lines	VoIP Hosted PBX Exchange – Trixbox (one time cost) – support 30 license users (one time)	£4544.00 (£4065.00 one cost and £478.00 for one year cost for 30 license lines regardless of the locations)
Inter offices communications (average usages/ month)	£500.00 / month (only gives 1500 minutes to Asia and 1000 minutes to US/Canada and Europe Landline (As per BT Tariff)	VoIP Service provide Free Communication between Inter Offices regardless of the geographical locations	£0.00 / month (No Cost)
Customers and Order‟s inquires return calls charges	£500.00 / month – for around 2000 outgoing minutes (Per BT Tariff)	VoIP Provider call charges	£300.00 / month (Ref. Vonage Comm.)
Line Rental for PSTN Business lines	£350.00 / month – for UK and International Line Rent for 20 lines	VoIP Rental	£0.00 (No Cost) for 30 lines
Internet Rental for	£100.00 / month	Internet for VoIP Service	£100.00 / month
Installation Period	2 – 3 weeks	VoIP Installation Period	2 – 3 days
Backup Support	Additional Cost (depends on support time)	VoIP Backup Support	Free
PSTN Customer Service	0.60 P charge / min	VoIP Customer Service	Free (24X7)
Offered Services	£3.00 / line monthly for Caller ID, Voice Mail (additional cost for additional services)	VoIP Offered Services	£0.00 / month Caller ID, Voice Mail, IVR, Auto Attendant, Call Forwarding, Auto Redial, Voice Message Forward to Email as Attachment,
PSTN QoS	Good	VoIP QoS	Better than PSTN
Length of Contract	1 Year or 18 months	VoIP Contract	Free (No Contract)
PSTN Compatibility with VoIP	No Compatibility	VoIP Compatibility with PSTN	Yes (easy to integrate)
Upgrade	Additional Cost depends on the number of lines	VoIP Upgrade	Upgrade any time without contract
PSTN Soft Image	No Image Available	VoIP Trixbox Image	Free
Emergency Service Call	Better than VoIP	VoIP Emergency Call Service	Good but PSTN quality is better
PSTN Denial Attacks	Secure	VoIP Denial Attacks	More chance to be attacked then PSTN
PSTN Redundancy	Not Available	VoIP Redundancy	Available in case of failure
PSTN IVR	Not Available	VoIP IVR	Available and modify according to the requirements
PSTN Night mode Call Transfer	Not Available	VoIP Night Mode Call Transfer	Available with Reliability
Number Portability	Not Available	VoIP DID portability	Freely Available
PSTN Dedicated Number for each employee and vendor	Costly	VoIP Dedicated Numbers	Economical (Incoming DID are free)
Legal and Ethical	Available	VoIP Legal and	Available
PSTN Phone Cost / Hosting Cost	£15.00 / per handset	VoIP Phone Cost / Hosting Cost	Free - Soft Phone & User can also use PSTN phone as a IP Phone £100.00 per month hosting cost
PSTN Bandwidth Compressed Device	Not Available	VoIP Bandwidth Compressed Device	£200.00 / month for 30 lines (VOIP BOX Hardware)
PSTN Cost	£2050.00 / month (additional cost for additional features)	VoIP Cost	£4478.00 / year and £4544.00 / onetime cost

Table 7 clearly shows that EGC saves more than 70% Cost if move towards new and latest VoIP technology, which also provide additional features.

b) Trixbox (IP PBX) Order Information

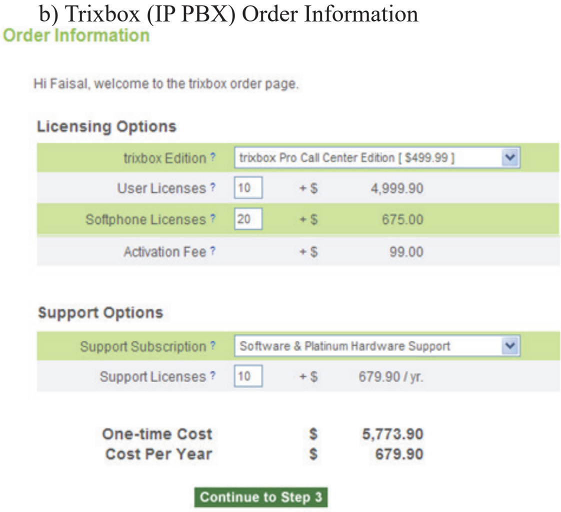

Figure 7. Price Information of Trixbox VoIP License

EVALUATION of DOS

The topology design given below is basically based on IP PBX image, virtual machines, built in PC connected to the Switch, Computers, IP Soft Phones, Laptops which are connected through routers and switches. The attacker launched the attack using TCP SYN technique and interrupts the voice communication and captures real time calls. The network is configured with class C private IP addresses with subnet mask of 28. The webpage can be access by entering the IP address of the IP PBX which is http://192.168.80.128/ in the Internet Explorer browser. In this case study, the tool VQManager is used to track the behavior of the traffic and QoS before and after the attack. During the attack attacker wants to intercept the IP Phone and exploit the other side of computer. In flooding attack, the first step is to analyze VoIP network packets and determine the IP address of the IP PBX, computer, soft phones and gateway etc. To conduct flooding attack the attacker is using VoIP TCP SYN. To capture real time traffic of legitimate user attacker use the command “nmap –sp 192.168.80.128/28”. After obtaining the important information, the attacker generates hundreds of messages in order to make the targeted system deadly.

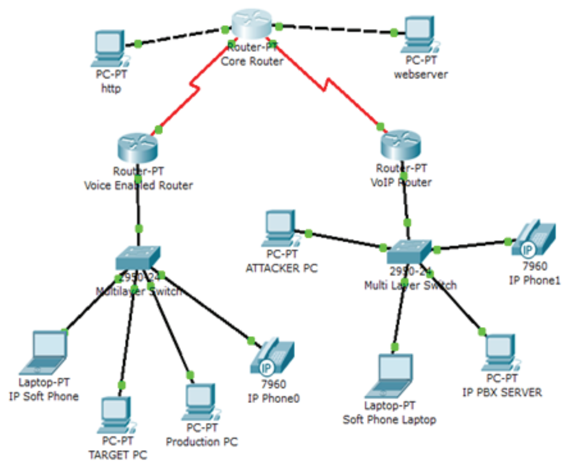

Figure 8. A Network Topology

By Using “nmap command with –sp option” which provides the complete information of all hosts connected to the network have weak security. Nmap command also sends TCP probes to port number 80 of each host on the network. By using –PI option with nmap command helps to stop the TCP probes toward port 80.

a) . Behavior of the Traffic after the Attack

The behavior of the voice traffic after the attack is shown in Figure 9.

Figure 9. Attacked Soft Phone

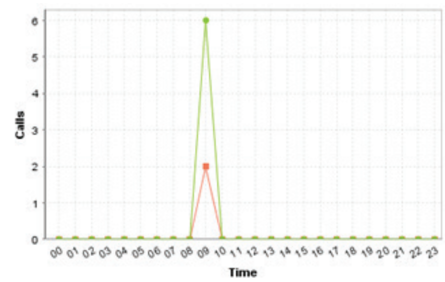

Figure 10. Graphs of Call Quality after Attack

	Min	Max	Avg
Delay (ms)	1001	1001	1001
Jitter (ms)	36	36	36
Loss (%)	0	0	0
MOS	4.2	4.2	4.2
R Factor	88	88	88

Good Tolerable Poor Configure

Figure 11. Behavior of Delay after Attack

RESULTS AFTER COUNTERMEASURES

The behavior of call status after countermeasures is illustrated in Figure 12below. In this scenario, special port numbers are integrated with the https address in order to secure the VoIP PBX Server, For example, https://www.yasir.com:9619/. Also, applied the internal firewall between the trusted and untrusted networks, LAN is usually considered as trusted network and WAN is considered as untrusted network.

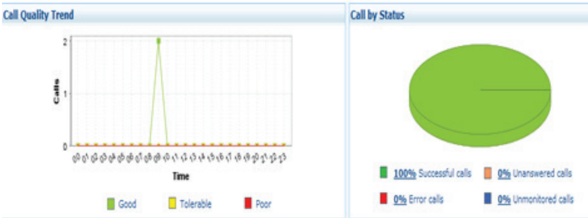

Figure 12. Graphs and Status of Calls after Applying Countermeasures

	Min	Max	Avg
☑ Delay (ms)	1	1	1
☑ Jitter (ms)	36	36	36
☑ Loss (%)	0	0	0
☐ MOS	4.4	4.4	4.4
☐ R Factor	93	93	93

Good Tolerable Poor Configure

Figure 13. Behavior of Delay after Applying Countermeasures

We evaluate the VoIP PBX, VoIP Box and DoS attacks. VOIP Box is used to improve the number of calls on the existing bandwidth and reduce the jitter. By supporting these arguments, real time survey analysis has been introduced. After these results EGS has decided to upgrade their existing network with VoIP in few months. Furthermore, DoS attack has been critically evaluated.

VoIP technology provides lot of benefits but still this technology faces many issues such as migration from PSTN to VoIP, implementation of VoIP, proper selection of tools and techniques, quality of service and regulatory challenges to the providers and users. Therefore, it is important to understand the project and quality management aspects for the successful execution of VoIP technology. It covers project scope, cost issues, quality of service, risks and other factors. The important concern is to make sure that VoIP users achieve same level of voice quality as a call made over traditional PSTN network. In Wide Area Network (WAN), Multi-Protocol Label Switching (MPLS) VPN, tagging techniques and QoS enabled voice routers are used to prioritize the voice traffic and achieve the best quality of service.

Session Initiation Protocols (SIP) is used to measure and optimize the performance of voice quality. It is difficult to decide the best VoIP PBX tool in the VoIP War. Each vendor has some good features with some drawbacks which affects the users as well as open the doors for hackers/attackers. As per my research and experience, Trixbox PBX is better than other vendors because it provides full manageability to their users, such as easy to understand, install and execute. Trixbox is also compatible with all other vendors available in the market and easy to integrate with VOIP Box to achieve the best voice quality on low bandwidth but still cost of VOIP Box is issue for the small companies. Also, Trixbox is not feasible to handle the large organizations due to their architecture. Engineers are working to upgrade the Trixbox architecture but still under process. VQ Manager is very good tool to record the QoS parameters, such as delay, jitter, packet loss, etc. but it does not meet the real time and update the results after 2 sec delay. ICMP and TCP SYN flooding attacks techniques are used to attack the VoIP network and I reduced the ACK time but attacker can also increase the time of spoofed packet. Therefore, it may be difficult to stop the attacks but my approach reduced the factor of attack. VoIP certification should be prerequisite for network engineers to meet economical, professional and social goals. It also covers the legal aspects of the VoIP technology but the cost of VoIP certifications for the individual and small businesses is still an issue.

Like the other technologies, VoIP and IP PBX have some legal, social and ethical issues. Most of the African region such as Ghana, Nigeria does not allow their IT geeks to use VoIP and some countries such as Saudi Arabia, UAE, Pakistan do not allow their private users to use VoIP service. Such countries control their VoIP technology under government influence and do not allow the private operators due to security and level of competitions. VoIP technology also integrates with mobile technology like I-Phone, Blackberry, etc. VoIP services are utilizing huge amount of bandwidth. Therefore, VoIP provider have to make sure that they are not violating any country legal, social and ethical issues meet the requirements issued by International Telecommunication Union (ITU). In this report, my recommended tools and techniques meet the standards of ITU and Data Protect Act 1998 and Act 2006.

CONCLUSION

We achieve multiple results; first, we provide comprehensive knowledge about PSTN and VoIP technologies and prove that VoIP is better than PSTN due to its flexibility, scalability and security. After the comparison of the PSTN and VoIP, IP PBX was introduced and justified the importance of IP PBX and growth rate of VoIP. Also, prove the importance of IP PBX for corporate and home users due to its versatile features and most importantly is cost effectiveness. The great challenge was to select the proper and economical IP PBX which supports the individual users and small and medium sized businesses. After the comparison report, Trixbox, IP PBX, is finalized as the best IP PBX among the other vendors. Installation and configuration of the Trixbox is a big challenge but that challenge has been handled successfully. The implementation of Trixbox also fulfils the legal, social and ethical rules and regulations. The comparison of signaling protocol proved that SIP is slightly better among other protocols but not always. Realistically, all protocols have some advantages and disadvantages. QoS of the VoIP network is justified by using multiple proper voice codec. Another challenge is to secure the network during DoS attacks. This task is also achieved successfully by using TCP SYN flooding attack which clearly showed the behavior of traffic being attacked and quality of the user calls after implementing the countermeasures. The countermeasures reduced the effect of the attacker towards the VoIP PBX network. Lastly, a survey report of the Express Gift Service, a UK based company, proved that VoIP technology is economical, reliable and scalable than regular PSTN network and meet Quality of Service (QoS) parameters. In summary, VoIP technology is thoroughly evaluated, implemented and integrated with IP PBX and VOIP PBX. The results show that QoS and level of security has been achieved. Also, survey report prove that VoIP technology has additional features and more than 50% economical than PSTN. Selected VoIP codec provides the appropriate solutions to improve the voice quality. Identified the problematic areas where network is under attack and introduce the countermeasures which reduce the attacks and improve the network life.

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