

VoIP (voice over IP) is the transmission of voice and multimedia content over Internet Protocol ([IP](#)) networks. VoIP is enabled by a group of technologies and methodologies used to deliver voice communications over the internet, enterprise [local area networks](#) or [wide area networks](#).

Download this free guide



## Download Our Buyer's Guide to Video Conferencing Systems

Video has become a mainstream method of collaboration for businesses of all sizes. In this buyer's guide, learn your options when it comes to video conferencing technology, and how to find the best system for your business.

Start Download

VoIP historically referred to using IP to connect private branch exchanges ([PBXs](#)), but the term is now used interchangeable with [IP telephony](#).

### How does VoIP work?

VoIP [encapsulates](#) audio via a [codec](#) into data packets, transmits them across an IP network and unencapsulates them back into audio at the other end of the connection. [VoIP endpoints](#) include dedicated desktop [VoIP phones](#), [softphone](#) applications running on PCs and mobile devices, and [WebRTC](#)-enabled browsers.

By eliminating the use of [circuit-switched](#) networks for voice, VoIP reduces network infrastructure costs, enables providers to deliver voice services over their [broadband](#) and private networks, and allows enterprises to operate a single voice and data network. VoIP also piggy-backs on the resiliency of IP-based networks by enabling fast [failover](#) around outages and redundant communications between endpoints and networks.

### VoIP protocols and standards

VoIP endpoints typically use [International Telecommunication Union \(ITU\)](#) standard codecs, such as [G.711](#), which is the standard for transmitting [uncompressed](#) packets, or [G.729](#), which is the standard for compressed packets. Many equipment vendors also use their own proprietary codecs. Voice quality may suffer when

compression is used, but compression reduces bandwidth requirements. VoIP typically supports non-voice communications via the [ITU T.38](#) protocol for sending faxes over a VoIP or IP network in real time.

## PRO+ Content

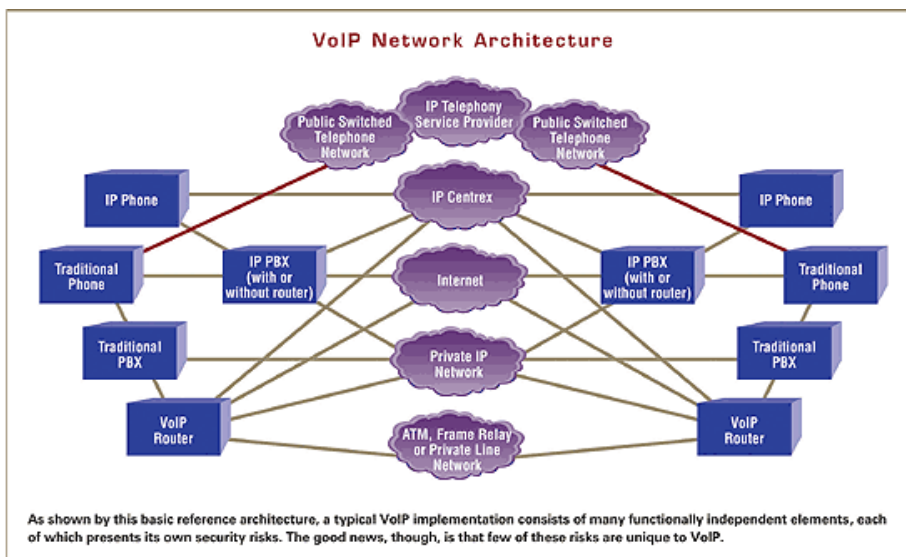
E-Zine

[The right call: Choosing the best VoIP telephone for your needs](#)

E-Handbook

[How to buy Voice over Internet Protocol services](#)

Once voice is encapsulated onto IP, it is typically transmitted via the real-time transport protocol or through its encrypted variant, [secure real-time transport](#) protocol. The [Session Initiation Protocol](#) (SIP) is most often used for signaling that is necessary to create, maintain and end calls. Within enterprise or private networks, [quality of service](#) is typically used to prioritize voice traffic over non-[latency](#)-sensitive applications to ensure acceptable voice quality.



Additional components of a typical VoIP system include the following: an [IP-PBX](#) to manage user phone numbers; devices; features and clients; gateways to connect networks and provide failover or local survivability in the event of a network outage; and [session border controllers](#) to provide security, call-policy management and network connections. A VoIP system can also include location-tracking databases for [E911](#) (enhanced 911)

call routing, and management platforms to collect call-performance statistics for reactive and proactive voice-quality management.