ISSN:2320-0790

COMPUSOFT
An International Journal of Advanced Computer Technology

Voice over Internet Protocol (VOIP): Future Potential

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Abstract: VoIP (voice over IP) delivers standard voice over telephone services over Internet Protocol (IP). VoIP is the technology of digitizing sound, compressing it, breaking it up into data packets, and sending it over an IP (internet protocol) network where it is reassembled, decompressed, and converted back into an analog wave form. Gateways are the key component required to facilitate IP Telephony. A gateway is used to bridge the traditional circuit switched PSTN with the packet switched Internet. The paper covers software, hardware and protocol requirements followed by weighing the VoIP advantages such as low cost, portability, free and advanced features, bandwidth efficiency, call recording and monitoring against the VoIP disadvantages such as power dependency, quality of voice and service, security, and reliability. With ever increasing internet penetration and better broadband connectivity, VoIP is going to expand further with businesses already using VoIP standalone or in a hybrid format, although our focus and scope here remains VoIP. Mobile VoIP, an infant with less than 4% market share, has so far been focusing on increasing active subscriptions without a sustainable revenue model, but has the potential and is going to see tussle with static VoIP for space in days ahead.

Keywords: VoIP, Internet Protocol (IP), Bandwidth, PSTN, Internet, Broadband Connectivity, Gateway, IP PBX, Protocol, Data Packets, PCM Interface, Network Management.

I. INTRODUCTION

VoIP (voice over IP - that is, voice delivered using the Internet Protocol) is a term used in IP telephony for a set of facilities for managing the delivery of voice information using the Internet Protocol (IP). In general, this means sending voice information in digital form in discrete packets rather than in the traditional circuit-committed protocols of the public switched telephone network (PSTN). A major advantage of VoIP and Internet telephony is that it avoids the tolls charged by ordinary telephone service

VoIP is therefore telephony using a packet based network instead of the PSTN (circuit switched). During the early 90's the Internet was beginning its commercial spread. The Internet Protocol (IP), part of the TCP/IP suite (developed by the U.S. Department of Defense to link dissimilar computers across many kinds of data networks) seemed to have the necessary qualities to become the successor of the PSTN.

The first VoIP application was introduced in 1995 - an "Internet Phone". An Israeli company by the name of "VocalTec" was the one developing this application. The application was designed to run on a basic PC. The idea was to compress the voice signal and translate it into IP packets for transmission over the Internet. This "first generation" VoIP application suffered from delays (due to congestion), disconnection, low quality (both due to lost and out of order packets) and incompatibility.

VocalTec's Internet phone was a significant breakthrough, although the application's many problems prevented it from becoming a popular product. Since this step IP telephony has developed rapidly. The most significant development is gateways that act as an interface between IP and PSTN networks.

II. UNDERSTANDING VOICE OVER IP

Voice over IP (VoIP) is a blanket description for any service that delivers standard voice telephone services over Internet Protocol (IP). Computers to transfer data and files between computers normally use Internet protocol.

"Voice over IP is the technology of digitizing sound, compressing it, breaking it up into data packets, and sending it over an IP (internet protocol) network where it is reassembled, decompressed, and converted back into an analog wave form." The transmission of sound over a packet switched network in this manner is an order of magnitude more efficient than the transmission of sound over a circuit switched network.

As mentioned before, VoIP saves bandwidth also by sending only the conversation data and not sending the silence periods. This is a considerable saving because generally only one person talks at a time while the other is listening. By removing the VoIP packets containing silence from the overall VoIP traffic we can reach up to 50% saving.In a circuit switched network, one call consumes the entire circuit. That circuit can only carry one call at a time.

In a packet switched network, digital data is chopped up into packets, sent across the network, and reassembled at the destination. This type of circuit can accommodate many transmissions at the same time because each packet only takes up what bandwidth that is necessary.. Internet Telephony simply takes advantage of the efficiencies of packet switched networks.

Gateways are the key component required to facilitate IP Telephony. A gateway is used to bridge the traditional circuit switched PSTN with the packet switched Internet.

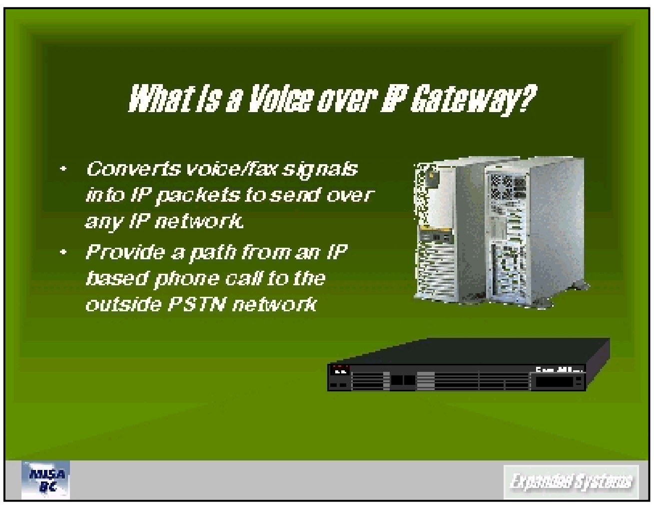

Fig. 1. What is VoIP Gateway

The gateway allows the calls to transfer from one network to the other by converting the incoming signal into the type of signal required by the network it is required to send it on. For example, A PC user wishes to call someone using a conventional phone. The PC sends the IP packets containing digitized voice to the gateway.

III. Requirements of a VoIP

The requirements for implementing an IP Telephony solution to support Voice Over IP varies from organization to organization, and depends on the vendor and product chosen. The following section aims to identify the fundamental requirements in the general case and is split into 3 sections:

- Software Requirements
- Hardware Requirements
- Protocol Requirements

A. Software Requirements

The software package chosen will reflect the organizational needs, but should contain the following modules as defined in the Technology Guide Series - Voice Over IP Publication, and other sources.

Voice Processing Module: This aspect of the software is required to prepare voice samples for transmission. The functionality provided by the voice processing module should support:

A PCM Interface is required to receive samples from the telephony interface (e.g. a voice card) and forward them to the Voice Over IP software for further processing.

Echo Cancellation is required to reduce or eliminate the echo introduced as a result of the round trip exceeding 50 milliseconds.

Idle Noise Detection is required to suppress packet transmission on the network when there are no voice signals to be sent. This helps to reduce network traffic as up to 60% of voice calls are silence and there is no point in sending silence.

A Tone Detector is required to discriminate between voice and fax signals by detecting DTMF (Dial Tone Multi frequency) signals.

The Packet Voice Protocol is required to encapsulate compressed voice and fax data for transmission over the network.

A **Voice Playback Module** is required at the destination to buffer the incoming packets before they are sent to the Codec for decompression.

Call Signalling Module is required to serve as a signalling gateway which allows calls to be established over a packet switched network as opposed to a circuit switched network (PSTN for example).

Packet Processing Module is required to process the voice and signalling packets ready for transmission on the IP based network.

Network Management Protocol allows for fault, accounting and configuration management to be performed.

B. Hardware Requirements

The exact hardware, which would be required, again, depends on organizational needs and budget. The list below highlights the most general hardware required.

The most obvious requirement is the existence (or installation) of an IP based network within the branch office gateway is required to bridge the differences between the protocols used on an IP based network and the protocols used on the PSTN.

The gateway takes a standard telephone signal and digitizes it before compressing it using a Codec. The compressed data is put into IP packets and these packets are routed over the network to the intended destination.

The PC's attached to the IP based network require the voice/fax software outlined above. They also require Full Duplex Voice Cards which allow both communicating parties to speak at the same time - as often happens in reality.
As an alternative to installing Voice Cards, IP Telephones can be attached to the network to facilitate Voice Over IP. A secondary gateway should be considered as a backup in the event of the failure of the primary gateway.

C. Protocol Requirements

There are many protocols in existence but the main ones are considered to be the following:

H.323: H.323 is an ITU (International Telecommunications Union) approved standard which defines how audio /visual conferencing data is transmitted across a network. H.323 relies on the RTP (Real-Time Transport Protocol) and RTCP (Real Time Control Protocol) on top of UDP (User Datagram Protocol) to deliver audio streams across packet based networks.

G.723.1: G.723.1 defines how an audio signal with a bandwidth of 3.4 KHz should be encoded for transmission at data rates of 5.3Kbps and 6.4Kbps. G.723.1 requires a very low transmission rate and delivers near carrier class quality. The VoIP Forum as the baseline Codec for low bit rate IP Telephony has chosen this encoding technique.

G.711: G.711 is an ITU standardised PCM (Pulse Code Modulation). This allows carrier class quality audio signals to be encoded for transmission at data rates of 56Kbps or 64Kbps. G.711 uses A-Law or Mu-Law for amplitude compression and is the baseline requirement for most ITU multimedia communications standards.

Real-Time Transport Protocol (RTP): RTP is the standard protocol for streaming applications developed within the IETF (Internet Engineering Task Force).

Resource Reservation Protocol (RSVP): RSVP is the protocol which supports the reservation of resources across an IP network. RSVP can be used to indicate the nature of the packet streams that a node is prepared to receive.

IV. HOW VOIP WORKS

Let us look at very simple VoIP call. Consider two VoIP telephones connected via an IP network. In this example both VoIP telephones are connected to a local LAN. Sally's phone has an IP address of 192.168.1.1. Bill's phone is 192.168.1.2, the IP addresses uniquely identify the telephones. Both our phones are configured to use a widely used VoIP standard called H.323.

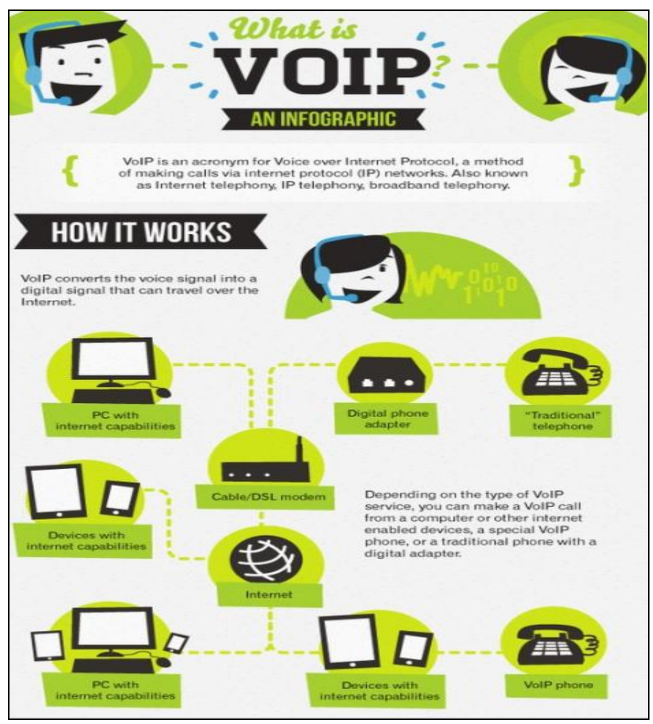

Fig. 2. How VoIP Works

Bill wants to talk to Sally and his phone knows the IP address of Sally's phone. Bill lifts the handset and 'dials' Sally, the phone sends a call setup request packet to Sally's phone, Sally's phone starts to ring, and responds to Bill's phone with a call proceeding message. When Sally lifts the handset the phone sends a connect message to Bill's phone. The two phones will now exchange the data packets containing the speech. At the end of the call Bill replaces his handset and phone stops sending voice data, sends a disconnect message and Sally's phone responds with a release message. The call is now complete. All the messages contain the Q931 ISDN protocol.

Having introduced VoIP,let us focus on three main 'types' of VoIP installed in the market place today.

VoIP over the Internet: This is probably the best known and most publicized, talking PC to PC. Basically free telephone calls. The call is only free if both parties to the call have access to the public Internet at zero cost.

- **Advantage**:Free calls regardless of distance or length of call.
- **Disadvantage**: Often the voice quality is bad due to the lack of bandwidth available for the call.
- **Other factors**: Have to use a PC or other computer running VoIP software.

Office to Office:A large multinational company will have offices across the whole country. They have a fixed data network connecting all the offices together. This allows every computer access to every other computer in the company. By installing a VoIP Gateway in each office and connecting it to the office legacy PBX and to the data network, employees use the data network for voice calls between offices.

- **Advantage**: Interoffice calls are free, since the company already has the bandwidth between offices. The technology is transparent to the user, and requires minimum training. The only new equipment required is a gateway at each office. Voice quality is good, because the company has control over the bandwidth.
- **Disadvantage**: Extra bandwidth may be required between offices, which offset the savings.
- **Other factors**: The carrier providing the interoffice bandwidth will almost certainly offer an alternative solution including management of the internal telephone traffic.

IP PBX: A traditional Private Branch Exchange (PBX) connects all the phones within an organization to the public telephone network. Essentially IP PBX replaces all the internal phones with VoIP telephones. The IP PBX has standard telephone trunk connections to the public telephone network. The IP PBX is a PBX with VoIP, but it also has the ability to support VoIP over the Internet and Office to Office VoIP.

- **Advantage**: Single cable infrastructure. The technology is transparent to the user, and requires minimum training. Future proof technology.
- **Disadvantage**: Primarily useful for Greenfield sites, but can be adapted to work with existing technology.

A. How VoIP Works: The Protocols

I have made an assumption that both ends of a VoIP telephone conversation are compatible. This compatibility only happens if both ends agree to use the same protocol. All manufacturers who claim to be producing industry standard voice over IP either support SIP or H.323 protocol.

What is H.323: Over the next few years, the industry will address the bandwidth limitations by upgrading the Internet backbone to asynchronous transfer mode (ATM), the switching fabric designed to handle voice, data, and video traffic. Such network optimization will go a long way toward eliminating network congestion and the associated packet loss. The Internet industry also is tackling the problems of network reliability and sound quality on the Internet through the gradual adoption of standards. Standards-setting efforts are focusing on the three central elements of Internet telephony: the audio codec format; transport protocols; and directory services.

H.323 Call Sequence: As such, H.323 addresses the core Internet-telephony applications by defining how delay-sensitive traffic, (i.e., voice and video), gets priority transport to ensure real-time communications service over the Internet.

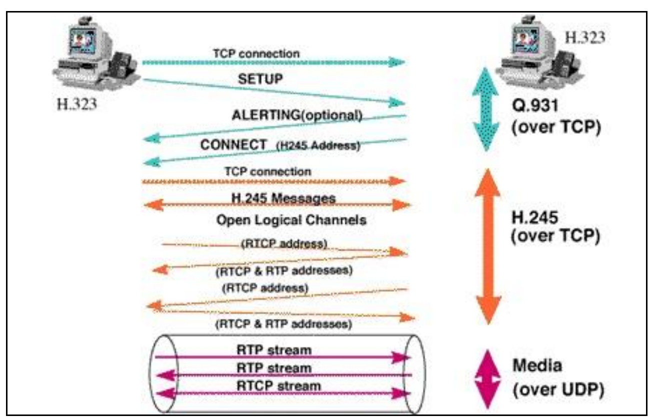

Fig. 3. VoIP Functioning: Protocols

The H.324 specification defines the transport of voice, data, and video over regular telephony networks, while H.320

defines the protocols for transporting voice, data, and video over integrated services digital network (ISDN).

B. How VoIP Works: Encoding

The call control part of H.323 sets up the parameters for the full duplex voice path between source telephone and destination telephone. I will continue with my analogies to explain how your voice gets transported across the Internet.

In terms of H.323 there is a trade-off between call quality and bandwidth, in general the higher the quality the greater the bandwidth required.

During the call setup portion of H.323 the phones have to decide which speech encoder/decoder to use when they send the speech to the other phone, Bill and Sally both have phones that support G.723.1, G.711 and G729.

The main difference between each of these encoders is the amount of bandwidth they use, G.711 uses 64kbit/s and G.723.1 can use as little as 5.3kbit/s. Although it would seem obvious to use the encoder with the lowest bandwidth, there is a loss of quality with a lower bandwidth. At the same time a stream of G723.1 encoded voice data starts being sent from each phone to the other phone.

C. How VoIP Works part 4: Hear the Quality

The performance of the speech encoders at each end, the number of packets lost on route, Latency and Jitter.

I have already talked about the encoders in the previous section. I also bundle into the encoding process echo suppression. In the early days of voice calls via satellite there would be an annoying echo. As the technology improved the echo disappeared. Echo suppression is very key to good quality VoIP calls. I do not dwell on the subject since the mathematics is beyond my comprehension. Good echo suppression makes for quality calls.

Be warned that because a manufacturer has a G.723.1 encoder it may not sound the same as another manufacturer who claims to have G.723.1, quality does vary. As a general rule the occasional lost packet will not affect too drastically the quality of a call, but lose 5 in a row and an entire word is lost and this will be a problem. So if you are going to have lost packets make sure they are only lost in a regular distributed manner. 5% lost packets distributed evenly will not result in the loss of words lose 5% of the words by clustering the packets and the effect is bad.

V. VoIP: Pros and Cons

A. Advantages of VoIP

There are many advantages to be gained from implementing an IP Telephony solution within the organization.

The following list aims to highlight some of the advantages of such a strategy:

Low cost:By far the single most important advantage over PSTN (Public Switched Telephone Network). When installing VoIP in the office only a single cable is required to the desk, for both telephone and data. Eliminating separate telephone wiring.

Portability: VoIP can be used anywhere in any corner of the world howsoever far the two locations are from each other as long as the internet connectivity is available, with no increase in cost irrespective to distance between caller and receiver. It is great for users with high mobility.

Fig. 4. VoIP: Advantages

Free and advanced features: While the traditional copper line phone companies charge for features like caller ID, call forwarding, voice mail, call waiting and distinctive ringing, VoIP service providers typically provide these features free of charge. Also since VoIP utilizes newer computer systems more advanced calling features are available which cannot be obtained via traditional copper POTS lines. E.g. A videophone provides 2-way video and audio over high-speed Ethernet connections. Currently Skype and Oovoo are two popular videophone service providers.

Call recording and monitoring: There may be neat call tracking and call recording features built into the VoIP phone system. However, the disadvantage is that this allows anyone off-site with technical skills the ability to monitor your company's calls and activity without the business owner's knowledge.

Single network infrastructure: When installing VoIP in the office only a single cable is required to the desk, for both telephone and data. Eliminating separate telephone wiring.

VoIP uses "soft" switching which eliminates most of the legacy PBX equipment. Reducing the cost of installing a communications infra-structure and the maintenance cost once installed.

Simple upgrade path: The VoIP PBX technology is software based. It is easier to expand, upgrade and maintain than its traditional telephony counterparts.

Bandwidth efficiency: VoIP can compress more voice calls into available bandwidth than legacy telephony. IP Telephony helps to eliminate wasted bandwidth by not transporting the 60% of normal speech which is silence.

IP - the underlying protocol: It is supported by most platforms and is independent of the transport protocol used.

Only one physical network is required to deal with both voice/fax and data traffic instead of two physical networks.

Having only one physical network has many advantages such as lower physical equipment cost and lower maintenance costs.

B. VOIP: Disadvantages:

While there are many aspects of VoIP which provide considerable benefits, the technology is developing and problems remain.

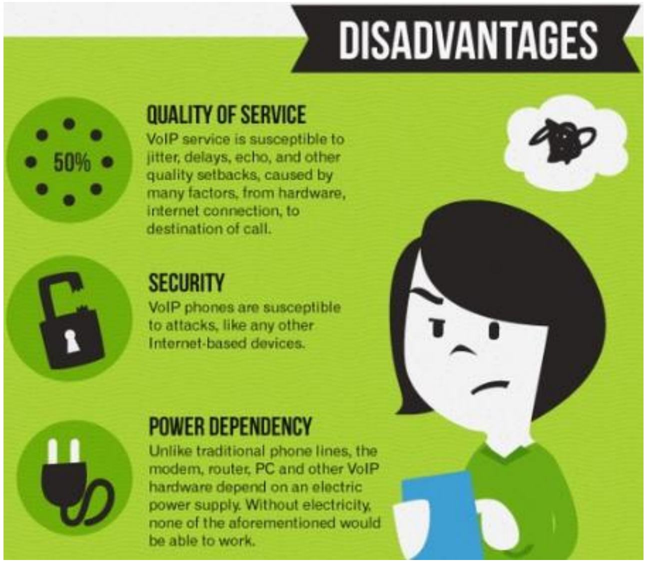

Fig. 5. VoIP: Disadvantages

The following section looks at some of the weaknesses of this technology and their consequences.

Power Dependency: If either the power or internet connection is lost due to power outage or otherwise, you are left with no phone service.

Voice and Service Quality: Heavy congestion on the network can result in considerable degradation of service as IP is not good at providing QoS (Quality of Service) guarantees.

VoIP has a bit to improve on Voice Quality, but not in all cases. VoIP QoS(Quality of service) depends on so many factors: your broadband connection, your hardware, the service provided by your provider, the destination of your call etc. More and more people are enjoying high quality of phone calls using VoIP, but still many users complain of hearing Martian, having to wait a lot before hearing an answer etc.

Reliability:The Internet is not the best medium for real time communications. Individual packets can take different routes and varying delays can be encountered and packets lost in transit. Waiting for delayed packets or retransmission of lost packets can result in considerable degradation of quality. Long delays in transit can affect quality so much that the technology can become unusable, though many vendors do have solutions which aim to negate the degradation suffered due to transit delays.

While some standards have been set by the ITU, the technology is not fully standardized and there is no guarantee that products from different vendors will be interoperable. Some vendors are trying to resolve this problem by forming groups and making guarantees about the products in the group but this is only a partial solution - vendors outwith the group cannot guarantee interoperability.

Security:This one is the last in this list, but it is not the least! Security is a main concern with VoIP, as it is with other Internet technologies. The most prominent security issues over VoIP are identity and service theft, viruses and malware, denial of service, spamming, call tampering and phishing attacks.

Feedback to Lucent Technologies customers reflect this worry. Major companies are planning to install IP Telephony capabilities at some point and have carried out initial investigations, however:

Since only one physical network for both data and voice/fax transmissions is required, failure of the network could be catastrophic, as all communications capabilities are lost.

C. VOIP: Opportunities:

Many vendors offer the ability to incorporate Virtual Private Networking (VPN) with relative ease into the IP Telephony solutions they provide. This allows any transmission to be encrypted using a number of cryptographic techniques and providing security by transmitting the communications through a 'tunnel' which is set up using PPTP (Point-to-Point Tunnelling Protocol) before commencing communications.

IP Telephony allows companies to exploit Computer Telephony Integration to its full extent.The convergence of communications technologies allows greater control over communications, most vendors provide logging and accounting facilities whereby all usage can be monitored.

Mobile VoIP: The mobile VoIP has accelerated the growth of VoIP market substantially. However, mVoIP market, which relies on VoIP subscriptions brought by mobile applications, has not achieved sustainable revenue growth owing to lack of credible monetisation models and stiff competition. Until now, the mVoIP market has witnessed race to bottom in which the effectiveness of the market is measured in terms of active subscribers but not in terms of revenues.

However, mVoIP providers have been able to disrupt the global telecommunications market by providing not only free OTT services, but also platforms that create higher customer engagement and provide more diverse revenue opportunities than standard voice and SMS services.

Contrastingly, static VoIP operators have not seen such a dramatic change in their number of subscriptions. Instead static VoIP market size and revenues have been growing at stable and healthy rates. But more importantly, this submarket is responsible for more than 96% of global VoIP revenues and these are expected to grow steadily in the next five years.

Due to the current disruption that telecom companies face all around the world, Visiongain expects incumbent operators to struggle at competing directly with OTT providers.

This can be witnessed in the way telecom companiesare applying pressure to create regulatory changes against OTT providers. One of these battles is being fiercely fought in the US as ISPs look to create a two tier internet whilst internet companies fight back against this anti-competitive initiative. On the contrary, more forward looking telecom companieswill understand the importance of embracing OTT providers and cooperating with the disruptors, in order to improve their service offerings and stop the erosion of ARPU. Others, especially those operators in the most developed markets will compete directly by offering VoLTE as the only way to increase ARPU and restore their lost market share from OTT providers.

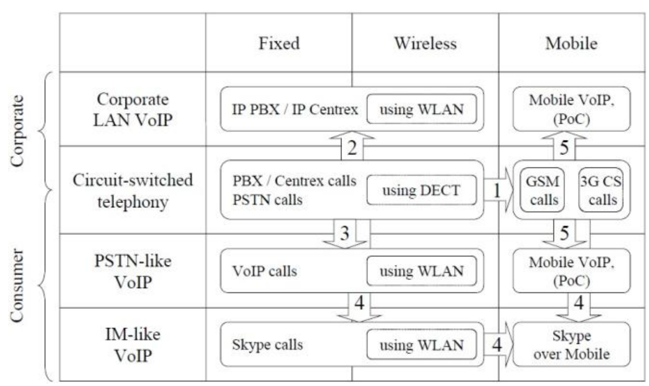

Fig. 6. Evolution Path of Voice Communications

VI. CONCLUSION:

Although, VoIP is extensively used and its usage is bound to grow immensely in times to come. However, its dependence on good internet connectivity, strong penetration, and reliance on power availability raises questions on its reliability.

With VoIP your voice is digitized into packets of data (ones and zeros). These packets are split up like pieces of a puzzle, sent over a data connection (for example, the internet), and then reassembled at the other end. The problem, in essence, is timing. The packets must arrive at the receiving end and be reassembled, in the exact same order, as they were disassembled. Unfortunately, packets may take multiple different routes to get from point A to point B hence there may be a delay of one packet over another. When this occurs, the recipient will hear a garbled voice.

Without a doubt, the data revolution will only gain momentum in the coming years, with more and more voice traffic moving onto data networks. Vendors of voice equipment will continue to develop integrated voice and data devices based on packetized technology. Users with ubiquitous voice and data service integrated over one universal infrastructure will benefit from true, seamless, transparent interworking between voice and all types of data. With ever increasing internet penetration and superior broadband connectivity, the advanced call features at negligible costs bodes well for the future of VoIP.

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Fixed-to-mobile substitution was recognized as the strongest barrierto VoIP evolution. Consumers and business users are increasingly using mobile phones as their onlycommunications devices, decreasing the market potential of VoIP services targeted mainly to the fixed domain. On the other hand, the emergence of WLAN-like technologies in private home andoffice networks, as well as in multi-mode mobile handsets can increase the potential of VoIPservices.

Overall, the global VoIP market is far from reaching maturity as in reality there are only 10 countries in the world which could be characterised as mature VoIP markets.

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