International Journal of Information & Computation Technology.
ISSN 0974-2239 Volume 3, Number 1 (2013), pp. 5-12

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VoIP Systems Interface with the Conventional PSTN and Mobile Phones

E. Kalaikavitha

M.C.A., M.Phil., Asst. Professor, Dept. of Information Technology.
Rathinam College of Arts and Science College, Coimbatore–21
E-mail: ekalaikavitha@gmail.com

Abstract

Voice over IP was introduced to me over a year ago by a friend who was already using the service because she travels a lot. Because of my work, most of my friends and family live in other cities and it costs me long distance to make phone calls during the day or night.

Long distance calling cards, making phone calls using my computer and the Internet, dialing certain numbers before I could make the long distance call and more. I was already set up with home phone and a cell phone. To make calls on my cell phone was absorb, the home phone was fine, but the package my local telephone company had me locked in to limited me to when I could make calls.

VoIP Definition

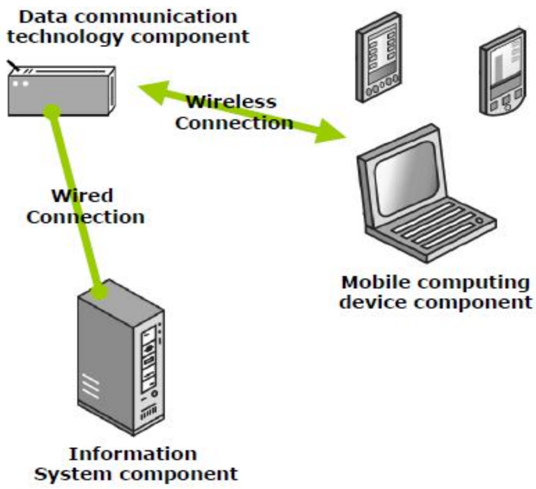
VoIP is short for Voice over IP. Other terms for VoIP also include IP Telephony, Internet telephony, Broadband Telephony, Broadband Phone, Voice over Broadband,

Voice over Internet Protocol, Digital Phone Service, Cable Phone Service, Managed IP Telephony and so on. Many industry experts see Voice over IP as a leading-edge technology for the future in telecommunication.

Who are the main users of VoIP service?

- Residential home users
- Small Business or Home Office

A Definition of Mobile Computing

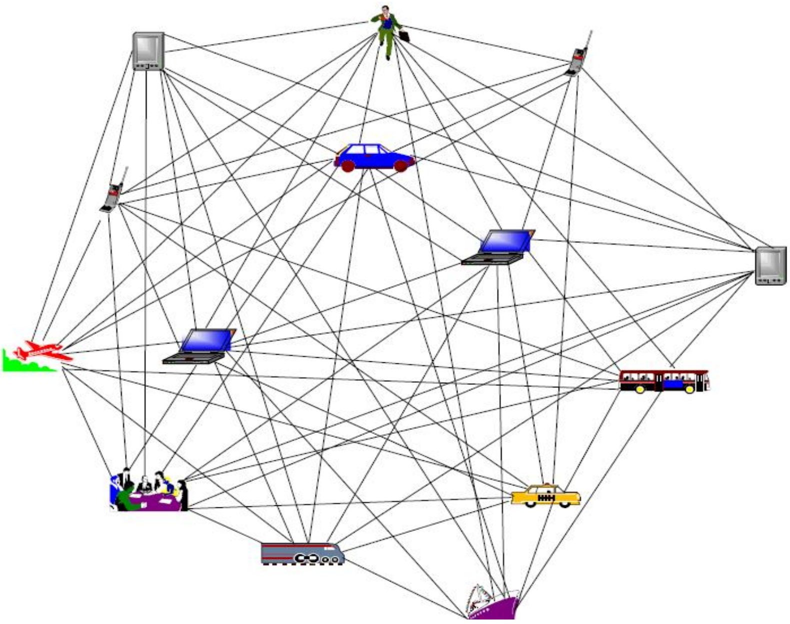


Display, collect, and transfer information from a mobile device to an information system using one or a combination of various data transfer methods**.**

Mobile connectivity

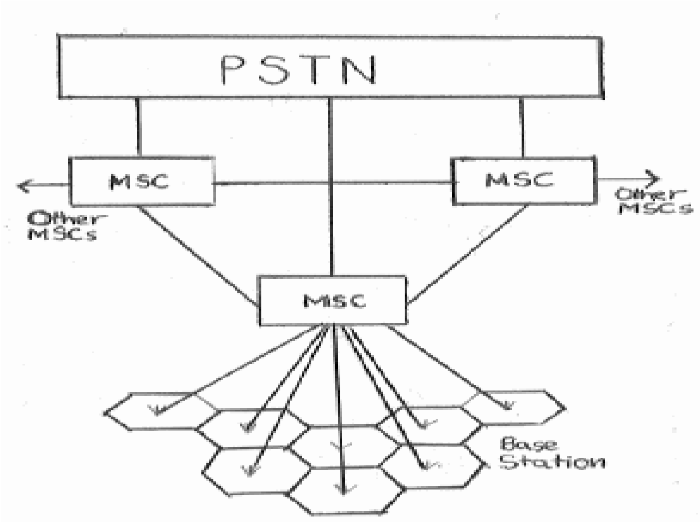
The mobile connectivity between two nodes exists if they are continuously connected through wireless channel, and can utilize the channel without being subjected to spatial and temporal constraints.

In addition to these technical challenges, mobile computing also faces business challenges. This is due to the lack of trained professionals to bring the mobile technology to the general people and development of pilot projects for testing its capabilities.



Existing Cellular Network Architecture

A cellular network consists of mobile units linked together to switching equipment, which interconnect the different parts of the network and allow access to the fixed Public Switched Telephone Network (PSTN). The technology is hidden from view; it's incorporated in a number of tranceivers called Base Stations (BS). Every BS is located at a strategically selected place and covers a given area or cell - hence the name cellular communications. A number of adjacent cells grouped together form an area and the corresponding BSs communicate through a so called Mobile Switching Centre (MSC).



The MSC is the heart of a cellular radio system. It is responsible for routing, or switching, calls from the originator to the destinator. It can be thought of managing the cell, being responsible for set-up, routing control and termination of the call, for management of inter-MSC hand over and supplementary services, and for collecting charging and accounting information. The MSC may be connected to other MSCs on the same network or to the PSTN.

Advantages of Using VoIP

VoIP technology uses the Internet's packet-switching capabilities to provide phone service. VoIP has several advantages over circuit switching. For example, packet switching allows several telephone calls to occupy the amount of space occupied by only one in a circuit-switched network. Using PSTN, that 10-minute phone call we talked about earlier consumed 10 full minutes of transmission time at a cost of 128 Kbps. With VoIP, that same call may have occupied only 3.5 minutes of transmission time at a cost of 64 Kbps, leaving another 64 Kbps free for that 3.5 minutes, plus an additional 128 Kbps for the remaining 6.5 minutes. Based on this simple estimate, another three or four calls could easily fit into the space used by a single call under the conventional system. And this example doesn't even factor in the use of data compression, which further reduces the size of each call.

Let's say that you and your friend both have service through a VoIP provider. You both have your analog phones hooked up to the service-provided ATAs. Let's take another look at that typical telephone call, but this time using VoIP over a packet-switched network:

Disadvantages of Using VoIP

The current Public Switched Telephone Network is a robust and fairly bulletproof system for delivering phone calls. Phones just work, and we've all come to depend on that. On the other hand, computers, e-mail and other related devices are still kind of flaky. Let's face it -- few people really panic when their e-mail goes down for 30 minutes. It's expected from time to time. On the other hand, a half hour of no dial tone can easily send people into a panic. So what the PSTN may lack in efficiency it more than makes up for in reliability. But the network that makes up the Internet is far more complex and therefore functions within a far greater margin of error. What this all adds up to is one of the major flaws in VoIP: reliability.

First of all, VoIP is dependant on wall power. Your current phone runs on phantom power that is provided over the line from the central office. Even if your power goes out, your phone (unless it is a cordless) still works. With VoIP, no power means no phone. A stable power source must be created for VoIP.

Another consideration is that many other systems in your home may be integrated into the phone line. Digital video recorders, digital subscription TV services and home security systems all use a standard phone line to do their thing. There's currently no way to integrate these products with VoIP. The related industries are going to have to get together to make this work.

Emergency 911 calls also become a challenge with VoIP. As stated before, VoIP uses IP-addressed phone numbers, not NANP phone numbers. There's no way to associate a geographic location with an IP address. So if the caller can't tell the 911 operator where he is located, then there's no way to know which call center to route the emergency call to and which EMS should respond. To fix this, perhaps geographical information could somehow be integrated into the packets. Because VoIP uses an Internet connection, it's susceptible to all the hiccups normally associated with home broadband services. All of these factors affect call quality:

- Latency
- Jitter
- Packet loss

Phone conversations can become distorted, garbled or lost because of transmission errors. Some kind of stability in Internet data transfer needs to be guaranteed before VoIP could truly replace traditional phones.

VoIP is susceptible to worms, viruses and hacking, although this is very rare and VoIP developers are working on VoIP encryption to counter this.

Another issue associated with VoIP is having a phone system dependant on individual PCs of varying specifications and power. A call can be affected by processor drain. Let's say you are chatting away on your softphone, and you decide to open a program that saps your processor. Quality loss will become immediately evident. In a worst case scenario, your system could crash in the middle of an important call. In VoIP, all phone calls are subject to the limitations of normal computer issues.

Packet loss

Packet loss on a VoIP phone line is similar to when a call on a cell phone begins to "break-up." Clip and phrases of words are dropped during the packet-transfer process, resulting in the person on the receiving end missing parts of the conversation.

These are the steps used when packet loss occurs:

- Know when the packet loss occurs. If your packet loss is occurring while you are playing an online game, for example, it is likely the result of the game using much of the available bandwidth. Deal with this issue by logging off your game when you use your phone.
- Configure your Ethernet and router properly. Improper configuration also leads to packet loss on your phone line. Reconfigure them to "full duplex."
- Conduct a bandwidth test on your computer. You don't have enough bandwidth for your VoIP phone if bandwidth is below 90 kbs. Call your service provider to see about getting a different phone package with more bandwidth
- Install traffic-shaping software. This software will give bandwidth priority to your VoIP line when in use. The result is less, if any, packet loss.

The effect of frame loss

Frame loss is a very common event in Ethernet because nodes drop frames as soon as the FCS does not match of any other serious defect is detected. Let's now analyze the consequences of this on performance.

If the network is transmitting isochronal information which is time-dependent, for example audio or video, then higher layers, such as UDP, do not guarantee delivery of all the packets. Furthermore, if real-time information in being streamed it does not make any sense to re-transmit lost packets. The slot time for that specific packet from a conversation or a movie would definitely be gone. So the main consequence is the loss of quality of the streamed information When packet delivery must be assured the higher level protocols must guarantee that every single packet arrives at the receiver. Connection oriented protocols, like TCP, have packet number ids which allow the transmission of acknowledge packets (ACK) when the packet has arrived correctly, or for requesting retransmission (NACK) if it does not. Transmitters do not assume that a packet has been received correctly until they receive the ACK, but if the packet not arrive a time-out indicates that retransmission of the packet is necessary. This strategy can have a devastating effect on the throughput if the algorithm is not properly adjusted.

The performance of TCP over wide area networks (the Internet) has been extensively modeled.

Jitter

Jitter refers to how variable latency is in a network. High jitter, greater than approximately 50 msec, can result in both increased latency and packet loss.

Let's see how.

When talking to someone it's important that they hear what you say in the same order that you say it, otherwise they won't understand what you're telling them. Unfortunately, jitter causes packets to arrive at their destination with different timing and possibly in a different order than they were sent (spoken), with some arriving faster and some slower than they should.

To correct the effects of jitter, VoIP endpoints collect packets in a buffer and put them back together in the proper timing and order before the receiver hears them. This works, but it's a balancing act. Processing that buffer adds delay to the call, so the bigger the buffer, the longer the delay. Remember the effects of latency? Keep in mind, no matter how big the buffer is, it is finite in size. If voice packets arrive when the buffer is full then packets are dropped and the receiver will never hear them. These are called discarded packets.

Latency

A measure of the delay in a call. We measure both the round-trip delay between when information leaves point A and when a response is returned from point B, and the one-way delay between when something was spoken and when it was heard. The largest contributor to latency is caused by network transmission delay. Round-trip latency affects dynamics of conversation and is used in our MOS calculations. One-way latency is used. With round trip latencies above 300 msec or so, users may experience annoying talk-over effects.

Conclusion

Corporate customer telephone support often use IP telephony exclusively to take advantage of the data abstraction. The benefit of using this technology is the need for only one class of circuit connection and better bandwidth use. Companies can acquire their own gateways to eliminate third-party costs, which is worthwhile in some situations.

VoIP is widely employed by carriers, especially for international telephone calls. It is commonly used to route traffic starting and ending at conventional PSTN telephones.

Many telecommunications companies are looking at the IP Multimedia Subsystem (IMS) which will merge Internet technologies with the mobile world, using a pure VoIP infrastructure. It will enable them to upgrade their existing systems while embracing Internet technologies such as the Web, email, instant messaging, presence, and video conferencing. It will also allow existing VoIP systems to interface with the conventional PSTN and mobile phones.

The Future

The future of Mobile Computing is very promising indeed, although technology may go too far, causing detriment to society.

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