VOIP

Voice-over-Internet protocol (VoIP, IPA: /voip/) is a protocol optimized for the transmission of voice through the Internet or other packet-switched networks. VoIP is often used abstractly to refer to the actual transmission of voice (rather than the protocol implementing it). This latter concept is also referred to as IP telephony, Internet telephony, voice over broadband, broadband telephony, and broadband phone.
VoIP providers may be viewed as commercial realizations of the experimental Network Voice Protocol (1973) invented for the ARPANET providers. Some cost savings are due to utilizing a single network to carry voice and data, especially where users have underused network capacity that can carry VoIP at no additional cost. VoIP-to-VoIP phone calls are sometimes free, while VoIP calls connecting to public switched telephone networks (VoIP-to-PSTN) may have a cost that is borne by the VoIP user.
Voice-over-IP systems carry telephony signals as digital audio, typically reduced in data rate using speech data compression techniques, encapsulated in a data-packet stream over IP.
There are two types of PSTN-to-VoIP services: Direct inward dialing (DID) and access numbers. DID will connect a caller directly to the VoIP user, while access numbers require the caller to provide an extension number for the called VoIP user.

Contents

History

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Voice-over-Internet Protocol has been a subject of interest almost since the first computer network. By 1973, voice was being transmitted over the early Internet. The technology for transmitting voice conversations over the Internet has been available to end-users since at least the early 1980s. In 1996, a shrink-wrapped software product called VocalTec Internet Phone (release 4) provided VoIP along with extra features such as voice mail and caller ID. However, it did not offer a gateway to the PSTN, so it was only possible to speak to other Vocaltec Internet Phone users. In 1997, Level 3 began development of its first softswitch (a term they invented in 1998); soft switches were designed to replace traditional hardware telephone switches by serving as gateways between telephone networks.

Revenue in the total VoIP industry in the US is set to grow by 24.3% in 2008 to $3.19 billion. Subscriber growth will drive revenue in the VoIP sector, with numbers expected to rise by 21.2% in 2008 to 16.6 million. The US's largest VoIP provider is Vonage.

Functionality

VoIP can facilitate tasks and provide services that may be more difficult to implement or more expensive using the PSTN. Examples include:

- The ability to transmit more than one telephone call over the same broadband connection. This can make VoIP a simple way to add an extra telephone line to a home or office.
- Conference calling, call forwarding, automatic redial, and caller ID; zero- or near-zero-cost features that traditional telecommunication companies (telcos) normally charge extra for.
- Secure calls using standardized protocols (such as Secure Real-time Transport Protocol.) Most of the difficulties of creating a secure phone connection over traditional phone lines, like digitizing and digital transmission, are already in place with VoIP. It is only necessary to encrypt and authenticate the existing data stream.
- Location independence. Only an Internet connection is needed to get a connection to a VoIP provider. For instance, call center agents using VoIP phones can work from anywhere with a sufficiently fast and stable Internet connection.
- Integration with other services available over the Internet, including video conversation, message or data file exchange in parallel with the conversation, audio conferencing, managing address books, and passing information about whether others (e.g. friends or colleagues) are available to interested parties.
- Advanced Telephony features such as call routing, screen pops, and IVR implementations are easier and cheaper to implement and integrate. The fact that the phone call is on the same data network as a user's PC opens a new door to possibilities.

Implementation

Because UDP does not provide a mechanism to ensure that data packets are delivered in sequential order, or provide Quality of Service (QoS) guarantees, VoIP implementations face problems dealing with latency and jitter. This is especially true when satellite circuits are involved, due to long round-trip propagation delay (400–600 milliseconds for links through geostationary satellites). The receiving node must restructure IP packets that may be out of order, delayed or missing, while ensuring that the audio stream maintains a proper time consistency. This function is usually accomplished by means of a jitter buffer in the voice engine.

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

VoIP challenges:

- Available bandwidth
- Network Latency

- Packet loss
- Jitter
- Echo
- Security
- Reliability
- In rare cases, decoding of pulse dialing

Many VoIP providers do not decode pulse dialing from older phones. The VoIP user may use a pulse-to-tone converter, if needed.

Fixed delays cannot be controlled but some delays can be minimized by marking voice packets as being delay-sensitive (see, for example, Diffserv).

The principal cause of packet loss is congestion, which can sometimes be managed or avoided. Carrier VoIP networks avoid congestion by means of teletraffic engineering.

Variation in delay is called jitter. The effects of jitter can be mitigated by storing voice packets in a jitter buffer upon arrival and before producing audio, although this increases delay. This avoids a condition known as buffer underrun, in which the voice engine is missing audio since the next voice packet has not yet arrived.

Common causes of echo include impedance mismatches in analog circuitry and acoustic coupling of the transmit and receive signal at the receiving end.

Reliability

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Please help improve this article by adding reliable references. Unsourced material may be challenged and removed. (February 2008)
Conventional phones are connected directly to telephone company phone lines, which in the event of a power failure are kept functioning by backup generators or batteries located at the telephone exchange. However, IP Phones and the IP infrastructure they connect to (routers and servers) typically depend on the availability of mains electricity or another locally generated power source.

Voice travels over the internet in almost the same manner as data does in packets. So when you talk over an IP network your conversation is broken up into small packets. The voice and data packets travel over the same network with a fixed bandwidth. This system is more prone to congestion and DoS attacks than traditional circuit switched systems. To increase the reliability of VoIP phones the VoIP provider needs to increase dedicated and redundant connectivity via T-1 access and backup DSL, with automatic failover at each location. The company can create a reliable network by reducing the number of single points of failure.

Quality of service

Some broadband connections may have less than desirable quality. Where IP packets are lost or delayed at any point in the network between VoIP users, there will be a momentary drop-out of voice. This is more noticeable in highly congested networks and/or where there are long distances and/or interworking between end points. Technology has improved the reliability and voice quality over time and will continue to improve VoIP performance as time goes on.

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

A number of protocols have been defined to support the reporting of QoS/QoE for VoIP calls. These include RTCP XR (RFC3611), SIP RTCP Summary Reports, H.460.9 Annex B (for H.323), H.248.30 and MGCP extensions. The RFC3611 VoIP Metrics block is generated by an IP phone or gateway during a live call and contains information on packet loss rate, packet discard rate (due to jitter), packet loss/discard burst metrics (burst length/density, gap length/density), network delay, end system delay, signal / noise / echo level, MOS scores and R factors and configuration information related to the jitter buffer.

RFC3611 VoIP metrics reports are exchanged between IP endpoints on an occasional basis during a call, and an end of call message sent via SIP RTCP Summary Report or one of the other signaling protocol extensions. RFC3611 VoIP metrics reports are intended to support real time feedback related to QoS problems, the exchange of information between the endpoints for improved call quality calculation and a variety of other applications.

Mobile Number Portability (MNP) in the Internet Telephony Environment
Mobile number portability (MNP) also impacts the internet telephony, or VOIP (Voice over IP) business. A voice call originated in the VOIP environment which is routed to a mobile phone number of a traditional mobile carrier also face challenges to reach its destination in case the mobile phone number is ported. Mobile number portability is a service that makes it possible for subscribers to keep their existing mobile phone number when changing the service provider (or mobile operator).

VoIP is clearly identified as a Least Cost Routing (LCR) voice routing system, which is based on checking the destination of each telephone call as it is made, and then sending the call via the network that will cost the customer the least. With GSM number portability now in place, LCR providers can no longer rely on using the network root prefix to determine how to route a call. Instead, they now need to know the actual current network of every number before routing the call.

Therefore, VoIP solutions also need to handle MNP when routing a voice call. In countries without a central database like UK it might be necessary to query the GSM network about the home network a mobile phone number belongs to. As VoIP starts to take off in the enterprise markets because of least cost routing options, it needs to provide a certain level of reliability when handling calls.

MNP checks are important to assure that this quality of service is met; by handling MNP lookups before routing a call and assuring that the voice call will actually work, VoIP companies give businesses the necessary reliability they look for in an internet telephony provider. UK-based messaging operator Tyntec provides a Voice Network Query service, which helps not only traditional voice carriers but also VoIP providers to query the GSM network to find out the home network of a ported number.

In countries such as Singapore, the most recent Mobile number portability solution is expected to open the doors to new business opportunities for non-traditional telecommunication service providers like wireless broadband providers and voice over IP (VoIP) providers.

In November 2007, the Federal Communications Commission in the United States released an order extending number portability obligations to interconnected VoIP providers and carriers that support VoIP providers.

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Difficulty with sending faxes

This article or section needs copy editing for grammar, style, cohesion, tone or spelling.
You can assist by editing it now. A how-to guide is available. (June 2008)
The support of sending faxes over VoIP is still limited. The existing voice codecs are not designed for fax transmission. (They are designed to digitize an analog representation of a human voice efficiently, but the inefficiency of digitizing an analog representation (modem signal) of a digital representation (a document image) of analog data (an original document) more than negates any bandwidth advantage of VoIP. In other words, the fax “sounds” simply doesn’t fit in the VoIP channel.) An effort is underway to remedy this by defining an alternate IP-based solution for delivering fax-over-IP, namely the T.38. This protocol is designed to work like a traditional fax machine. It can work using several configurations. The fax machine could be a traditional FAX machine connected to the PSTN, an ATA box, or similar; it could be a FAX machine with an RJ-45 connector plugged straight into an IP network; it could be a computer pretending to be a FAX machine. Originally, T.38 was designed to use UDP and TCP transmission methods across an IP network. The main difference between using UDP and TCP methods for a FAX is the real time streaming attributes. TCP is better suited for use between to IP devices. However, older fax machines connected to an analog system benefits from UDP near real time characteristics. There have been updated versions of the T.30 to resolve the FoIP issues, which is the core Fax protocol. Some new fax machines have T.38 built in capabilities, which allows the user to just plug right into the network with minimal configuration changes. A unique feature of T.38 is that each packet contains a copy of the main data in the previous packet. This is an option, but most implementations seem to support it. This forward error correction scheme makes T.38 far more tolerant of dropped packets than using VoIP. It requires two successive lost packets to actually lose any data. The data you lose will only be a small piece, but with the right settings and error correction mode, there is a high probability that you will receive the whole transmission. Tweaking the settings on the T.30 and T.38 protocols could also turn your unreliable fax into a robust machine. Some fax machines pause at the end of a line, to allow the paper feed to catch up. This is good news for packets that were lost or delayed, because it gives them a chance to catch up. However, if this was to happen on every line, your FAX transmittal would take a long time. Another possible solution to overcome the drawback is to treat the fax system as a message switching system, which does not need a real-time data transmission—such as sending a fax as an email attachment (see Fax) or remote printout (see Internet Printing Protocol). The end system can completely buffer the incoming fax data before displaying or printing the fax image.

Emergency calls

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WikiProject Telecommunications may be able to help recruit one.
If a more appropriate WikiProject or portal exists, please adjust this template accordingly.
The nature of IP makes it difficult to locate network users geographically. Emergency calls, therefore, cannot easily be routed to a nearby call center. Sometimes, VoIP systems may route emergency calls to a non-emergency phone line at the intended department. In the US, at least one major police department has strongly objected to this practice as potentially endangering the public.

VoIP E911 is another method by which VoIP providers in the US are able to support emergency services. In the US, the VoIP E911 emergency-calling system associates a physical address with the calling party's telephone number as required by the Wireless Communications and Public Safety Act of 1999. All "interconnected" VoIP providers (those that provide access to the PSTN system) are required to have E911 available to their customers. VoIP E911 service generally adds an additional monthly fee to the subscriber's service per line, similar to analog phone service. Participation in E911 is not required and customers can opt-out or disable E911 service on their VoIP lines if desired. VoIP E911 has been successfully used by many VoIP providers to provide physical address information to emergency service operators.

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One shortcoming of VoIP E911 is that the emergency system is based on a static table lookup. Unlike in cellular phones, where the location of an E911 call can be traced using aGPS or other methods, the VoIP E911 information is only accurate so long as the subscriber is diligent in keeping their emergency address information up to date. In the US, the Wireless Communications and Public Safety Act of 1999 leave the burden of responsibility upon the subscriber and not the service provider to keep their emergency information up to date.
A tragic example of a miscommunication with VoIP, is an 18-month-old boy named Elijah Luck. In an emergency, 911 services were called. An ambulance was sent to the former home of the Lucks. The Voice over Internet Protocol Telephone Company knew the correct address, as they were paying their bill from the correct current billing address the company had on record. "It's up to subscribers to ensure the company has up-to-date contact information" was the response from the VoIP company. After about a half hour wait, the Lucks called from a neighbors land line, 911 services arrived in six minutes. Elijah Luck was pronounced dead at the Alberta Children's Hospital. CBC story of Elijah Luck.

Integration into global telephone number system

While the wired public switched telephone network (PSTN) and mobile phone networks share a common global standard (E.164) which allocates and identifies any specific telephone line, there is no widely adopted similar standard for VoIP networks. Some allocate an E.164 number which can be used for VoIP as well as incoming and external calls. However, there are often different, incompatible schemes when calling between VoIP providers which use provider-specific short codes.

VoIP phone accessibility and portability

If using a software based soft-phone, calls can only be placed from the computer on which the soft-phone software resides. Thus with a soft-phone the caller is typically limited to a single point of calling. When using a hardware based VoIP phone-device/phone-adapter it is possible to connect traditional analog phones directly to a VoIP phone-adapter without the need to operate a computer. The converted analog phone signal can then be connected to multiple house phones or extensions, just as any traditional phone company signal can be connected. A second VoIP hardware configuration option involves the use of a specially designed VoIP telephone which incorporates a VoIP phone adapter directly into the phone itself, and which also does not require the use of a computer. A third VoIP hardware configuration option involves the use of a WiFi router and a WiFi SIP phone which can extend a service range throughout a home or office. WiFi SIP phones can also be used at any location where an "unauthenticated" open hotspot Wi-Fi signal is available. However, note that many hotspots require browser-based authentication, which most SIP phones do not support.

Mobile phones and hand-held devices

Telcos and consumers have invested billions of dollars in mobile phone equipment. In developed countries, mobile phones have achieved nearly complete market penetration, and many people are giving up landlines and using mobiles exclusively. Given this situation, it is not entirely clear whether there would be a significant higher demand for VoIP among consumers until either public or community wireless networks have similar geographical coverage to cellular networks (thereby enabling mobile VoIP phones, so called WiFi phones or VoWLAN) or VoIP is implemented over 3G networks. However, "dual mode" telephone sets, which allow for the seamless handover between a cellular network and a WiFi network, are expected to help VoIP become more popular.

Phones like the NEC N900iL, and later many of the Nokia Eseries and several WiFi enabled mobile phones have SIP client's hardcoded into the firmware. Such clients operate independently of the mobile phone network unless a network operator decides to remove the client in the firmware of a heavily branded handset. Some operators such as Vodafone actively try to block VoIP traffic from their network and therefore most VoIP calls from such devices are done over WiFi.

Several WiFi only IP hard phones exist, most of them supporting either Skype or the SIP protocol. These phones are intended as a replacement for PSTN based cordless phones but can be used anywhere where WiFi internet access is available.

Another addition to hand held devices are ruggedized bar code type devices that are used in warehouses and retail environments. These types of devices rely on "inside the 4 walls" type of VoIP services that do not connect to the outside world and are solely to be used from employee to employee communications.

Security

Many consumer VoIP solutions do not support encryption yet, although having a secure phone is much easier to implement with VoIP than traditional phone lines. As a result, it is relatively easy to eavesdrop on VoIP calls and even change their content. An attacker with a packet sniffer could intercept your VoIP calls if you are not on a secure VLAN. This security vulnerability could lead to Denial of Service (DoS) attacks to you and anyone on your network. The DoS would devastate your phone network by creating a continuing busy signal and forced disconnects. Viper Lab predicts VoIP attacks against service providers will escalate since unlicensed mobile access technology becomes more widely deployed to allow calls to switch from cell networks to VoIP networks, Viper Labs warns that “service providers are, for the first time, allowing subscribers to have direct access to mobile core networks over IP, making it easier to spoof identities and use illegal accounts to launch a variety of attacks. There is no such thing as a 100% secure solution to network security. The implementation of voice over internet protocol just adds to that complexity, by giving hackers another means to access your system. Customers can secure their network by limiting access to the virtual local area network, thus hiding their voice data network from the users. If the customer maintains a secure and properly configured gateway, you can keep most of the hackers out. There are several open source solutions that facilitate sniffing of VoIP conversations. A modicum of security is afforded due to patented audio codecs that are not easily available for open source applications; however such security through obscurity has not proven effective in the long run in other fields. Some vendors also use compression to make eavesdropping more difficult. However, real security requires encryption and cryptographic authentication which are not widely available at a consumer level. The existing secure standard SRTP and the new ZRTP protocol is available on Analog Telephone Adapters (ATAs) as well as various softphones. It is possible to use IPsec to secure P2P VoIP by using opportunistic encryption. Skype does not use SRTP, but uses encryption which is transparent to the Skype provider.

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

Caller ID

Caller ID support among VoIP providers varies, although the majority of VoIP providers now offer full caller ID with name on outgoing calls. When calling a PSTN number from some VoIP providers, caller ID is not supported.

In a few cases, VoIP providers may allow a caller to spoof the caller ID information, potentially making calls appear as though they are from a number that does not belong to the caller. Business grade VoIP equipment and software often makes it easy to modify caller ID information. Although this can provide many businesses great flexibility, it is also open to abuse.

VoIM

Voice over Instant Messaging (VoIM) presents VoIP as one communication mode among several, with an IM user interface (contact list and presence) as the primary user experience. Many instant messenger services added client-to-client or client-to-PSTN VoIP in the mid-2000s.

Adoption

Mass-market telephony

A major development starting in 2004 has been the introduction of mass-market VoIP services over broadband Internet access services, in which subscribers make and receive calls as they would over the PSTN. Full phone service VoIP phone companies provide inbound and outbound calling with Direct Inbound Dialing. Many offer unlimited calling to the U.S. and some to Canada or selected countries in Europe or Asia as well, for a flat monthly fee.

These services take a wide variety of forms which can be more or less similar to traditional POTS. At one extreme, an analog telephone adapter (ATA) may be connected to the broadband Internet connection and an existing telephone jack in order to provide service nearly indistinguishable from POTS on all the other jacks in the residence. This type of service, which is fixed to one location, is generally offered by broadband Internet providers such as cable companies and telephone companies as a cheaper flat-rate traditional phone service. Often the phrase "VoIP" is not used in selling these services, but instead the industry has marketed the phrases "Internet Phone", "Digital Phone" or "Softphone" which is aimed at typical phone users who are not necessarily tech-savvy. Typically, the provider touts the advantage of being able to keep one's existing phone number.

At the other extreme are services like Gizmo Project and Skype which rely on a software client on the computer in order to place a call over the network, where one user ID can be used on many different computers or in different locations on a laptop. In the middle lie services which also provide a telephone adapter for connecting to the broadband connection similar to the services offered by broadband providers (and in some cases also allow direct connections of SIP phones) but which are aimed at a more tech-savvy user and allow portability from location to location. One advantage of these two types of services is the ability to make and receive calls as one would at home, anywhere in the world, at no extra cost. No additional charges are incurred, as call diversion via the PSTN would, and the called party does not have to pay for the call. For example, if a subscriber with a home phone number in the U.S. or Canada calls someone else within his local calling area, it will be treated as a local call regardless of where that person is in the world. Often the user may elect to use someone else's area code as to minimize phone costs to a frequently called long-distance number.

For some users, the broadband phone complements, rather than replaces, a PSTN line, due to a number of inconveniences compared to traditional services. VoIP requires a broadband Internet connection and, if a telephone adapter is used, a power adapter is usually needed. In the case of a power failure, VoIP services will generally not function. Additionally, a call to an emergency services number may not automatically be routed to the nearest local emergency dispatch center. Some VoIP providers only handle emergency call for one country. Some VoIP providers offer users the ability to register their address so that emergency services work as expected.

Another challenge for these services is the proper handling of outgoing calls from fax machines, DVR boxes, satellite television receivers, alarm systems, conventional modems or FAXmodems, and other similar devices that depend on access to a voice-grade telephone line for some or all of their functionality. At present, these types of calls sometimes go through without any problems, but in other cases they will not go through at all. And in some cases, this equipment can be made to work over a VoIP connection if the sending speed can be changed to a lower bits per second rate. If VoIP and cellular substitution becomes very popular, some ancillary equipment makers may be forced to redesign equipment, because it would no longer be possible to assume a conventional voice-grade telephone line would be available in almost all homes in North America and Western Europe. The TestYourVoIP Web site offers a free service to test the quality of or diagnose an Internet connection by placing simulated VoIP calls from any Java-enabled Web browser, or from any phone or VoIP device capable of calling the PSTN.

Corporate and Telco use

Although few office environments and even fewer homes use a pure VoIP infrastructure, telecommunications providers routinely use IP telephony, often over a dedicated IP network, to connect switching stations, converting voice signals to IP packets and back. The result is a data-abstracted digital network which the provider can easily upgrade and use for multiple purposes.

Because of the bandwidth efficiency and lowcosts that VoIP technology provides, businesses are slowly beginning to migrate from traditional copper-wire telephone systems to VoIP systems to reduce their monthly phone costs.

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 phone] s.

Electronic Numbering (ENUM) uses standard phone numbers (E.164), but allows connections entirely over the Internet. If the other party uses ENUM, the only expense is the Internet connection. Virtual PBX (or IP PBX) allows companies to control their internal phone network over an existing LAN and server without needing to wire a separate telephone network. Users within this environment can then use standard telephones coupled with an FXS, IP Phones connected to a data port or a Softphone on their PC. Internal VoIP phone networks allow outbound and inbound calling on standard PSTN lines through the use of FXO adapters.

Use in amateur radio

Sometimes called Radio over Internet Protocol or RoIP, Amateur radio has adopted VoIP by linking repeaters and users with Echolink, IRLP, D-STAR, Dingotel and EQSO. In fact, Echolink allows users to connect to repeaters via their computer (over the Internet) rather than by using a radio. By using VoIP Amateur Radio operators are able to create large repeater networks with repeaters all over the world where operators can access the system with actual ham radios.

Ham Radio operators using radios are able to tune to repeaters with VoIP capabilities and use DTMF.

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