VoIP - IP Telephony

IP telephony (short for Internet protocol telephony and Internet telephony) or Voice over IP, is telephony over computer networks which are structured according to Internet standards. Typical telephony information, i.e. voice and control information, for example for setting up a connection, is transmitted via a data network. The call participants can be connected to computers, telephones specialising in IP telephony or classic telephones connected via special adapters.

IP telephony is a technology that makes it possible to implement the telephone service on IP infrastructure so that it can replace conventional telephone technology including ISDN and all components.

The aim of using IP telephony for communications network operators is to reduce costs through a uniformly structured and operated network. Due to the long period of use of classic telephony systems and the necessary new investments for IP telephony, the change for existing providers is often implemented as a long, smooth transition ("smooth migration"). Meanwhile, both technologies exist in parallel. This results in a clear need for solutions to connect both telephony systems (e.g. via VoIP gateways) and the necessity for targeted planning of the system change, taking into account the respective possibilities for cost and performance optimisation. The number of providers using only new technology (i.e. IP telephony instead of conventional telephones) is increasing. At the end of 2016, around 25.2 million people in Germany were using Voice over IP technology.

Switching VoIP telephone calls - Switching service

The switching of telephone calls is an essential task in computer networks. Since many users are dynamically connected to the Internet, so that the IP address changes frequently, the IP address itself is not a "telephone number" for contacting VoIP telephones. An exchange service in the form of a server takes over this task and enables telephony with changing IP addresses of the IP telephones.

- VoIP phones log on to the server (for example, SIP server), so the server knows the current IP address of the phones.
- Using the IP address of the telephone, which has been made known to the server, it can take over the switch, and the dialed IP telephone rings depending on this IP address (i.e. anywhere in the world, if the IP telephone has registered itself from there with the switch server via the Internet).
- Communication between the IP telephones can take place independently of the server.
- There are commercial services that offer a local telephone with an account for the exchange server, which can also be reached via the fixed network. IP calls are usually free of charge.
- If a fixed IP address exists, it is possible to operate a switching server on the corresponding computer (for example OpenSIPS) in order to connect several switching servers to each other in a similar way to the connection of several local networks in the fixed network. Commercial solutions often include partner networks that establish a free connection between VoIP partner networks. Network selection is often limited because companies have to generate revenue from VoIP phone connections to the regular fixed network. Free, self-operated open-source telephony servers can technically form a network of exchanges independent of these economic limits on the Internet. Even though SIP telephony servers function technically well, there is currently no institutionalized networking of such SIP switch servers.

Functional Principle

VoIP encapsulation

Telephoning with IP telephony can be the same for the subscriber as in classic telephony. As with conventional telephony, the telephone call is divided into three basic processes: call setup, call transmission and call termination. In contrast to traditional telephony, VoIP does not connect through dedicated "lines", but instead digitizes the voice and transports it in small data packets using the Internet protocol.

Signalling protocols

Connections (call control, signalling) are established and terminated using a protocol that is separate from voice communication. The negotiation and exchange of parameters for voice transmission takes place via protocols other than those of the call control.

In order to establish a connection to a call partner in an IP-based network, the current IP address of the called party must be known within the network, but not necessarily on the caller’s side. Geographically fixed connections such as in the fixed network (PSTN) do not exist in purely IP-based networks. The accessibility of the called party is made possible, similar to in mobile networks, by a previous authentication of the called party and a related announcement of his current IP address. In particular, a connection can be used regardless of the location of the user, which is referred to as nomadic use.
A fixed assignment of telephone numbers to IP addresses is not possible due to a change of location of the subscriber, a change of user on the same PC or dynamic address assignment when setting up a network connection. The general solution is that the subscribers or their terminals store their current IP address on a service computer (server) under a user name. The computer for connection control, or sometimes the terminal device of the caller himself, can request the current IP address of the desired call partner from this server via the selected user name and thus establish the connection.

Common signalling protocols are:

- **SIP** - Session Initiation Protocol, IETF RFC 3261
- **SIPS** - Session Initiation Protocol over SSL, RFC 3261
- **H.323** - Packet-based multimedia communications systems, an ITU-T standard
- **IAX** - Inter-Asterisk eXchange protocol
- **ISDN** over IP - ISDN/CAPI-based protocol
- **MGCP** and Megaco - Media Gateway Control Protocol H.248, joint specification of ITU-T and IETF
- **MINET** - from Mitel
- **Skinny Client Control Protocol** - from Cisco Systems (not to be confused with SCCP (Q.71x) of ITU-T)
- **Jingle** - extension of [XMPP protocol](https://xmpp.org), based on Google Talk

Related Links:

- ADPCM - Adaptive Differential Pulse Code Modulation ([Snom Service Hub](https://www.snom.com)
- Basics of VoIP ([Snom Service Hub](https://www.snom.com)
- Codec ([Snom Service Hub](https://www.snom.com)
- CSTA - Computer-supported telecommunications applications ([Snom Service Hub](https://www.snom.com)
- CTI - Computer Telephony Integration ([Snom Service Hub](https://www.snom.com)
- ENUM - Telephone Number Mapping ([Snom Service Hub](https://www.snom.com)
- Hot Desking ([Snom Service Hub](https://www.snom.com)
- How can I obtain a SIP trace from the phone ([Snom Service Hub](https://www.snom.com)
- How to obtain a SIP trace from a deskphone ([Snom Service Hub](https://www.snom.com)
- IP-PBX ([Snom Service Hub](https://www.snom.com)
- IVR - Interactive Voice Response ([Snom Service Hub](https://www.snom.com)
- Jitter ([Snom Service Hub](https://www.snom.com)
- LLDP - Link Layer Discovery Protocol ([Snom Service Hub](https://www.snom.com)
- PBX - Private Branch Exchange ([Snom Service Hub](https://www.snom.com)