IP Telephony Tutorial

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Introduction

IP Telephony is a vast new field that principally solves a simple need: Enabling use of LAN and Internet WAN infrastructure for telephony, replacing more or less the use of PSTN and local (and proprietary) media for that purpose. What will become apparent is that IP Telephony is not limited to just this. The technologies that IP telephony is based on can, beyond pure voice, achieve:

- Data and video conferencing
- Enhanced services enabled by the fact that it's all digital
    - Integration with Internet and intranet messaging services will become obvious
- Multi-line telephony via a single broadband link
- No cost for certain applications (e.g. peer-to-peer telephony/conferencing)

IP Telephony will evolve beyond devices and services we normally associate with telephony, like e.g. entry phones, integration in notebooks, PDA's, access/web pads, printers and copiers (e.g. for helpdesk access), dedicated audio and video surveillance devices, etc.

Terms

IP Telephony is known by many names, including Internet Telephony and Voice over IP (VoIP), etc. The term IP Telephony is used here because Internet Telephony indicates specific use of the Internet as transport media, and VoIP indicates any kind of voice over IP, not just telephony.

The Public Switched Telephony Network (PSTN) is the global telephony system we use every day.

ITU stands for International Telecommunication Union, the standards body for most things telephony and fax related.

The Internet Engineering Task Force (IETF) deals with Internet protocols, including also IP Telephony and IP Faxing.

Technology

The market is clearly heading towards standard technologies for achieving IP Telephony, with the goal to achieve complete interoperability. The first attempt in this direction was ITU's H.323 that stems from a set of specifications for voice (G.7xx), video (H.26x) and data (T.12x) conferencing over ISDN (H.320). H.323 is quite flexible but also considered very complex and has different options forcing implementations to implement them all to secure interoperability. The audio and video streams are transferred over IETF Real Time Protocols (RTP), that is designed to favor timeliness before integrity. In other words, it's considered worse that audio and video streams come out of sync than drop a few packets. Version 2 of H.323 adds better security, faster calling, more PBX-like services, etc. and H.323 is now developed even further.

An upcoming alternative is the Session Initiation/Description/Announcement Protocol (SIP/SDP/SAP) set of specifications, that is more optimized to the task of IP Telephony, has equal or better functionality than H.323, and also has less functionality redundancy. These protocols use the ITU audio and video codecs over RTP like H.323, hence providing compatibility in terms of media streaming, but not signalling.

Media Gateway Control Protocol (MGCP and Megaco) is a protocol optimized for inter-gateway communication. This means SxP and MGCP/Megaco complement each other. Still, some claim SxP will evolve into the only protocol set needed, which seems realistic at least for terminals. Some though claim MGCP will become the one-and-only winner. It's likely both, and H.323, will co-exist and even complement each other.

All mentioned protocols are in a specification flux, where H.323 is at version 2, but a version 3 is on the way, and initial versions of SxP and MGCP have been released, but new extended versions are continually developed.

Many believe SxP and MGCP will one day rule out H.323, yet today all products support H.323, but still few support SxP and MGCP. Information from Voice on the Net (VON) Fall 99 indicates that middle 2000 we should expect products to support also the newer protocols (from 3Com, Cisco and others).

The digital audio codecs transfer audio at anything from 5 kbps (G.723.1) to 64 kbps (G.711; also used for PSTN). Technically G.711 is very simple to implement and uses very little CPU power (enabling low-cost/power CPU's to be used for IP Telephony without DSP support), but also takes too much bandwidth for the existing Internet (even though it should work OK within a switched LAN), and dropped packets can not be re-constructed, causing annoying drop-outs. G.723.1 (5 and 6 kbps) and G.729 (8 kbps) are the most popular highly compressed codecs, that provide almost the same voice quality as G.711 but at a much lower data rate and with higher latency. For these codecs a DSP is needed even for a single-line device. It's likely G.729 will become the most used codec in the future, unless completely new and better ones are designed.

Note that neither of the protocols mentioned are limited to just TCP/IP, but in the scope of this report, and in real life, TCP/IP will be the choice.

In a way related to IP Telephony, IP Faxing is now standardized via IETF and ITU, where ITU has agreed to adopt the RFC's proposed by IETF, published as the T.37 (store and forward via SMTP and TIFF) and T.38 (real-time via T.30 (the protocol for faxing via the PSTN) and optionally H.323) ITU standards.

System components

H.323

An H.323 system consists of the following components.

Terminal or End Point

Digital voice (and video) terminals in the shape of PC's, telephones, picture phones and the like. Wireless access points, entry phones, etc. with integrated IP Telephony may also apply here.

Gateway

Converts between different protocols, typically H.323 to PSTN, H.320 (ISDN) or SxP, MGCP etc.

Gatekeeper

Provides call control for the terminals within the network. Translates between mnemonical terminal names to IP addresses, manages bandwidth, authenticates terminals, etc.

A gatekeeper is not required, but simplifies call control considerably.

Multipoint Control Unit (MCU) and Multipoint Processor (MP)

Handle setting up of conferences (audio, video and/or data) between multiple terminals/users. The MCU deals with the setup, the MP with the funneling of data streams.

SIP

User agent

The terminal device, as for H.323. Without a SIP server one can call directly between user agents, in a peer-to-peer fashion.

Proxy server

Acts as a "middle man" for the call signalling. When the call has been set up the media data is transferred peer-to-peer.

Redirect server

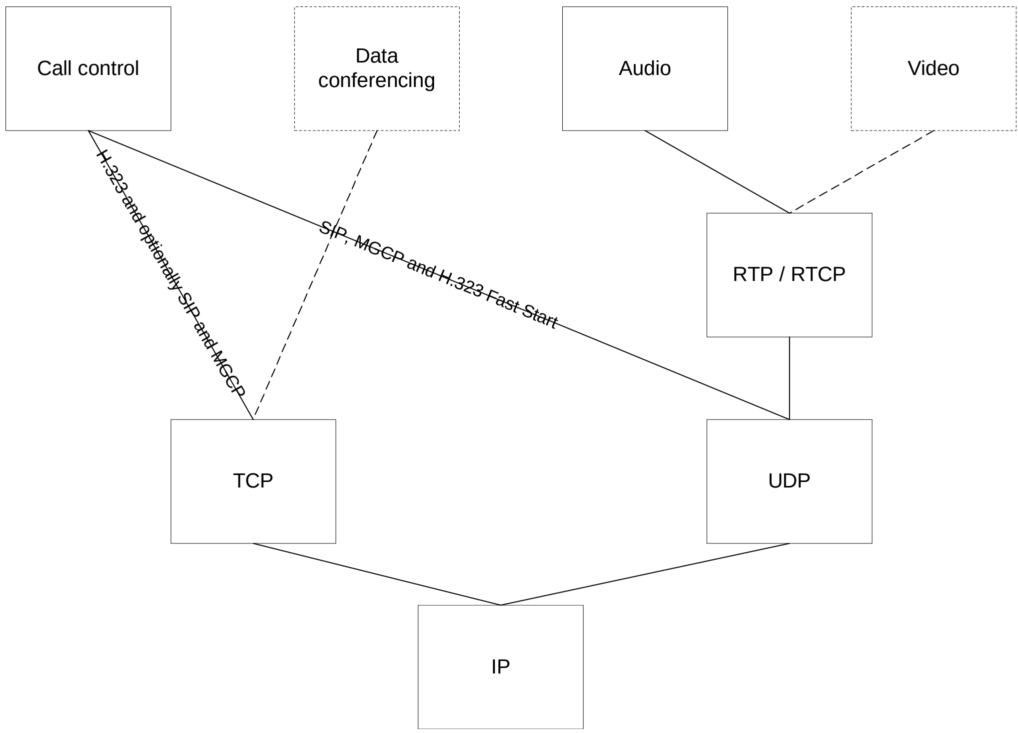
As the name indicates tells the user agent what other user agent(s) to call based on a given address.

MGCP

TBA

Protocol elements

What follows is a principal drawing indicating what protocol elements an IP Telephony terminal needs. Dashed line indicates optional elements.



Products

Don't see this list as more than a snapshot of what will surface, as the market and technologies are developing as I write.

IP telephone

Attaches directly to the LAN or WLAN infrastructure, and communicates with a local IP telephony gateway or an Internet service, for access to the PSTN. 8x8 and others have developed low cost chips that integrate a CPU and DSPs. Codec software is also included, altogether making development of an IP Telephony appliance a "snap".

Expect mobile phones and PDA's (and combinations) to long term integrate IP Telephony over Bluetooth and future wireless technologies (including GPRS and UMTS). Expect also dedicated IP phones for local use. Symbol Technologies is the first company with such devices (for IEEE 802.11). Ericsson (and other mobile phone manufacturers) are rumoured to already be developing such phones for Bluetooth and UMTS.

Audio/video surveillance

Existing digital surveillance cameras don't support IP Telephony protocols nor audio, but could very well do so for enabling synchronized audio and video, as well as bidirectional communication (entry phone being a special case; more below). As client software for that is free (at least for H.323, e.g. MS NetMeeting; expect also SIP clients to surface), there's no cost induced on the client side.

Wireless access point with IP Telephony

Probably mainly applies to Bluetooth, utilizing TCS for voice telephony. This device would enable mobile phones to be used as cordless phones. The IP Telephony functionality in an access point would serve as a gateway between Bluetooth voice and IP Telephony via LAN. Call control and voice communicated via the mobile phone would be funneled via the access point to a telephony gateway (in an office) or to an Internet Telephony service (in a home or possibly also a smaller office utilizing a Centrex service).

IP Telephony gateway for small and branch offices

The main need for such a device is believed to be branch offices that want to transparently integrate all offices into the same telephony infrastructure. Effectively the gateway would link together all local analogue phones, and ensure that the same switchboard and local numbers can be used both at the main office and in the branches. If all telephony is performed via IP phones (dedicated wired or wireless such, or the earlier mentioned access point with integrated IP Telephony) the gateway is though not needed at the branch.

Preferably a gatekeeper (in the case of H.323) is integrated with the gateway, so that a separate such is not needed.

IP Telephony gateway for homes

This solution is focusing on the existence of one or maybe two PSTN lines only, which limits the hardware requirements. This type of gateway would work well as a feature of a broadband gateway, as it wouldn't add that much cost to a such, and without broadband it's not of much use anyway. One would connect the single line PSTN chain to the PSTN connector on the gateway. After setting up the gateway to the appropriate Internet Telephony service calls to and from the home would go through the broadband network and the Internet service instead of via the PSTN connection.

Some (most?) cable modems are already equipped with this functionality.

Other voice appliances

Like IP phones would connect directly to the LAN infrastructure, but have more specific functions than IP phones. Applications may be entry phones (possibly with bidirectional or unidirectional video), audio points (possibly integrated in office devices like printers and copiers; would be very cheap) for helpdesk access, network speakers (for "Muzac"), etc.

IP Telephony will also be integrated as software in user terminals, like notebooks, PDA's, access pads, etc. Currently such exist for H.323 (e.g. NetMeeting) but comparably few for SIP. Microsoft hasn't committed to bring out a SIP client for Windows yet.

IP Faxing

There exists a few Internet faxes and network scanners on the market, yet they are way too expensive to reach a mainstream market. It is though believed the savings are so great (no-cost transmission of paper-based information, more reliable than fax, better image quality than fax (including color), etc.) that there is a yet unexploited market.

Most such devices use standard e-mail and send the scanned images as attachments, but the previously mentioned IP Faxing protocols also apply, simplifying interoperability.

Most high capacity IP Telephony gateways support fax machines and conversion to T.38.

IP Telephony pros

Lower costs and less bandwidth use are often mentioned as the key reasons to adopt IP Telephony. I can agree that is the case today, but as Local Exchange Carriers (LEC's) can lower prices on existing telephony services, and data is anyway already taking the most bandwidth of the WAN "pipes", the long term advantage of IP Telephony must be relating to the data/voice integration. Hence, potentially better and new services can be provided without any hassle. We are moving from a voice world to a data world in any case, and future generations will expect voice services to be fully integrated with information ditto, and be completely free.

Lower cost

It's short term certainly still valid as a key argument from an operator point of view as Competitive LEC's (CLECs) can easily compete with established LECs by sinking much less costs in establishing the infrastructure needed to set up an exchange service. GlocalNet is an example, that in part uses IP Telephony between POP's. Of course you can already communicate with a peer for no cost at all between two PCs, but this will probably continue to be a very small application of IP Telephony. For new offices, not the least branch offices, IP Telephony can provide considerable cost savings, as the company needs only to install LAN's, and with the mentioned Bluetooth Access Point one can use a mobile phone as the only phone both externally and internally. There's no need for a PBX at the branch office if the company runs IP Telephony to the main office and integrates with the PBX there. This also results in location transparency to external callers.

Less bandwidth need

The reason why this is so often mentioned is that the operators pay for WAN bandwidth, and hence find themselves able to squeeze more calls into a certain "pipe width" both due to better compression and the packet-based transfer (no circuit switching that locks up bandwidth). In a LAN and increasingly also on a WAN this is less of a concern as anyway data transfer is passing voice transfer as the main bandwidth user. A few years from now voice will be a very small part of the data flow. This is though a very good reason for using IP Telephony in the first place, as it can easily ride on the same pipes as the data.

Use of cheaper infrastructure components

This is a very valid point, as it's very expensive to grow and maintain the PSTN. IP Telephony uses standard networking infrastructure components, both in the local situation as well as in the "PSTN replacement" situation.

Enhanced features

Already established on the Internet, and increasingly also as software and hardware solutions for organizations, are CTI and IP Telephony services that in part also integrate with Unified Messaging services. Hence, beyond normal voice services like re-direction, call waiting, voice mail etc. one could also get voice-mail via e-mail, faxing via the Internet (including conversion of incoming faxes to e-mails), e-mail notifications via voice calls etc. One thing is certain: "We've seen nothing yet". As

PCs and workstations (due to sheer performance, but also through specialized hardware for voice coding etc.) can be used as IP Telephony gateways and service platforms, suddenly developing telephony services becomes mainly a software job on platforms the bulk of software developers already know. A feature many hope will be key for increasing customer service is web calling. This means you can establish a “hot line” directly from a company’s web page (or within a company from e.g. the help desk’s web page), without cost for neither the customer or the called company, and without having to know who or how to call. A call center (possibly outsourced) then takes care of the call. Due to Internet performance limitations this is yet not working well, but expect e-commerce sites to use this extensively in the future.

A seemingly very important step was taken by Microsoft when they integrated IP Telephony (for the time being only H.323) into TAPI 3, being part of Windows 2000 and likely also Millennium (the follow-up to Windows 98, optimized for home applications), but so far Microsoft hasn’t indicated any plans for SIP and MGCP.

Higher quality

This is just a potential advantage. Right now the situation is actually the reverse. For a few reasons why, see the “cons” section. In the long term it should be possible to get close to CD quality audio. Of course this is not that very significant for voice alone, as the current PSTN quality (which is actually already digital, using the G.711 codec) is quite sufficient for just that, but when considering possible future radios and music via IP Telephony this comes into play.

IP Telephony cons

The world of IP Telephony is not just a rosy one. Here are a few issues, in cases quite severe, that might later and in cases altogether disqualify IP Telephony for certain applications. It must also be remembered that both the market and the technologies are yet immature, which though indicates there is room still to make a mark in the marketplace.

Bandwidth and latency

That TCP/IP is the network protocol used the most today doesn’t mean that it’s good at everything, especially not bandwidth and latency sensitive applications. Also, even though one could add features to TCP/IP that would secure that, doesn’t mean the Internet would in practice be able to provide it. Within a LAN this is less of a problem (where the bandwidth and latency is through “brute force” much better than on the Internet). At least for voice the Internet will be up and running for the task in a few years time, not the least because of Cisco (that provides most of the Internet infrastructure) pushing the envelope.

Interoperability

Even though H.323 and upcoming SxP and MGCP are standards doesn’t mean interoperability is simple:

- Specifications are interpreted differently. Especially H.323 allows different interpretations and subsets, yet there’s no subset profiles defined. The latter is currently discussed in the industry to secure compatibility and interoperability.
- SxP and MGCP are very much moving targets, yet it's believed specifications will be stable and functionality-rich around middle 2000.
- As the specifications are yet so limited (compared to actual needs), each manufacturer adds proprietary features, that are 100% certain to not work with any other manufacturer’s products.

PBX capabilities

What you can do with contemporary analogue PBX's is far superior compared to what you can currently do with any manufacturer’s IP Telephony system. Clearly there’s a trend towards providing so called iPBX's, but they have at least a year to mature before companies sensibly would use them extensively instead of PBX's.

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  - Media Gateway Control (megaco)
  - Multiparty Multimedia Session Control (mmusic)
  - Internet Fax (fax)
- Internal resources:
  - IP Telephony
  - Multimedia Conferencing
  - Fax client