Report

Future of VoIP

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1 Historical Setting

In the early days of long distance communication consumers were heavily dependant on telecommunication companies as means of a rapid and reliable communication. However this was soon going to change with the publication of ”A Mathematical Theory of Communication” in 1948 by the American mathematician Dr. Claude Shannon, in which he drafted the concept of communicating in binary code. An idea later used to created first digital networks[23].This paper was going to change the way humans are going to communicate, as digital communications revolution set off to shape the world we are living in today.

Also during this time the American government required a solution to strengthen weak points in existing communications networks. This was a growing concern for American military command, as a nuclear strike could break a communication link between important command and control facilities and severely damage coordination of American troops and prevent them from organising a coordinated response to an attack Twenty years after release of Shannon's paper ARPANET (Advanced Research Projects Agency Network) was developed by USA Dept. of Defence. It was first computer network that later evolved into modern day Internet. Only five years later the first VoIP calls where pioneered on ARPANET as part of research project. However those calls were bounded and restricted to one private network. But the foundation for development of modern VoIP were set[12].

It was until 1995, the first company was to release VoIP software product. VocalTec a small Israel company released their first commercial product ”Internet Phone”. It was designed for a domestic PC user utilising commodity hardware[28]. By 1998 espite obstacles including but not limited to the lack of a high speed network infrastructure, according to PriMetrica inc. VocalTec handled 0.2% of all international phone calls[3].

In tests with circuit-switched and packet-switched networks, it has been shown that packet-switched networks can carry five to ten times the number of voice calls over the same bandwidth[5]. Better bandwidth utilisation combined with the development growth in the VoIP sector, has attracted numbers of investors and companies who started to work and build on this technology. Cisco and Lucent created a device that could route the VoIP traffic. And as a direct result in 2000 4.3% of worlds voice traffic was carried over IP and by 2003 this figure had increased to 12.8%. [28] [3]

It became evident to many that VoIP could be used to improve internal communication within large organisations and bring down cost of telecommunication bills. In 2004 Boeing at that time having 157000 employees and offices in 70 countries decided that all phone calls within company are going to be handled via VoIP. Soon many other companies became to adopt same strategy.

More recently in 2003 the Swedish entrepreneurs Niklas Zennstrm and the Dane Janus Friis started a company called Skype which was based on generation three peer-to-peer network, which originated in the Kazaa music sharing technology. Currently Skype is one of the leading VoIP provider to businesses and private users.

2 Adoption

The adoption of VoIP in the domestic market has been largely driven by the availability of cheap broadband as well as the ubiquity of personal computers, and this is predicted to increase with mobile internet connected handsets becoming prevalent.

It is difficult to assess the actual number of domestic users as this technology is offered primarily through a number of commercial providers, being integrated into non-traditional telephony platforms such as gaming platforms and finally the use of private servers.

According to TeleGeography Research, Skype in 2009 accounted for 12 percent of international calling minutes, a 50 percent increase over 2008[8]. A more interesting trend is that 36 percent of Skype-to-Skype calls were video based[8]. Which indicates that telephony customers are willing to move to a more immersive conversation using video when appropriate and suitably convenient. In addition to this its is not unusual for a Skype user to be informed that more that 20 million users are currently logged into the network. These numbers are quite impressive when you consider that Skype was founded in 2003 and only recently has consumer technology enabled a reasonable quality of service as well as the ability to use this service on the mobile handset.

In the enterprise environment, VoIP based systems have been gaining popularity internally due their benefits over legacy systems, these include less infrastructure, lower service charges, integrated messaging, global mobility and new tools[25].

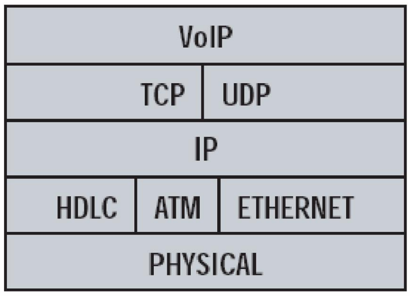
Migrating to these VoIP systems enable convergence on a single network plane with IP end-to-end as well as integration with existing user directory's via Microsoft Active Directory or *nix based LDAP (Lightweight directory access protocol) which offer centralised logging and access control. But its not just in the administration and back-end where enterprise customers gain advantage, directory integration enables an employees phone line to be diverted to their current location, integration of instant messaging, email, calendaring systems and mobility by offering these services to the mobile handset.

Cisco, Avaya and Nortel, are the biggest players in this market with all three company's producing both hardware and software components of an enterprise level PBX VoIP system. Due to the open nature and interoperability of VoIP, these firms are moving their focus from hardware to their software offerings. All of the above companies now offer "unified communication" as opposed to IP telephony, as they try to become the centralised communication hub for today's office. But in this move to a software based solution the market has been opened up to companies like Microsoft who now offer "Office Communications Server"[21] and IBMs Tivoli Netcool Enterprise VoIP Manager.

Telecommunications providers have largely completed the move to digital switching in the mid 1980s[9] this technology is not directly part of the VoIP protocol stack, but it acts as an enabler and is more commonly referred to as "IP backhaul", further discussion of the backhaul network is beyond the scope of this document.

3 VoIP Standards

Contrary to common belief VoIP is not a single protocol or protocol stack but a collection of technologies that maybe be assembled together to provide real-time voice, video and data over a packet switched data network. Simpler VoIP implementations concern themselves with the session and presentation layers of the OSI seven layer model, as illustrated. But when scaling this technology across enterprise or internet sized systems many deeper grained issues begin to appear such as fault tolerance, security, latency, mobility and quality of service. Below we will examine three of the most common VoIP standards/platforms.



3.1 H.323 - Packet-based multimedia communications systems

A standard defined by the International Telecommunication Union (ITU) and was initially published in 1996[15] and was the first standards based VoIP stack[16]. This standard encapsulates a number of ITU-T and IETF protocols used in the transmission of voice, video and data over a packet switched network.

The standard consists of a number of classes of components, below is a breakdown and description of these elements:

Terminals* Voice and video handsets, high-definition videoconferencing systems, voicemail systems, softphones.

Multipoint Control Units* (MCUs) Responsible for managing multi-point conferences (two or more endpoints involved in a conference).

Gateways* Interfaces to other networks e.g. PSTN, H.320 (VoIP using ISDN) or other H.323 networks (proxy).

Gatekeeper An optional component which manages admission control and address resolution and can be used to implement features such as follow-me/find-me or forward on busy.

Border Elements and Peer Elements Exchange addressing information and participate in authorisation across administrative domains. Peer elements are used to aggregate and reduce the volume of routing information exchanged across domains.

*often referred to as endpoints

While all of the elements described above are not required, at minimum two terminals are required to facilitate communication between two people and in most deployments a gatekeeper is deployed to facilitate address resolution among other functions.

H.323 is described as a framework document which defines how various protocols fit together. Common implementations of this standard may employ twelve or more protocols in to perform basic VoIP functionality. The standard covers all aspects of communication from T.120 for data conferencing to T.38 for fax transmission and from H.235 defining security to H.225.0 defining call signaling between endpoints.

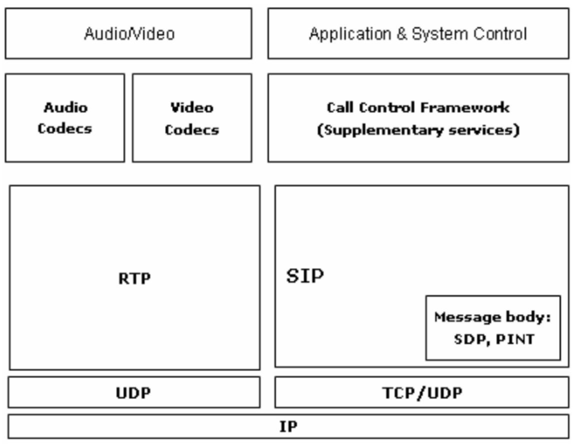
From this brief overview of the standard it is quite clear to see that H.323 is a robust and complex standard defined by the telecommunications industry for the telecommunications industry. From its inception it has been positioned as a next generation to the POTS network offering voice, video, data, conferencing, roaming and other features naively while maintaining compatibility with legacy systems by way of support for PSTN numbering for example.

One criticism of this standard is its complexity due to its rigid specification for areas of the standard, which has slowed the deployment of these H.323 networks.

3.2 SIP - Session Initiation Protocol

The initial SIP draft document was published during the same period as the first revision of the H.323 standard was published in 1996. The SIP signaling protocol is published by a working group of the IETF (Internet Engineering Task Force), and due to this affiliation the standard relies much more heavily on existing internet protocols such as HTTP, in contrast H.323 utilises ISDN Q.931 protocol for signaling as well as numerous other telecommunication industry protocols.

SIP is regarded as a much simpler protocol when compared to H.323[27], while maintaining a somewhat similar feature set[22] when communicating over IP. The IETF approach to telephony signaling reuses many mechanisms from HTTP including authentication, encoding and error codes.

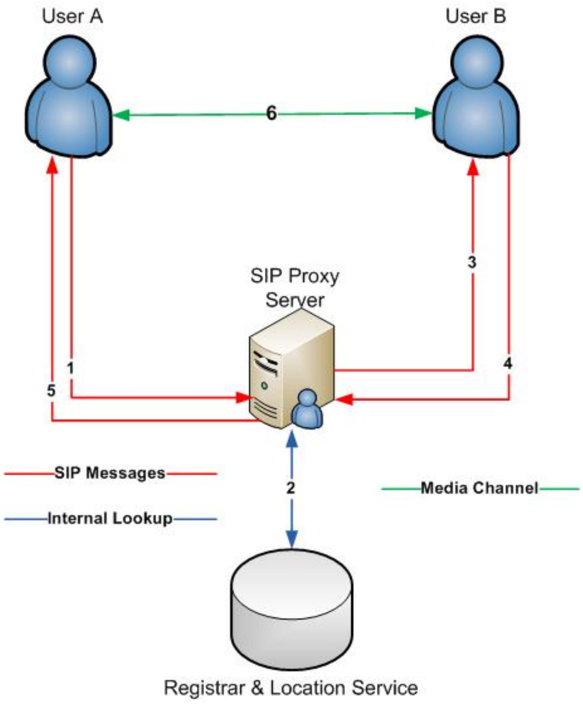


SIP was designed to control unicast or multicast multimedia communication sessions over IP, example applications of the protocol would include voice and video streaming, conferencing, online gaming and instant messaging to name a few.

The protocol is only involved in the signaling portion of a VoIP session, the design of the protocol borrows many elements from the HTTP protocol, utilising its request/response model and the packet header is largely the same. For the transmission most typically UDP or TCP are used to connect to other SIP endpoints and encryption is provided utilising TLS (Transaction Layer Security) the same mechanism used to secure HTTP transactions.

Unlike H.323 the design goal of SIP was to provide a signaling and call setup protocol and not to rigorously define all the potential features of the protocol, while offering expandability to offer features far beyond a traditional PSTN network. Depending on specific implementation SIP can be peer-to-peer, enabling the creation of a scalable and decentralised network.

Below you can see an example of the SIP architecture, communication is initiated using a SIP server but the actual media transmission is on a peer-to-peer basis, with only external lookups for authentication, authorisation and user lookup/discovery.



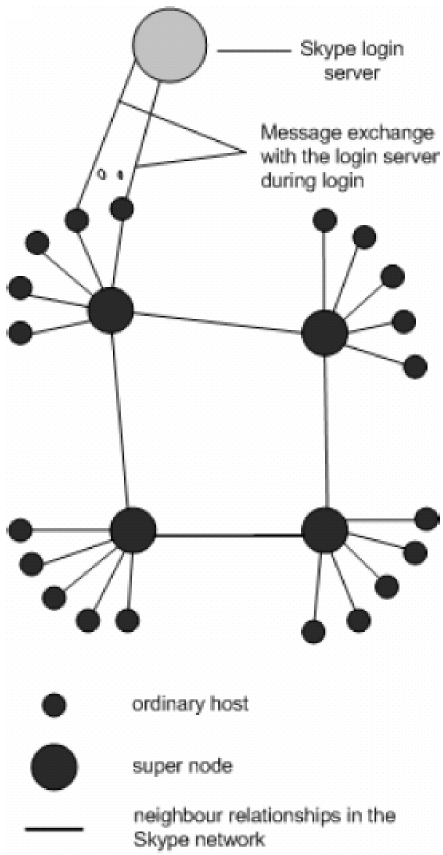
Proponents of SIP distinguish the protocol from existing VoIP protocols such as H.323 to SIPs roots laying in IP and being governed by the IETF and not by the telecommunications industry. H.323 is the more mature standard at this point but lacks in flexibility, for example it does not support presence information. SIP appears to be gaining more deployments in the corporate environment, but has issues with interoperability due to vendor specific additions.

3.3 Skype

Skype unlike H.323 and SIP is a closed source propriety protocol, network and business model. Due to the nature of Skype complete details of the protocol remain undisclosed but due to research efforts the operation of the system is largely understood, considering the usage numbers mentioned earlier in this paper omitting an overview of what is known about the Skype protocol would leave this document lacking.

The system is designed as a peer-to-peer network which allows the company to operate a small core network of login servers, reducing their bandwidth requirements and aiding in maintaining a high degree of uptime. The remainder of the network comprises of Skype clients, these clients have two operating modes, either as a node or a super-node.

Below is an architectural diagram of a Skype based network.



Any node with a public IP address having sufficient CPU, memory, and network bandwidth is a candidate to become a super node[1]. These super nodes may relay communications for regular nodes located behind a NAT or Firewall configuration, which is a common issue for many peer-to-peer networks. For a regular node to connect to to the network it must register itself with the login servers and connect to a super-node.

With a peer-to-peer approach such as this, authentication and privacy become important elements to the protocol. As we have already discussed all authentication is managed via the Skype central login servers, Skype ensures privacy in the channel between two or more communicating users by transparently encrypting the data stream. For this encryption the system uses RSA 1024bit for authentication, RC4 for signaling and 256bit AES for VoIP, offering the user a quite reasonable level of security[6].

Skype recently submitted their Skype SILK wideband audio codec to the IETF for standardisation [14], but the Skype is also believed to use G.729 and SVOPC in transmission of audio. For video transmission it is believed that Skype utilises VP7 from On2 Technologies, but since Googles acquisition of the company and the removal of their website its impossible at this point to confirm this.

While the Skype protocol may not be as all encompassing as H.323 and SIP it feature set is large enough to be positioned as a viable VoIP option. It has support for voice, video, data and text messaging as well as conferencing, mobile usage and presence information. And offering an interesting third architectural option in its useage of super-nodes for a VoIP implementation.

4 Emerging Applications

4.1 Entertainment

In current times VoIP is reaching its capabilities as alternative to the traditional telephone service. Slowly the software industry is trying to integrate the VoIP protocol into their products. Currently integrated VoIP is used to enhance the user experience by allowing for intuitive and direct way of communicating with fellow users. And the biggest beneficent of this process is gaming industry.

As high speed Internet connection became wide spread, the gaming industry began to focus on creating multiplayer games. Users playing early multiplayer games did not have a chance to communicate verbally; instead they were forced to write short messages through chat boxes. The introduction of VoIP allowed to overcome slow and cumbersome communication and created additional way of enhancing the game play[4].

However the early implementations of VoIP in multiplayer games were far from perfect. Jitter, lost packets, microphone echo effect, exiting from the game environment which resulted in the termination of voice chat were amongst main complaints from players. As a solution third party companies: Mumble, Ventrilo and Teamspeak, came up with more advanced implementations of VoIP for games. Allowing them to create their teams, adjust max input output volumes, regulate voice transmission thresholds and/or echo cancelation. But again there was a drawback to these solutions, to manage your voice communication players were forced to minimise the game prior to adjusting voice chat options.

Currently there is no golden solution to the above problems and gamers are forced to choose either functionality or ease of use. The gaming market changes rapidly though and there might be a possibility that VoIP services could be reliable, user-friendly and be integrated in all of games. There are two main gamimg platforms: consoles and the traditional PC.

First let's discuss consoles, as this is the fastest growing gaming sector[24]. Consoles by nature are a unified platform, can provide an integrated VoIP service and expose it to game developers so they have an option of utilising that console's VoIP implementation in their product. As the console market is very competitive all the main console companies: Sony, Microsoft and Nintendo are trying to develop good quality integrated voice services to gain a competitive advantage[13].

The PC as a common platform is not so easy to consolidate as many different operating systems exist side-by-side. Building an integrated VoIP solution for all OS, old and new would be unfeasible. However in 2004 Valve Corporation released a platform called Steam. It provides an easy way to buy and store your games online. It also has an exposed API that provides a wrapper for any game it sales. Wrapper allows users to access Steam's resources such as friend's lists, groups or instant messaging. Steam grew popular over the last few years and became a predominant games distributor for the PC platform (estimated 70% market share[10]). In 2007 Valve implemented a VoIP service. It's far from being perfect but the lack of a competitor does not stimulate Valve to further work on enhancing its implementation. Although Steam provides users with a clean interface, abilities to create ”chat rooms” and manage transmission/receive volumes.

All of the platforms however were faced with one big problem how to serve a large client base without a large dedicated server farm and large bandwidth demand. In 2010 Steam reached 30 million users [26], Xbox Live 23 million users[29] and Playstation3 33.5 million users[2]. To provide reliable VoIP services the platforms (similary to Skype) use a peer-to-peer architecture for providing VoIP. That way service providers are distributing workload to each client and only controlling how service is provided via authorisation and discovery services.

4.2 Smart-phone Integration

After establishing a solid foundation in the PC market VoIP took a natural step and began spreading to mobile devices. Data plans are now a common option for contract and pre-paid mobile users. Combined with a widespread 3G (3rd generation) mobile data network, handsets are equipped with fast data access that VoIP clients may take advantage of in order to allow mobile users gain cheaper calls.

However VoIP implementations are still suffering from many issues namely packet loss, delay and jitter. Despite that the biggest thing crippling VoIP is the handset manufacturers companies themselves. Apple’s iPhone has restricted to VoIP transmission only on Wi-Fi networks, blocking 3G’s data channel as means of VoIP communication, this of course limits where and when user can use VoIP. Microsoft with its Windows Mobile also limits the capabilities of VoIP, in contrary to iPhone it allows VoIP to access 3G but it limits access to internal handset earpiece. Resulting in bad quality of received voice, especially if external speaker is located on side or back of phone[17].

There is no definite statement by mobile producers why such a actions were taken. However one commonly shared opinion is that telecom companies which are closely cooperating with mobile producers to reduce the capabilities of VoIP in mobiles. To protect their market and force mobile customers to use only their services. This state of affairs would be maintined but the introduction of a new OS by Google is set to upset the market dynamics. Android is gaining popularity as it is an open-source product as it does not restrict the functionality of the device. As this alternative is gaining more and more mobile market share, proprietary OS’s will have to remove their restrictions on VoIP to stay competitive.

Other issue that can affect mobile VoIP is the lack of bandwidth. If the bandwidth pipe is too small for continuous packet transmission the audio quality of the conversation will be degraded. If UDP packets were used to encode voice then call receiver would hear noticeable gaps in conversation. If TCP were used than all packets that did not made through would have to be retransmitted causing pipe to be overblown with packets. This could possibly be a real issue; however any standard Wi-Fi network has bandwidth large enough to accommodate for number of clients using VoIP services. Bandwidth in Vo3G (Voice over 3G), is usually smaller to Wi-Fi but it is capable of caring a VoIP conversation. The designed/specified bandwidth that 3G can reach is 2.4 Mb/s, however realistically a stationary user can expect to reach 2 Mb/s. Pedestrian can expect 384Kb/s, while a vehicle passenger only 144Kb/s [11]. Most of VoIP codecs require less than 40 Kb/s to sustain reasonable level of service quality[19], and it confirms that currently bandwidth should not be a concern affecting VoIP services.

4.3 VoIP and Video Conferencing

Every business is looking to reduce the cost of operating and encourage its employees work more effectively. On top of that many companies have their branches spread across the whole world. VoIP has proven to be very successful as a cheap method of bringing people from across the world together. But this was not enough for businesses who demanded a easier and more direct way for its employees to communicate. That is why high quality video conferencing is becoming quite popular across geographically diverse organisations. But there are issues and tradeoffs tied to this solution, ranging from technical to psychological[20].

To psychological problems with video conferencing we can account the discomfort in delays of responses. Delays as small as 300ms can be spotted by users and create a feeling of discomfort and bad communication. Another issue is related to facial expressions, because of hardware limitations users are looking at the monitor rather than at the camera creating bad image in eye of the other person as it looks like person is avoiding an eye contact[7]. Although video conferencing is used widely in businesses, it is also popular method of communication for domestic users, as it allows for more direct and natural way to talk to other people. Video conversations provides means of expressing emotions easier (especially facial or gestures) that cannot be expressed through traditional phone.

Tandberg is a telecommunication company that in 2009 released a video conferencing phone with LCD screen and camera. This is an attempt to create a new generation of stationary phones that allow for quick and more direct video calls. This product was lunched to target offices, however if this concept is going to adopted, this concept might also be brought to domestic use.

4.4 Unified Communications

An underlying theme in the development of VoIP technologies is the integration with existing non-real-time forms of communication such as email, voicemail and fax services.

Initially focused at business customers unified communications can provide potentially great value when integrated with business processes, enabling decision makers to be more responsive to real-time data from production and work-flow systems no matter the persons location. The combining of presence information, messaging, conferencing, collaboration, email, calendaring and business systems with IP technology and the plethora of new form factor devices becomes a very powerful concept and provides new opportunities to communication system providers. A number of major firms are competing currently in this area namely Microsoft, Cisco, Google and IBM - with each firm attempting to merge it with their existing platforms.

The concept of unified communications is also making its way slowly into the domestic market, an early leader in this area was RIM with their Blackberry device which enabled mobile phone users to receive emails to their mobile device instantly. Recently with the increased adoption of IP on the mobile handset users are becoming familiar with sending and receiving Twitter messages, mobile email, uploading photos to social networks. While these communication streams have not yet be integrated they pave the way to having your presence and communications available on whatever device is appropriate at the time, with the network dynamically routing messages to the relevant location.

Recently Google has attempted to position itself as a unified communication provider with a new product called Google Voice. They offer integration of email, sms, traditional mobile phone services, VoIP, conferencing, switching calls to other devices mid conversation and numerous other features to users in both a desktop and mobile client.[18]

This area is still in its infancy, but with the increasing adoption of capable IP connected mobile handsets and the adoption cloud based services it appears to be a logical step forward.

5 Future

Attempting to predict the future of a particular technology is a notoriously hazardous pass-time, with the potential of a new previously unseen disruptive technology being just on the horizon. But ignoring this we will attempt to identify trends for VoIP and beyond in this section. We say VoIP and beyond because each of the major standards H.323, SIP and although not a public standard Skype, treat voice traffic as an equal to text, data and video. This progression to unified communications has already begun in the corporate domain and as pointed out earlier in this document the early steps are already being made for domestic users.

5.1 Follow me communication

At this point is quite clear to see a trend towards mobile connected devices and the provision of cloud based internet services. With more and more of our data being stored in the cloud, the era of the internet terminal may not be so far away. With such devices a user may just pick up a terminal at random and login, gaining access to all their data and services through these online services. Today VoIP standards allow for a call to be initiated on a particular device and without dropping the call move to another device say from a desk phone to mobile phone and from mobile phone to car phone through the course of a single conversation. But in future we will require these protocols to become device and even location aware not only presenting real-time data in suitable format for the device, but prioritising when to notify the user of these messages by looking at calendar information possibly mixed with location coordinates. This context aware notification and smart transcription/formatting of messages will become very important as time progresses, eventually it will be an essential mechanism for managing a constant torrent of data.

5.2 Network Changes

In the last ten years the fixed line telecoms providers have migrated their networks from largely voice based carriers to IP data networks, but mobile telecoms providers have been insulated from this demand for mobile data until recently largely due to mobile data technology lagging behind wired technology. AT&T in the USA have been on the leading edge of this migration since the introduction of the iPhone in 2007, with many of the customers of the phone being tech savy and regular internet users. Since the launch of the iPhone AT&T have spend almost $37 billion dollars on upgrades to the network to increase its capacity.

Over the next number of years the battle between the traditional telecommunications providers and that of the data providers will be key in defining how we gain access to these communication networks. Currently in the USA we are reading about network neutrality and how the telecoms providers want to offer a tiered internet experience to their customers, much like the la carte TV service Sky Digital offer us today. To date mobile network providers have remained silent on this matter, but with mobile data becoming more important than the traditional voice service, how are the network providers going to operate a financially successful business.

5.3 Costs

The price of call between two computers using a VoIP implementation is currently free. This is due to the fact that companies like Skype do not handle the communication traffic. They have only several control servers that allow users to set up link between two parties. Currently the only costs a user incurs is that of the internet connection, which is still quite cheap as high speed broadband becomes almost ubiquitous. This creates a threat for telecom companies as their main market is drifting away from standard traditional mean of communication towards IP based communication. This is going to be tough challenge deal with as data plans for mobiles are cheap and calls through Vo3G are cheaper then standard mobile phone companies rates. Vo3G also allows users to avoid extra cost related to calls between different networks. Mobile companies might end up providing only data plans as they wont be able to compete with VoIP. The cost of international calls are going down and they will continue due to the fact that customers are more likely to set up a VoIP call if they believe that its too expensive for them to use conventional phone. The telecom companies have no choice but to keep reducing the prices of calls in hope that customers would consider it as cheap and price it in a way that convenience wins out over the alternatives. However the cost of the services might be reduced to such low level that telocom companies would not break even. This would certainly make them increase the fees for renting their infrastructure. Which means that at the end of the day its going to be the customers who will pay for continually decreasing prices.

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