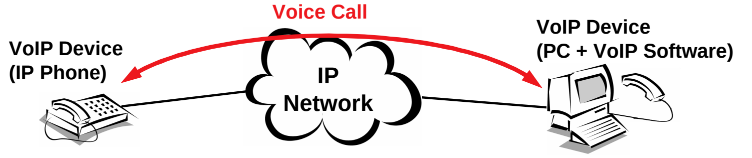
Voice over IP Solutions (VoIP)

VoIP Introduction
What is VoIP?



A-VOIP-2

The fundamental idea behind **Voice over IP (VoIP)** is that a phone call is sent via a series of data packets using an IP communication infrastructure. This is done to achieve a better total bandwidth utilization by allowing voice and other digital data to be “mixed” on the same IP network.

The term Voice over IP is used to describe the fundamental technology, concepts and standards. The terms IP Telephony and Internet Telephony or LAN Telephony describe specific implementations of VoIP.

IP Telephony provides traditional and new advanced telephony functions over a managed or private network that can guarantee the necessary quality, predicatbility and low packet delay from one end of an network to the other. These networks are typically owned and run by a single organization like carriers and ISPs.

Internet Telephony or **Voice On the Net (VON)** refers to an implementation of the same kind of technology using the Internet as a replacement for PSTN services. But although the Internet is a communication infrastructure based on IP it is a patchwork of many, mostly unmanaged networks, owned and operated by disparate organizations. The resulting network infrastructure therefore does not guarantee the quality, low delays and predictability necessary to provide high quality voice services. The idea of making almost cost free calls all over the Internet nevertheless has found an constantly increasing number of users all over the world even if Internet telephony still lacks many of the traditional PSTN capabilities.

LAN Telephony refers to technical solutions where VoIP technology is used to replace conventional PBX technology in a high bandwidth local area network.

VoIP Introduction
Advantages of VoIP

Advantages for end users / companies:

- **Dramatic cost reduction for telephony**
- **Only one communication infrastructure internally (companies)**
- **Better scalability of the communication infrastructure**
- **New integrated voice / video / data applications**
- **Better and reduced management of the infrastructure**
- **Future proof**

Advantages for providers and carriers:

- **Dramatic cost reduction (infrastructure)**
- **New integrated and value add services**
- **New business models**

SMC Networks

A-VOIP-3

There is one main motivation for implementing VoIP: cost reduction. But it is not only about cheaper phone calls, there are many advantages for customers and providers:

VoIP Introduction
Comparison PSTN / VoIP

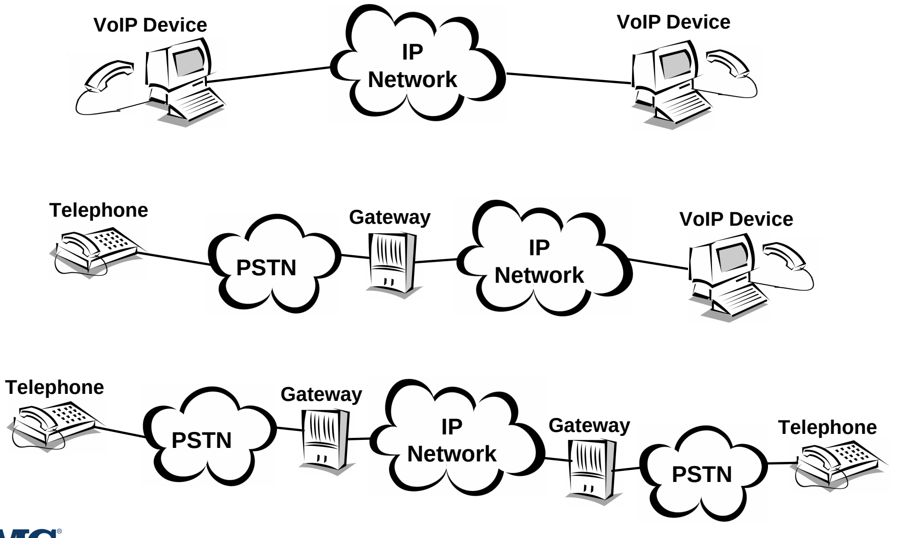
	PSTN	VoIP
Communication type	channel oriented	packet oriented, best-effort delivery
Communication channel	fixed path, out-of-band signaling	virtual channel
Data flow	continuous	variable
Bandwidth (single call)	64kbps each direction	depends on compression algorithm used
Compression	no	yes - various, depending on implementation
Delay	minimal, not relevant	significantly higher, value depends on many variables
Quality	very high	depends on many variables
Error Rate	minimal, practically zero	depends on many variables
Standardization	stable, comprehensive	many different approaches, techn. alternatives, standardization bodies

A-VOIP-4

VoIP technology is very young compared to the traditional technology behind the PSTN. VoIP technology promises dramatic cost reduction for voice communication and is the basis for the development of new powerful services and applications. But there are also some trade-offs when we compare the fundamental technologies. Although the PSTN infrastructure and technology is relative complex (and expensive) the PSTN delivers very reliable voice services at a very high quality with minimal delays. The technology is proven, reliable and generally accepted.

It is the underlaying packet oriented networking technology that makes VoIP the more efficient solution for voice communication. But this is also the reason for significantly higher delays, variable data flows, sometimes unpredictable network bandwidth and most of the time lower quality. This technology approach is very flexible and allows a broad range of different implementations. These range from simple Internet Telephony solutions with no guarantes for acceptable quality for the voice communication to reliable high quality IP Telephony services or LAN Telephony that offer advanced and supplementary (CTI) services.

VoIP Introduction
VoIP Scenarios



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There are three fundamentally different VoIP scenarios when looking at typical telephony usage:

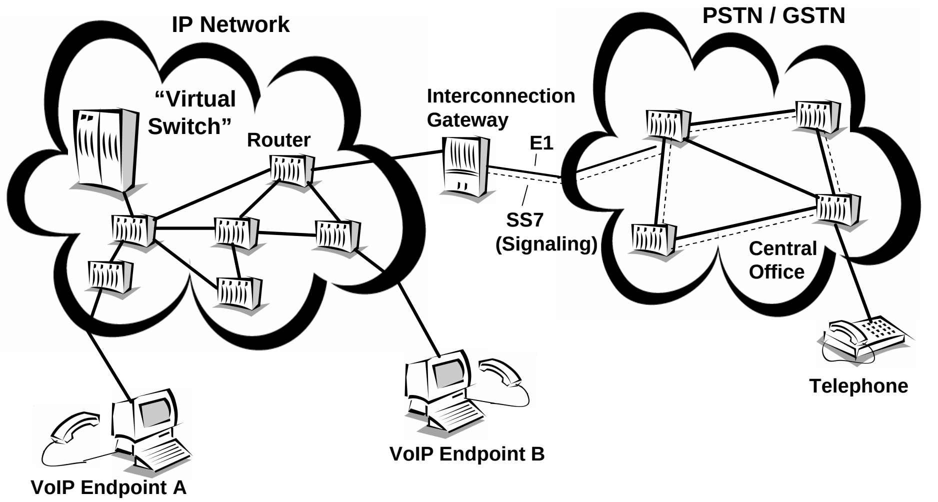
- VoIP device to VoIP device
- VoIP device to telephone
- Telephone to telephone

The scenaro “**VoIP device to VoIP device**” is referring to applications that are only using an IP network as means to interconnect the end devices. These are computers of some kind. Classic Internet Telephony and also enterprise LAN Telephony are typical examples for this scenario.

In many instances it will be necessary to establish communication with end users (devices) that are not connected to the IP network but to are conventional telephone (or other devce) connected within a PSTN. In this “**VoIP to telephone**” scenario the two very different communication infrastructures (IP network vs. PSTN) have to be interconnected to allow calls to pass from one side to the other. The devices used to connect these fundamentally different working networks are called (interconnection) gateways.

A third VoIP communication scenario whch we best describe as “**telephone to telephone**” is becoming more and more important although, in most cases invisible (transparentely deployed by the voice carier) to the users. In this scenario a “communication channel” is established between conventional access infrastructures for analogue or ISDN telephones (or other devices) using an IP network infrastructure to typically to reduce cost for long distance calls.

Building Blocks and Elements
Overview



SMC Networks

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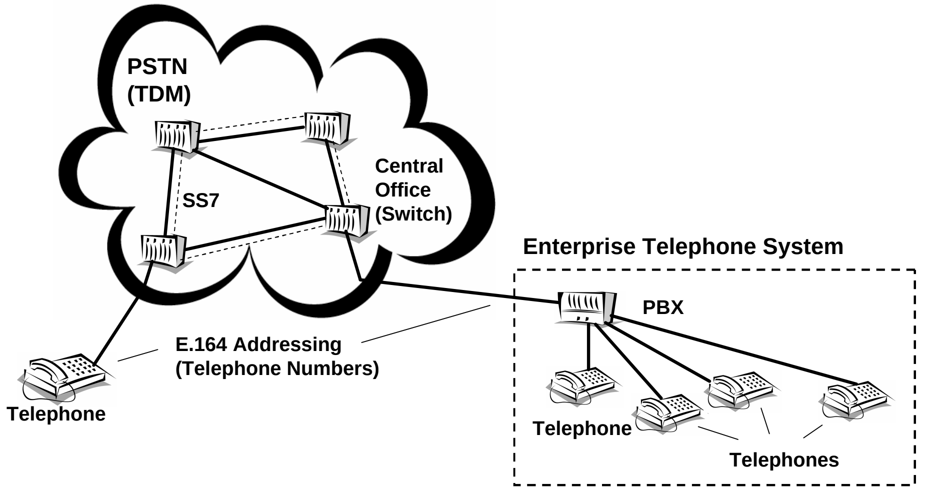
To describe the VoIP environment it is best to identify first the fundamental elements or building blocks, to be able to understand their principal structure and functions.

The above picture shows on the left side a typical **IP based network** used for VoIP applications. It can be basically any kind of IP network offering acceptable performance and reliability. On the user side VoIP endpoints (such as IP telephones or PCs) are attached to the network and can be used for voice, video and other communication applications. Special devices (switches, software PBXes, etc.) are used to provide call control and other important functions in the network.

Several application scenarios require communication between VoIP users/devices and the users/devices connected to **traditional PSTN** infrastructures. A PSTN works fundamentally in a different way to an IP networks when it comes to establishing and providing communication channels for communication endpoints. Because of this it is necessary to implement (interconnection) **gateway systems** that convert signalling and data formats and therefore can "bridge" between the fundamentally different communication infrstructures.

Building Blocks and Elements
PSTN / GSTN Overview



A-VOIP-7

PSTN (public switched telephone network) is the worlds collection of interconnected voice oriented public telephone networks that are often also referred to as the Plain Old Telephone Service (POTS). Today, these networks are almost entirely digital in technology. The final link from the central (or local) telephone office to the user (subscriber) can be digital (ISDN) but in the majority of cases still analogue. End devices connected to the PSTN have to comply to the technology used on this final link to the user.

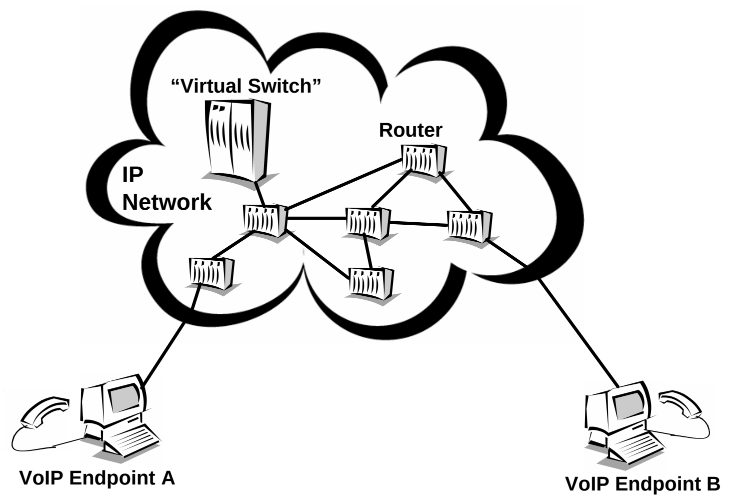
Although the PSTN today is typically digital it is by no means packet oriented(like IP networks are). From a communication technology point of view the PSTN is based on **TDM (Time Division Multiplexing)** a type of multiplexing technology. TDM repeatedly transmits a fixed sequence of time slots over a single transmission channel and combines multiple voice or data streams by assigning each stream a different time slot in the sequence.

Adressing is also different between a PSTN and an IP network. In a PSTN a telephone numbers consists of up to 15 digits, in IP networks IP addresses are used to identify individual nodes.

To ensure efficient and secure worldwide telecommunications a special network and protocol named **SS7 (Signaling System 7)** is used in the PSTN to perform call control functions (setup, closing, etc.). It also provides enhanced call features and services (toll-free and toll services, call forwarding, local number portability, etc.) SS7 is a telecommunications protocol defined by the International Telecommunication Union (ITU) and is characterized by high-speed circuit switching and out-of-band signaling.

Out-of-band signaling is signaling that does not take place over the same path as the data transfer but uses a separate digital channel (signaling link), where messages are exchanged between network elements such as telephone switches.

Building Blocks and Elements
IP Network Overview



A-VOIP-8

The above picture shows a typical IP based network used for VoIP applications. It can be basically any kind of **IP network** which offers acceptable performance and reliability.

Depending on the requirements and expectations of the users these IP networks can be either **a managed or private MAN/WAN (IP Telephony), a companies internal network (LAN/WAN) or the worldwide Internet (Internet Telephony, VOI).**

Addressing is based on **IP addresses**. This makes it necessary to provide special directory services to translate names (for example e-mail addresses) or “real” assigned telephone numbers to the IP addresses, which are used to identify an individual VoIP endpoint.

On the user side VoIP **endpoints** like IP telephones or PCs with VoIP communicateion software and hardware (for example softphones) are attached to this infrastructure. They can be used for voice, video and other communication applications. Special intelligent devices (switches, software PBXes, etc.) are used to provide call control and other important functions in the network.

Due to the fact that there are different ways to implement a VoIP network, names and functions of individual devices in the network differ. The two predominant technology alternatives are either based around H.323 or SIP standards (recommendations, RFCs, etc.).

Building Blocks and Elements
VoIP Endpoints / Devices

Examples of VoIP endpoints / devices for users:
- PC with VoIP hardware and software (soft client)
  - PC with headset, handset, etc.
  - Laptop with sound system and microphone
- IP telephones
- VoWLAN phones
- VoIP adapter (to connect analogue phones, etc.)
- VoIP PBX (Private Branch Exchange)
- IAD with integrated VoIP gateway (Barricade)
- CPEs
- …

Examples for VoIP endpoints that provide additional services:
- VoIP gateways
- Media server
- Proxies
- …

A-VOIP-9

A broad range of VoIP products are today available not only as VoIP endpoints (or devices) for the user that wishes to communicate via VoIP. Also there are systems that provide an increasing number of advanced communication services to the users:

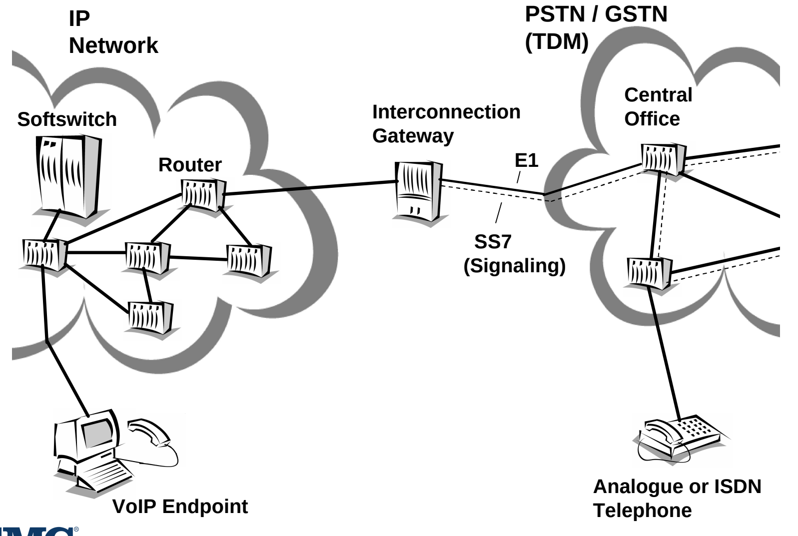
Examples of VoIP endpoints / devices for users:

- PC with VoIP hardware and software (soft client)
- PC with headset, handset, etc.
- Laptop with sound system and microphone
- IP telephones
- VoWLAN phones
- VoIP adapter (to connect analogue phones, etc.)
- VoIP PBX (Private Branch Exchange)
- IAD with integrated VoIP gateway (Barricade)
- CPEs
- …

Examples for VoIP endpoints that provide additional services:

- VoIP gateways (interconnection between TDM and VoIP networks)
- Media server (voice mail, interactive voice response applications, network announcements)
- Proxies (interconnection between private and public IP networks)
- …

Building Blocks and Elements
Interconnection Gateways



SMC Networks

A-VOIP-10

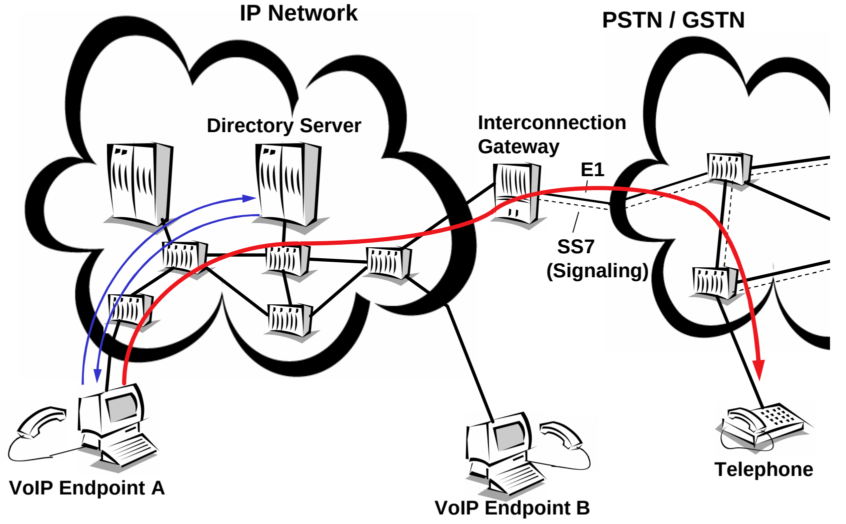
(Interconnection) gateway systems convert signalling and/or data formats and therefore can “bridge” between fundamentally different communication infrastructures.

Typical examples for such gateways are:

- **H.323-to-SS7 gateway**
- **SIP-to-SS7 gateway**
- **SIP-to-H.323 gateway**

Gateway functionality can also be found incorporated into other devices like VoIP access router (IAD) or PBX solutions.

Building Blocks and Elements
VoIP / PSTN Addressing



SMC Networks

A-VOIP-11

A **numbering plan** is a type of numbering scheme that defines the structure and use of the telephone numbers used for intra-site, inter-site and outbound telephone calls.

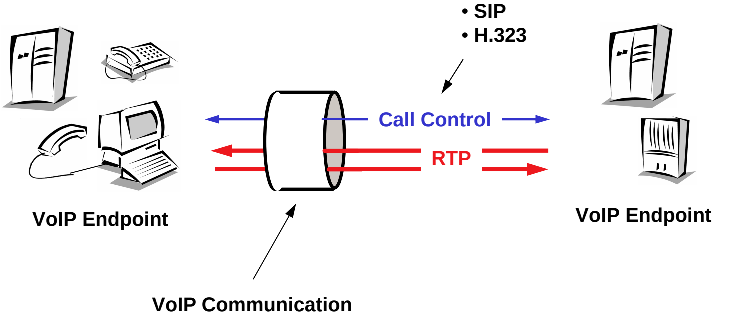
In the **PSTN** the numbering plan is clearly defined. A **telephone number** consists of up to 15 digits, made up of a one to three digit country code (CC), followed by the subscriber number (SN). The ITU recommendation **E.164** specifies the number structure and functionality/service of the numbers used for international public telecommunication. E.164 also defines all international country codes are defined.

Within IP networks **IP addresses** are used to identify individual systems. When connected to the Internet every addressable node has to have its own unique IP address. An additional level of complexity is created by the fact many users are conneted to the Internet through dial-up services where the address is dynamically assigned. For plain VoIP solutions techniques based on DNS or other directory services technologies are available.

In most cases it will be necessary to dials out of the VoIP network into the PSTN and vice versa. There are solutions and services available that offer gateways to the PSTN from a VoIP phone. By simply dialing a conventional telephone number the telephone call will be routed over the IP network to the defined gateway.

Electronic Numbering (ENUM) makes it possible to dial traditional E.164 phone numbers, but be connected entirely over the Internet. This allows E.164 adresses to be used iwithin a DNS. Directory structure. ENUM allocates a specific zone (suffix) for use with E.164 numbers. Any phone number, can be then transformed into a hostname by reversing the numbers, separating them with dots and adding the suffix. DNS can then be used to look up internet addresses for VoIP telephony services.

VoIP Protocols
Overview



A-VOIP-12

The **predominant VoIP technology alternatives** today are **H.323** and **SIP** (Session Initiation Protocol). Although fundamentally trying to solve the same problem - to deliver voice and other multimedia services over IP - these two technology approaches differ in some ways.

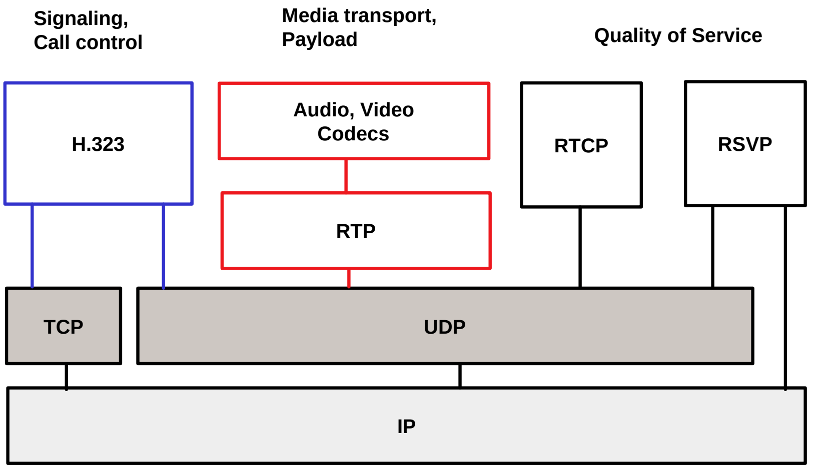
H.323 a “product” of the telecommunication industry. It is today an **ITU (International Telecommunication Union)** umbrella standard, defining a broad range of audio, video and data compression standards and protocols. Beeing an ITU standard the technology is sound and reliable but also very complex.

SIP on the other hand was and is still being developed by the **IETF**. SIP is a fast and simple signalling and application layer control protocolo for IP (Internet) multimedia applications and telephony. It is gaining fast in popularity and has be come a very attractive alternative to the much older and previously widely implemented H.323 technology.

H.323 and SIP although designed by very different standardization bodies have nevertheless a lot in common. They differ fundamentaly in signaling / call control and management but both rely on RTP (Real-Time Transport Protocol) for transporting the voice (multimedia) information. They also share many of the fundamental technologies for digitizing analogue signals.

H.323 and SIP are by no means the only protocols to build VoIP solutions. There are a broad range of technology alternatives including many proprietary solutions of individual manufacturers as well as standards like Megaco/H.248 or MGCP. **MGCP (Multimedia Gateway Control Protocol)** is typically used in VoIP solutions utilizing TV cable network infrastructures.

Signaling Protocols
H.323 Introduction



A-VOIP-13

H.323 is an interoperability standard that describes the modes of operation required for various audio, video, and/or data terminals to work together.It is one of the important standards for IP / Internet voice and video applications including IP/Internet phones as well as audio and video conferencing equipment.

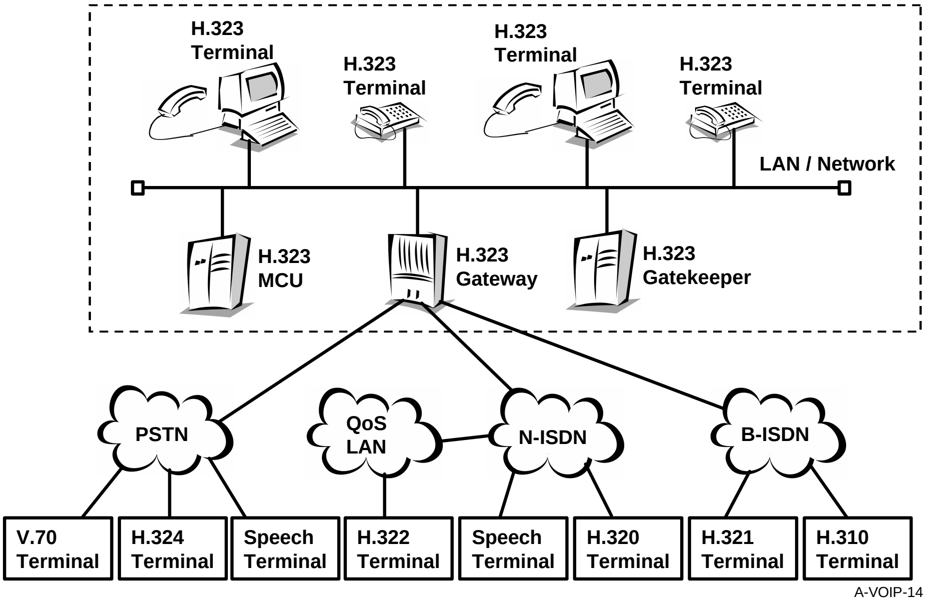
The development of H.323 originally was focussed on LAN video conferencing applications. When it was ratified in 1996 by the ITU (International Telecommunication Union) it became an umbrella standard, defining a broad range of audio, video and data compression standards and protocols.

The H.323 standard specifies the following protocols and technologies:

- Codecs for audio and video
- Registration, admission and status (H.225)
- Call signalling (H.225)
- Control signalling (H.245)
- RTP (Real-time Transfer Protocol)
- RTCP (Real-time Control Protocol)

H.323 can be implemented in various different ways. Depending on the application it can be used for audio/voice only (telephony), audio and video (video conferencing, video telephony), audio and data (whiteboaarding), or all of these applications combined (full video and data conferencing).

Signaling Protocols
H.323 Architecture



The architecture of H.323 contains several componente (or elements) that perfom specific functions in the network:

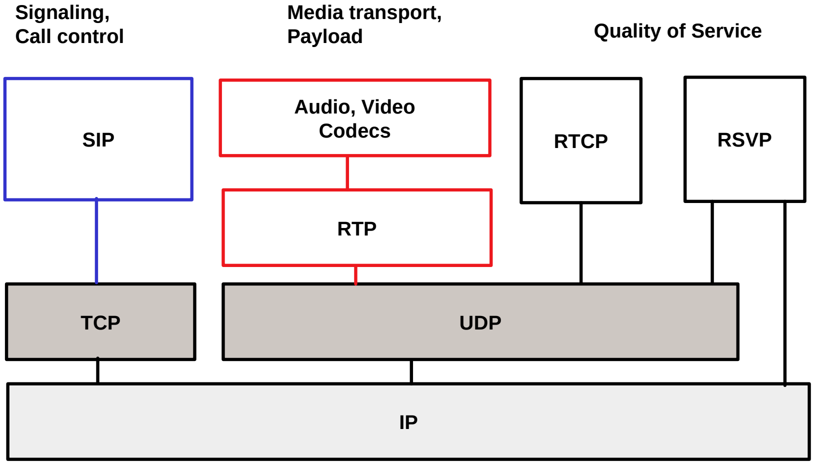
Endpoint Terminals are used for bidirectional multimedia communications. They are either a PC or a dedicated device supporting H.323 and one or more applications (like an IP phone for example). A H.323 terminal consists of at least a network interface, a System Control Unit (SCU), the H.255 Layer and an audio codec unit for voice applications. Video codecs and other user data applications are optional components. H.323 terminals are compatible with H.324 terminals on POTS, H.320 terminals on ISDN, H.310 terminals on B-ISDN, H.322 terminals on guaranteed QoS LANs and V.70 GSTN (Global Switched Telephone Network) terminals.

H.323 **Gateways** are used to interconnect various different telephony (and video conferencing) technologies. These gateways convert digitized voice into IP packetand, translate call control information.

H.323 **Gatekeeper** function as virtual PBX or switch. H.323 does not require a Gatekeeper to be implemented. If deployed it acts as the central point for switching all calls within its domain. Terminals will first contact the Gatekeeper, which then controls the calls by granting (or denying) permission for the call and by instructing endpoints about how to perform call control and other functions.

Although multipoint conferencing on the H.232 network can be done in a decentralized way, often **Multipoint Control Units (MCUs)** or Multipoint Controllers (MCs) are used for centralized multipoint conferences. MCUs perform audio mixing and voice switching between several endpoints and coordinate the call control and codec negotiation between all participating devices.

Signaling Protocols
SIP Introduction



A-VOIP-15

The **Sesssion Initiation Protocol (SIP)** is a relatively new Internet standard that basically is a simple signaling and application layer control protocol for multimedia conferencing and telephony. It is today besides H.323 the most important standard for IP / Internet voice applications.

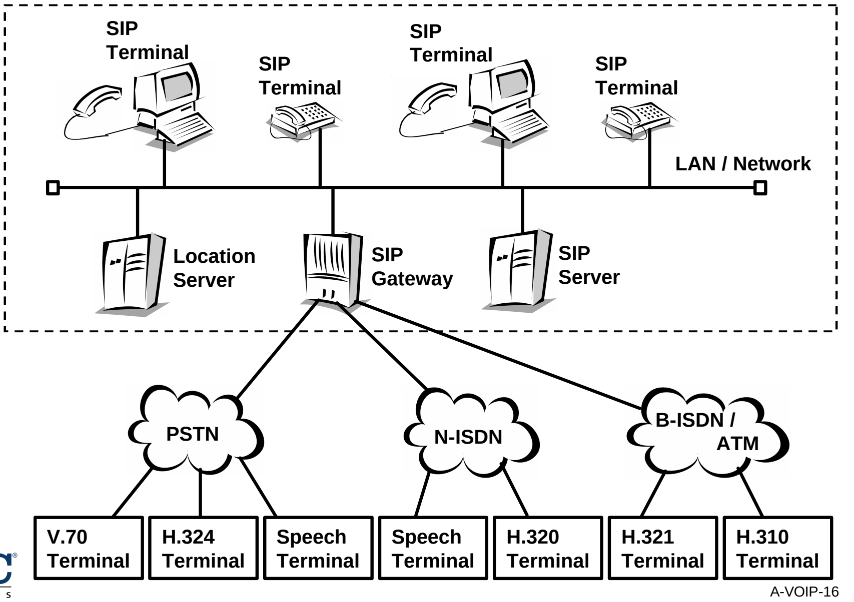
SIP is defined in RFC 2543, as part of the **MMUSIC (Multiparty Multimedia Session Control) working group** of the IETF. This group is working on a complete framework of protocols to solve abroad range of multimedia conferencing and communication solutions. These include SDP (Session Description Protocol), SAP (Session Announcement Protocol), RTSP (Real-Time Stream Protocol), SCCP (Simple Conference Control Protocol) and SIP.

SIP is a simple, low-level protocol for establishing, modifying or terminating multimedia sessions or Internet telephony calls between two or more users. Such sessions can include voice, video, chat, other multimedia data (such as interactive games).The protocol can also invite participants to unicast or multicast sessions.

SIP supports name mapping and redirection services. This makes it possible to identify users regardless where they are and allows users to initiate and receive connections and services from any location.

SIP is a simple request-response protocol, dealing with requests from clients and responses from servers. Users are identified by SIP URLs. SIP determines the end system to be used for the session, the communication media and media parameters. It then initiates the communication. It establishes call parameters at each end of the communication, and handles call transfer and termination. Like H.323 SIP uses the RTP (Real-Time Transport Protocol) for transporting the voice (multimedia) information.

Signaling Protocols
SIP Architecture



A-VOIP-16

The SIP architecture contains several componente (or elements) that perfom specific functions in the network:

SIP Endpoint Terminals are used for bidirectional multimedia communications and are either a PC or a dedicated device supporting SIP and one or more applications (like an IP phone for example). SIP terminals are compatible with H.324 terminals on POTS, H.320 terminals on ISDN, H.310 terminals on B-ISDN, H.322 terminals on guaranteed QoS LANs and V.70 GSTN (Global Switched Telephone Network) terminals.

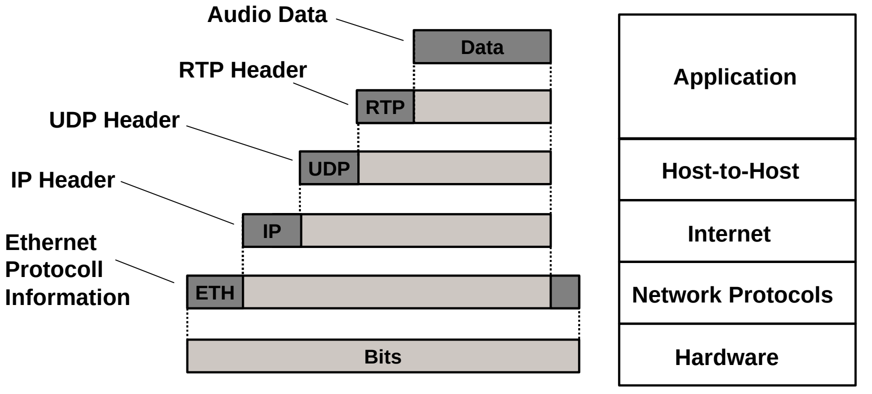
A **Location Server** maintain the IP addresses of all users and is, for example responsible for translating alias names to IP addresses.

The **SIP Server** is responsible for call handling and call establishment. Call control can be implemented in one of two ways - either in proxy mode or in redirect mode.

Just like H.323 Gateways, **SIP Gateways** are used to interconnect various different telephony (and video conferencing) technologies. These gateways convert digitized voice into IP packets and translate call control information.

Media Transport Protocols

RTP



Structure of an Ethernet RTP Packet

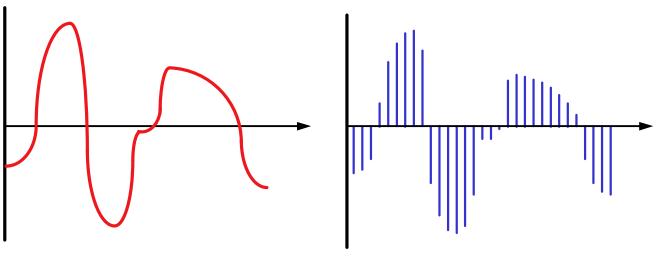
TCP/IP Architecture

A-VOIP-17

RTP (Real-time Transport Protocol) in conjunction with RTCP (Real-Time Transport Control Protocol) has been used by the research and university community for years on the Mbone. This network is the Internet's still somewhat experimental multicast backbone, for video conferencing and telephony over IP. It never gained general acceptance because of its very limited signalling capabilities.

The **(RTP)** was specifically designed to allow the transport of isochrounous data across a packet network. It addresses some of the problems occuring in such networks such as jitter and desequenced packets. Typically it is used in combination with UDP. On top of UD,P it can be sent in multicast IP packets, which means that a RTP stream originating from a single source node can be simultaneously sent to multiple destination nodes.

Codecs
Introduction



Analogue Signal

Sampled and quantisized analogue Signal

SMC Networks

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In order to be transmitted across computer networks voice (the same is true in principle for video) has to be converted from its analogue form into a digital format. This digitizing of data is done by discretizing the signal in time (sampling) and then discretizing the signal in amplitude.

The systems that perform this conversion process (in both directions) are called **codecs**. In adition to the actual digitalization of analogue signals these codecs further processes the data to further reduce the amount of data further and to optimize it for the transmission. Parameters like the sampling rate, the number of bits used to encode the data and the methods used to optimize the transmission greatly influence the quality of the digitized speech.

VoIP systems may have several different codecs implemented. This means that negotiating and agreeing on the codec actually used is part of any VoIP communication.

Typical examples of codes are

- **G.711 PCM**
- **G.723.1 MP-MLQ**
- **G.729 CS-ACELP**
- **G.726 ADPCM**
- G.723 MP-MLQ
- G,728 LD-CELP

Quality of Service
Introduction

To transmit voice with an apropriate quality several requirements have to be met:

- Enough bandwidth has to be provided reliably and any time
- Delay has to be kept constantly below a certain level
- Changes in the delay (jitter) have to be kept to a minimum
- Good voice codecs (appropriate to the avail. bandwidth)
- Echo cancellation
- Miminal or no packet loss

SMC Networks

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When it comes to quality expectations for telephony applications, the digital PSTN of today sets a very high standard. Short delays, a clear signal and high reliability is what we expect of our daily used telephone services. Although VoIP solutions have the potential to deliver comparable communication quality in theory, the actual quality we get depends on many differrent factors.

To improve the communication quality in packet oriented networks many technical solutions are either already available or under development. QoS technologies play an increasing role in improving the acceptance of VoIP.

“Qualty of Service” (QoS) might be best defined as the ability of a (communication) system or network to guarantee specific parameters or values when it comes to uptime, throughput, delays, error rate, etc.

The term QoS as used in the context of IP networks and VoIP refers to a broad collection of networking protocols, technologies and techniques that provide and control buffering, voice packet priorization, echo cancellation and guaranteed bandwidth.

Quality of Service

Layer 2/3 Networking Technologies

QoS solutions can be roughly divided into the following groups:

- WAN/MAN technologies with QoS capabilities
- Layer 2 switching technologies
- Layer 3 switching and routing protocols and technologies
- Internet / IP QoS protocols
- Rate control solutions

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A-VOIP-20

QoS can be implemented at very different levels in the network. Solutions largely depend on the underlaying network technology and structure.

WAN MAN protocols and technologies used by carriers and other providers of WAN/MAN services quite often have comprehensive QoS functionality built in. Typical examples of such technologies are **ATM (Asynchronous Transfer Mode, SONET (Synchronous Optical Network)** and **Frame Relay**.

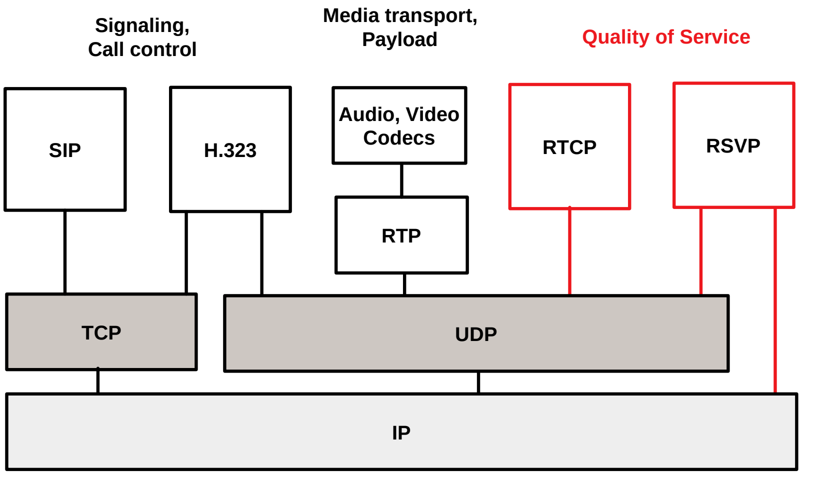
Layer 2 as well as Layer 3 switching technology is available today to deliver QoS functionality in typical enterprise, but also some carrier networking infrastructures. These technologies range from simple **CoS (Class of Service)** implementations in LAN switches (different priority queues) to sophisticated layer 2/3 switching solutions. Typical examples for such QoS switching solutions are either prorietary approaches like IP Switching or Tag Switching or standards based technologies such as **MPLS (Multiprotocol Label Switching).**

Of special interest for VoIP solutions in general are technologies and protocols providing Layer 3 QoS solutions that allow the limitations and problems in (also large) IP network including the Internet to be overcome. Typical examples for such QoS solutions or approaches are **RTCP (Real-Time Transport Control Protocol), RSVP (Resource ReServation Protocol)** and **RTSP (Resource Reservation Protocol).**

In order to be able to solve some QoS problems at specific bottlenecks in the network infrastructure, special **rate control solutions** have been developed. These **bandwith management** and **traffic shaping** solutions typically work transparently and allow the priorization, reservation, limitation and blocking of IP communication at an application/protocol and/or network/host level.

Quality of Service

Internet / IP QoS Protocols



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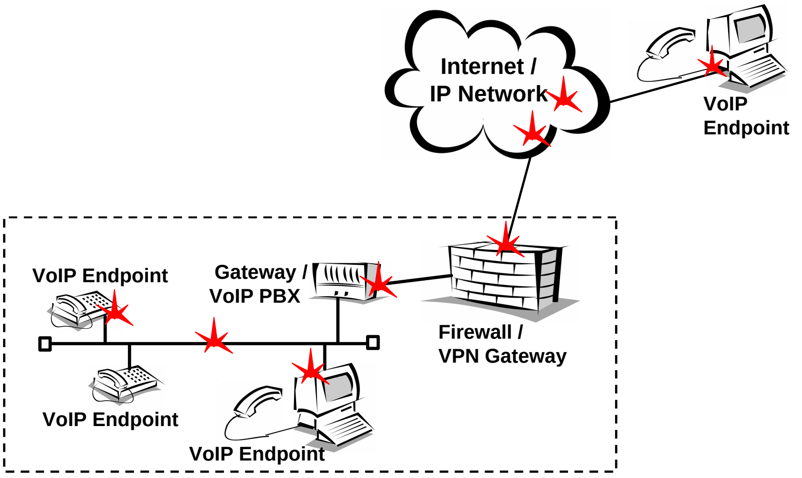
Before multimedia and VoIP applications can be widely used, the IP networks must be modified to support real-time QoS and provide controlled end to end delays. This is especially true for the Internet infrastructure.

In typical VoIP communications **RTP (Real-Time Transport Protocol)** will be used for transporting the actual voice (multimedia) information. Because RTP does not provide the QoS functionality needed various protocols have been developed or are under development. These offer additional levels of control and better QoS for RTP communications in IP networks and the Internet.

RTCP (Real-Time Transport Control Protocol) is a protocol to exchange control information from time to time bewteen the participants of a particular RTP session. RTCP control packets can include information about the mapping of participants to specific single stream sources and about the quality of the transmission in the network.

The **Resource ReServation Protocol (RSVP),** is an Internet control protocol specially designed to enable applications to request a specific quality of service from the network. It is used by both network hosts and routers. RSVP treats an application flow as a one-way connection sending QoS request from the sender to the receiver. RSVP is a transport layer protocol that uses IP as its network layer and is designed to operate with other unicast and multicast routing protocols.

Security
Overview



SMC Networks

A-VOIP-22

The fast deployment of VoIP solutions not only offers new possibilities and opportunities but also introduces **new risks**. The fundamental technology changes for voice communication introduce new threats and new challenges for security specialists and network administrators.

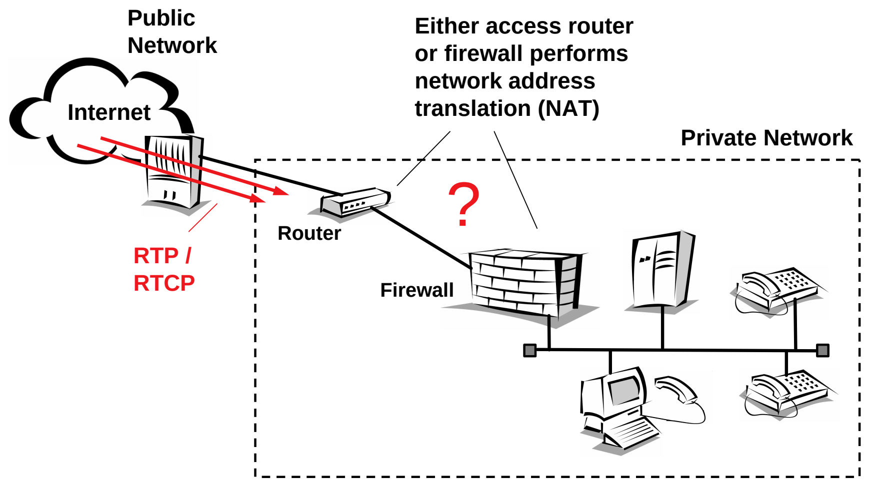
Eavesdropping on conversations in the network by intercepting a VoIP connection is only one example of the new security threats. However, security measures in general to not differ very much from networks without VoIP. Whether the goal of the attacker is to gain information, steal resources or to disrupt business processes, the used approaches and tools are pretty much the same.

But there are certainly security issues to be adressed that result from the specific **VoIP technology** implemented. The SIPstandard for example does include functions to enhance media security (encryption), message exchange security and authentication. Also in H.323 there are functions and protocols (described in H.235) that are designed to provide better levels of authentication, privacy and integrity.

In situations where higher levels of privacy and security are needed, technologies like **firewalls, authentification systems and VPN technology** can be implemented.

Some of the "classic" attack patterns or techniques like DDoS attacks, will probably also be directed at VoIP servers and gateways. Firewall systems need to be configured or even upgraded to be able to prevent damage to the network and its components.

Security
VoIP and Firewalls / NAT



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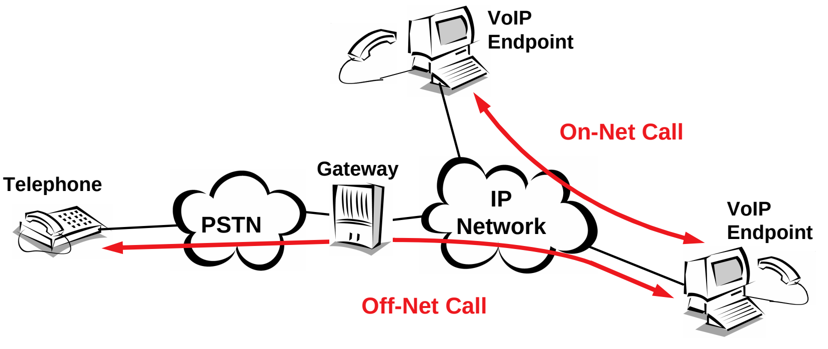
Although not a security problem, special attention has to be given to communication issues and problems . In particular, those that have to do with techniques (mainly network address translation) used in many firewall solutions and IADs (Internet Access Devices) - such as access routers.

NAT (Network Address Translation) techniques allow more than one system (for example a LAN) to be connect to the Internet using a single (often even dynamically assigned) IP address. The address translation device converts the outgoing IP address of each LAN device into its single Internet address and vice versa. Because of serious limitations related to incoming connections that cannot be simply directed to the individual systems in the internal network, these network address translation devices need special software that work at an application/protocol level (for example **SIP signaling proxies** and **H.323 proxies**) to overcome these issues.

Other issues or problems that have to be taken into account when using systems like firewalls or VPN gateways in VoIP solutions have to do with the **additional latency** that these systems would cause. This means that a lot of attention has to be given to the overall network design and the selection of network and security devices that are in the communication path between VoIP systems to keep latency at a minimum.

VoIP Scenarios
Introduction



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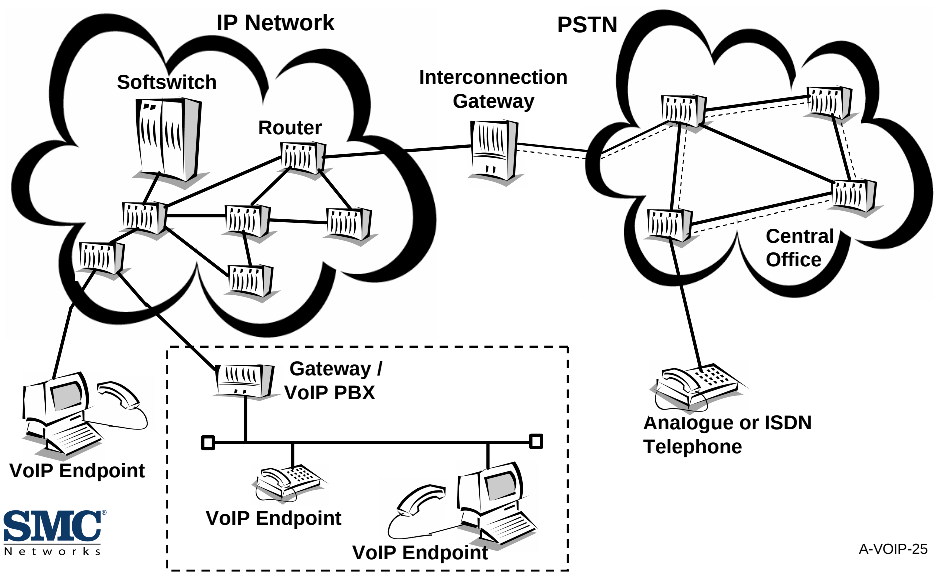
VoIP can be deployed and implemented in many different ways and application scenarios. The complexity of VoIP results to some degree from the fact that several different VoIP technologies, standards and implementations are often combined.

The terms on-net and off-net for example refer to two different levels of complexity in VoIP communications. An **on-net call** originates and terminates within a single enterprises or providers network. **Off-net calls** on the other hand cross the bounbdaries of the providers respectively enterprise network and require therefore special technology (gateways, etc.) and have other compatibility aspects (signaling, coding, etc.) taken into account.

A good way of simplifying the subject further is to look at specific application scenarios or implementations. **Typical VoIP scenarios or applications** are for example :

- Residential user / branch office with ITSP
- (On-net) Internet Telephony solutions
- Long distance call services (PSTN to PSTN)
- LAN Telephony / small PBX replacement
- WLAN hotspot with ITSP
- VoIP Access Router / IAD for SME networks

VoIP Scenarios
Internet Telephone Services



SMC Networks

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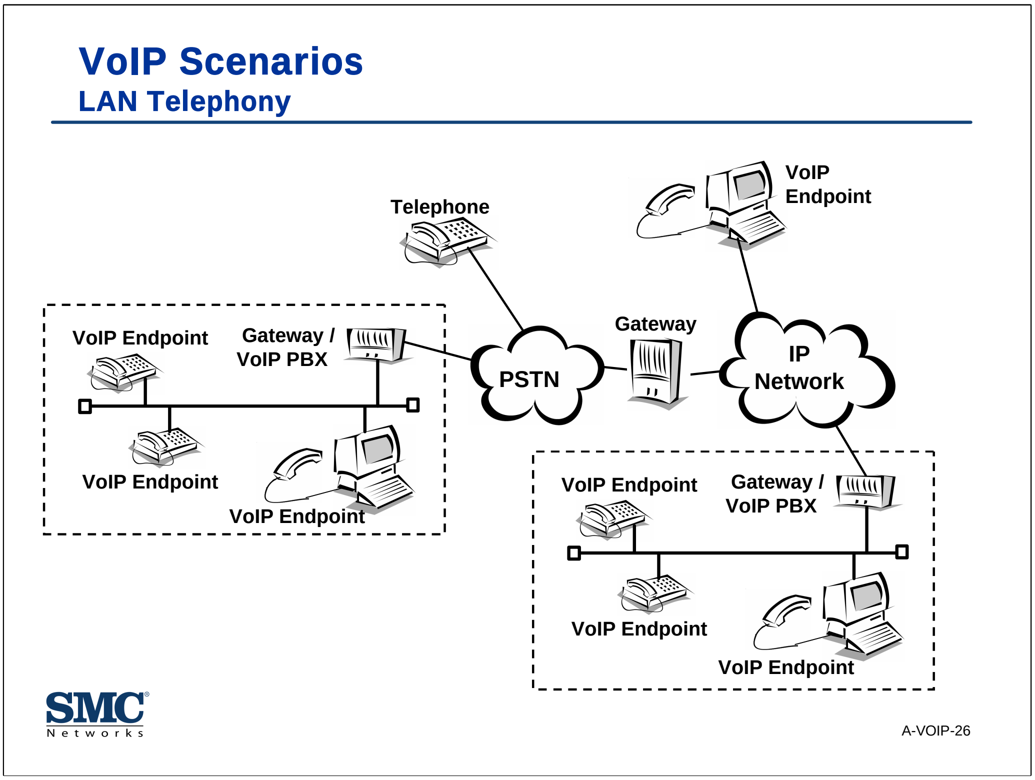
An **ITSP (Internet Telephony Service Provider)** interconnects the PSTN with the Internet by implementing interconnection gateways. Residential users as well as enterprise users can make very cheap (sometimes cost free) on-net calls within the provider IP network, or the Internet.

Most ITSPs offer telephony services that allow their users/subscribers to establish telephone communications also with subscribers that are connected to the PSTN (off-net calls) at comparatively low charges.

Most ITSPs therefore assign a normal telephone number to their customers. Every call that is made from the PSTN to that number will be routed by the interconnection gateway to the customer's IP phone.

The same technology approach is the basis for special long distance call service offerings where an IP network infrastructure is used to interconnect two PSTN users/subscribers. Extreme cost reduction for long distance services/calls are possible in these cases .



LAN Telephony refers to VoIP solutions that are designed to replace conventional PBX technology in both the small office or enterprize environment.

LAN Telephony is today a well accepted technology. This is mainly because of its easy implementation and management and the dramatic cost reduction by not having to implement an expensive completely separate communication for telephony solutions next to the data network.

But cost reduction is only one good reason for LAN Telephony. This technology allows much faster implementation of new and advanced communication applications (CTI, whiteboarding, etc.) and plays in many businesses an important role in increasing competitiveness and efficiency.

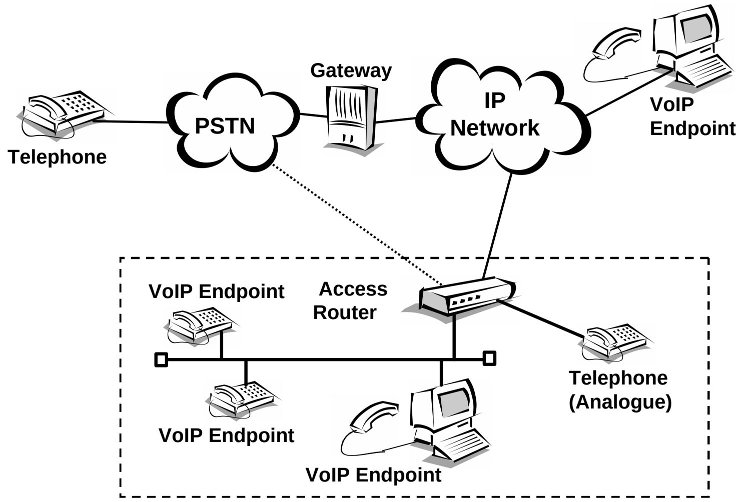
Many problems and difficulties that only have been partly solved in large IP WAN infrastructures and the Internet can be ignored in todays switched high bandwidth local area networks. There are hardly any constrains when it comes to delay, network error rate or bandwidth in a well designed switches Ethernet LAN.

The actual technical implementation will depend on the choosen VoIP technology. SIP and H.323 solutions for example differ in architecture and setup.

LAN Telephony solutions not necessarily have to be connected to the Internet or an IP network. In many cases the VoIP PBX systems are connected to the PSTN - exactly the same way as conventional PBX systems are. Often the VoIP PBX is connected to both, the PSTN and an IP network (Internet). In this cases the user is able to benefit from the low costs of Internet or IP Telephony solutions but still can leverage the specific benefits that PSTN technology offers (availability, special services, etc.).

VoIP Scenarios

VoIP IAD / Access Router (for SMEs)



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For SME customers and smaller offices the perfect VoIP solution consitst of an **IAD (internet access device)** or access router **with integrated VoIP gateway functionality**.

Good devices offer all the IP routing, NAT, security and management functions to provide Internet access for one or more computer systems in the local network. Additional QoS features and the necessary server and gateway functions allow the customer to quickly configure a complete VoIP (LAN Telephony, IP Telephony, Internet Telephony) solution. These VoIP IADs will support one or several of the VoIP standards (H.323, SIP, MGCP, etc).

Some of these devices will support an additional ISDN or analogue PSTN telephone connection. This allows the user to alternatively use this connection in case of a problem with the Internet connection or to use special voice services that are not available from within the VoIP network. The user can define in a dialing plan which numbers should be routed into either the PSTN or the VoIP (Internet, etc.) network

For users that wish to use existing analogue devices (telephones, DECT wireless phones, fax machines, etc) in their VoIP solution, some access devices have integrated VoIP adapters that provide one (or more) analogue ports.

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