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VOICE, DATA & VIDEO TRANSMISSION & THEIR IMPACTS ON COMMUNICATION AS WELL AS SOCIATY

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Abstract- This paper mainly deals with the transmission & communication of voice, data, and video with the existing & more improved versions of equipments & processes.

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I am also thankful to my colleague as well as my trainer at Wbeidc to do this kind of project.

Index Terms :
EPABX,VOICE TRANSMISSION,PBX FUNCTIONS, INTERFACE STANDARDS,HOSTED PBX,IP-PBX, DATA, VPN, VIDEO, IMPACTS
.

1 INTRODUCTION

Communication on the IP network is inherently less reliable in contrast to the circuit-switched public telephone network, as it does not provide a network-based mechanism to ensure that data packets are not lost, or delivered in sequential order. It is a best-effort network without fundamental Quality of Service (QoS) guarantees. Therefore, VoIP implementations may face problems mitigating latency and jitter.

By default, IP routers handle traffic on a first-come, first-served basis. Routers on high volume traffic links may introduce latency that exceeds permissible thresholds for VoIP. Fixed delays cannot be controlled, as they are caused by the physical distance the packets travel; however, latency can be minimized by marking voice packets as being delay-sensitive with methods such as DiffServ.

A VoIP packet usually has to wait for the current packet to finish transmission, although it is possible to preempt (abort) a less important packet in mid-transmission, although this is not commonly done, especially on high-speed links where transmission times are short even for maximum-sized packets. An alternative to preemption on slower links, such as dialup and DSL, is to reduce the maximum transmission time by reducing the maximum transmission unit. But every packet must contain protocol headers, so this increases relative header overhead on every link along the user's Internet paths, not just the bottleneck (usually Internet access) link.

ADSL modems provide Ethernet (or Ethernet over USB) connections to local equipment, but inside they are actually ATM modems. They use AAL5 to segment each Ethernet packet into a series of 48-byte ATM cells for transmission and reassemble them back into Ethernet packets at the receiver. A virtual circuit identifier (VCI) is part of the 5-byte header on every ATM cell, so the transmitter can multiplex the active virtual circuits (VCs) in any arbitrary order. Cells from the *same* VC are always sent sequentially.

However, the great majority of DSL providers use only one VC for each customer, even those with bundled VoIP service. Every Ethernet packet must be completely transmitted before another can begin. If second PVC were established, given high priority and reserved for VoIP, then a low priority data packet could be suspended in mid-transmission and a VoIP packet sent right away on the high priority VC. Then the link would pick up the low priority VC where it left off. Because ATM links are multiplexed on a cell-by-cell basis, a high priority packet would have to wait at most 53 byte times to begin transmission. There would be no need to reduce the interface MTU and accept the resulting increase in higher layer protocol overhead, and no need to abort a low priority packet and resend it later.

This doesn't come for free. ATM has substantial header overhead: 5/53 = 9.4%, roughly twice the total header overhead of a 1500 byte TCP/IP Ethernet packet (with TCP timestamps). This "ATM tax" is incurred by every DSL user whether or not he takes advantage of multiple virtual circuits - and few can.

ATM's potential for latency reduction is greatest on slow links, because worst-case latency decreases with increasing link speed. A full-size (1500 byte) Ethernet frame takes 94 ms to transmit at 128 kb/s but only 8 ms at 1.5 Mb/s. If this is the bottleneck link, this latency is probably small enough to ensure good VoIP performance without MTU reductions or multiple ATM PVCs. The latest generations of DSL, VDSL and VDSL2, carry Ethernet without intermediate ATM/AAL5 layers, and they generally support IEEE 802.1p priority tagging so that VoIP can be queued ahead of less time-critical traffic.

Voice, and all other data, travels in packets over IP networks with fixed maximum capacity. This system is more prone to congestion[*citation needed*] and DoS attacks[13] than traditional circuit switched systems; a circuit switched system of insufficient capacity will refuse new connections while carrying the remainder without impairment, while the quality of real-time data such as telephone conversations on packet-switched networks degrades dramatically.

Fixed delays cannot be controlled as they are caused by the physical distance the packets travel. They are especially problematic when satellite circuits are involved because of the long distance to a geostationary satellite and back; delays of 400-600 ms are typical.

When the load on a link grows so quickly that its queue overflows, congestion results and data packets are lost. This signals a transport protocol like TCP to reduce its transmission rate to alleviate the congestion. But VoIP usually does not use TCP because recovering from congestion through retransmission usually entails too much latency. So QoS mechanisms can avoid the undesirable loss of VoIP packets by immediately transmitting them ahead of any queued bulk traffic on the same link, even when that bulk traffic queue is overflowing.

The receiver must resequence IP packets that arrive out of order and recover gracefully when packets arrive too late or not at all. Jitter results from the rapid and random (i.e., unpredictable) changes in queue lengths along a given Internet path due to competition from other users for the same transmission links. VoIP receivers counter jitter by storing incoming packets briefly in a "de-jitter" or "playout" buffer, deliberately increasing latency to increase the chance that each packet will be on hand when it's time for the voice engine to play it. The added delay is thus a compromise between excessive latency and excessive dropout, i.e., momentary audio interruptions.

Although jitter is a random variable, it is the sum of several other random variables that are at least somewhat independent: the individual queuing delays of the routers along the Internet path in question. Thus according to the central limit theorem, we can model jitter as a gaussian random variable. This suggests continually estimating the mean delay and its standard deviation and setting the playout delay so that only packets delayed more than several standard deviations above the mean will arrive too late to be useful. In practice, however, the variance in latency of many Internet paths is dominated by a small number (often one) of relatively slow and congested "bottleneck" links. Most Internet backbone links are now so fast (e.g. 10 Gb/s) that their delays are dominated by the transmission medium (e.g. optical fiber) and the routers driving them do not have enough buffering for queuing delays to be significant.

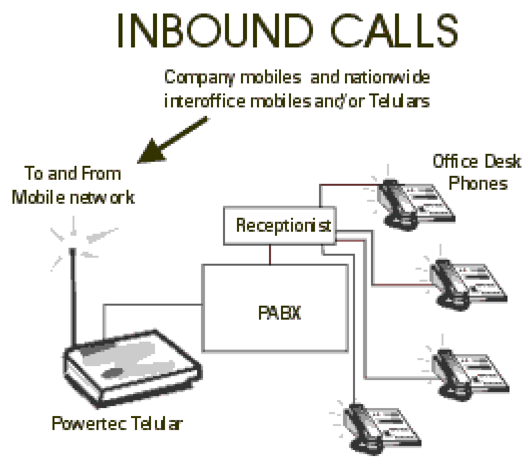
It has been suggested to rely on the packetized nature of media in VoIP communications and transmit the stream of packets from the source phone to the destination phone simultaneously across different routes (multi-path routing). In such a way, temporary failures have less impact on the communication quality. In capillary routing it has been suggested to use at the packet level Fountain codes or particularly raptor codes for transmitting extra redundant packets making the communication more reliable.

A number of protocols have been defined to support the reporting of QoS/QoE for VoIP calls. These include RTCP Extended Report (RFC 3611), SIP RTCP Summary Reports, H.460.9 Annex B (for H.323), H.248.30 and MGCP extensions. The RFC 3611 VoIP Metrics block is generated by an IP phone or gateway during a live call and contains information on packet loss rate, packet discard rate (because of jitter), packet loss/discard burst metrics (burst length/density, gap length/density), network delay, end system delay, signal / noise / echo level, Mean Opinion Scores (MOS) and R factors and configuration information related to the jitter buffer.

RFC 3611 VoIP metrics reports are exchanged between IP endpoints on an occasional basis during a call, and an end of call message sent via SIP RTCP Summary Report or one of the other signaling protocol extensions. RFC 3611 VoIP metrics reports are intended to support real time feedback related to QoS problems, the exchange of information between the endpoints for improved call quality calculation and a variety of other applications.

2 EPABX (Electronic Private Automated Branched Exchange)

A term used synonymously with Electronic Private Automated Branched Exchange(EPABX) & Private Automated Branch Exchange(PABX).Similar to a central office exchange but smaller.A central office exchange can accomodatye 10,000 subscribers.PBX systems are typically designed to accommodate from 20-10,000 subscribers or stations.
Residing in an enterprise, a PBX is a piece of equipment that is responsible for switching calls between enterprise users.The PBX allows the users to share a specific no of external phone lines, saving the added cost of having an external phone line for each user.
With the help of these Piece of telecommunication equipments installed in a company's office which makes it possible to connct many internal telephone lines to city telephone lines.
A PBX(also caleed Private Business Exchange or PABX for Private Automatic Branch Exchange) is a telephone exchange that serves aparticular business or office as opposed to one a common carrier or telephone company operates for many business(es) or for the general public.



PBXs make connections among the internal telephones of a private organization—usually a business—and also connects them to the public switched telephone network (PSTN) via trunk lines. Because they incorporate telephones, fax machines, modems, and more, the general term "extension" is used to refer to any end point on the branch.

PBXs are differentiated from "key systems" in that users of key systems manually select their own outgoing lines, while PBXs select the outgoing line automatically. Hybrid systems combine features of both.

Initially, the primary advantage of PBXs was cost savings on internal phone calls: handling the circuit switching locally reduced charges for local phone service. As PBXs gained popularity, they started offering services that were not available in the operator network, such as hunt groups, call forwarding, and extension dialing. In the 1960s a simulated PBX known as Centrex provided similar features from the central telephone exchange.

3 Voice Transmission

Voice over Internet Protocol (**Voice over IP**, **VoIP**) is any of a family of methodologies, communication protocols, and transmission technologies for delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. Other terms frequently encountered and often used synonymously with VoIP are *IP telephony*, *Internet telephony*, *voice over broadband* (VoBB), *broadband telephony*, and *broadband phone*.

Internet telephony refers to communications services — voice, fax, SMS, and/or voice-messaging applications — that are transported via the Internet, rather than the public switched telephone network (PSTN). The steps involved in originating a VoIP telephone call are signaling and media channel setup, digitization of the analog voice signal, optionally compression, packetization, and transmission as Internet Protocol (IP) packets over a packet-switched network. On the receiving side similar steps reproduce the original voice stream.[1]

VoIP systems employ session control protocols to control the set-up and tear-down of calls as well as audio codecs which encode speech allowing transmission over an IP network as digital audio via an audio stream. The codec used is varied between different implementations of VoIP (and often a range of codecs are used); some implementations rely on narrowband and compressed speech, while others support high fidelity stereo codecs.

4 IMPACTS OF THE TECHNOLOGY

Susceptibility to power cut

Telephones for traditional residential analog service are usually connected directly to telephone company phone lines which provide direct current to power most basic analog handsets independently of locally available power.

IP Phones and VoIP telephone adapters connect to routers or cable modems which typically depend on the availability of mains electricity or locally generated power. Some VoIP service providers use customer premise equipment (e.g., cablemodems) with battery-backed power supplies to assure uninterrupted service for up to several hours in case of local power failures. Such battery-backed devices typically are designed for use with analog handsets.

PSTN integration

E.164 is a global FGFnumbering standard for both the PSTN and PLMN. Most VoIP implementations support E.164 to allow calls to be routed to and from VoIP subscribers and the PSTN/PLMN. VoIP implementations can also allow other identification techniques to be used. For example, Skype allows subscribers to choose "Skype names" (usernames) whereas SIP implementations can use URIs similar to email addresses. Often VoIP implementations employ methods of translating non-E.164 identifiers to E.164 numbers and vice-versa, such as the Skype-In service provided by Skype and the ENUM service in IMS and SIP.

Echo can also be an issue for PSTN integration. Common causes of echo include impedance mismatches in analog circuitry and acoustic coupling of the transmit and receive signal at the receiving end.

Security

VoIP telephone systems are susceptible to attacks as are any internet-connected devices. This means that hackers who know about these vulnerabilities (such as insecure passwords) can institute denial-of-service attacks, harvest customer data, record conversations and break into voice mailboxes.

Another challenge is routing VoIP traffic through firewalls and network address translators. Private Session Border Controllers are used along with firewalls to enable VoIP calls to and from protected networks. For example, Skype uses a proprietary protocol to route calls through other Skype peers on the network, allowing it to traverse symmetric NATs and firewalls. Other methods to traverse NATs involve using protocols such as STUN or ICE.

Many consumer VoIP solutions do not support encryption, although having a secure phone is much easier to implement with VoIP than traditional phone lines. As a result, it is relatively easy to eavesdrop on VoIP calls and even change their content. An attacker with a packet sniffer could intercept your VoIP calls if you are not on a secure VLAN.

There are open source solutions, such as Wireshark, that facilitate sniffing of VoIP conversations. A modicum of security is afforded by patented audio codecs in proprietary implementations that are not easily available for open source applications [; however, such security through obscurity has not proven effective in other fields. Some vendors also use compression, which may make eavesdropping more difficult.[*citation needed*] However, real security requires encryption and cryptographic authentication which are not widely supported at a consumer level. The existing security standard Secure Real-time Transport Protocol (SRTP) and the new ZRTP protocol are available on Analog Telephone Adapters (ATAs) as well as various softphones. It is possible to use IPsec to secure P2P VoIP by using opportunistic encryption. Skype does not use SRTP, but uses encryption which is transparent to the Skype provider. In 2005, Skype invited a researcher, Dr Tom Berson, to assess the security of the Skype software, and his conclusions are available in a published report.

The Voice VPN solution provides secure voice for enterprise VoIP networks by applying IPSec encryption to the digitized voice stream.

Securing VoIP

To prevent the above security concerns government and military organizations are using Voice over Secure IP (VoSIP), Secure Voice over IP (SVoIP), and Secure Voice over Secure IP (SVoSIP) to protect confidential and classified VoIP communications. Secure Voice over IP is accomplished by encrypting VoIP with Type 1 encryption. Secure Voice over Secure IP is accomplished by using Type 1 encryption on a classified network, like SIPRNet. Public Secure VoIP is also available with free GNU programs.

TELEMEDICINE

Telemedicine is a rapidly developing application of clinical medicine where medical information is transferred through interactive audiovisual media for the purpose of consulting, and sometimes remote medical procedures or examinations.

Telemedicine may be as simple as two health professionals discussing a case over the telephone, or as complex as using satellite technology and videoconferencing equipment to conduct a real-time consultation between medical specialists in two different countries. Telemedicine generally refers to the use of communications and information technologies for the delivery of clinical care.

Care at a distance (also called *in absentia* care), an old practice which was often conducted via post. There has been a long and successful history of in absentia health care which, thanks to modern communication technology, has evolved into what we know as modern telemedicine.

In its early manifestations, African villagers used smoke signals to warn people to stay away from the village in case of serious disease. In the early

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1900s, people living in remote areas in Australia used two-way radios, powered by a dynamo driven by a set of bicycle pedals, to communicate with the Royal Flying Doctor Service of Australia.

The terms eHealth and telehealth are at times incorrectly interchanged with telemedicine. Like the terms "medicine" and "health care", telemedicine often refers only to the provision of clinical services while the term telehealth can refer to clinical and non-clinical services such as medical education, administration, and research. The term eHealth is often, particularly in the U.K. and Europe, used as an umbrella term that includes telehealth, electronic medical records, and other components of health IT.

DISTANCE EDUCATION

In the twentieth century, radio, television, and the Internet have all been used to further distance education. Computers and the Internet have made distance-learning distribution easier and faster. The private, for-profit University of Phoenix, which is primarily an online university, now has 200,000 students and expects to serve 500,000 by 2010, yet little is known about student success or lack of success in such a fast-growing institution. In 1996, cable pioneer Glenn Jones and Bernard Luskin launched Jones International University as the first accredited fully online university accredited by a regional accrediting association. JUI is accredited by the North Central Association of Schools and Colleges.

EDUSAT or GSAT-3 was launched on 2004-09-20 by the Indian Space Research Organisation. EDUSAT is the first Indian satellite built exclusively to serve the educational sector. It is mainly intended to meet the demand for an interactive satellite-based distance education system for the country.

SOFTWARE AS A SERVICE

Software as a service (**SaaS**, typically pronounced [sæs]), sometimes referred to as "software on demand," is software that is deployed over the internet and/or is deployed to run behind a firewall on a local area network or personal computer. With SaaS, a provider licenses an application to customers either as a service on demand, through a subscription, in a "pay-as-you-go" model, or (increasingly) at no charge. This approach to application delivery is part of the utility computing model where all of the technology is in the "cloud" accessed over the Internet as a service.

Advantages

- Pay per use
- Anytime, anywhere accessibility
- Pay as you go
- Instant scalability
- Security
- Reliability
- APIs

SaaS was initially widely deployed for sales force automation and Customer Relationship Management (CRM). Now it has become commonplace for many business tasks, including accounting software, computerized billing, ERP software, invoicing, human resource management, financials, content management, collaboration, document management, and service desk management.

PBX functions

Functionally, the PBX performs four main call processing duties:

- Establishing connections (circuits) between the telephone sets of two users (e.g. mapping a dialled number to a physical phone, ensuring the phone isn't already busy)
- Maintaining such connections as long as the users require them (i.e. channelling voice signals between the users)
- Disconnecting those connections as per the user's requirement
- Providing information for accounting purposes (e.g. metering calls)

In addition to these basic functions, PBXs offer many other calling features and capabilities, with different manufacturers providing different features in an effort to differentiate their products. Common capabilities include (manufacturers may have a different name for each capability):

- Auto attendant
- Auto dialing
- Automatic call distributor
- Automated directory services (where callers can be routed to a given employee by keying or speaking the letters of the employee's name)
- Automatic ring back
- Call accounting
- Call blocking
- Call forwarding on busy or absence
- Call park
- Call pick-up
- Call transfer
- Call waiting
- Camp-on
- Conference call
- Custom greetings

- Customised Abbreviated dialing (Speed Dialing)
- Busy Override
- Direct Inward Dialing
- Direct Inward System Access (DISA) (the ability to access internal features from an outside telephone line)
- Do not disturb (DND)
- Follow-me, also known as find-me: Determines the routing of incoming calls. The exchange is configured with a list of numbers for a person. When a call is received for that person, the exchange routes it to each number on the list in turn until either the call is answered or the list is exhausted (at which point the call may be routed to a voice mail system).
- Interactive voice response
- Music on hold
- Night service
- Public address voice paging
- Shared message boxes (where a department can have a shared voicemail box)
- Voice mail

Interface standards

Interfaces for connecting extensions to a PBX include:

- POTS (plain old telephone service) - the common two-wire interface used in most homes. This is cheap and effective, and allows almost any standard phone to be used as an extension.
- Proprietary - the manufacturer has defined a protocol. One can only connect the manufacturer's sets to their PBX, but the benefit is more visible information displayed and/or specific function buttons.
- Digital Enhanced Cordless Telecommunications (DECT) - a standard for connecting cordless phones.
- Internet Protocol - For example, H.323 and SIP.

Interfaces for connecting PBXs to each other include:

- Proprietary protocols - if equipment from several manufacturers is on site, the use of a standard protocol is required.
- ISDN PRI - Runs over T1, 23 bearer channels + 1 signalling channel
- QSIG - for connecting PBXs to each other, usually runs over T1 (T-carrier) or E1 (E-carrier) physical circuits.
- DPNSS - for connecting PBXs to trunk lines. Standardized by British Telecom, this usually runs over E1 (E-carrier) physical circuits.
- Internet Protocol - H.323, SIP and IAX protocols are IP based solutions which can handle voice and multimedia (e.g. video) calls.

Interfaces for connecting PBXs to trunk lines include:

- standard POTS (plain old telephone service) lines - the common two-wire interface used in most domestic homes. This is adequate only for smaller systems, and can suffer from not being able to detect incoming calls when trying to make an outbound call.
- ISDN - the most common digital standard for fixed telephony devices. This can be supplied in either Basic (2 circuit capacity) or Primary (24 or 30 circuit capacity) versions. Most medium to large companies would use Primary ISDN circuits carried on T1 or E1 physical connections.
- RBS (robbed bit signaling) - delivers 24 digital circuits over a four-wire (T1) interface.
- Internet Protocol - H.323, SIP, MGCP, and Inter-Asterisk eXchange protocols operate over IP and are supported by some network providers.

Interfaces for collecting data from the PBX:

- Serial interface - historically used to print every call record to a serial printer. Now an application connects via serial cable to this port.
- Network Port (listen mode) - where an external application connects to the TCP or UDP port. The PBX then starts streaming information down to the application.
- Network port (server mode) - the PBX connects to another application or buffer.
- File - the PBX generates a file containing the call records from the PBX.

The call records from the PBX are called SMDR, CDR, or CIL. It is possible to use a Voice modem as FXO card.

Hosted PBX systems

A **hosted PBX** system delivers PBX functionality as a service, available over the Public Switched Telephone Network (PSTN) and/or the internet. Hosted PBXs are typically provided by the telephone company, using equipment located in the premises of the telephone company's exchange. This means the customer organization doesn't need to buy or install PBX equipment (generally the service is provided by a lease agreement) and the telephone company can (in some configurations) use the same switching equipment to service multiple PBX hosting accounts.

Instead of buying PBX equipment, users contract for PBX services from a hosted PBX service provider, a particular type of application service provider (ASP). The first hosted PBX service was very feature-rich compared to most premise-based systems of the time. In fact, some PBX functions, such as follow-me calling, appeared in a hosted service before they became available in hardware PBX equipment. Since that introduction, updates and new offerings from several companies have moved feature sets in both directions. Today, it is possible to get hosted PBX service that includes far more features than were available from the first systems of this class, or to contract with companies that provide less functionality for simple needs.

In addition to the features available from premises-based PBX systems, hosted-PBX:

- Allows a single number to be presented for the entire company, despite its being geographically distributed. A company could even choose to have no premises, with workers connected from home using their domestic telephones but receiving the same features as any PBX user.
- Allows multimodal access, where employees access the network via a variety of telecommunications systems, including POTS, ISDN, cellular phones, and VOIP. This allows one extension to ring in multiple locations (either concurrently or sequentially).
- Supports integration with custom toll plans (that allow intra company calls, even from private premises, to be dialed at a cheaper rate) and integrated billing and accounting (where calls made on a private line but on the company's behalf are billed centrally to the company).
- Eliminates the need for companies to manage or pay for on-site hardware maintenance.
- Allows scalability so that a larger system is not needed if new employees are hired, and so that resources are not wasted if the number of employees is redu

A **mobile PBX** is a hosted PBX service that extends fixed-line PBX functionality to mobile devices such as cellular handsets, smartphones and PDA phones by provisioning them as extensions. Mobile PBX services also can include fixed-line phones. Mobile PBX systems are different from other hosted PBX systems that simply forward data or calls to mobile phones by allowing the mobile phone itself, through the use of buttons, keys and other input devices, to control PBX phone functions and to manage communications without having to call into the system first.

A mobile PBX may exploit the functionality available in smartphones to run custom applications to implement the PBX specific functionality.

In addition, a mobile PBX may create extension identifiers for each handset that allow dialing other cell phones in the PBX via their extension shortcut, instead of a PSTN number.

IP-PBX

An IP PBX handles voice signals under Internet protocol, bringing benefits for computer telephony integration (CTI). An IP-PBX can exist as physical hardware, or can carry out its functions virtually, performing the call-routing activities of the traditional PBX or key system as a software system. The virtual version is also called a "Soft PBX".

Important informations :

- Basic bandwidth reservation for Voice can be implemented on setting:
  - Outgoing Bandwidth Control.
  - B-Channels limitation.
- No external Firewall is required at site
- If, at some point, *normal* Internet traffic (like Web navigation, Mail server access, etc.) is required by the Customer, then VPN Site-to-Site networking implementation must be designed according to VPN with normal Internet traffic scenario. In that case:
  - Customer need to ensure itself with additional ISP connection to Internet.
  - Customer need to ensure additional devices/services like:
    - Router, necessary to route LAN hosts normal traffic to Internet.
    - Firewall, necessary to protect LAN hosts and implement DMZ at site.
    - Proxy, necessary to implement Internet traffic policies.

VPN with normal Internet traffic :

DATA:

- Normal Internet traffic completely separated from Voice traffic by means of second Internet connection to ISP:
  - Voice quality over VPN is preserved from normal Internet traffic side effects (No Voice/Date traffic congestion, no burstly data traffic or high downstream can jam Voice connections).
  - Manage VPN Bandwidth entirely (No Bandwith share with normal Internet traffic).
  - Lightweight processing power requirements handling and routing only Voice traffic over VPN Trunk.
- Routing/Security policies demanded to external Proxy/Firewall devices for normal unencrypted Internet traffic handling.

VPN with normal Internet traffic and QoS support :

Important informations:

- The ISP must support QoS to ensure sufficient voice packet quality (According to specific SLA)
- VPN Site-to-Site gateway to a single ISP access, used for both:
  - Voice and Data traffic.
  - Normal Internet traffic.
- a cost-effective gateway to route:
  - VoIP traffic by performing CorNet-IP trunking between sites over VPN trunk.
  - Normal Internet traffic to ISP.
  - Traffic to DMZ at site.
- External additional Proxy/Firewall:
  - must be capable to handle traffic carried by VPN transparently.
  - offer high level of Security for normal unencrypted Internet traffic
- Acts as gateway to access enterprise resources and applications available on the company's VPN networked sites.
- The ISP access can be overloaded (Available bandwidth could be used by massive or bursty downloads) affecting Voice quality first (Bursty data traffic, high downstream can jam Voice connections).
- Operation of public servers on (DMZ) is only recommended QoS can be provided by ISP and access Router is able to support bandwidth control for both Voice and Data traffic.
- Not all ISPs are able nor have a suitable infrastructure ready to offer adequate/any QoS features on their Internet access.

Echo cancellation :

A fundamental feature of professional videoconferencing systems is acoustic echo cancellation (AEC). Echo can be defined as the reflected source wave interference with new wave created by source. AEC is an algorithm which is able to detect when sounds or utterances reenter the audio input of the videoconferencing codec, which came from the audio output of the same system, after some time delay. If unchecked, this can lead to several problems including:

1. the remote party hearing their own voice coming back at them (usually significantly delayed)
2. strong reverberation, rendering the voice channel useless as it becomes hard to understand and
3. howling created by feedback. Echo cancellation is a processor-intensive task that usually works over a narrow range of sound delays.

CLOUD COMPUTING

Cloud computing is Internet-based computing, whereby shared resources, software, and information are provided to computers and other devices on demand, as with the electricity grid.

Cloud computing is a natural evolution of the widespread adoption of virtualization, Service-oriented architecture and utility computing. Details are abstracted from consumers, who no longer have need for expertise in, or control over, the technology infrastructure "in the cloud" that supports them. Cloud computing describes a new supplement, consumption, and delivery model for IT services based on the Internet, and it typically involves over-the-Internet provision of dynamically scalable and often virtualized resources. It is a byproduct and consequence of the ease-of-access to remote computing sites provided by the Internet. This frequently takes the form of web-based tools or applications that users can access and use through a web browser as if it were a program installed locally on their own computer. NIST provides a somewhat more objective and specific definition here. The term "cloud" is used as a metaphor for the Internet, based on the cloud drawing used in the past to represent the telephone network, and later to depict the Internet in computer network diagrams as an abstraction of the underlying infrastructure it represents. Typical cloud computing providers deliver common business applications online that are accessed from another Web service or software like a Web browser, while the software and data are stored on servers.

Several other effect on society

1. *Impact on medicine and health*
2. *Impact on education*
3. *Impact on business*
4. *Impact on law :*
5. *Impact on media relations :*

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