Voice over IP and Skype for Business with SIP Trunking in Business Scenarios

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ABSTRACT *Voice over Internet protocol (VoIP), there is an existing way of communication over any network. The Users can make thetelephone calls over an IP network using this technology. This paper will describe Voice over Internet Protocol (VoIP) to a level that allows discussion of security issues and concerns. There are two kinds of spoofing attacks are possible, first one is IP spoofing attack and another is URI spoofing attack, which are described in this paper. The Implementation of VoIP concerned by businesses, components of a VoIP system, and relevant security issues. The business concerns will be those which are used to affect the Quality of Service (QoS). The network components call processors, gateways and two of the more common architectures are held by VoIP. Skype for business from Microsoft is used in Day to Day conjunction in Business environments is vastly now being deployed with SIP Trunking nowadays.*

Key words: *VoIP, H.323, SIP, MGCP, QoS, Spoofing Attacks.*

I. INTRODUCTION

To transmit voice conversations over a data network using IP, VoIP technology is used. Such data network may be the Internet or a corporate Intranet or managed networks which are specially used by long distance and local service traditional providers and ISPs (Internet Service Provider).

Voice over Internet Protocol (VoIP) is a form of communication that allows end-user to make phone calls over a broadband internet connection. Basic VoIP access usually allows you to call others who are also receiving calls over the internet. Interconnected VoIP services also allow you to make and receive calls to and from traditional landline numbers, usually for a service fee. A special type of adapter is used in some VoIP services which required a computer and a dedicated VoIP telephone. Other services allow to end-users to use own landline phone, it is used to replace VoIP calls. All these paradigms are held by a special adapter.

Voice over IP refers to the diffusion of voice traffic over internet-based networks. Internet Protocol (IP) was originally designed for data networking for purpose of its success, VoIP protocol has been adapted to voice networking. The history of VoIP began with conversations by a few computer users over the Internet. Initially, VoIP required a headset to be plugged into the computer, and the participants could only speak with others who had a similar set up. They had to phone each other ahead or sent a text message, in order to alert the user at the other end of the incoming call and the exact time [2].

In November 1977, the IETF published the Specifications for the NVP (network voice protocol)'. In the preface to this document, the objectives for the research were explained as the development and the demonstration of the 'feasibility of secure, high-quality, low-bandwidth, real-time, full-duplex1 digital voice communications over packet-switched computer communications networks [3].

In the mid-90s, IP networks were growing, the technology had progressed and the use of personal computers had grown extensively. The belief that VoIP could start to make some impact on the market resulted in high expectations and the distribution of the first software package. In its early stages, the VoIP technology was not sufficiently mature. There was a big gap between the marketing structure and the technological reality. It results in an overall agreement that technical shortages stopped any major transition to VoIP. However, VoIP is continued to make technical and commercial progress. The most of the technical problems have been solved by VoIP technology. There are no restrictions in the limited market conditions [4].

The communications network providers are used to adopt IP in their infrastructure, enterprises are adopting IP for private corporate networks. The communication between employees facilitate by using VoIP technique, whether working at corporate locations, working at home, or travelling. VoIP can also augment corporate efficiencies.There are several enterprises which are used to test VoIP, doing a tryout, or engaging in incremental upgrades. The majority of multinational corporations use VoIP instead of remote possibility. The business opportunity will be a major part of their business operations in the near future [5].

This paper is divided into seven parts. Starting with introduction (Section-I), next section covers the implementation of VoIP (Section-II). Moving ahead, Configuration of VoIP is discussed (Section-III). After that VoIP attacks are discussed (Section-IV), How to Protect against Risks are discussed (Section-V). More over Requirements, Availability and Service Limitations are discussed (Section-VI) and finally, conclusions summarizes the last section (Section-VII).

II. IMPLEMENTATION OF VoIP

In this section first we will discuss VoIP protocols and after that data processing in VoIP, at last we will discuss about quality of service in VoIP systems.

a. *Protocols*

There are currently three types of protocols which are widely used in VoIP implementations: the H.323 family of protocols, the Session Initiation Protocol and the media Gateway Controller Protocol (MGCP). The discussion of these protocols is as follows:

- *H.323 Family of Protocols*

H.323 [8], [9] is a set of recommendations from the International Telecommunication Union (ITU) and consists of family of protocols that are used for call set-up, call termination, registration, authentication and other functions. These protocols are transported over TCP or UDP protocols. The following figure.1 shows the various H.323 protocols with their transport mechanisms. H.323 family of protocol consists of H.225 which is used for registration, admission, and call signaling. H.245 is used to establish and control the media sessions. T.120 is used for conferencing applications in which a shared white-board application is used. The audio codec is defined by G.7xx series by H.323, while video codec is defined by H.26x series of specifications. H.323 uses RTP for media transport and RTCP is used for purpose of controlling RTP sessions. The following figure.2 & figure.3 shows the H.323 architecture and call set-up process.

- *Session Initiation Protocol (SIP)*

The modification and termination sessions between two or more participants the IETF is used which is defined by SIP (session initiation protocol) [9]. These sessions are not limited to VoIP calls. The SIP protocol which is a text-based protocol, it is similar to HTTP and offers an alternative to the complex H.323 protocols. SIP protocol become more popular in comparison to H.323 family of protocol because it is more similar than it. The following figure.4 and figure.5 shows the SIP architecture, call set-up and tear down process.

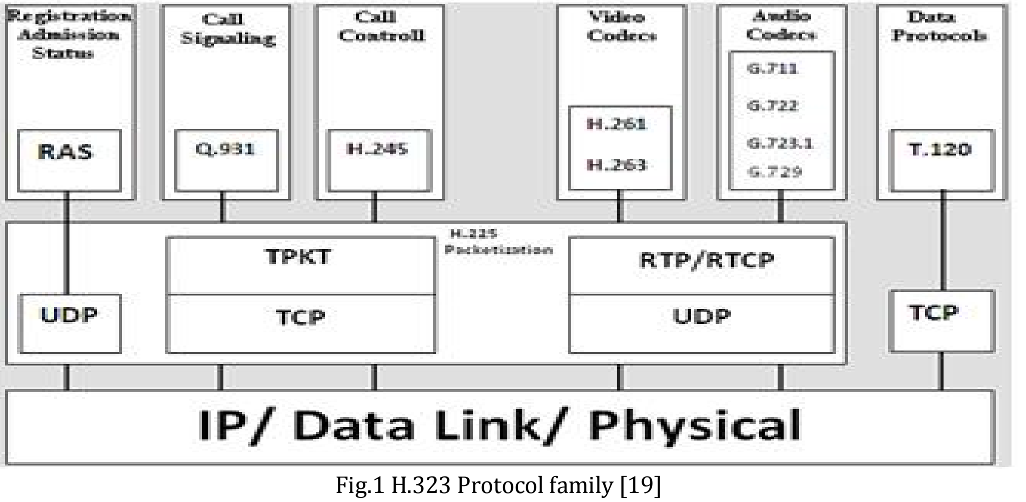

Fig.1 H.323 Protocol family [19]

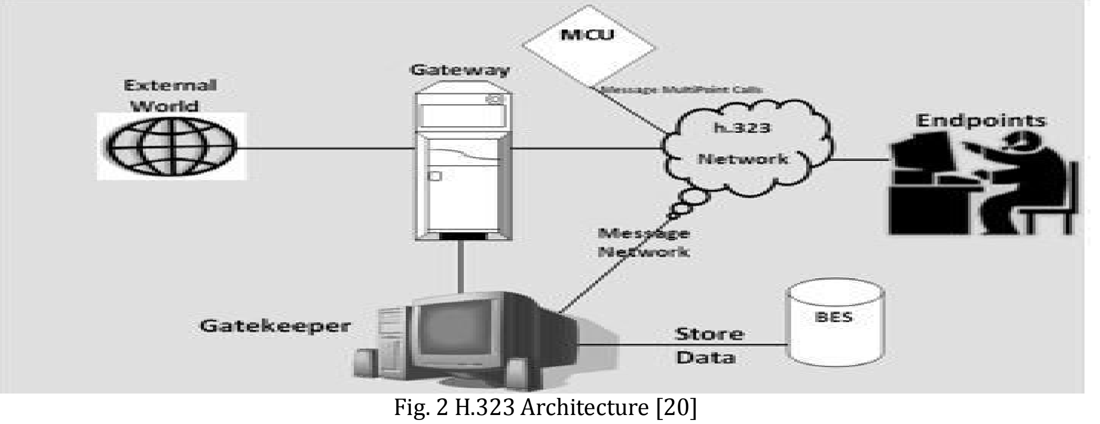

Fig. 2 H.323 Architecture [20]

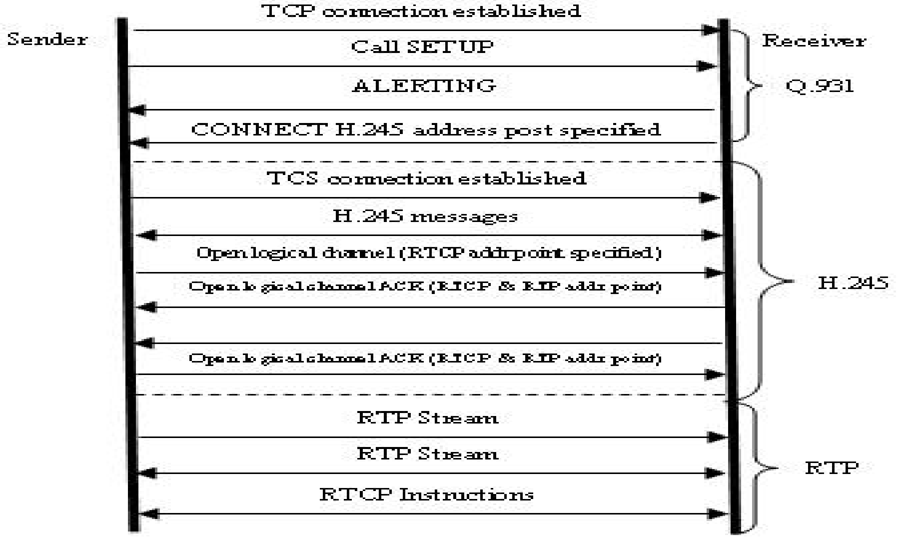

Fig. 3 Call Setup Process in H.323 [20]

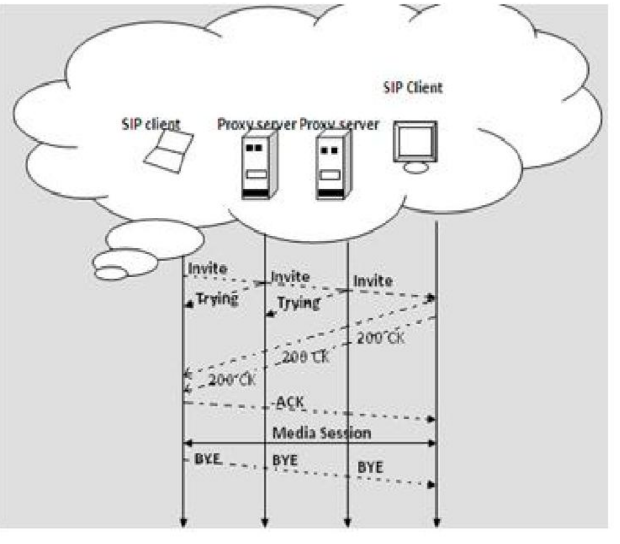

Fig. 4 SIP Network Architecture [9]

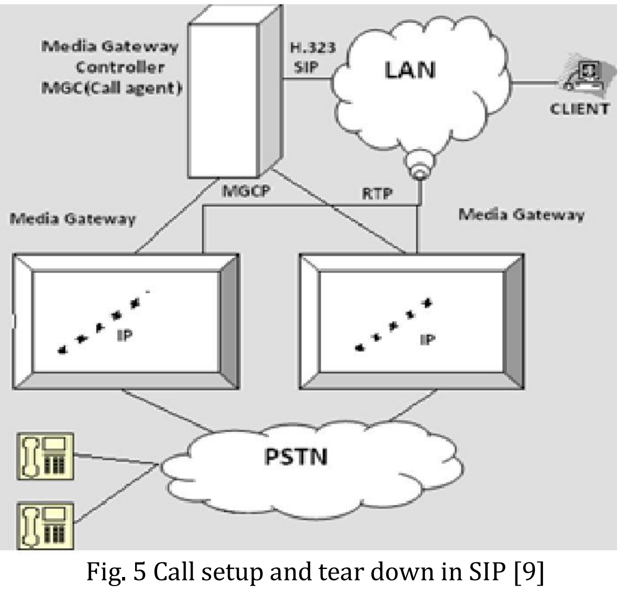

Fig. 5 Call setup and tear down in SIP [9]

 Media Gateway Control Protocols (MGCP)

The communication between the separate components of a decomposed VoIP gateway is done by media gateway control protocol. It is a complementary protocol to SIP and H.323. “Call agent” is mandatory and manages calls and conferences, when we are using MGCP and MGC server (Figure 6). The MG endpoint is not responsible for calls and conferences. It does not maintain call states. MGs are responsible to execute commands sent by the MGC call agents. MGCP assumes that call agents will synchronize with each other sending coherent commands to MGs under their control. MGCP does not define a mechanism for synchronizing call agents. MGCP is a master/slave protocol with a closely coupling between the MG (endpoint) and MGC (server).

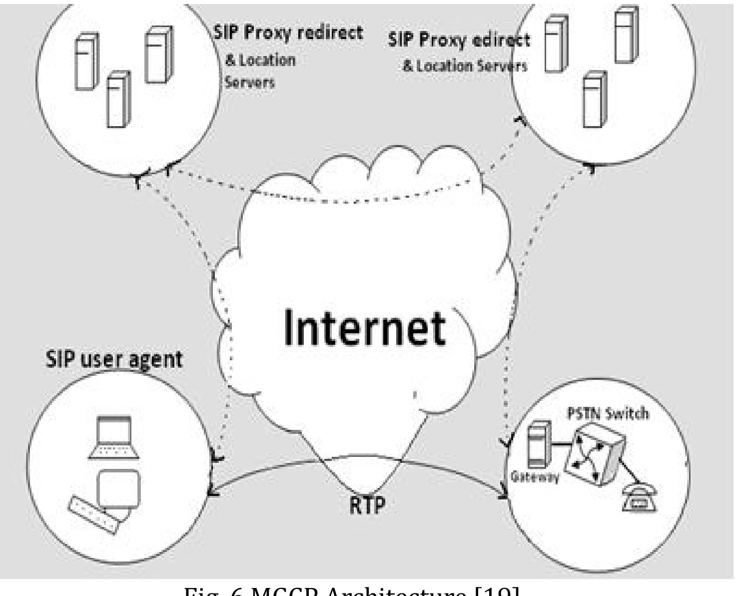

Fig. 6 MGCP Architecture [19]

b. *Data Processing in VoIP Systems*

There are three types of essential components in VoIP: CODEC (Coder/Decoder), packetizer and playout buffer [10], [11]. The analog voice signals are converted into digital signals at sender’s side, after that these digital signals are compressed and then encoded into a predetermined format using voice codec. There are various voice codecs developed and standardized by International Telecommunication Union-Telecommunication (ITU-T) such as G.711, G.729, and G.723 etc. The packetization process is performed by distributing fragmented encoded voice into equal size of packets.

Furthermore, in each packet, some protocol headers from different layers are attached to the encoded voice. Protocols headers added to voice packets are of Real-time Transport protocol (RTP), User Datagram Protocol (UDP), and Internet Protocol (IP) as well as Data Link Layer header. In addition, RTP and Real-Time Control Protocol (RTCP) were designed to support real-time applications at the application layer.

Although TCP transport protocol is commonly used in the internet, UDP protocol is preferred in VoIP and other delay-sensitive real-time applications. TCP protocol is suitable for less delay-sensitive data packets and not for delay-sensitive packet due to the acknowledgement (ACK) scheme that TCP applies. This scheme introduces delay as receiver has to notify the sender for each received packet by sending an acknowledgement. The UDP protocol cannot be applied to VoIP technology. It is more suitable for VoIP applications.

The packets are then sent out over IP network to its destination where the reverse process of decoding and de-packetizing of the received packets is carried out. The time variations of packet delivery (jitter) may occur in transmission process. Hence, a play out buffer is used at the receiver end to migrate the package without any interruption. Packets are queued at the playout buffer for a playout time before being played. However, these packets continued to arrive until the playout time is discarded. The fig.7 shows the end –to-end transmission of voice in VoIP system.

Besides, there are signaling protocols of VoIP namely Session Initiation Protocol (SIP) and H.323. These signaling protocols are required at the very beginning to establish VoIP calls and at the end to close the media streams between the clients.

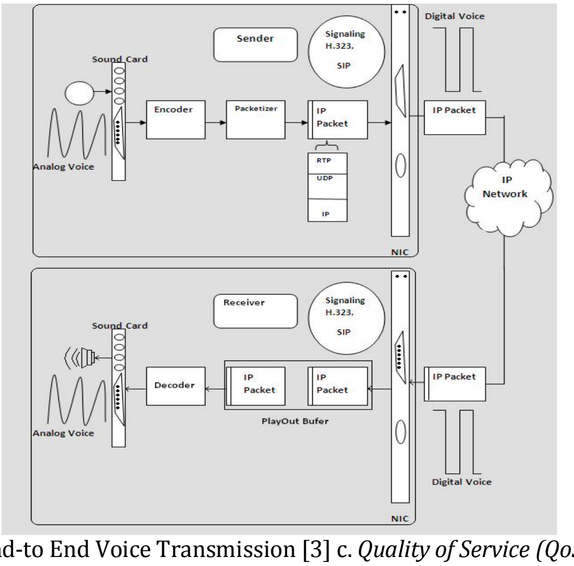

Fig. 7 End-to End Voice Transmission [3] c. *Quality of Service (QoS) in VoIP Systems*

Quality of service (QoS) [3] can be defined as the network ability to provide good services that satisfy its customers. In other words, QoS is used for measurement of the degree of user satisfactions. When degree of user satisfactions is higher than it means the QoS is also higher. QoS are briefly described as given below:

- *Delay*
  Delay can be defined as the total time it takes since a person, communicating another person, speaks words and hearing them at the other end. Delay can be categorized into three categories: delay at the source, delay at the receiver and network delay [3].
- *Jitter*
  IP network does not guarantee of packets delivery time which introduces variation in transmission delay. This variation is known as jitter and it has more negative effects on voice quality [3], [4].
- *Packet Loss*
  Packets transmitted over IP network may be lost in the network or arrived corrupted or late. Packets would be discarded, when they arrive late at the jitter buffer of the receiver or when there is overflow in jitter buffer or router buffer. Therefore, packet loss is equal to the total loss occurs during congestion of network and late arrival [5]. During the packet loss, the sender is informed to retransmit the lost packets. It causes more packet delay and it affects transmission QoS.
- *Echo*
  In VoIP, Echo occurs when a caller at the sender side hears the reflection of his own voice after he talked on the phone or the microphone, whereas the callee does not notice the echo. Echo is the term of the reflections of the sent voice signals by the far end. Echo could be electrical echo which exists in PSTN networks or echo of sound which is an issue in VoIP networks [6].
- *Throughput*
  The throughput may be defined as the maximum number of bits received out of the total number of bits sent during an interval of time.

  Unified Communications

Unified communications (**UC**) is a business term describing the integration of enterprise communication services such as instant messaging (chat), presence information, voice (including IP telephony), mobility features (including extension mobility and single number reach), audio, web &video conferencing, fixed-mobile convergence (FMC), desktop sharing, data sharing (including web connected electronic interactive whiteboards), call control and speech recognition with non-real-time communication services such as unified messaging (integrated voicemail, e-mail, SMS and fax). UC is not necessarily a single product, but a set of products that provides a consistent unified user interface and user experience across multiple devices and media types.[1]

In its broadest sense, UC can encompass all forms of communications that are exchanged via a network to include other forms of communications such as Internet Protocol Television (IPTV) and digital signage Communications as they become an integrated part of the network communications deployment and may be directed as one-to-one communications or broadcast communications from one to many. UC allows an individual to send a message on one medium and receive the same communication on another medium. For example, one can receive a voicemail message and choose to access it through e-mail or a cell phone. If the sender is online according to the presence information and currently accepts calls, the response can be sent immediately through text chat or a video call. Otherwise, it may be sent as a non-real-time message that can be accessed through a variety of media.

In a day to day scenario

Modern knowledge workers use many ways to communicate with colleagues, customers, suppliers, and collaborators in the course of their daily routine. Telephones, instant messages, presence, voicemail, videoconferencing, web conferencing, document sharing, collaborative electronic whiteboarding, and e-mail can all be used for conducting business. UC brings all these disparate tools into a cohesive system, allowing users to quickly and easily transition between communication modes, allowing a simple telephone call to smoothly morph into a videoconference with the ability to add new parties and share documents and whiteboards when additional expertise is required. By unifying communications, corporations can realize measurable productivity gains and streamline the flow of information throughout the enterprise

Skype forBusiness

Skype for Business (formerly **Microsoft Office Communicator** and **Microsoft Lync**) is an instant-messagingclient used with Skype for Business Server or with Skype for Business Online (available with Microsoft Office 365 Skype for Business is enterprise software; compared to Skype it has different features. On 11 November 2014, Microsoft announced that in 2015, Skype for Business would replace Lync. The latest version of the communication software combines features of Lync and of the consumer software Skype. There are two user interfaces – organizations can switch their users from the default Skype for Business interface to the Skype for Business (Lync) interface.

Features

Basic features of Skype for Business include:

- Instant messaging
- Voice Over IP(VoIP)
- Video conferencing inside the client software

Advanced features relate to integration with other Microsoft software:

- Availability of contacts based on Outlook contacts stored in a Exchange server
- Users can retrieve contact lists from a local directory service such as Microsoft Exchange Server
- Microsoft Office can show if other people are working on the same document
- All communication between the clients takes place through the Skype Servers. This makes communications more secure, as messages do not need to leave the corporate intranet, unlike with the Internet-based Windows Live Service. The server can be set to relay messages to other instant messaging networks, avoiding installation of extra software at the client side.
- A number of client types are available for Microsoft Lync, including mobile clients.
- Uses SIP as the basis for its client communication protocol
- Offers support for TLS and SRTP to encrypt and secure signaling and media traffic
- Allows sharing files

Note: With the release of Lync Server 2013 in October 2012, a new collaboration feature "Persistent Group Chat" which allows multi-party chat with preservation of content between chat sessions was introduced. However, only the native Windows OS client and no other platform supports this feature at this time. The main new features of this version are the addition of real-time multi-client collaborative software capabilities, (which allow teams of people to see and simultaneously work on the same documents and communications session). Lync implements these features as follows:

- Collaboration through Whiteboard documents, where the participants have freedom to share text, drawing and graphical annotations.
- Collaboration through PPT documents, where the participants can control and see presentations, as well as allow everybody to add text, drawing and graphical annotations.
- Polling lists, where Presenters can organize polls and all participants can vote and see results.
- Desktop sharing, usually by allowing participants to see and collaborate on a Windows screen
- Windows applications sharing, by allowing participants to see and collaborate on a specific application.

All collaboration sessions get automatically defined as conferences, where clients can invite more contacts. Conference initiators (usually called "organizers") can either promote participants to act as presenters or

demote them to act as attendees. They can also define some basic policies about what presenters and attendees can see and do. Deeper details of policy permissions are defined at server level.
Following Microsoft's acquisition of Skype in May 2011, the Lync and Skype platforms could be connected, but sometimes only after lengthy provisioning time.

Extensions

Lync uses a number of extensions to the SIP/SIMPLE instant-messaging protocol for some features. As with most instant-messaging platforms, non-Microsoft instant-messaging clientsthat have not implemented these publicly available extensionsmay not work correctly or have complete functionality. Lync supports federated presence and IM to other popular instant message services such as AOL, Yahoo, MSN, and any service using the XMPP protocol. Text instant-messaging in a web browser is available via Lync integration within Exchange Outlook Web App.
Although other IM protocols such as Yahoo do have wider support by third-party clients, these protocols have been largely reverse-engineered by outside developers. Microsoft does offer details of its extensions on MSDN and provides an API kit to help developers build platforms that can interoperate with Lync Server and clients.
SIP Trunking
SIP trucking is Voice over Internet Protocol (VoIP) and streaming media service based on the Session Initiation Protocol by which Internet telephony service providers (ITSPs) deliver telephone services and unified communications to customers equipped with SIP-based private branch exchange (IP-PBX) and Unified Communications facilities. Most Unified Communications software applications provide voice, video, and other streaming media applications such as desktop sharing, web conferencing, and shared whiteboard.

Domains

The architecture of SIP trucking provides a partitioning of the Unified Communications network into two different domains of expertise.

- Private Domain: refers to a part of the network connected to your PBX or unified communications server (typically everything you are responsible for).
- Public Domain: refers to the part of the network which allows access into the PSTN (Public Switched Telephone Network) or PLMN (Public Land Mobile Network). This is usually the responsibility of your internet telephone service provider (ITSP).

The interconnection between the two domains must occur through a SIP trunk.
The interconnection between the two domains, created by transport via the Internet Protocol(IP), involves setting specific rules and regulations as well as the ability to handle some services and protocols that fall under the name of SIP trunking.
The ITSP is completely responsible to the applicable regulatory authority regarding all the following law obligations of the Public Domain.

- Tracking traffic;
- Identification of users;
- Implementation of the lawful interceptionmechanisms.

The private domain instead, by nature, is not subject to particular constraints of law, and may be either the responsibility of the ITSP, the end user (enterprise), or of a third party who provides the voice services to the company.

Architecture

In each domain there are elements that perform the characteristic features requested to that domain, in particular the result (as part of any front-end network to the customer) is logically divided into two levels:

- The control of access (Switch)
- Network-border elementsthat separate the Public Domain from the Private Domain, implementing all the appropriate ITSP phone security policies.

The private domain consists of three levels:

- Corporate-Border Elements that separate the Public Domain from the Private Domain, implementing all the appropriate company security policies.
- Central Corporate Switching Node;
- IP-PBXs

Example of UC

During the preparation of this paper, a conference call was held between a group of Skype for Business users and an outside party. To set up the call, one Skype for Business user sent out an invitation that included the time and date of the call and list of participants. The external party received an automatically generated message that included a telephone number to access a voice bridge using a toll-free number along with a passcode that was unique to the planned meeting. (Unbeknownst to both parties, the dial-in number was provided by using the Level 3 SIP Trunking service.) The external party also received a link to a website that allowed a download of a desktop client program for the user's PC in order to support full Skype for Business functionality, including video calling, document sharing and white boarding. During the conversation, additional expertise was needed, so the Skype for Business user searched the corporate directory for an expert with the right skills. Before the expert was added to the discussion, active presence was used to verify that the expert was available, even though the expert was connected to the Skype for Business system over an Internet connection from a home office, which was necessitated by a snow day in the local school district. As different parties joined and left the call, audio levels were automatically adjusted to ensure that the participants could hear each other comfortably. When the call ended, each user had a copy of the whiteboard diagram available for their use.

Benefits of Combining Skype for Business to SIP Trunking

Convergence and Access

The purpose of SIP Trunking is to replace the public telephone network connections provided by traditional PBXs with a solution that is more efficient and easier to integrate. It provides the same basic functions as a PRI (Primary Rate Interface) trunk on an ISDN system or a T-1 trunk on a traditional digital telephone system, which is to supply a path for both communication (such as voice calls) and signaling (i.e. the dialing instructions and call supervision/control) between a customer's equipment and a carrier's equipment. SIP Trunking works by converging voice and data access onto a single network, thereby reducing the cost of maintaining duplicate backbones for communications.

Pooling Concurrent Call Paths

Carrier contracts for SIP Trunking will normally specify a maximum number of Concurrent Call Paths (CCPs) that will be supported. One CCP is occupied during each active SIP call that is routed through the carrier and then released back into the pool of available CCPs when the call ends. The total number of CCPs indicates the total number of simultaneous calls that can be transmitted over the SIP trunk.

When multiple sites are connected with SIP telephony over a corporate WAN, the CCPs from all the sites can be pooled into a single SIP Trunking connection to the carrier. This standard feature from Level 3 (which is an optional extra from a few other carriers and not available from most others) is particularly beneficial for smaller company sites. Instead of requiring a 24 voice-channel T-1 or a 23 voice-channel ISDN PRI telephone company connection to a remote office.that may have only a dozen employees (and therefore a need to support perhaps three to seven concurrent calls), these callers can be grouped together with all of an enterprise's other sites, and share a common pool of CCPs. The benefits are similar to the practice of centralizing and aggregating Internet trunks in a data centre rather than buying direct Internet access into each branch.

SIP Trunking works by converging voice and data access onto a single network, thereby reducing the cost of maintaining duplicate backbones for communications.

Interoffice Calling

Cost savings can also be generated by routing calls among different locations of an enterprise using SIP telephony over the corporate WAN. The Skype for Business server can automatically recognize and route calls to destinations that are within the enterprise. Plus, unlike services from many other carriers, SIP Trunking services from Level 3 offer free calling among enterprise locations, even when the calls are routed over the Level 3® Network instead of through the enterprise WAN.

Teleconferencing

Carriers and third parties offer commercial-grade teleconferencing services for what seem like small per-minute fees. However, given the number of hours that employees typically spend on conference calls each month, these costs can accumulate over time and become a major expense item. With Skype for Business, these expenses are eliminated, because the servers that form the core of a Skype for Business system installation provide the voice bridging functions that are required for conference calling. These services can also be expanded beyond the boundaries of the enterprise to enable external customers and collaborators to easily participate in a teleconference.

Web Conferencing

As in the market for teleconferencing, a number of service providers and technology suppliers have emerged over the past decade to supply web conferencing services that allow employees to deliver rich media presentations, including video, application sharing and white boarding to viewer PC.Building and operating these systems can be expensive, both in terms of technology license fees and in terms of support and other recurring costs. Skype for Business natively incorporates a full suite of these functions, eliminating the expense of purchasing, installing, operating and or leasing third party tools for web conferencing.

IT Labour Savings

Because of the simplicity of installing and maintaining a single, unified platform with Skype for Business and SIP Trunking, overall IT staff support costs can be significantly reduced, as compared to managing multiple legacy applications on different platforms. Here are five examples of ways in which Skype for Business with SIP Trunking can lower IT costs:

b. Call admission control (CAC): CAC allows sophisticated control of media flow to ensure balance between real time media and corporate applications. By routing videoconferencing traffic over the corporate IP backbone to the Internet and voice traffic over SIP trunks, CAC allows IT staff to observe call traffic volumes in real time. IT staff can use this information to help decide if a reported call failure is due to exceeding traffic limits or due to a failure that requires troubleshooting, such as quality issues resulting from insufficient bandwidth.
c. PowerShell scripting: This tool supplies a scripting language and a command-line shell that allows Skype for Business to be customized to blend seamlessly into existing corporate work procedures. It is also incredibly powerful, allowing for very complex operations to be performed on the system quite simply. Combined with Role based Access Control (RBAC), administrator activity can be scoped to precisely the areas they are responsible for.

Skype for Business's Web-based control panel: This provides a mechanism for monitoring the status of a Skype for Business system from multiple locations without special monitoring software. Thecontrol panal also offers a monitoring capability that provides near-realtime reporting on dozens of detailed metrics about the status of every call in the network.

**Survivable Branch Appliance (SBA): Used in place of legacy PBX equipment, these devices (which are available from multiple manufacturers) provide survivable communication services in remote end offices, even during network outages Level 3's Voice Complete solution leverages SIP trunking with native PRI handoff eliminates the need for a secondary local loop and enables inbound and outbound calls.

Simplified capacity management: SIP trunks from Level 3 do not block calls, and are limited only by the bandwidth of the converged data backbone. Capacity management is thereby simplified to a single network

Fewer Help Desk calls

Skype for Business uses one integrated client package on each user's desktop to provide access to multiple UC and SIP Trunking functions, thereby reducing the number of desktop applications that help desk staff need to support.

Costs of Skype for Business and SIP Trunking

The primary costs of SIP Trunking are billed as monthly recurring costs from a carrier like Level 3, and consist of three main components: Network Access, Concurrent Call Paths, and Usage.

Network Access represents the cost of physically connecting the enterprise's telecommunications network to the carrier's network and the cost of additional bandwidth required to carry voice over the enterprise WAN. In traditional telephone systems, connections would normally be made by way of dedicated ISDN PRI circuits or T-1 circuits at each site. With SIP Trunking, these costs are replaced by an IP or VPN connection from the enterprise to the carrier and WAN or Internet capacity at each office location that is typically underutilized. (In many cases existing WAN or Internet capacity at sites is underutilized and no additional capacity is required.) As a result, the overall costs are normally significantly lower with SIP Trunking. Concurrent Call Paths represent the cost of providing capacity in a carrier's network for the total number of calls that can be active at any one time. With SIP Trunking, the CCPs are pooled across the entire organization, thereby reducing the number of peak call paths required for the same level of service provided by legacy T-1 or ISDN lines. Usage represents the cost of long-distance minutes of telephone call traffic consumed by the enterprise, which can be reduced in two ways through the use of SIP Trunking. First, all charges for calls between enterprise locations are eliminated, even if they are from an office on one side of the country to the other, because these conversations are carried over the enterprise WAN using UC/SIP. Second, the costs of calling people outside the enterprise are often reduced, due to the attractive rates offered for SIP Trunking customers. The Business Case Scenario included later in this whitepaper includes the net effect of conversion from legacy telco interfaces to the SIP Trunking.

Skype for Business Hardware Requirements

A combination of centralized equipment and distributed equipment is required for a complete Skype for Business installation. Database servers are required to support the active directory and user lookup/presence functions. Standard servers are required to process the various forms of media content, including audio, video and desktop application sharing as well as signal conversion between different bit rates and device types.

Each user needs an audio device to use telephone functions; a number of choices are available. For employees who need a device that operates independent of a computer workstation, IP-enabled phones connect directly to the IP network and may have advanced capabilities such as color displays to provide directory information. Lower-end desktop devices may also have a look and feel similar to a traditional telephone handset with a dial pad, but are connected to a user's PC by a USB port and depend on the PC for directory and other functions. An even more economical option is to use a PC equipped with a USB headset where voice processing is done internally to the PC.

Optionally, a Survivable Branch Appliance (SBA) can be installed in one or more remote offices to provide local call processing support and to provide backup connections to the local public telephone network in the event that communications with the central servers are interrupted.

Skype for Business Software License Requirements Two different tiers of software licenses are required for a fully-equipped Skype for Business system: server licenses and user licenses. Server licenses are required for the central call processing and database management systems. A user software license is required either for the interoperability of Skype for business Solution within Industry.

Other Services

Configuring and installing a Skype for Business system can be complicated, so it may make sense to enlist the aid of a partner that has appropriate training and experience. In many cases, suitable suppliers of professional services can be found that are third-party Microsoft Partners.

Internal company IT resources can be used for planning, setup, testing and coordinating professional services in conjunction with the third-party partner. These resources can also be used to help train end users in system operation and routine usage.

Assuming that each end user will require four hours of training, the primary cost of teaching new users how to use Skype for Business will be complementary cost. In other words, the time spent in training classes by staff members will not be used by the company that they would perform. Interestingly, in the economic analysis provided later in this whitepaper, the opportunity costs of staff training is one of the largest components of the overall system installation cost which is met by the enterprise.

Hosted Services

As an alternative to purchasing hardware and acquiring software licenses, it is possible to lease Skype for Business services from a provider in a "hosted" environment. In this scenario, up-front costs are converted into monthly recurring costs, which may be financially preferable for some enterprises. Note that the cost impact of SIP Trunking does not depend on whether the Skype for Business system is purchased or hosted by a third party.

SIP Trunking Benefits

These calculations show the net impact of SIP Trunking – i.e. the difference between the costs of a legacy ISDN PRI/T-1 infrastructure and the costs of a SIP trunking infrastructure. As described earlier in this paper, the savings are due to eliminating many of the local access lines needed to support each location, pooling the Concurrent Call Paths (CCPs) across the organization, and virtually eliminating the costs of voice calls between locations. This calculation also adds the impact of a lower long distance (LD) rate, which will apply to most customers switching to Level 3 voice services. Note that the savings in each category are phased in over the three-year analysis period; this is based on the assumption that fifty percent of the potential yearly savings will be achieved in the first year, seventy-five percent in the second year, and the full amount of savings will be achieved in the third year.

Skype for Business Benefits

Descriptions of most of the benefits of installing a Skype for Business system were provided previously in this whitepaper. For these calculations, the assumptions on savings are intended to be conservative. For example, the analysis assumes that each person will spend only four hours per year using teleconference services.

This analysis adds cost savings due to the elimination of annual license/upgrade fees for the PBX, and removing the need for labour to perform adds/ moves/changes to the PBX extensions, since Skype for Business allows essentially unlimited mobility for users within the network. Also note that there are negative savings (i.e. costs) in Year 1 for Help Desk Call Reduction – this assumes that there will actually be more help desk calls in the first year of deployment as users become accustomed to the new system.As in the case of the SIP Trunking analysis, the benefits of Skype for Business increase over time as the deployment spreads throughout the enterprise. For example, the reduction of business travel was assumed to be five percent of the total corporate travel budget in years 2 and 3, but only half that amount in year 1.

Skype for Business Costs

The major cost factors of a Skype for Business system consists of hardware, software, services and training. Since most of these costs are incurred a project startup, the analysis shows these costs as part of startup. Of all the cost items, user training is the largest single expense; this is calculated based on four hours of opportunity cost for 5,000 employees at an average hourly wage of $65.00.

Actual Payback

Based on this scenario, the payback of the original investment will occur in just over ten months from the project launch. There are, however, a number of factors that can impact the actual payback period for an enterprise. One factor that can impact the payback period is the number of offices that are present in the organization – as this number increases, the savings from SIP Trunking increase rapidly, due to both the elimination of telco connections and due to the increased efficiency of sharing CCPs across the organization. One factor that could decrease the savings of presented in the scenario would be if the WAN bandwidth had to be significantly expanded to support the extra demands of the UC features and functions.

Conclusion

To create a meaningful business case, a variety of inputs are needed. First, a solid understanding of employee populations at each location is required, along with some concept of the calling patterns within the company and with parties outside the company. An understanding of current expenditures on services such as teleconferencing and web conferencing is also important to know.

Many times, the results of the analysis will show significant hard-cost savings and significant soft-cost savings. The hard savings (i.e. those that can be reliably quantified) are concentrated in the areas of cost avoidance, such as replacing legacy telephone company services with SIP Trunks. The soft-cost savings (i.e. those that are harder to quantify or may be small changes in large numbers) are more likely to come from productivity and efficiency improvements, which depend on the adoption rate of Skype for Business functionality by users in the enterprise. Fortunately, the hard-cost savings generated by the SIP Trunking service and reduced expenditures on teleconferencing and Web conferencing can go a long way towards justifying the costs of a Skype for Business installation, making the soft-cost savings of Skype for Business Unified Communications a welcome, but not essential, benefit of deployment.

Microsoft has a long history in the Unified Communications field, with products such as NetMeeting first released in 1996 as an add-on to Internet Explorer 3.0. Level 3 has been a pioneer inVoIP and SIP technology over the past 4 years, and is currently a leading provider in the United States. With over 125 patents issued and pending for VoIP and Soft Switch technology, Level 3 has the experience and capacity to process over 13 billion voice call minutes per month. Microsoft and Level 3 make an ideal team to handle UC and SIP requirements for any enterprise, large or small.

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