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Implementing VoIP In a Small Enterprise Network

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Abstract: This paper discusses the implementation of VoIP in a small enterprise network. VoIP which can be described as the process by which multimedia and voice communication are digitally delivered over the internet protocol is being continuously implemented in different enterprise networks be it small or large. It could be deployed as an open source or a proprietary or virtual/hosted solution. This implementation is carried out in several ways based on different IP-PBX such as Asterisk and 3CX and VoIP protocols such as H.323, Media Gateway Control Protocol (MGCP), Real-time Transport Protocol (RTP), RTP Control Protocol (RTCP), Secure Real-time Transport Protocol (SRTP), Session Description Protocol (SDP) and Session Initiation Protocol (SIP). However, to implement VoIP in a small enterprise network, factors such as cost, bandwidth, flexibility and availability must be considered. Therefore 3CX which is a proprietary and cost effective IP-PBX and SIP trunking which is a cost effective VoIP protocol compared to other conventional trunking protocols are considered suitable for a small enterprise VoIP and are discussed in this paper.

Keywords- VoIP; SIP trunking; IP-PBX; Design

1. INTRODUCTION

With the influence of Voice over IP on telecommunications market based on its flexibility as well as cost reduction, users of VoIP expect call quality to be as good as if not better than the traditional Public Switched Telephone Network (PSTN) [1]. An enterprise with an installed IP-based Private Branch Exchange (IP-PBX) such as 3CX, Asterisk etc. can make cheap calls within its network. When making a VOIP call from an enterprise to an outside destination (say another enterprise), conventionally, the call leaves the IP based network and it travels through a Circuit-Switch network. This cost more compared to having the circuit-switching system replaced by an IP-based switching system such as SIP trunking. This is because SIP trunking provides a virtual connection to an ITSP (Internet Telephony Service Provider) through internet protocols [4]. This enables the ITSP to provide VoIP connectivity between an enterprise network and the PSTN. Therefore implementing VoIP base on SIP trunk in a small enterprise network is a good step in order to simplify the enterprise's telecommunications requirements to reduce cost.

With several enterprises moving towards implementing VoIP while many are also working on a continuous improvement of their already implemented VoIP services, this paper discusses the three types of VoIP solutions and focuses on 3CX which is a proprietary VoIP solution based on SIP and SIP trunking as good IP-PBX and VoIP protocol that should be deployed by small enterprises. Their advantages, design considerations when being deployed and some best practices are also discussed.

VoIP SOLUTIONS

OPEN SOURCE IP-PBX: The term open source simply implies that all the source code of the soft PBX are available and can be accessed by users. This way, the PBX application can be modified by users and this modification can be made available for reuse by the open source community [8]. The open source IP-PBX VoIP solution can be used on any hardware as well as any non proprietary standards. Therefore the cost of developing a new PBX and implementation cost are reduced [13]. An example of this is Asterisk. Most IP-PBX manufacturers base their PBX solutions on Asterisk. This has made Asterisk the most tested, documented and the most popular open source IP-PBX for VoIP platforms in the telecommunications market today. Therefore, for a small medium business considering an open source IP-PBX solution, the benefits include cost reduction, several features, ability to operate with an existing hardware and according to business requirements [13]. However, the implementation requires a specialist who understands the modification process of the source code and without whom nothing can be done.

VIRTUAL/HOSTED VOIP SOLUTION: This is a VoIP solution in which the user has nothing to do with the PBX system. The PBX is hosted by the VoIP provider instead of having it installed and monitored within the enterprise. This can be operated from anywhere there is internet access. It is location independent. The only thing needed on the user equipment is just the user interface which can be easily downloaded and requires no professional skills or a technician to operate. With this, users can manage their communications and add several features offered by the system. An example of this is Skype. So many people make use of Skype today from their PCs, Laptops, telephones and Androids but do not know what a PBX is. This shows how simple it is to make use of hosted PBX. It also support the features offered by premises based PBX such as Call forwarding, conference call, auto attendance, video conferencing etc. But this does not give the enterprise an absolute control over the PBX.

PROPRIETARY VOIP SOLUTION: This is a VOIP solution in which the IP-PBXs are vendor specific. Here, all modifications and upgrades can only be carried out by the manufacturers. Mostly, these modifications and upgrades are carried out automatically without interrupting the users existing configurations [9]. The IP-PBX comes with many features out of which a user can choose. It does not require a special technician to deploy it and there is absolute control over it from within the enterprise. Example of this include Cisco and 3CX which is the main focus of this report and will be discussed in the next chapter.

2. 3CX

This is among the first set of Windows-based PBX system. Although 3cx is a proprietary IP-PBX, it has versions for all Windows operating systems that are mostly used today. It has a very good user friendly GUI and the installation is very easy and straight forward. Upgrades are not complex and changes can easily be made without affecting existing configurations, [9]. It offers all the functions offered by a normal traditional phone systems. It allows communications between phones connected to it within a network in which 3CX is being implemented and allow these phones to communicate with external phones be it on a PSTN or another VoIP network through SIP trunk.

3CX supports multiple gateway devices that are used in translating PSTN to SIP. This way, users are free to decide on what to use. One of the major reasons why people should go for 3CX is that it is very easy to implement even by someone who has no background knowledge compare to IP-PBX like Asterisk that require you to understand the concept of programming. 3CX can either be free or commercial edition. The free edition does not expire and it is been used in some homes and by some small businesses. It however does not have all the features available in the commercial edition [9].

Another advantage of 3CX is that it is open to be used and supports many hardware devices such as phones and gateways manufactured by different vendors. This is because 3CX was designed based on the industrial standard protocol 'SIP'. With this, all devices that support SIP are compatible with 3CX.

THE 3CX COMPONENTS

The components that make up 3CX include: SQL database for data storage, Windows services and web interface for configuration by the administrator or whoever the user is. There is 3CX VoIP client and VoIP phone which are software that make usage as a phone on a computer with headset possible. It consist of an operator panel called 3CX Call Assistance where phone status can be seen, control calls, and provides room for simple chat with operator and caller. It also consist of a Call Reporter used for printing graphs and monitoring call details.

THE 3CX PHONE

Virtually anything that can be done on a normal phone can be done on a 3CX phone. These include placing calls, taking calls and handling multiple of three calls at once, transferring of calls and showing calls. It can be used to dial Microsoft Outlook using the Telephony Application Programming Interface (TAPI) drivers but this is not a free package. It is compatible with any SIP-based IP PBX because it is a standard SIP phone. It has a call recording button for saving sounds to the local PC and supports auto-answering.

Figure1: 3CX Softphone [9].

3CX ASSISTANCE

This allows the user to see what is happening with the phone and carryout call control using drag and drop. This is like the hardware used by receptionist in the olden days to direct calls. It features include handling all call queues, parked calls and connected extensions. It provides voicemail indicator. It visually groups extensions and also integrates inbound call with CRM (Customers Relationship Management).

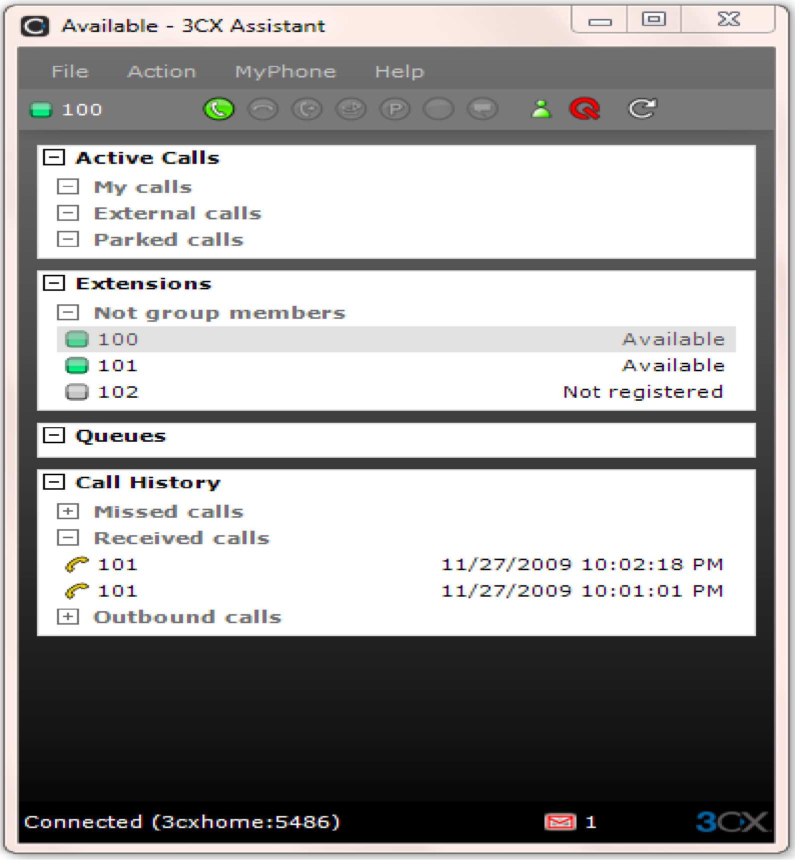

Figure2: 3CX Assistance [9].

3CX GATEWAY FOR SKYPE

This allows Skype which is a virtual VoIP solution to be used in making call on 3CX Phones. To use this, a Skype trunk line will be added as it is being done with SIP trunking and PSTN.

FEATURES AND CHARACTERISTICS OF 3CX

Since it is windows based it makes it easy almost immediately to be used by users since Windows is a widely used operating system. It is open and does not depend on vendors. It is not expensive to implement. To be factual, some small companies or homes make use of the free version. The commercial edition is also priced reasonably because the mode of licensing is based on per current call. Another advantage is that it is not a turnkey hardware phone system this makes the cost lower and can be used with almost all SIP standard hardware. For any enterprise that wishes to fully utilize IP-PBX like 3CX in order to communicate through an IP network with an outside network not within its network, the solution is SIP trunk. This is fully discussed below.

3. SIP TRUNKING

SIP can be described as an application layer signalling protocol that is designed to be independent on existing transport layer and can run on TCP/IP, UDP/IP or Stream Control Transmission Protocol (SCTP). This feature allows SIP to offer services such as call control, mobility, interoperability with existing telephony system and many more. SIP is commonly used in controlling multimedia communication sessions which include voice and video calls across Internet Protocol. SIP sessions may include one or more participants and can also be used to create, modify and terminate two participants (unicast) or multi participants (multicast) sessions by consisting of one or more media streams [5]. SIP trunk can be used to describe any of the following cases [3]:

* A VoIP service provided by an Internet Telephony Service Provider to virtually connect an enterprise and Public Switched Telephone Network (PSTN), by replacing the physical connectivity provided by the traditional circuit-switching.
* A port on an enterprise server which provides connectivity between different enterprise's server-based system.
* An interconnection of different IP-PBXs that replaces the conventional TDM trunk.

Session Initiation Protocol (SIP) can be described as a communications protocol that enables communication between devices in an enterprise network. This communication could be within a LAN, a WAN or over the internet. SIP trunking offers a new form of connectivity to a service provider both for outgoing and incoming calls over the internet while avoiding the traditional circuit-switching such as the ISDN.

To better understand how the SIP trunk system works, it is good to know how the traditional circuit-switching network handles calls. Here, connections to the service provider (telephone company) were possible only using a dedicated telephone line, such as an ISDN connection [2]. This require a physical connectivity for both outgoing and incoming calls and therefore set of dedicated hardware connections are required for the phone system. Figure1 below illustrates a circuit-switch network. Calls are connected to the service provider through the ISDN trunk.

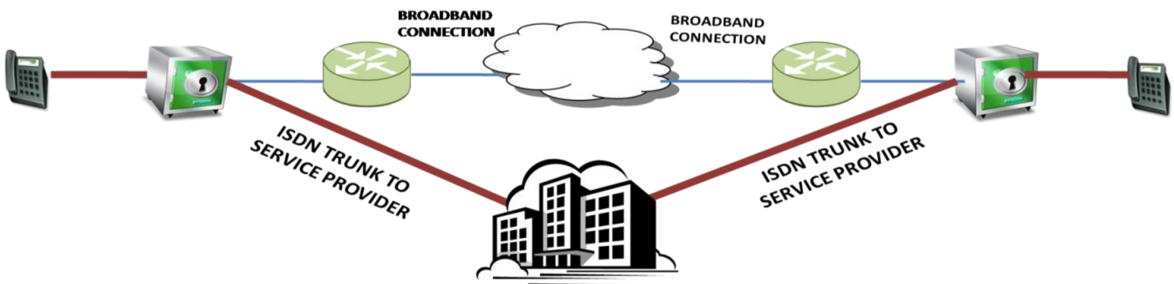

Figure3: ISDN Trunk (source: self-drawn, inspired by [2])

With SIP trunk, the dedicated hardware connectivity is no longer important. Connection to the internet service provider (ITSP) is now through the broadband connection (internet) and this can handle more calls without any additional physical lines. Figure2 below illustrates a SIP trunk network. Here, calls are connected to the internet service provider (ITSP) through SIP trunk while making use of the internet.

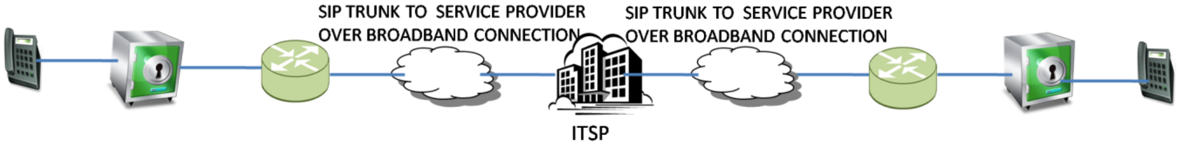

Figure4: SIP Trunk (source: self-drawn, inspired by [2])

ADVANTAGES OF SIP TRUNK

There are many enterprises with VoIP already implemented within their network. Some use it just to enable communication within the enterprise LAN, some use it within and outside their LANs, while some are yet to have it implemented. In early VoIP communication, any call going outside of the enterprise LAN goes through an integrated service digital network (ISDN) trunk link. Although the enterprises achieved a good return on their investment when compare to the cost of the old communication method, but with the introduction of SIP trunking, the return on the enterprise investment became much more greater. This is because SIP trunking has taken VoIP further beyond the LAN applications by making use of a full IP communication over the internet [4]. Explained below are some of the benefits of SIP trunking.

1. COST REDUCTION

One of the major benefit of SIP trunking is cost reduction. Enterprises can join voice and data services in the same network without the need to pay for an expensive ISDN subscription cost. Making VoIP enabled telephone call is cheaper than calling over an ISDN infrastructure most especially international calls. SIP trunking supports business growth because extra phones can be added without an installation of extra line which cost more [2]. The tables below shows two different example of cost reduction as a result of SIP trunking in the U.S.A and U.K.

Traditional Phone Lines and Broadband Internet Service			SIP Trunking and Broadband Internet Service	
3 lines for service with 1000 minutes local and domestic long distance	$135		SIP service with 3 concurrent call sessions and 1000 minutes local and domestic long distance	$57
250 minutes international long distance	$36		250 minutes international long distance	$8
High-speed Internet	$90		High-speed Internet	$90
Monthly charges	$261		**Monthly charges**	$155
Annual costs	$3,131		**Annual costs**	$1,860
			Annual cost savings with SIP trunking	**$1,272**

Table1: showing cost reduction with SIP Trunking in U.S.A [2].

Traditional Phone Lines and Broadband Internet Service			SIP Trunking and Broadband Internet Service	
Recurring ISDN charges	£272		16 SIP trunk channels	£144
Traffic	£2,380		Traffic (local, long distance, international)	£1,694
Monthly charges	£2,652		**Monthly charges**	£1,738
Annual costs	$31,824		**Annual costs**	£20,856
			Annual cost savings with SIP trunking	**£10,968**

Table2: showing cost reduction with SIP Trunking in U.K [2].

2. BANDWIDTH UTILIZATION

In the ISDN trunking system, two lines are used, one for telephony srevices (the ISDN trunk) and the other for internet services (the broadband connection). This way, bandwidth utilization on both line is often low. Voice traffics are much during peak periods in an office environment. But during non peak period, the line bandwidth is underutilized. The data traffics on the other hand has lots of traffics happening throughout the day. This can be seen in figure3 below.

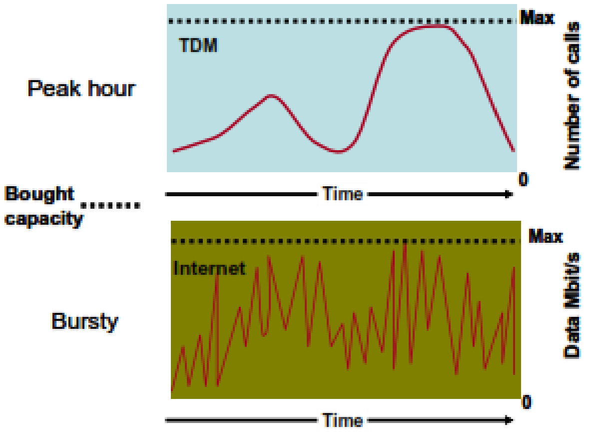

Figure5: typical bandwidth utilization on a TDM trunk [4].

From figure5 above, if the time periods are rearranged with the highest usage on the left with lowest to the right in a decreasing order, this reveals the total bandwidth that is being wasted. This is shown in Figure6 below.

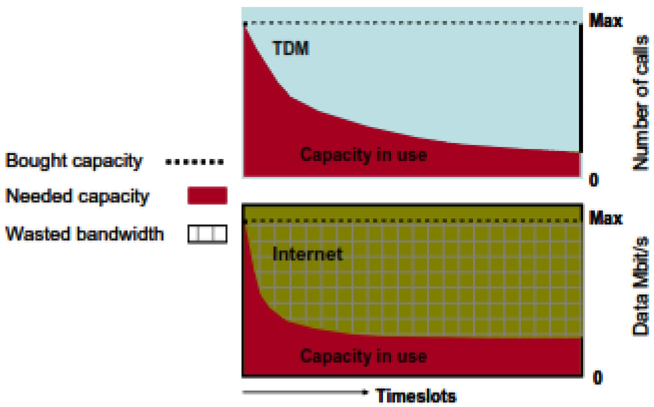

Figure6: Comparison of the bandwidth in use [4].

Data traffics when compared to voice traffic which is a real time communication, is not time critical. Therefore combining the traffics on the same line using SIP trunk while applying the correct quality of service with the time critical voice communications prioritized over the non time critical data communication at all time, will provide a maximum utilization of the bandwidth and the capacity needed on request is always available. This can be seen in figure7 below.

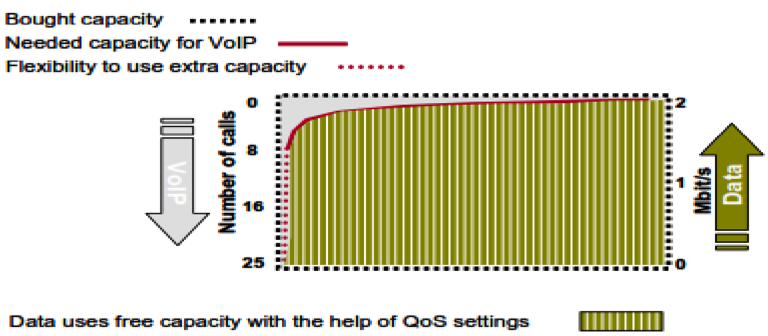

Figure7: Bandwidth utilization with SIP trunk [4].

The capacity of a trunk line can be determined by the number of calls it can handle simultaneously and the available bandwidth. ISDN/TDM trunks are bounded to a particular timeslot capacity. Therefore the volumes of calls at any point in time is dependent on the available timeslots [4]. This is a different case in the SIP trunk line. Here, calls are bounded to the available bandwidth. The bandwidth can be allocated dynamically based on demand and therefore offers an efficient capacity scale.

3. FLEXIBILITY IN HANDLING AN INCREMENT IN TRAFFIC

For a ISDN/TDM trunk to support more voice traffics, there must be an increment in the trunk capacity. Therefore there is need for an additional PSTN gateway and/or Primary Rate Interface (PRI) subscription fee. To move from one E1/T1 to two or more requires additional hardware installation [4]. The E1 which are copper lines that make up a TDM trunk in Europe and the rest of the world apart from the North America and Japan that make use of the T1, is only available in bulk of 23/30 lines. When an available capacity has been used up, the enterprise will have to buy a whole bulk of E1/T1 even when just one voice traffic line is needed since the E1/T1 are only available in bulk of 23/30.

With the introduction of SIP trunking, an enterprise will not have to purchase any additional hardware or pay subscription fees for any additional line(s) due to increment in traffic. More bandwidth just need to be allocated to any increment in call traffic.

4. NUMBERING FLEXIBILITY

With SIP trunking, an enterprise can have a very good numbering scheme that is independent on physical location. In the ISDN/PSTN network, the telephone operators provide telephone numbers based on geographical location of local exchange. This linkage of both numbering and location is broken by SIP trunking. It allows cost effective solution which cannot be replicated using ISDN network to be delivered to customers [12]. For example, a U.K number can be provided in overseas country in order to reduce call cost on internal calls and when customers call the overseas office. Call cost can also be saved on a customer who takes his number with him when his office is being relocated. Also, numbers from different geographical locations are not only supported from a single site. For example, a Sunderland number can be hosted in London office.

5. AVAILABILITY OF FAST SERVICES

The time taken to put an ISDN network in place is much because it goes through many processes as physical lines has to be installed. Trunk rolls are required and there must be technical inspection of the enterprise premises before the implementation can take place [6]. As for SIP trunking, it has list of features that are immediately available in as much as the initial configurations as been done. Take for instance, to enable an extra capacity to an initial bandwidth setup, it is just a matter of an adjustment in the initial configuration and the extra capacity will be available instantly. Another advantage here is that an enterprise does not have to pay an additional money for the additional features after paying for the SIP trunk itself.

6. MULTIHOMING ADVANTAGE ON INTERNATIONAL CALLS

Multihoming is a technology that allows an enterprise to subscribe to two or more service providers to achieve good performance and service reliability. For several years up till now, multihoming has been increasingly leveraged for improving WAN performance, reduce cost of bandwidth and optimize the usage of upstream link [6]. Because the SIP trunking is linked to the ITSP through the internet, it gives enterprises the opportunity to utilize the advantages of multihoming so as to achieve an improvement in the performance of VoIP calls. By considering low cost routing (LCR) model, enterprises can subscribe their SIP trunk to two or more ITSPs that offer a high quality performance VoIP for a reasonable cost [7]. Therefore, an enterprise may consider subscribing to ITSPs in countries where it has the highest call traffics. With this, calls can be routed through the cheapest ITSP based on the country of call destination and therefore saving cost. Multihoming also provides failover within the network and this ensures that VoIP traffics are always transported without disruption. Figure8 below illustrate a multihomed network.

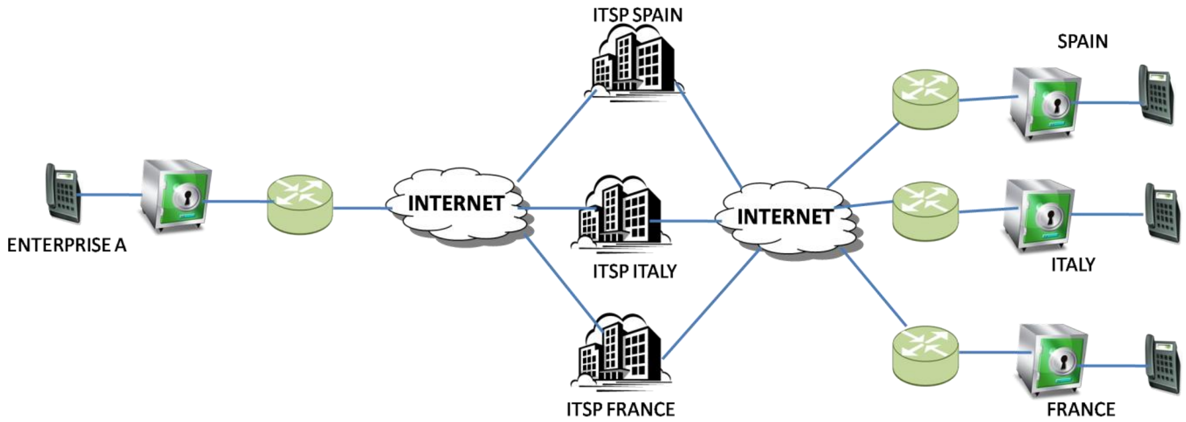

Figure8: Multihoming in SIP Trunking (source: self-drawn, inspired by [2])

7. SIP TRUNKING IMPROVES PRODUCTIVITY

SIP is a standard protocol which give an enterprise the opportunity to increase productivity by delivering the fastest return on investment (ROI) compared to other protocols. It has great effect on workers performance, collaboration and communication [2]. It supports video conferencing, instant messaging, file transfer, application sharing, real-time communications between machines, alarm distribution, checking online availability and whiteboarding.

8. ALL IP CALLS FROM ONE IP TO ANOTHER

Earlier VoIP communication from one IP were not transported in its digital form to the other IP. It is transcoded into an analogue form at the PSTN and to an ITSP. The ITSP transcodes it back to digital and then sends to the PSTN of the call destination where it is converted to analogue and then back to digital at the destination. This caused more cost on calls and reduces the call qualities dues to the series of conversion. [4].

Figure9: ISDN VoIP (source: self-drawn, inspired by [4])

As for SIP trunk, it allows a direct IP to IP calls in its digital form without passing through the PSTN. This helps retain the call qualities and also reduces call cost.

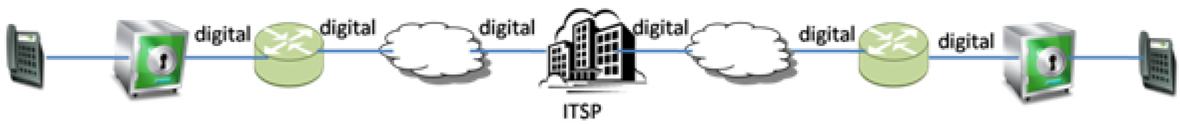

Figure10: SIP trunk VoIP (source: self-drawn, inspired by [4])

4. VoIP DESIGN CONSIDERATIONS

1. CHOOSING A SERVICE PROVIDER

To select a service provider for VoIP, the enterprise must look for a SIP trunking service provider that supports the IP-PBX in use because all PBX are not supported by all providers [10]. However, 3CX is an IP-PBX based on SIP therefore supports any SIP trunking provider. Area covered by the provider, reliability and how consistent must be considered.

The enterprise must consider the kind of business it does and the number of locations it has cause this determines what call plan they will choose and such provider must offer guarantee on customer satisfaction and installation. Finally additional features that will be provided by the provider must be considered.

2. DESIGN CONSIDERATION WITH QoS

Since the internet is used for transportation of many realtime applications such as voice and video which are delay sensitive and data, there must be a measurable, reliable, predictable and guaranteed network design. This can be achieved by proper management of bandwidth, delay, jitter and packet loss within the network [11]. With QoS, an enterprise can manage its network with good convergence. That is, voice, video and data convergence will be transparent to users. QoS make different kind of network traffic to make use of the network resource based on priority level. Voice, video, and critical applications will be given higher priority so that they can be transported across the network in real time to avoid delay, jitter or packet loss.

3. VoIP SECURITY.

Since the enterprise will connect to networks outside its private network or private WAN, there is need for adequate security features and facilities within its network. In a VoIP network, security starts from the installation of good anti-virus and make sure it is regularly updated. Intrusion detection and prevention system should also be introduced into the network [1]. The introduction of an application layer gateway or firewall between a trusted and an untrusted zone maintains good security and also prevents UDP flooding. SIP security can also be ensure by the use of the AAA (Authentication, Authorization, and Accounting) [12]. In other to Isolate VoIP segments, it is good to create policy-based security zone.

Security issues such as eavesdropping on critical information can be prevented by the use of VPN such as IPsec with adequate encryption. And finally, VLANs should be used to secure Voice traffic and separate it from data traffics.

CONCLUSION.

In this paper, different VoIP solutions were discussed and 3CX which is a proprietary solution was recommended and discussed. SIP trunking, its features and advantages were discussed with necessary diagram. Factors to be considered in selecting a service provider, implementation of QoS and how to secure a VoIP network were also identified. Finally, since cost, bandwidth, flexibility and availability are important factors to be considered when setting up VoIP within a network, a combination of 3CX and SIP trunking are considered suitable for the implementation of VoIP whitin a small enterprise network.

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