# VOIP Solution

**Voice over Internet Protocol** (also **voice over IP**, **VoIP** or **IP telephony**) is a methodology and group of technologies for the delivery of [voice communications](https://en.wikipedia.org/wiki/Voice_communication) and [multimedia](https://en.wikipedia.org/wiki/Multimedia" \o "Multimedia)sessions over [Internet Protocol](https://en.wikipedia.org/wiki/Internet_Protocol) (IP) networks, such as the [Internet](https://en.wikipedia.org/wiki/Internet). The terms **Internet telephony**, **broadband telephony**, and **broadband phone service** specifically refer to the provisioning of communications services (voice, [fax](https://en.wikipedia.org/wiki/Fax), [SMS](https://en.wikipedia.org/wiki/Short_Message_Service), voice-messaging) over the public Internet, rather than via the [public switched telephone network](https://en.wikipedia.org/wiki/Public_switched_telephone_network) (PSTN).

The steps and principles involved in originating VoIP telephone calls are similar to traditional digital [telephony](https://en.wikipedia.org/wiki/Telephony) and involve signaling, channel setup, digitization of the analog voice signals, and encoding. Instead of being transmitted over a [circuit-switched network](https://en.wikipedia.org/wiki/Circuit-switched_network), however, the digital information is packetized, and transmission occurs as IP packets over a [packet-switched network](https://en.wikipedia.org/wiki/Packet-switched_network). They transport audio streams using special media delivery protocols that encode audio and video with [audio codecs](https://en.wikipedia.org/wiki/Audio_codec), and [video codecs](https://en.wikipedia.org/wiki/Video_codec). Various codecs exist that optimize the media stream based on application requirements and network bandwidth; some implementations rely on [narrowband](https://en.wikipedia.org/wiki/Narrowband) and [compressed speech](https://en.wikipedia.org/wiki/Speech_coding), while others support [high-fidelity](https://en.wikipedia.org/wiki/High_fidelity) stereo codecs. Some popular codecs include [μ-law](https://en.wikipedia.org/wiki/%CE%9C-law) and [a-law](https://en.wikipedia.org/wiki/A-law) versions of [G.711](https://en.wikipedia.org/wiki/G.711), [G.722](https://en.wikipedia.org/wiki/G.722), an [open source](https://en.wikipedia.org/wiki/Open_source) voice codec known as [iLBC](https://en.wikipedia.org/wiki/ILBC" \o "ILBC), a codec that only uses 8 kbit/s each way called [G.729](https://en.wikipedia.org/wiki/G.729), and many others.

Early providers of voice-over-IP services offered business models and technical solutions that mirrored the architecture of the legacy telephone network. Second-generation providers, such as [Skype](https://en.wikipedia.org/wiki/Skype), built closed networks for private user bases, offering the benefit of free calls and convenience while potentially charging for access to other communication networks, such as the PSTN. This limited the freedom of users to mix-and-match third-party hardware and software. Third-generation providers, such as [Google Talk](https://en.wikipedia.org/wiki/Google_Talk), adopted the concept of [federated VoIP](https://en.wikipedia.org/wiki/Federated_VoIP)—which is a departure from the architecture of the legacy networks.[[1]](https://en.wikipedia.org/wiki/Voice_over_IP#cite_note-1) These solutions typically allow dynamic interconnection between users on any two domains on the Internet when a user wishes to place a call.

In addition to [VoIP phones](https://en.wikipedia.org/wiki/VoIP_phone), VoIP is available on many [smartphones](https://en.wikipedia.org/wiki/Smartphone" \o "Smartphone), personal computers, and other Internet access devices. Calls and SMS text messages may be sent over [mobile data](https://en.wikipedia.org/wiki/Mobile_data) or [Wi-Fi](https://en.wikipedia.org/wiki/Wi-Fi).[[2]](https://en.wikipedia.org/wiki/Voice_over_IP#cite_note-EBSCOhost-2)

# VoIP Protocols

Voice over IP has been implemented in various ways using both proprietary protocols and protocols based on open standards. These protocols can be used by a VoIP phone, special-purpose software, a mobile application or integrated into a web page. VoIP protocols include:

* Session Initiation Protocol (SIP), connection management protocol developed by the IETF
* H.323, one of the first VoIP call signaling and control protocols that found widespread implementation. However, since the development of newer, less complex protocols such as MGCP and SIP, H.323 deployments are increasingly limited to carrying existing long-haul network traffic.[citation needed]
* Media Gateway Control Protocol (MGCP), connection management for media gateways
* H.248, control protocol for media gateways across a converged internetwork consisting of the traditional public switched telephone network (PSTN) and modern packet networks
* Real-time Transport Protocol (RTP), transport protocol for real-time audio and video data
* Real-time Transport Control Protocol (RTCP), sister protocol for RTP providing stream statistics and status information
* Secure Real-time Transport Protocol (SRTP), encrypted version of RTP
* Session Description Protocol (SDP), file format used principally by SIP to describe VoIP connections
* Inter-Asterisk eXchange (IAX), protocol used between VoIP servers
* XMPP, instant messaging, presence information, and contact list maintenance
* Jingle, adds peer-to-peer session control to XMPP
* Skype protocol, proprietary Internet telephony protocol suite based on peer-to-peer architecture

# Quality of Service

Communication on the IP network is perceived as less reliable in contrast to the circuit-switched public telephone network because it does not provide a network-based mechanism to ensure that data packets are not lost, and are delivered in sequential order. It is a best-effort network without fundamental [Quality of Service](https://en.wikipedia.org/wiki/Quality_of_Service) (QoS) guarantees. Voice, and all other data, travels in packets over IP networks with fixed maximum capacity. This system may be more prone to congestion[[*citation needed*](https://en.wikipedia.org/wiki/Wikipedia:Citation_needed)] and [DoS attacks](https://en.wikipedia.org/wiki/Denial-of-service_attack" \o "Denial-of-service attack)[[12]](https://en.wikipedia.org/wiki/Voice_over_IP#cite_note-12) than traditional [circuit switched](https://en.wikipedia.org/wiki/Circuit_switched) systems; a circuit switched system of insufficient capacity will refuse new connections while carrying the remainder without impairment, while the quality of real-time data such as telephone conversations on packet-switched networks degrades dramatically.[[13]](https://en.wikipedia.org/wiki/Voice_over_IP#cite_note-cisco-13) Therefore, VoIP implementations may face problems with [latency](https://en.wikipedia.org/wiki/Latency_(engineering)), packet loss, and [jitter](https://en.wikipedia.org/wiki/Jitter).[[13]](https://en.wikipedia.org/wiki/Voice_over_IP#cite_note-cisco-13)[[14]](https://en.wikipedia.org/wiki/Voice_over_IP#cite_note-14)

By default, network routers handle traffic on a first-come, first-served basis. Fixed delays cannot be controlled as they are caused by the physical distance the packets travel. They are especially problematic when satellite circuits are involved because of the long distance to a [geostationary satellite](https://en.wikipedia.org/wiki/Geosynchronous_satellite) and back; delays of 400–600 ms are typical. Latency can be minimized by marking voice packets as being delay-sensitive with QoS methods such as [DiffServ](https://en.wikipedia.org/wiki/DiffServ" \o "DiffServ).[[13]](https://en.wikipedia.org/wiki/Voice_over_IP#cite_note-cisco-13)

Network routers on high volume traffic links may introduce latency that exceeds permissible thresholds for VoIP. When the load on a link grows so quickly that its switches experience queue overflows, congestion results and data packets are lost. This signals a transport protocol like [TCP](https://en.wikipedia.org/wiki/Transmission_Control_Protocol) to reduce its transmission rate to alleviate the congestion. But VoIP usually uses [UDP](https://en.wikipedia.org/wiki/User_Datagram_Protocol) not TCP because recovering from congestion through retransmission usually entails too much latency.[[13]](https://en.wikipedia.org/wiki/Voice_over_IP#cite_note-cisco-13) So QoS mechanisms can avoid the undesirable loss of VoIP packets by immediately transmitting them ahead of any queued bulk traffic on the same link, even when that bulk traffic queue is overflowing.

VoIP endpoints usually have to wait for completion of transmission of previous packets before new data may be sent. Although it is possible to preempt (abort) a less important packet in mid-transmission, this is not commonly done, especially on high-speed links where transmission times are short even for maximum-sized packets.[[15]](https://en.wikipedia.org/wiki/Voice_over_IP#cite_note-ciscoPacket-15) An alternative to preemption on slower links, such as dialup and [digital subscriber line](https://en.wikipedia.org/wiki/Digital_subscriber_line) (DSL), is to reduce the maximum transmission time by reducing the [maximum transmission unit](https://en.wikipedia.org/wiki/Maximum_transmission_unit). But every packet must contain protocol headers, so this increases relative header overhead on every link traversed, not just the bottleneck (usually Internet access) link.[[15]](https://en.wikipedia.org/wiki/Voice_over_IP#cite_note-ciscoPacket-15)

The receiver must resequence IP packets that arrive out of order and recover gracefully when packets arrive too late or not at all. [Jitter](https://en.wikipedia.org/wiki/Jitter) results from the rapid and random (i.e. unpredictable) changes in queue lengths along a given Internet path due to competition from other users for the same transmission links. VoIP receivers counter jitter by storing incoming packets briefly in a "de-jitter" or "playout" [buffer](https://en.wikipedia.org/wiki/Buffer_(telecommunication)), deliberately increasing latency to improve the chance that each packet will be on hand when it is time for the [voice engine](https://en.wikipedia.org/wiki/Voice_engine)to play it. The added delay is thus a compromise between excessive latency and excessive [dropout](https://en.wikipedia.org/wiki/Dropout_(electronics)), i.e. momentary audio interruptions.

Although jitter is a random variable, it is the sum of several other random variables that are at least somewhat independent: the individual queuing delays of the routers along the Internet path in question. Thus according to the [central limit theorem](https://en.wikipedia.org/wiki/Central_limit_theorem), we can model jitter as a [gaussian random variable](https://en.wikipedia.org/wiki/Gaussian_random_variable" \o "Gaussian random variable). This suggests continually estimating the mean delay and its standard deviation and setting the playout delay so that only packets delayed more than several standard deviations above the mean will arrive too late to be useful. In practice, however, the variance in latency of many Internet paths is dominated by a small number (often one) of relatively slow and congested "bottleneck" links. Most Internet backbone links are now so fast (e.g. 10 Gbit/s) that their delays are dominated by the transmission medium (e.g. optical fiber) and the routers driving them do not have enough buffering for queuing delays to be significant.

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).[[16]](https://en.wikipedia.org/wiki/Voice_over_IP#cite_note-16) In such a way, temporary failures have less impact on the communication quality. In [capillary routing](https://en.wikipedia.org/wiki/Capillary_routing) it has been suggested to use at the packet level [Fountain codes](https://en.wikipedia.org/wiki/Fountain_code) or particularly [raptor codes](https://en.wikipedia.org/wiki/Raptor_code) for transmitting extra redundant packets making the communication more reliable.[[*citation needed*](https://en.wikipedia.org/wiki/Wikipedia:Citation_needed)]

A number of protocols have been defined to support the reporting of [quality of service](https://en.wikipedia.org/wiki/Quality_of_service) (QoS) and [quality of experience](https://en.wikipedia.org/wiki/Quality_of_experience) (QoE) for VoIP calls. These include [RTCP](https://en.wikipedia.org/wiki/RTP_Control_Protocol) Extended Report ([RFC 3611](https://tools.ietf.org/html/rfc3611)), [SIP](https://en.wikipedia.org/wiki/Session_Initiation_Protocol) RTCP Summary Reports, H.460.9 Annex B (for [H.323](https://en.wikipedia.org/wiki/H.323)), [H.248.30](https://en.wikipedia.org/wiki/Media_Gateway_Control_Protocol_(Megaco)) and [MGCP](https://en.wikipedia.org/wiki/Media_Gateway_Control_Protocol_(MGCP)) extensions. The [RFC 3611](https://tools.ietf.org/html/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 (because of jitter), packet loss/discard burst metrics (burst length/density, gap length/density), network delay, end system delay, signal / noise / echo level, [Mean Opinion Scores](https://en.wikipedia.org/wiki/Mean_Opinion_Score) (MOS) and R factors and configuration information related to the jitter buffer.

[RFC 3611](https://tools.ietf.org/html/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. [RFC 3611](https://tools.ietf.org/html/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.

Rural areas in particular are greatly hindered in their ability to choose a VoIP system over PBX. This is generally down to the poor access to superfast broadband in rural country areas. With the release of 4G data, there is a potential for corporate users based outside of populated areas to switch their internet connection to 4G data, which is comparatively as fast as a regular superfast broadband connection. This greatly enhances the overall quality and user experience of a VoIP system in these areas.