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Voice over Internet Protocol – The Technology and Its Application

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ABSTRACT:- The paper discusses the implementation of Voice over IP (VoIP). The objective of this paper is to identify and characterize the primary issues that must be addressed to define a large scale VoIP network that is capable of supporting (but not necessarily limited to) full PSTN equivalence. It presents a high-level network architecture that acts as a starting point from which PSTN equivalent VoIP networks can be built The term "Voice over IP" (VoIP) describes the transport of voice over IP based networks, it is a generic term that covers deployments ranging in complexity from hobbyists using the internet to get free phone calls on a peer to peer basis or we can say end to end basis, to full scale PSTN replacement networks.

Voice over Internet Protocol (VOIP) refers to the transmission of speech across data-style networks. This form of transmission is conceptually superior to conventional circuit switched communication in many ways. This publication introduces VOIP, it's working, its all features, architecture, all issues of network with network interconnection, explaining its advantages and disadvantages with applications, etc.

Index Terms—VoIP, BICC, ISDN, MGCP, MPLS, PSTN, SS7, SIP, TDM

1 INTRODUCTION

VoIP refers only to the underlying transport protocol that encapsulates voice traffic or voice media streams and allows them to be carried over data networks, using IP network technologies or internet protocols. VoIP, however, is not IP Telephony, nor is it the more widely used industry terminology called IP Communications that refers to an even broader definition of communications networking applications and technologies. VoIP can be understood as simply a transport protocol for carrying voice over any packet network, usually between sites. The term convergence, also sometimes referred as a multi-service network, refers to the integration of data, voice, and video solutions onto a converged network infrastructure.

2 OVERVIEW OF VOIP

At its simplest, Voice over IP is the transport of voice using the Internet Protocol (IP), however this broad term hides a multitude of deployments and functionality and it is useful to look in more detail at what VoIP is being used for today. Currently the following types of VoIP applications are in use:

- Private users who are using voice over IP for end to end phone calls over the public internet. These users typically trade quality, features and reliability for the fact that the service is very low cost and are generally happy with the service. Although globally the numbers of users taking advantage of this technology is large the density of such users is very low and when compared with the PSTN the call volumes are negligible.
- Business users on private networks provided by telecom and datacom providers. These services offer relatively high quality and reliability and are feature rich but come at a price. When compared with the PSTN the call volumes supported by these services are small, however such services are nonetheless commercially successful.
- IP trunking solutions used by long haul Voice providers. Typically these offerings At its simplest, Voice over IP is the transport of voice using the Internet Protocol (IP), however this broad-term hides the multitude of deployments and functionality and it is useful to look in more detail at what VoIP is being used for today. Currently the following types of VoIP applications are in use:
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  - Business users on private networks provided by telecom and datacom providers. These services offer relatively high quality and reliability and are feature rich but come at a price. When compared with the PSTN the call volumes supported by these services are small, however such services are nonetheless commercially successful.
  - IP trunking solutions used by long haul Voice providers. Typically these offerings use private IP networks to connect islands of the PSTN

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together, e.g. a low cost way of calling the USA from the UK. Customers access these services using traditional black phones but the voice is carried over an IP network. Although these voice over IP deployments have been successful and each will continue to have its place in the future they have not yet faced the issue of how the wider PSTN could be migrated to an end-to-end voice over IP infrastructure. Providing a voice over IP solution that will scale to PSTN call volumes, offer PSTN call quality and equivalent services, as well as supporting new and innovative services is a significant challenge.

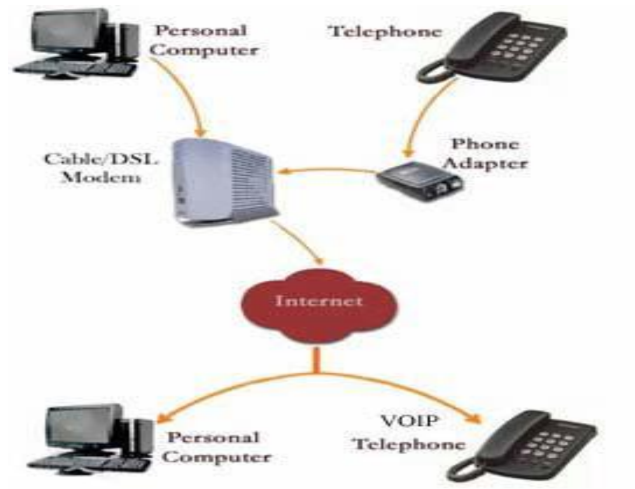
3 VoIP Working

VoIP converts the voice signal from your telephone into a digital signal that can travel over the Internet. If you are calling a regular telephone number, the signal is then converted back at the other end. Depending on the type of VoIP service, you can make a VoIP call from a computer, a special VoIP phone, or a traditional phone with or without an adapter. In addition, new wireless "hot spots" in public locations such as airports, parks and cafes allow you to connect to the Internet, and may enable you to use VoIP service wirelessly. If your VoIP service provider assigns you a regular telephone number, then you can receive calls from regular telephones that don't need special equipment, and most likely you'll be able to dial just as you always have.

3.1. Equipment:

- Broadband Internet connection
- Computer
- a special VoIP telephone or a regular telephone with an adapter.

3.2. One example of VoIP service works:



4 Network Architecture

VoIP can be deployed in many different network segments. To date, it has been mostly deployed in the backbone and enterprise networks.

This section describes the function of the network components listed in Figure 2. Depending upon the particular network architecture some of these network components may be combined into a single solution, for example a combined signaling and trunking gateway.

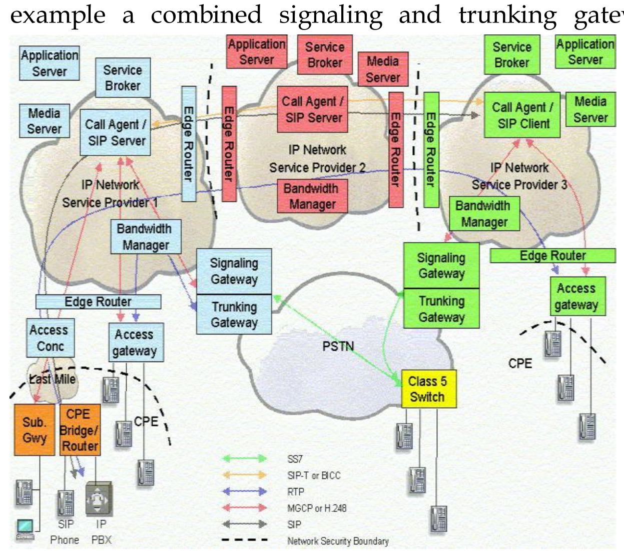


Figure 2: Next Generation VoIP Network

4.1. Call Agent/SIP Server/SIP Client

The Call Agent/SIP Server/SIP Client is located in the service provider's network and provides call logic and call control functions, typically maintaining call state for every call in the network. Many call agents include service logic for supplementary services, e.g. Caller ID, Call Waiting, and also interact with application servers to supply services that are not directly hosted on call agent. The Call Agent will participate in signaling and device control flows originating, terminating or forwarding messages. There are numerous relevant protocols depending upon the network architecture including SIP, SIP-T, H.323, BICC, H.248, MGCP/NCS, SS7, AIN, ISDN, etc. Call Agents also produce details of each call to support billing and reconciliation. A SIP Server provides equivalent function to a Call Agent in a SIP signaling network, its primary roles are to route and forward SIP requests, enforce policy (for example call admission control) and maintain call details records. For example the SIP Server in Service Provider 1's network will route and forward SIP requests from SIP Phones belonging to customers. A SIP Client provides similar function to a SIP Server, but originates or terminates SIP signaling rather than forwarding it to a SIP Phone or other CPE device. For example a SIP call is shown in figure 2 between a SIP Phone in Service Provider 1's network and the SIP server in Service Provider 3's network. The Call Agent/SIP Server terminates the SIP signaling and converts it to H.248 or MGCP to set up a call to the correct subscriber. Call Agents are also known as Media Gateway Controllers, Soft switches and Call Controllers. All these terms convey a slightly different emphasis but maintaining call state is the common function.

4.2 Service Broker

The service broker is located on the edge of the service provider service network and provides the service distribution, coordination, and control between application servers, media servers, call agents and services that may exist on alternate technologies (i.e. Parlay Gateways and SCP s). The service broker allows a consistent repeatable approach for controlling applications in conjunction with their service data and media resources to enable services to allow services to be reused with other services to create new value added services.

4.3. Application Server

The Application Server is located in the service provider's network and provides the service logic and execution for one or more applications or services that are not directly hosted on the Call Agent. For example, it may provide voice mail or conference calling facilities. Typically the Call Agent will route calls to the appropriate application server when a service is invoked that the Call Agent cannot itself support.

4.4. Media Server

This Media Server is located in the service provider's network. It is also referred to as an announcement server. For voice services, it uses a control protocol, such as H.248 (Megaco) or MGCP, under the control of the call agent or application server. Some of the functions the Media Server can provide are

- playing announcements
- Mixing – providing support for 3-way calling etc
- Codec transcending and voice activity detection
- Tone detection and generation
- Interactive voice response (IVR) Processing
- Fax processing.

4.5. Signaling Gateway

The Signaling Gateway is located in the service provider's network and acts as a gateway between the call agent signaling and the SS7-based PSTN. It can also be used as a signaling gateway between different packets based carrier domains. It may provide signaling translation, for example between SIP and SS7 or simply signaling transport conversion e.g. SS7 over IP to SS7 over TDM.

4.6. Trunking Gateway

The Trunking Gateway is located in the service provider's network and as a gateway between the carrier IP network and the TDM (Time Division Multiplexing)-based PSTN. It provides trans-coding from the packet based voice, VoIP onto a TDM network. Typically, it is under the control of the Call Agent/Media Gateway Controller through a device control protocol such as H.248 (Megaco) or MGCP.

4.7. Access Gateway

The Access Gateway is located in the service provider's network. It provides support for POTS phones and typically, it is under the control of the Call Agent / Media Gateway Controller through a device control protocol such as H.248 (Megaco) or MGCP.

4.8. Access Concentrator

The Access Concentrator is located in the service provider's network and terminates the service provider end of the WAN links used over the "last mile".

4.9. Bandwidth Manager

The Bandwidth Manager is located in the service provider's network and is responsible for providing the required QoS from the network. It is responsible for the setting up and tearing down of bandwidth within the network and for controlling the access of individual calls to this bandwidth. It is responsible for installing the appropriate Policy in edge routers to police the media flows on a per call basis.

4.10. Edge Router

The Edge Router is located in the service provider's network and routes IP traffic onto the carrier backbone network. Typically the edge router will provide many other functions and can be combined with the Access Concentrator.

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4.11. Subscriber Gateway

The Subscriber Gateway is located at the customer premises and terminates the WAN (Wide Area Network) link (DSL, T1, fixed wireless, cable etc) at the customer premises and typically provides both voice ports and data connectivity. Usually, it uses a device control protocol, such as H.248 (Megaco) or MGCP/NCS, under the control of the Call Agent. It provides similar function to the Access Gateway but typically supports many fewer voice ports. Subscriber Gateways are also known as IADs, Residential Gateways, or MTAs (in acable network).

4.12. Bridge/Router

The Bridge/Router is located at the customer premises and terminates the WAN (Wide Area Network) link (DSL, T1, fixed wireless, cable etc) at the customer premises. The difference between this and the Subscriber Gateway is a bridge/router does not provide any native voice support, although voice services for example SIP phones, can be bridged/routed via this device.

4.13. IP Phone/PBX

IP Phones and PBX systems are located at customer premises and provide voice services. They interact with the Call Agent/SIP Server using a signaling protocol such as SIP, H.323 or a device control protocol such as H.248 (Megaco) or MGCP.

5 ISSUES IN A VOIP NETWORK

There are several issues that need to be addressed in order to provide a toll-quality, PSTN equivalent end-toend VoIP network. These include:

5.1. Choice of Signaling Protocols

Numerous different signaling protocols have been developed that are applicable to a VoIP solution. They include:

- Device control protocols such as H.248 (Megaco), MGCP, NCS, etc
- Access services signaling protocols such as SIP, H.323, etc
- Network service signaling protocols such as SIP, SIP-T, BICC, CMSS, etc The choice of which protocol to use in a service provider network is dependent upon both the service set being offered and the equipment available to provide these services. For example a network must support SIP in order to provide access to SIP phones.

5.2. Quality of Service

One of the key requirements for the widespread deployment of VoIP is the ability to offer a toll quality service equivalent to the existing PSTN. Indeed some carriers are even looking for Next-Generation Networks as a means for delivering much higher voice quality as a service.
Perceived Voice quality is very sensitive to three key performance criteria in a packet network, in particular:

- Delay
- Jitter
- Packet loss

IP, by its nature, provides a best-effort service and does not provide guarantees about the key criteria. Therefore it is necessary to implement a suitable QoS solution in the majority of cases where simple over provisioning cannot guarantee success. There are a large number of technologies that can be chosen to provide QoS support such as Diffserv, RSVP, MPLS and even ATM. However the objective of such a solution is always to guarantee prioritization of voice media streams over best-effort data, and to ensure that the voice service is not compromised by unforeseen traffic patterns.

5.3. Reliability/Availability

The PSTN achieves five-nine reliability, equivalent to fewer than five minutes per year downtime and it handles millions of simultaneous calls. A VoIP network needs to achieve similar levels of reliability and scalability.
The required reliability and scalability can be achieved in a VoIP network by using redundant and load sharing equipment and networks. The call agent, access gateway, trunk gateway, signaling gateway and media server need to be fault tolerant. The types of functionality often used to achieve fault tolerance include:

- Redundant hardware
- Redundant network connections
- Hot-swap capability
- No single point of failure
- Software and firm ware that can be upgraded without loss of service

5.4. Call Routing and Number Plans

The PSTN is able to route calls between telephones anywhere in the world, for example a user can call Australia from Canada. This is achieved by having a well- defined number plan both nationally and internationally. Routing tables can be built using this numbering plan to provide end-to- end connectivity.

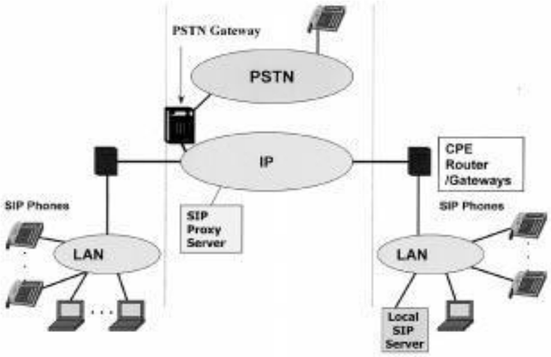

Fig 3: VoIP from end to end.

5.5. Network Interconnection

The PSTN is not a single network but a collection of networks operated by thousands of service providers. At each network boundary a network interconnection is required. Network interconnection agreements are put in place to cover items such as interconnection points, signaling, timing, billing and tariffs, bearer transport, regulatory requirements, etc. In addition these normally require approval from the relevant regulator. Constraints of scalability and established business models mean the next generation VoIP network will, like the PSTN, be a collection of networks and network interconnection agreements will still be required. This will need to cover similar topics to existing interconnect agreements but in addition address items such as security, QoS, signaling protocols (SIP, SIP-T, BICC). These additions may require regulatory approval as well.

6 ADVANTAGES AND DISADVANTAGES

6.1. Advantages

- Cost – a VOIP system is usually cheaper to operate than an equivalent office telephone system with a Private Branch Exchange and conventional telephone service.
- Integration with other services – Innovative services are emerging that allow customers to combine web access with telephone features through a single PC or terminal. For example, a sales representative could discuss products with a customer using the company's web site. In addition, the VOIP system may be integrated with video across the Internet, providing a teleconferencing facility.

6.2. Disadvantages:

- Startup cost – although VOIP can be expected to save money in the long run, the initial installation can be complex and expensive. In addition, a single standard has not yet emerged for many aspectsof VOIP so an organization must plan to support more than one standard, or expect to make relatively frequent changes as the VOIP field develops.
- Security – the flexibility of VOIP comes at a price: added complexity in securing voice and data. Because VOIP systems are connected to the data network, and share many of the same hardware and software components, there are more ways for intruders to attack a VOIP system than a conventional voice telephone system.

7 FUTURE OF VOIP

Call centers are a great use of VoIP; the technology is here to stay. More and more residential customers are moving to VoIP as a primary means of communication, though there is still less than one percent of homes that have gone to VoIP. The business community has embraced using VoIP more completely, as there are more dollars to be saved by connecting their largest centers together without having toll charges or having to maintain dual data and voice facilities.

The "connected" office worker in the near future will be able to remain connected regardless of where they are located. Wi-Fi / Wi-Max enabled portable devices and VoIP enabled applications will extend the enterprise to them anywhere, any time. In moving to a converged voice, data, and video network, organizations will need to "future-proof" their infrastructure in order to support multi-media interactions with constituents and support the connected tele-worker. (See Figure 4)

Figure 4: Technology Support for the "Connected" Tele-worker

8 CONCLUSION

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In this paper, we discussed Voice Over Internet Protocol is a set of software, hardware and standards designed to make it possible to transmit voice over packet switched networks, either an internal Local Area Network, or across the Internet. The transmission of voice over packet-switched IP networks – is one of the most important emerging trends in electronic communications. As with many new technologies, VOIP introduces both security risks and opportunities. VOIP has a very different architecture than traditional circuit-based telephony, and these differences result in significant security issues. Lower cost and greater flexibility are among and delay variation (jitter). VOIP is still an emerging technology, so it is difficult to develop a complete picture of what a mature worldwide VOIP network will one day look like.

the promises of VOIP. VOIP systems take a wide variety of forms, including traditional telephone handsets, conferencing units, and mobile units. In addition to end-user equipment, VOIP systems include a variety of other components, including call processors/call managers, gateways, routers, firewalls, and protocols. Quality of Service (QoS) is fundamental to the operation of a VOIP network that meets users' quality expectations. However, the implementation of various security measures can cause a marked deterioration in QoS. These complications range from firewalls delaying or blocking call setups to encryption-produced latency

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