Overview

What is Internet Telephony?

Definition: Internet telephony (IPT) is transport of telephone calls over the Internet, no matter whether traditional telephony devices, multimedia PCs or dedicated terminals take part in the calls and no matter whether the calls are entirely or only partially transmitted over the Internet.

What is Internet Telephony Good For?

The most significant **benefit** of IPT and driver of its evolution is money-saving and easy implementation of innovative services:

- In the future, Internet Telephony Service Providers (ITSP) may use a single infrastructure for providing both, Internet access and Internet telephony. Only data-oriented switches could be deployed for switching data as well as packetized voice. Multiplexing data and voice could also result in better bandwidth utilization than in today's over-engineered voice-or-nothing links. Not only the providers, but also their clients will profit of lower costs eventually.
- Now, customers may take advantage of flat Internet rating vs. hierarchical PSTN rating and save money while letting their long-distance calls be routed over Internet. This is especially true in Europe, where the prices of long-distance calls are still higher than in US. But: according to some estimations, the prices of the traditional and the Internet telephony will equalize together with the convergence of quality of services provided by them.
- The IPT users may also profit of its software-oriented nature: software solutions may be easily extended and integrated with other services and applications, e.g. whiteboarding, electronic calendar, or WWW. Deployment of new IP telephony services requires significantly lower investment in terms of time and money than in the traditional PSTN environment.

But: The wide business deployment is still hindered by lower quality of voice over IP, particularly by higher delay and jitter. Also many technical aspects of accounting, billing, charging , roaming etc. remain open yet.

Internet Telephony Scenarios

The IPT usage scenarios are commonly **classified** by the type of devices terminating an Internet call. Because there may be either a PSTN device or a data-oriented terminal on each side of a call, there are 4 generic classes. Note, that although "PC" is a well established term, any device capable of transmitting voice over data network may apply in this context. See for example the dedicated device Aplio/phone.

Caller’s Terminal	Callee’s Terminal	Notes	Costs Paid By Caller
PC	PC	This class is attractive especially for private users who already have an Internet access and an audio-capable PC. Necessary software is available for free . This pure-IP scenario is likely to take advantage of integration with other Internet services, such as WWW, instant messaging, E-mail, etc.	○ Costs of ownership and maintenance of the hardware (PC with modem and sound or a dedicated device) and software (IPT software is often provided for free). ○ Costs of Internet access (incl. the local call).
PC	telephone (POTS/ISDN/GSM...)	This is an extension of the previous class in that the PC-callers may reach also the PSTN callees. A gateway converting the Internet call into a PSTN call has to be used and located as near to the callee as possible to minimize the price for the gateway-to-callee connection. This scenario is commercially provided by gateway operators like AccessPower, Delta Three.	○ Costs of ownership and maintenance of the hardware (PC with modem/dedicated device) and software (IPT software is often provided for free). ○ Costs of Internet access (incl. the local call). ○ Costs charged by the gateway operator. (~ 5-12 cents per minute to the U.S. in August 98) The costs charged by the operator are determined mainly by the costs of the call placed from the gateway to the callee.

telephone (POTS/ISDN/GSM...)	telephone (POTS/ISDN/GSM...)	This class is attractive for those who want to save on long-distance call and do not have/want to use a PC. For example, mobile phone users certainly prefer to carry only the mobile phone without any additional boxes. The call has to pass two gateways: GSTN-to-Internet and Internet-to-GSTN. This solution has been comercialy provided by gateway operators like AccessPower, DeltaThree or Paegas CZ.	○ Costs charged by both gateway operators.(~ 7-17 cents per minute to the U.S. in August 98) The costs charged by the destination gateway are determined mainly by the costs of the call placed from the gateway to the callee. ○ Local Call Costs
telephone (POTS/ISDN/GSM...)	PC	This class is useful for those who want to reach Internet users with an ordinary telephone. The scenario is investigated by Tiphon. Telenor provides this service commercially in Norway under the name "Interfon".	○ Costs charged by a gateway operator. ○ Local Call Costs

Architecture

Architecture: the Internet telephony systems are composed of these elements:

- end devices; these may be either traditional telephones (analog/GSM/ISDN/...), audio-equipped personal computers, or single use appliances
- gateways; if a traditional telephone is used at either calling side the call (i.e. its transmission format, signaling procedures, audio codecs) has to be translated to/from the format for transport over Internet; this is the task of the gateways
- gatekeepers/proxies; the gatekeepers/proxies provide centralized call management functions; they may provide call admission control, bandwidth management, address translation, authentication, user location, etc.
- multipoint conference units; these manage multiparty conferences

The components may be implemented as hardware or software and may be integrated into single units optionally.

They communicate with each other over signaling and voice-transporting **protocols**. To ensure interoperability between products of different vendors, standardization bodies have elaborated standards for both classes of protocols. See the section "Players and ..." for more details.

Future

Making predictions is difficult and it belongs to the competence area of oracles, magicians and marketing managers. But let us at least summarize some important factors.

The law of supply and demand works also in the Internet telephony. An article has been published by Communications Industry Researchers, which claimed the prices of the traditional and the Internet telephony will equalize as soon as the quality of the both standards will do so. We believe, that the most significant obstacles in reaching the equilibrium are the still unsatisfactory **voice quality** and the lack of **means of commercial deployments**. Both of them are under investigation. The voice quality will increase with special QoS means and generic increasing bandwidth. The means of commercial deployments are being designed by both, commercial and academic world. For example, the gateway discovery architecture which enables open market of gateway operators is being proposed by IETF.

IPT may also become a subject to **government regulations**. Such efforts are very welcome to traditional telcos - a good example is the action brought by Czech Telecom against Paegas' "Internet call". According to Bruce Jacobs, some governments intend to regulate even the PC-based telephony (India, Pakistan), other have indicated they will treat IPT as simple resale (Canada) and others have recognized that action is premature (see the decision by EU and FCC). Look at the VON Coalition's pages for additional information on the regulations.

Another legal issue is **wiretapping**. A pretty contraversial discussion about the justification and standardization of wiretapping took place on the Raven mailing list of IETF. Eventually, IAB and IESG issued a RFC 2804 which justifies why IETF does not include such a functionality in its standards-track.

Players and Standards (Who Is Who)

Standardization Bodies

Standardization Body	Note
	The Internet Engineering Task Force (IETF) is a large open international community of network designers, operators, vendors, and researchers concerned with the evolution of the Internet architecture and the smooth operation of the Internet. See also the list of IETF's efforts and documents related to Internet telephony.
	The International Telecommunication Union (ITU) is the leading publisher of telecommunication technology, regulatory and standards information. Its published H.323 standard for multimedia terminals in networks with non-guaranteed QoS is gaining increasing popularity in the world of IPT vendors. The H.323 is an umbrella for many other standards covering signaling, real-time voice transports, codecs, etc. The standards are available at PictureTel's site.

	Standards organizations and other related bodies have agreed to co-operate for the production of a complete set of globally applicable Technical Specifications for a 3rd Generation Mobile System based on the evolved GSM core networks and the radio access technologies supported by 3GPP partners (i.e., UTRA both FDD and TDD modes). The Project is entitled the Third Generation Partnership Project and may be known by the acronym 3GPP. (Note that 3gpp anticipates use of SIP as telephony signaling protocol in all-IP networks.)
	The project "Telecommunications and Internet Protocol Harmonization over Networks" (Tiphon) has been established by the European Telecommunications Standards Institute (ETSI, well known for its GSM standards) to ensure that users connected to IP based networks can communicate with users in Switched Circuit Networks (such as PSTN, ISDN, GSM, SS7). Tiphon documents are located here. Tiphon has also launched an initiative TIPIA (Tiphon IP-Telephony Implementation Association) to support interoperability of Tiphon-compliant products.

Consortia, Coalitions, Associations, etc.

	SIP Forum is a non profit association whose mission is to promote awareness and provide information about the benefits and capabilities that are enabled by SIP.

	The mission of the International Multimedia Teleconferencing Consortium, Inc. (IMTC) is to bring together all organizations involved in the development of interactive, multimedia teleconferencing products and services to help create and promote the adoption of industry-wide interoperability standards. The IMTC is currently focused on multimedia teleconferencing standards adopted by the ITU and interoperability of products claimed to be ITU-conform. IMTC formed a Conferencing over IP (CoIP) Activity Group as a part of the IMTC Contributions. IMTC contributions are available here. The formet iNow has merged with IMTC. iNow! is a multi-vendor initiative established to quickly provide interoperability among IP telephony platforms. The iNOW! Standards-Based IP Telephony Interoperability Profile provides equipment vendors with the blueprint for chieving real world, revenue-generating gateway to gateway and gatekeeper to gatekeeper interoperability.
	The MIT Internet Telephony Consortiumconsists of member firmsand selected academics who collaborate on research into technical, economic, strategic and policy issues that arise from the convergence of telecommunications and the Internet.
	The VON Coalition's mission is twofold: actively advocate the viewpoint that the IP Telephony industry should remain as free of governmental regulations as possible, and to educate consumers and the media on Internet communications technologies. (See also the list of members.)
	The Telecommunications Industry Association represents the telecommunications industry. It is also focusing on stadardization of IP phones and their particular features (TR-41.3.4. spec).

	CableLabs is a membership organization consisting of cable television system operators serving cable subscribers in the North and South America. Its mission is to plan and to fund research and development projects; to transfer relevant technologies to member companies and industry suppliers; and to serve as a clearinghouse in providing technological information to its members. It has established a project PacketCable aimed at identifying, qualifying, and supporting Internet-based voice and video products over cable systems. Master-slave approach to iptel signaling is favoured.
	The Softswitch Consortium is the international organization for global cooperation and coordination of internetworking technologies in the field of internet-based real-time interactive communications and related applications. The purpose of the Consortium is to support rapid advancement of application development for the evolving Internet protocol networks which support both voice and multimedia communications. Internet protocol networks are built on distributed call control servers generally called "call agents," "media gateway controllers," "softswitches," and "media gateways". The Consortium promotes worldwide compatibility and interoperability; identifying, selecting, augmenting as appropriate, the development and distribution of standard interfaces for "call agents," media gateways, and applications.

The Parlay Group	The Parlay Group's objective is to promote industry acceptance of the Parlay API, a specification designed to enable carriers and independent software vendors to write applications to provide services across wireless, Internet, and wireline networks. Faster time-to-market and less complex development cycles are some of the key benefits of the Parlay API. Founded in 1998, the Parlay Group focused initial development of its API on functions such as call control, messaging, and security. The current Parlay Specification paves the way forward in developing usable, real-world product implementations of the API. The current members of the Parlay Group are AT&T, BT, Cisco Systems, Ericsson, IBM, Lucent Technologies, Microsoft, Siemens AG and Ulticom. The specification has been published at http://www.parlay.org .
The JAIN Initiative	Organized by Sun in 1998, the JAIN initiative addresses the needs of next-generation telecom networks by developing a set of industry-defined APIs for Integrated Networks. Network services today are typically built using proprietary interfaces that inhibit the marketplace for new services. Members of the JAIN community have joined forces to define open APIs based on Sun's Java platform, thus allowing service providers to rapidly create and deploy new flexible, revenue-generating services. Information about the JAIN program can be found at http://java.sun.com/products/jain/ .

Other related organizations involved in standardization/promotion of Internet telephony are SIP Forum, TTT, VoiceXML Forum.

Companies

Today almost all major companies providing Internet services and products are getting involved in the emerging Internet telephony market (to name at least some of them: 3Com, Ascend, Cisco, Clarent, Ericsson, Hitachi, Intel, Lucent, Microsoft, Motorola, Netscape, RADVision, Siemens, VocalTec and many more). Many of them have declared strategic alliances (e.g. Cisco with Hitachi, Gric, HP, OzEmail; Ascend with Mind CTI; NetSpeak with Motorola). See the list of products and Pulver.com's list of Internet telephony providers for more references.

Others

Pulver.com, a leading Internet telephony consulting firm, collects all information on Internet telephony on its web-site, i.a. a list of IPT Telcos, recent publications, gateway providers and much more. Pulver.com also organizes the "Voice over Net" conference and administers a "VoN" website.

PictureTel Corporation administers a website about videoconferencing and telecommunications standards. The site is intended to provide the standards community with a single point of access to the many industry activities associated with the development of videoconferencing standards.

Selected IETF Internet Drafts and RFCs Relevant to the Internet Telephony

Important note

These I-Ds/RFCs are mirrored from www.normos.org. Check their website for on-line database of IETF (and many other) standards. Fast full-text and database search is available.

Notification Service

You may want to subscribe to our notification service . Then you would receive an email notification whenever new related document appears or an existing one is updated.

Jump to:

- Accounting
- All Internet-Drafts with '-sip' in filename
- Call Services
- Call Signaling
- Configuration and Management
- Emergency Services
- Firewall Traversal
- Gateway Control
- Interworking with PSTN and H.323
- Geographical Services
- Media Transport
- Miscellaneous
- Mobility and Server Location
- Numbering and Call Routing
- Presence, Instant Messaging
- QoS Support
- Security

Selected ITU-T Standards Relevant to the Internet Telephony

Important note: these standards are mirrored from PictureTel's standard website. Paul Jones maintains a webpage about current state of H.323 documents.

Standard	Abstract
H.323 V2	H.323 is an umbrella recommendation that sets standards for multimedia communications over Local Area Networks (LANs) that do not provide a guaranteed Quality of Service (QoS). H.323 is part of a larger series of communications standards that enable videoconferencing across a range of networks. Known as H.32X, this series includes H.320 and H.324, which address ISDN and PSTN communications, respectively.
H.225	H.225 specifies call signaling (Q.931 subset), RAS, multimedia transport (RTP/RTCP).
H.450	H.450 specifies supplementary services. .0 is a framework description, the following recommendations specify individual services: Transfer (.2), Diversion (.3), Hold (.4), Park & Pickup (.5), Call Waiting (.6), Message Waiting Indication (.7), Name Identification(.8), Call Completion on Busy (.9).
H.235	Security, encryption, authentication, etc. There is only a placeholder for non-repudiation.
H.245	Multimedia signaling.
H.GCP	Proposed Recommendation Gateway Control Protocol.
H.323 Annex E (R4.1)	Call Signaling over UDP
H.323 Annex F	Single Use Audio Device (SUD) -- this document defines SUDs that operate using a well-defined subset oh H.323 protocols with a restricted functionality range.
H.225 Annex G	Interzone communication. Extended to include not only address resolution but also pricing information exchange, access authorization, and usage reporting.

1998-2001, maintained by Jiri Kuthan.

Issues in the Internet Telephony

This webpage presents information on current technical issues of the Internet telephony. Because the issues are still being discussed, mutually different points of view may and actually do appear in these webpages. Potential authors are encouraged to submit their contributions (see the notes to authors).

Signaling: H.323 vs. SIP

The IETF and ITU-T have designed different signaling protocols lacking interoperability. ITU-T reused and extended its signaling norms while IETF proposed a simple HTTP-like protocol fitting in the traditional Internet protocol family.

- SIP vs. H.323 Telephony
  - What is wrong with the previous article (SIP vs. H323)

Call Services

The traditional telecommunication operators have been providing additional call services on top of their infrastructures. This has been accomplished with the Intelligent Networks architecture. Mechanisms for creation of call services are being developed also for the Internet telephony. There are two different approaches which differ in where the service intelligence is located. The network centric approach pushes the intelligence in the operator-maintained units (like Call Agents in the Intelligent Networks) whereas the end-system-centric approach tends to locate considerable intelligence within the end systems.

Relevant work and papers:

- J. Lennox, H. Schulzrinne, T. Porta: Implementing Intelligent Network Services with the SIP
- J. Rosenberg: Internet Multimedia Conferencing: What now?
- Internet Drafts on Call Processing
- ITU-T: H.450.x standards specify supplementary services like transfer (.2), Diversion (.3), Hold (.4), Park & Pickup (.5), Call Waiting (.6), Message Waiting Indication (.7)
- see also GCP for more information on the network-centric approach
- TIA is also concerned with standardization of IP phone features

GCP

Internet telephony gateways consist of two functional parts - a dumb *media gateway* which converts audio data and an intelligent *media gateway controller* which communicates with the rest of the world over signaling protocols and controls 1-N media gateways over a *gateway control protocol (GCP)*. Standards are being developed by industry, IETF and ITU-T. IETF's Megaco has merged with ITU-T's H.248, a de facto standard MGCP co-exists.

Relevant work, papers and postings:

- ITU-T:H.248 (formerly H.GCP)
- IETF: Internet Drafts/RFCs on Gateway Control Protocols
- ITU-T: Rex Coldren of Lucent: Gateway Decomposition Control Models (point2point, MDCP, MGCP)
- Nancy Greene: Comparing versions of MGCP
- postings by Nancy Green, Francoies Menard and Bob Bell: : explanations of relations of the GPC-like protocols

Some people also defend the idea of the gateways controllers controlling the IP telephones. This approach is backed especially by the cable industry (see PacketCable, CableLabs). Needless to say, using master-slave protocol for call signaling instead of peer-to-peer protocols like SIP undermines the distributed concept of the Internet architecture. (more detailed arguments attached)

Interdomain communication

Interdomain operation raises additional issues such as call routing, charging & settlement, etc.

- ITU-T: H.225 - Annex G. H.323 was engineered for operation in local networks originally. This annex fixes this limitation and addresses issues like address resolution and communication between administrative domains.
- IETF: Internet Drafts and RFCs on Numbering and Call Routing and interdomain AAA

Mobility

Significant effort has been put into introducing the mobility to the Internet telephony. Several mobility approaches exist:

- Transparent usage of the existing wireless networks.
- Transparent usage of mobile IP.
- Explicit application-level support for the mobile Internet telephony.
- Combination of any of the aforementioned technologies

For more references see:

- IETF: Seamless MobilityWG
- ETSI/Tiphon: WG 07, Wireless and Mobility Aspects
- E. Wedlund, H. Schulzrinne: Mobility Support Using SIP
- ITU-T: Enhancements to ITU-T Recommendation H.323 to support User and Service Mobility
- Internet Drafts on SIP Mobility
- H. Schulzrinne: SIP for Mobile Applications
- Siemens's Vision Statement: IP Cellular Phones
- 3G.PP is developing all-IP SIP-based solutions for UMTS. A libary of 3gpp specifications is available.
- Adam roach reports on the current 3GPP status wrt SIP
- SIP & WAP

Multimedia: Quality of Service (QoS)

The Internet telephony still lacks the bussiness audio quality known from the PSTN world. Packet delay, loss and jitter are the main negative factors. Typically, router congestion is the main evil causing these factors to grow. Currently, several approaches exist to improve the audio quality of voice transported over the Internet.

- Resource Reservation - see the homepages of RSVP, RSVP IETF WG and IntServ IETF WG for more details on this stateful reservation approach.
- Differentiated Services - as opposed to the previous alternative the audio packets do not travel along a reserved path but get prefered treatment if tagged as real-time data. This approach is stateless. See the homepage of the DiffServ IETF WG for more details.
- Forward Error Correction - this class of algorithms reduces the impact of data loss by sending redundant data along with the audio data. The redundant data helps to reconstruct lost data. See SPB-FEC for an excellent example.
- Loss Concealment - this class of algorithms tries to reduce the impact of data loss by replacing the lost audio with an approximation. See an excellent example: APC.

You may want to check related documents:

- Internet Drafts related to Internet telephony and QoS.
- Schulzrinne et al.: Interaction of Call Setup and Resource Reservation Protocol in Internet Telephony
- Web page of the Tiphon working group focusing on QoS.

Interaction with Firewalls

The Internet telephony applications require firewalls to pass their RTP streams. The problem is the UDP port numbers of these streams are not fixed. Instead, they are negotiated during the session setup. If a firewall applies a "default-deny-explicit-allow" packet filtering policy, an additional mechanism for opening dynamic pinholes must exist. Typically, an entity that understands SIP/SDP or H.323 must distill port numbers during the session setup and use a control protocol to open the pinholes in firewalls. Alternatively, a kind of packet authorization might also help.

- Related Internet Drafts
- Firewall Control
- An article by Cisco explains why interaction of firewalls with H.323 is tricky.
- A similar article is also available from Intel.
- Microsoft suggests to pass connections on all dynamically assigned ports in its configuration guide for Netmeeting. Far away from the ’default-deny’ policy.
- Embedded Linux ALGs are available for both H.323 and SIP.

All of these references focus on H.323. Although SIP faces the same problem, a consensus seems to exist that the same task is easier with SIP.

Notes to Authors

- Only accepted submissions are presented on this site.
- Compact papers introducing topical issues in a self-explanatory way are encouraged.
- HTML format with links to other relevant sites is strongly preferred.
- Send your submissions to the administrator of this webpage.

H.323 vs. SIP Telephony

The IETF standards are interoperable with the ITU-T standards on the voice transport level because ITU-T incorporated IETF's RTP protocol in its H.323 umbrella standard. However, different signaling protocols are proposed by both institutions: ITU-T uses the H.323 standard ("Visual Telephone Systems and Equipment for Local Areas Networks which Provide a Non-guaranteed Quality of Service") whereas IETF pushes the SIP signaling. Currently, there are many contraversial discussions and predictions on which approach will gain greater popularity.

ITU-T backers claim the H.323 to have gained greater support from multiple vendors (including Microsoft and its NetMeeting). This is aparently result of early publication of the standard. The first version of SIP appeared later. However, fast work does not necessarily have to be the best one. Many SIP backers doubt that H.323 has addressed all challenging issues sufficiently. Number of H.323 versions (as for January 2001, already the fourth version has been approved) seems to confirm such concerns.

On the other hand, SIP designers have kept the following crucial aspects in their minds from the early beginning: Internet-wide issues, integration with Internet services, extensibility, modularity and simplicity. In the meanwhile, SIP products are developed by all industry leaders: Cisco, 3com, Ericsson, Nokia, Nortel just to name a couple of names. See product resources on SIP website and pulver.com. SIP has been adopted by the next generation cell phone industry for its 3gpp standars.

In the following paragraphs we examine both, ITU and IETF, approaches from the technical point of view. The discussion is an excerpt from comparison made by the SIP authors Henning Schulzrinne's and Jonathan Rosenberg's . The original documents are located here (regularly updated document) and here(outdated).) Yet another comparison was written by I. Dalgic and H. Fang. Yet another comparison was written within 3gpp -- it justified use of SIP in 3g mobile networks: s2-000505.

A relevant article "SIP Rules! appeared in May/2000 issue of Computer Telephony Magazine. An article on deploying SIP was issued in CWI World News.

	SIP	**H.323**
set of supported service	roughly the same	
media transport	equivalent (RTP, identical codecs)	
Call set-up delay	1.5 RTT	6-7 RTT (set-up delay may also increase significantly in a lossy network due to a TCP property; see a time-line for more details of H.323 V1) Note: this has been improved with H.323V2 which allows for transportation of the H.245 messages over the signaling H.225 channel.

Complexity	adequate: HTTP-like protocol	high: ASN, use of several different protocols (H.450, H.225.0, H.245)
Extensibility	the protocol is open to new protocol features	ASN.1 vendor specific ’nonstandardParam’ at predefined positions only; lack of negotiation of the extended capabilities
Codec support	any IANA registered codecs	ITU registered codecs (currently, i.e. ITU developed codecs)
Third-party call control (3PCC allows for additional services as blind transfer, operator assisted transfer, three-party calling, forwarding variations, etc.)	yes	none
Architecture	modular: SIP encompasses basic call signaling, user location and registration; other functions (QoS, directory accesses, service discovery, session content description) reside in separate orthogonal protocols	monolithic: The mix of services provided by the H.323 components encompass capability exchange, conference control, maintenance operations, basic signaling, QoS, registration, and service discovery.
Server stale-ful/less	stateless	stateful (servers are supposed to keep call state for the entire duration of a call; they also have to keep the TCP states) -> lower reliability and scalability
Conference control	distributed multicasting support	centralized (MC may become a bottleneck for larger conferences and additional features as MC cascading have to be employed); unicast signaling only -> lower reliability and scalability, additional complexity of special handling of large scale conferences
Loop detection	yes	none -> a redirection may cause infinite request forwarding
Firewall support	accomplished by SIP Proxy	complicated by its complexity, usage of dynamic ports and multiple UDP streams (see articles by Intel and Cisco)

Multicast capable signaling	yes -> this simplifies user location, group invitations, call center applications; the bandwidth is spared	no
Addressing	any URL including E-mail address, H.323, http, ...	host (without username!), gatekeeper-resolved alias (arbitrary case-sensitive string, e.g. E-mail address), E.164 telephone numbers
Transport protocol	any, allowing for connectionless protocols (UDP) which result in lower call-setup time	reliable protocol required
Web-integration	○ integration with other Internet services (e.g. a caller may send an E-mail to an unreachable callee) ○ click-to-dial feature	?
Inter-domain user location	by existing Internet services (DNS, LDAP, ...)	weak

Conclusion: The primary reason of existence of two non-interoperable signaling protocols is the both, the telecommunication and the Internet world, wanted to have protocols meeting their traditions. ITU wanted to have a sophisticated norm utilizing their other sophisticated norms, whereas IETF defined a protocol well fitting its puzzle of simple and powerful tools (see, what Henry Sinnreich of MCI says). The Internet telephony is located on the border of the both worlds and it is difficult to predict which approach will gain the most popularity eventually. However, if the technical aspects discussed in this section and introduction of novel integrated service will have the last word SIP’s chances are high.

Network Bibliography Search Results: Internet telephony

C. Agapi, C. Chiu, T. Chong, H. Phillips and B. Willingham, "**Internet Telephony** Gateway Location Service Protocol," Internet Engineering Task Force, {Internet Draft}, Nov. 1998.

> **Abstract:** The development of IP to PSTN gateways has made it possible for users of the two disparate networks to place 'telephone' calls between the two networks with little effort. Users placing calls from an IP network to the PSTN network must choose a gateway to act as a bridge between the two networks. Selection of a gateway can be based on any number of criteria, such as price, codecs, version, billing, etc. In this draft we propose a gateway location service for this purpose. The gateway location service protocol is an instantiation of the Service Location Protocol that has been modified to run in a wide area network across many administrative domains.

J. Dobrowolski, W. Montgomery, K. Vemuri, J. Voelker and A. and Brusilovsky, "IN Technology for **Internet Telephony** Enhancements," Internet Engineering Task Force, {Internet Draft}, Jun. 1999.

> **Abstract:** The purpose of this Internet Draft is to start discussion on the issues of making available existing Intelligent Network (IN) capabilities to the Voice over IP (VOIP) Internet application and other Internet applications. In addition to benefitting from accessing existing IN services, interworking with IN will expedite development of new Internet applications.

V. Gurbani and V. Rastogi, "Accessing IN services from SIP networks," Internet Engineering Task Force, {Internet Draft}, Feb. 2001.

> **Abstract:** In **Internet telephony**, the call control functions of a traditional circuit switch are replaced by a IP-based call controller that must provide features normally provided by the traditional switch, including operating as a SSP for IN features. A traditional switch is armed with an IN call model that provides it a means to reach out and make service decisions based on intelligence stored elsewhere. Internet call controllers, by contrast, do not have an IN call model. Furthermore, since there are many Internet call models with varying number of states than the IN call model, there has to be a mapping from the IN call model states to the equivalent states of the Internet call model if existing services are to be accessed transparently. To leverage the existing IN services from the Internet domain, this draft proposes a mapping from the states of the IN call model to the states of SIP, an Internet call signaling protocol.

J. Rosenberg and H. Schulzrinne, "An RTP Payload Format for User Multiplexing," Internet Engineering Task Force, {Internet Draft}, May 1998.

> **Abstract:** This memo describes an RTP payload format for multiplexing data from multiple users into a single RTP packet. Such multiplexing is especially useful for transporting voice data between **Internet telephony** gateways. It causes significant reductions in header overheads and improves scalability.

J. Lennox and H. Schulzrinne, "CPL: A Language for User Control of **Internet Telephony** Services," Internet Engineering Task Force, {Internet Draft}, Jul. 2000.

Abstract: The Call Processing Language (CPL) is a language that can be used to describe and control **Internet telephony** services. It is designed to be implementable on either network servers or user agent servers. It is meant to be simple, extensible, easily edited by graphical clients, and independent of operating system or signalling protocol. It is suitable for running on a server where users may not be allowed to execute arbitrary programs, as it has no variables, loops, or ability to run external programs. This document is a product of the IP Telephony (IPTEL) working group of the Internet Engineering Task Force. Comments are solicited and should be addressed to the working group's mailing list at iptel@lists.research.bell-labs.com and/or the authors.}

J. Lennox and H. Schulzrinne, "Call Processing Language Framework and Requirements," Internet Engineering Task Force, {Internet Draft}, Jan. 2000.

Abstract: A large number of the services we wish to make possible for **Internet telephony** require fairly elaborate combinations of signalling operations, often in network devices, to complete. We want a simple and standardized way to create such services to make them easier to implement and deploy. This document describes an architectural framework for such a mechanism, which we call a call processing language. It also outlines requirements for such a language.

H. Schulzrinne and J. Lennox, "Call Processing Language Requirements," Internet Engineering Task Force, {Internet Draft}, Aug. 1998.

Abstract: A large number of the services we wish to make possible for **Internet telephony** require fairly elaborate combinations of signalling operations, often in network devices, to complete. We want a simple and standardized way to create such services to make them easier to implement and deploy. This document describes an architecture for such a method, which we call a call processing language. It also outlines requirements for such a language.

J. Rosenberg, H. Salama and M. Squire, "Attributes for a Gateway Location Protocol," Internet Engineering Task Force, {Internet Draft}, Jun. 1999.

Abstract: The Gateway Location Protocol (GLP) provides a mechanism for distributing and maintaining call routing tables between multiple **internet telephony** providers. GLP is currently under development by the iptel WG.

J. Rosenberg and H. Schulzrinne, "A Framework for Telephony Routing over IP," Internet Engineering Task Force, {Internet Draft}, Nov. 1999.

Abstract: This document serves as a framework for Telephony Routing over IP (TRIP), which supports the discovery and exchange of IP telephony gateway routing tables between providers. The document defines the problem of telephony routing exchange, and motivates the need for the protocol. It presents an architectural framework for TRIP, defines terminology, specifies the various protocol elements and their functions, overviews the services provided by the protocol, and discusses how it fits into the broader context of **Internet telephony**.

C. Bormann, "Providing integrated services over low-bitrate links," Internet Engineering Task Force, {Internet Draft}, Jun. 1999.

Abstract: This document describes an architecture for providing integrated services over low-bitrate links, such as modem lines, ISDN B- channels, and sub-T1 links. It covers only the lower parts of the Internet Multimedia Conferencing Architecture [1]; additional components required for application services such as **Internet Telephony** (e.g., a session initiation protocol) are outside the scope of this document. The main components of the architecture are: a real-time encapsulation format for asynchronous and synchronous low- bitrate links, a header compression architecture optimized for real- time flows, elements of negotiation protocols used between routers (or between hosts and routers), and announcement protocols used by applications to allow this negotiation to take place.

C. Huitema, "The multipart/sip-id media type," Internet Engineering Task Force, {Internet Draft}, Feb. 1999.

Abstract: This document proposes the definition of a multipart/sip-id media type, according to the rules defined in RFC 2048. This media type is intended to be carried by the session invitation protocol messages, when these messages are used to route calls between **Internet Telephony** domains.

T. Seth, A. Broscius, C. Huitema and H. Lin, "Performance Requirements for TCAP Signaling in Internet Telephony," Internet Engineering Task Force, {Internet Draft}, Mar. 1999.

Abstract: To allow interoperability between the existing telephone network and **Internet Telephony** (IT) it is necessary for the signaling performance to be comparable to that of the current standards to avoid introducing degradation in the service. In this Internet Draft, we discuss the performance requirements for TCAP signaling across an IP network. We also highlight the dependency on the SCP database location and thus problems related in providing high-quality service for TCAP based appli- cations.

C. Lee and M. Orsic, "A Framework for E.164 Number to IP Address Mapping," Internet Engineering Task Force, {Internet Draft}, Nov. 1998.

Abstract: This internet draft describes a framework for mapping the E.164 number of **internet telephony** (IT) subscribers to an IP addresses so that calls can be delivered to IT subscribers. The draft describes: - assumptions that the framework is based on - goals that the framework is designed for - functionality of network entities Several scenarios are included to illustrate the procedure.

J. Lennox, J. Rosenberg and H. Schulzrinne, "Common Gateway Interface for SIP," Internet Engineering Task Force, {Internet Draft}, Jun. 2000.

Abstract: In **Internet telephony**, there must be a means by which new services are created and deployed rapidly. In the World Wide Web, the Common Gateway Interface (CGI) has served as popular means towards programming web services. Due to the similarities between the Session Initiation Protocol (SIP) and the Hyper Text Transfer Protocol (HTTP), CGI seems a good candidate for service creation in a SIP environment. This draft proposes a SIP-CGI interface for providing SIP services on a SIP server.

J. Li and J. Mule, "SIP T.38 Call Flow Examples And Best Current Practice," Internet Engineering Task Force, {Internet Draft}, Mar. 2001.

Abstract: The Session Initiation Protocol allows the establishment of real- time Internet fax communications as defined by the ITU-T T.38 recommendation. This document attempts to clarify the options available to **Internet telephony** gateway vendors to handle real-time fax calls using SIP.

H. Schulzrinne, "Providing Emergency Call Services for SIP-based Internet Telephony," Internet Engineering Task Force, {Internet Draft}, Mar. 2001.

Abstract: If **Internet Telephony** is to offer a full replacement for traditional telephone services, it needs to provide emergency call services. In the United States, emergency calls are known as 911 services, based on the number dialed. This note desccribes some options for providing enhanced emergency service, i.e., emergency calls that allow emergency response centers to determine the address where the caller is located. This is made more difficult by the temporary nature of IP addresses, the large number of ISPs and their lack of legal responsibility for emergency services and the ability of many Internet terminals to be connected to the Internet at different locations. This note explores some of the requirements and design choices.

T. Seth, A. Broscius, C. Huitema and H. Lin, "Performance Requirements for Signaling in **Internet Telephony**," Internet Engineering Task Force, {Internet Draft}, Nov. 1998.

Abstract: To allow interoperability between the existing telephone network and **Internet Telephony** (IT) it is necessary for the signaling performance to be comparable to that of the current standards to avoid introducing degradation in the service. In this Internet Draft, we highlight the problem of providing high-quality signaling across an IP network that is built on a SONET infrastructure. We show that there are cases where the current PSTN standards are not satisfiable by a naive mapping of the IT signaling directly to the UDP or TCP transport protocols, even neglect- ing packet loss in router queues.

H. Sinnreich and F. Menard, "Service Requirements for **Internet Telephony** Signaling and Device Control Protocols," Internet Engineering Task Force, {Internet Draft}, Nov. 1998.

Abstract: This memorandum discusses the requirements for telephony signaling and device control over the Internet from the perspective of meeting the needs of Internet service providers (ISPs) and telecom carriers wishing to provide Internet services that include telephony. The requirements apply equally to various **Internet telephony** and non-**Internet telephony** devices. For the purpose of this Internet draft, the notion of telephony is broadened to include control of other types of streaming media sessions, such as RTSP based media server device control. Rather than severely restricting the device control framework to a particular set of devices, such as IP telephony gateways and telephony network access servers, this Internet draft presents requirements that are broad enough to satisfy the needs of any device that is expected to provide telephony and related services on the Internet.

R. Stewart and Q. Xie, "MULTI_NETWORK DATAGRAM TRANSMISSION PROTOCOL," Internet Engineering Task Force, {Internet Draft}, Sep. 1998.

Abstract: This Internet Draft discusses an experimental protocol, namely the Multi-network Datagram Transmission Protocol (MDTP), that is intended to provide fault-tolerant reliable/unreliable data transfer between communicating processes over IP networks [1]. MDTP is proposed as an application-level protocol which is designed with a high emphasis on supporting redundant networks and transparent fault management. MDTP also gives the application a great degree of timing control and configuration flexibilities. The motivation of developing MDTP is to establish a framework for supporting Internet-based high reliability real-time commercial applications such as signaling and call control for **Internet telephony**.

International Telecommunication Union, "Control protocol for multimedia communication," Telecommunication Standardization Sector of ITU, Geneva, Switzerland, Recommendation H.245, Feb. 1998.

Abstract: This Recommendation specifies syntax and semantics of terminal information messages as well as procedures to use them for in-band negotiation at the start of or during communication. The messages cover receiving and transmitting capabilities as well as mode preference from the receiving end, logical channel signalling, and Control and Indication. Acknowledged signalling procedures are specified to ensure reliable audiovisual and data communication.

Keywords: H.323; signaling; **Internet telephony**; teleconferencing; negotiation

International Telecommunication Union, "Security and encryption for H-Series (H.323 and other H.245-based) multimedia terminals," Telecommunication Standardization Sector of ITU, Geneva, Switzerland, Recommendation H.235, Feb. 1998.

Keywords: H.323; signaling; **Internet telephony**; teleconferencing; negotiation;security

International Telecommunication Union, "Interworking of H-Series multimedia terminals with H-Series multimedia terminals and voice/voiceband terminals on GSTN and ISDN," Telecommunication Standardization Sector of ITU, Geneva, Switzerland, Recommendation H.246, Feb. 1998.

Keywords: H.323; signaling; **Internet telephony**; teleconferencing; interworking

International Telecommunication Union, "H.323 extended for loosely-coupled conferences," Telecommunication Standardization Sector of ITU, Geneva, Switzerland, Recommendation H.332, Sep. 1998.

Keywords: H.323; signaling; **Internet telephony**; teleconferencing; SDP

International Telecommunication Union, "Generic functional protocol for the support of supplementary services in H.323," Telecommunication Standardization Sector of ITU, Geneva, Switzerland, Recommendation H.450.1, Feb. 1998.

Keywords: H.323; signaling; **Internet telephony**; teleconferencing; supplementary services

International Telecommunication Union, "Call Diversion Supplementary Service for H.323," Telecommunication Standardization Sector of ITU, Geneva, Switzerland, Recommendation H.450.3, Sep. 1997.

Abstract: Recommendation H.450.3 describes the procedures and the signalling protocol for the call diversion supplementary services (SS-DIV) in H.323 (Packet Based Multimedia Communications Systems) networks. This recommendation comprises the services call forwarding unconditional (SS-CFU), call forwarding busy (SS-CFB), call forwarding no reply (SS-CFNR) and call deflection (SS-CD). SS-CFU, SS-CFB, SS-CFNR and SS-CD are supplementary services which apply during call establishment providing a diversion of an incoming call to another destination endpoint. The procedures and the signaling protocol of this recommendation are derived from the call diversion supplementary service specified in ISO/IEC 13872 and 13873.

Keywords: call forwarding; **Internet telephony**; H.323; signaling

International Telecommunication Union, "Call waiting supplementary service for H.323," Telecommunication Standardization Sector of ITU, Geneva, Switzerland, Recommendation H.450.6, May 1999.

Keywords: call services; call waiting; **Internet telephony**; H.323

Aravind Srinivasan, K. G. Ramakrishnan, Krishnan Kumaran, Murali Aravamudan and Shamim Naqvi, "Optimal Design of Signaling Networks for **Internet Telephony**," in *Proceedings of the Conference on Computer Communications (IEEE Infocom)*, (Tel Aviv, Israel), Mar. 2000.

Abstract: We present an approach for efficient design of a signaling network for a network of Lucent SoftSwitches supporting Internet Telephony. While one may take an Integer Programming approach to solve this problem, it quickly becomes intractable even for modest-sized networks. Instead, our topology design uses random graphs that we show to be nearly optimal in cost, highly connected, and computationally efficient even for large networks. (Prior work~\cite{FCB99} has addressed topology design using random graph techniques. We identified some gaps in this work, for which we provide resolutions.) We then formulate a {\em Quadratic Assignment Problem} (QAP) to map the abstract topology into the physical network to achieve optimal load balancing for given demand forecasts, which we solve using randomized heuristics. Numerical results on several example networks illustrate the performance and computational efficiency of our method. A graphical design tool has been developed based on our algorithms.

Keywords: Network architectures (protocols, algorithms, intelligent networks, reliability); Network management and control; Internet and web applications

Olivier Hersent, David Gurle and Jean-Pierre Petit, "IP telephony," Reading, Massachusetts, 2000.

Keywords: Internet telephony; SIP; H.323

Jonathan Lennox and Henning Schulzrinne, "Feature Interaction in Internet Telephony," in *Proc. of Feature Interaction in Telecommunications and Software Systems VI*, (Glasgow, United Kingdom), May 2000.

Abstract: While **Internet telephony** aims to provide services at least equal to traditional telephony, the architecture of **Internet telephony** is sufficiently different to make it necessary to revisit the issue of feature interaction in this context. While many basic feature interaction problems remain the same, **Internet telephony** adds additional complications. Complications arise since functionality tends to be more distributed, users can program the behavior of end systems and signaling systems, the distinction between end systems and network equipment largely vanishes and the trust model implicit in the PSTN architecture no longer holds. On the other hand, **Internet telephony** makes end point addresses plentiful and its signaling makes it easy to specify in detail the desired network behavior. Many techniques for resolving interactions in the PSTN are no longer easily applied, but several new techniques, {\em explicitness}, {\em authentication}, and {\em verification testing}, become possible in the Internet environment.

Keywords: Feature interaction; **internet telephony**

Maria Stachelek, "PacketCable Network Architecture," in *Carrier Class IP Telephony*, (San Diego, California), Jan. 2000.

Keywords: packet cable; VoIP; **Internet telephony**

Ronald J. Wocjik, "PacketCable Network Architecture," in *Carrier Class IP Telephony*, (San Diego, California), Jan. 2000.

Keywords: IN; intelligent network; **Internet telephony**

Ravi Ravishankar, "Carrier Class IP Telephony," in *Carrier Class IP Telephony*, (San Diego, California), Jan. 2000.

Keywords: signaling network; **Internet telephony**; TALI; SCTP; SS7

Paul Kerney, "Building the New Public Network," in *Carrier Class IP Telephony*, (San Diego, California), Jan. 2000.

Abstract: Describes Spanish trial VoIP network architecture.

Keywords: Internet telephony

Ivan Gorgeon, "Implications of VoIP on the New Carrier Market," in *Carrier Class IP Telephony*, (San Diego, California), Jan. 2000.

Abstract: Graphs packet-switched voice minutes as fraction of total voice traffic. Claims 44,972 million voice minutes for 1999, growing to 219431 million in 2005, with packet-switched growing from 1,124 million to 83,384 million minutes.

Keywords: Internet telephony

Tony Eyers and Henning Schulzrinne, "Predicting **Internet Telephony** Call Setup Delay," in *Proceedings of the 1st IP-Telephony Workshop (IPtel 2000)*, (Berlin, Germany), Apr. 2000.

Abstract: Internet telephony has been the focus of much recent effort by ITU and IETF standards bodies, with initial, albeit small-scale deployment in progress. While **Internet telephony** voice quality has been studied, call setup delay has received little attention. This paper outlines a simulation study of **Internet Telephony** Call Setup delay, based on UDP delay/loss traces. The focus is signaling transport delay, and the variations arising from packet loss and associated retransmissions. Of particular interest are the differences arising from H.323 signaling, which uses TCP, and SIP, which can use UDP with additional error recovery. Results show that during high error periods, H.323 call setup delay significantly exceeds that of SIP. We also consider PSTN/**Internet telephony** interworking, and show that high blocking rates are likely if either H.323 or SIP are used across the public Internet.

Keywords: Internet telephony; SIP; H.323; signaling; call Setup Delay; QoS

Kundan Singh and Henning Schulzrinne, "Interworking Between SIP/SDP and H.323," in *Proceedings of the 1st IP-Telephony Workshop (IPtel 2000)*, (Berlin, Germany), Apr. 2000.

Abstract: There are currently two standards for signaling and control of Internet telephone calls, namely ITU-T Recommendation H.323 and the IETF Session Initiation Protocol (SIP). We describe how a signaling gateway can allow SIP user agents to call H.323 terminals and vice

versa. Our solution addresses user registration, call sequence mapping and session description. We also describe and compare various approaches for multi-party conferencing and call tranfer.

Keywords: Internet telephony; interworking; SIP; SDP; H.323; signaling gateway

Jonathan Lennox and Henning Schulzrinne, "The Call Processing Language: User Control of **Internet Telephony** Services," in *Lucent Technologies XML Day*, (Murray Hill, NJ), Feb. 2000.

Keywords: call processing; CPL; XML

Anders Kristensen, Anders Byttner and Roman Kurmanowytsch, "Programming SIP Services," in *Proceedings of the 1st IP-Telephony Workshop (IPtel 2000)*, (Berlin, Germany), Apr. 2000.

Abstract: As the number of communication modalities available to people increase, the ability for service providers and end users to author and provision communications services will become increasingly important. Programmability of **Internet Telephony** services will arguably need to be more like Web services than traditional telephony service environments. We have proposed a Java API based on the concept of Java servlets as an extension mechanism for SIP servers. This paper gives an overview of this API and our prototype implementation of it, including a description of how we support the Call Processing Language on top of it.

Keywords: SIP; Java; **Internet telephony**

Ralf Ackermann Utz Roedig and Ralf Steinmetz, "Evaluating and Improving Firewalls for IP-Telephony Environments," in *Proceedings of the 1st IP-Telephony Workshop (IPtel 2000)*, (Berlin, Germany), Apr. 2000.

Abstract: Firewalls are a well established security mechanism for providing access control and auditing at the borders between different administrative network domains. Their basic architecture, techniques and operation modes did not change fundamentally during the last years. On the other side new challenges emerge rapidly when new innovative application domains have to be supported. IP-Telephony applications are considered to have a huge economic potential in the near future. For their widespread acceptance and thereby their economic success they must cope with established security policies. Existing firewalls face immense problems here, if they - as it still happens quite often - try to handle the new challenges in a way they did with ‘‘traditional applications’’. As we will show in this paper, IP telephony applications differ from those in many aspects, which makes such an approach quite inadequate. After identifying and characterizing the problems we then describe and evaluate a more appropriate approach. The feasibility of our architecture will be shown. It forms the basis of a prototype implementation that we are currently working on.

Keywords: Firewalls; H.323; **Internet telephony**; network security; VoIP

Stefan Gessler, Oliver Haase and Andreas Schrader, "A Service Platform for Internet Telephony," in *Proceedings of the 1st IP-Telephony Workshop (IPtel 2000)*, (Berlin, Germany), Apr. 2000.

Abstract: Inevitably, two formerly separated kinds of communication networks - public switched telephone networks (PSTN) and packet data communication networks - are meeting under the umbrella of IP telephony. In this paper we present I2lN (Intelligent **Internet Telephony**) as a novel platform for IP telephony, which takes the best of the network centric approach of PSTN and the edge centric ap-proach of packet data networks. I2N provides various layers of a comprehensive IP telephony system, from basic call signalling, via access to user directories and support of various aspects of mobility, to the rapid integration of value-added services. Together with the integrated AQUARIUS QoS framework, I2N is perfectly suited to realise user-tailored communication applications with high quality media support. Interworking with related standards is provided by multi-level gateway technology.

Keywords: AQUARIUS QoS framework; multi-level gateway technology; CORBA; QoS; Java; IN

Henning Schulzrinne, "**Internet Telephony**: A Second Chance," in *Proceedings of the 1st IP-Telephony Workshop (IPtel 2000)*, (Berlin, Germany), Apr. 2000.

Lee W. McKnight, "**Internet Telephony** Markets: 2000 -- 3001," in *Proc. of Carrier Class IP Telephony*, (San Diego, California), Jan. 2000.

Abstract: Describes players and projections for IP telephony equipment. ‘‘True costs of telephony are 0.1 to 0.2 cents/minute. Voice traffic will be 2--10\% of total traffic by 2007.’’ Gives statistics for AT\&T and MCI POPs. Plots Mbone usage (or lack thereof).

Keywords: IP telephony; VoIP

Bülent Yener, "Smart Box Architecture," in *Special Workshop on Intelligence*, (San Francisco, California, USA), Mar 2000.

Abstract: Fuandametnally the IP-based networking is designed for delivering data traffic with best-effort service, thus it is not capable of providing end-to-end QoS. Several architectures have been proposed for providing QoS in the Internet: The integrated services (Intserv) model is based on reservations and can provide QoS, however; it is not scalable. The differentiated services (Diffserv) approach is scalable but falls short of ensuring deterministic guarantees - in particular for the services that belong to the same class. Finally, the multi protocol label switching (MPLS) architecture provides mechanisms for QoS-based routing but does not have the necessary resource management and scheduling support to ensure it. This work proposes a hybrid solution which combines the best of these technologies. First, at the network boundary Diffserv like Service Level Agreements (SLA) are provided to users by intelligent edge routers called the SBoX servers. An SBoX server uses Class Based Queuing (CBQ) with a hierarchy of flow aggregation. At the top a commodity-flow is defined for the aggregate flow between a pair of egress points. The packets of the same commodity-flow are marked by an MPLS label, which is globally unique within an Autonomous System (AS). Each commodity flow is partitioned to a set of macro-flows which are offered to users as SLAs. An SBoX server manges macro-flows and commodity flows only, and leaves the management of each macro-flow (at the micro-flow level based on some policies) to the enterprise/ users which signed the SLA. Second, the commodity-flows are managed and supported inside the network by an add-on Label Switching Router (LSR) called the SBoX router which performs MPLS of commodity-flows with CBQ. The main reason for an add on solution is the lack of end-to-end deployment of LSRs, and the vertically integrated architecture of the legacy routers. This paper explains the SBoX architecture and reports experimental results obtained on a prototype network

John Sundstrom and Eric Aupperle, "**Internet Telephony** GVSU to WMU," in *Proc. of Net@EDU Annual Member Meeting}*, (Tempe, Arizona), Feb. 2000.

Abstract: 11\% of telephone traffic goes to other universities. Uses trunk connection to MERIT network, costing \$19,000.

Keywords: IP telephony; VoIP

Masataka Ohta, Kenji Fujikawa, Manolo Sola and Kaz Satoh, "The Simple Internet Phone," in *Proc. of INET*, (Yokohama, Japan), Jul. 2000.

Abstract: The ‘‘Simple Internet Phone’’ has an architecture tuned for a future situation in which non-Internet networks, such as IP-based private telephone networks, will disappear. While the ‘‘Simple Internet Phone’’ is a form of voice over Internet Protocol (VoIP), most, if not all, VoIP protocols are designed placing the priority in the affinity to the telephone network. However, it is obvious that the telephone network will be replaced by the Internet, and will eventually disappear. At that time, most of the features of VoIP protocols will become obsolete. Instead, the ‘‘Simple Internet Phone’’ is designed placing the priority in the affinity to the Internet and its architectural principles as an ‘‘end-to-end,’’ ‘‘globally connected’’ and ‘‘scalable’’ IP network. As a result, most features of VoIP are substituted by the existing Internet protocols. With Internet phones, callees are required to have persistent connection to the Internet with globally unique addresses, which helps to promote the healthy development of the Internet.

Keywords: Internet telephony; NOTASIP

Xiaotao Wu and Henning Schulzrinne, "Where Should Services Reside in Internet Telephony Systems?," in *IP Telecom Services Workshop*, (Atlanta, Georgia), Sep. 2000.

Abstract: Internet telephony end systems can take on a much larger role in providing services than in the PSTN. However, there are still a large number of services that are better provided by servers residing in the ’network’. We analyze some sample services and discuss how they can be created in both architectures, using the SIP (Session Initiation Protocol) and DFC (Distributed Feature Creation) architectures as examples.

Keywords: Internet telephony services; SIP CGI; DFC; SIP; service creation

Henning Schulzrinne and Jonathan Rosenberg, "The Session Initiation Protocol: Internet-Centric Signaling," *IEEE Communications Magazine*, vol. 38, no. 10, Oct. 2000.

Abstract: The Session Initiation Protocol (SIP) provides advanced signaling and control functionality for a wide variety of multimedia services. SIP can efficiently and scalably locate resources based on a location-independent name and then negotiate session characteristics. It can find use in applications ranging from **Internet telephony** and conferencing to instant messaging, event notification and the control of networked devices. We summarize the main protocol features and describe a range of extensions currently being discussed within the Internet Engineering Task Force.

Keywords: signaling; session initiation protocol; SIP; Internet telephony; Internet multimedia; presence

Kundan Singh and Henning Schulzrinne, "Unified Messaging using SIP and RTSP," in *IP Telecom Services Workshop*, (Atlanta, Georgia), pp. 7, Sep. 2000.

Abstract: Traditional answering machines and voice mail services are closed systems, tightly coupled to a single end system, the local PBX or local exchange carrier. Even simple services, such as forwarding voice mail to another user outside the local system, are hard to provide. With the advent of **Internet telephony**, we need to provide voice and video mail services. This also offers the opportunity to address some of the shortcomings of existing voice mail systems. We

list general requirements for a multimedia mail system for **Internet telephony**. We then propose an architecture using SIP (Session Initiation Protocol) and RTSP (Real-Time Streaming Protocol) and compare various alternative approaches to solving call forwarding, reclaiming and retrieval of messages. We also briefly describe our prototype implementation.

Keywords: voice mail; video mail; unified messaging; SIP; RTSP; Internet telephony

Mike Pluke, "User identification solutions in converging networks," in *Report of IP-Telecoms Interworking Workshop (Numbering, Naming, Addressing and Routing)*, (Geneva, Switzerland), pp. IPW-10, Jan. 2000.

Abstract: Describes naming mechanisms for **Internet telephony**.

Keywords: naming; numbering; E.164

Cengiz Alaettinoglu, Van Jacobson and Haobo Yu, "Towards Millisecond IGP Convergence," in *Proc. of NANOG*, (Washington, DC), Oct. 2000.

Abstract: On currently deployed IP networks, ‘‘convergence’’ times, or the ability to reroute, is often cited as one of the key issues in providing new services and larger scale. It is, however, possible for link-state routing protocols to converge in link propagation time scales, that is, in tens of milliseconds. Why then are deployments of IS-IS, a link-state routing protocol, not anywhere near this point? In this talk, we present some analyses of IS-IS convergence by showing its behavior upon link/router failures and repairs, and its scaling properties to large networks, both in terms of number of nodes and links. We then explore changes needed in the IS-IS specification and implementations to reach IGP convergence in milliseconds. Our results are based on experimentation done with IS-IS, but some of the findings may apply to OSPF as well.

Keywords: IGP; routing

Mohsen Guizani, Ammar Rayes and Mohammed Atiquzzaman, "**Internet Telephony**," *IEEE Communications Magazine*, vol. 38, no. 4, pp. --, Apr. 2000.

Bo Li, Mounir Hamdi, Dongyi Jiang, Xi-Ren Cao and Y. Thomas Hou, "QoS-Enabled Voice Support in the Next-Generation Internet: Issues, Existing Approaches and Challenges," *IEEE Communications Magazine*, vol. 38, no. 4, pp. --, Apr. 2000.

Abstract: The Internet is under rapid growth and continuous evolution in order to accommodate an increasingly large number of applications with diverse service requirements. In particular, **Internet telephony**, or voice over IP is one of the most promising services currently being deployed. Besides the potentially significant cost reduction, Internet telephony can offer many new features and easier integration with widely adopted Web-based services. Despite these advantages, there still exist a number of barriers to the widespread deployment of **Internet telephony** such as the lack of control architectures and associated protocols for managing calls, a security mechanism for user authentication, and proper charging schemes. The most prominent one, however, is how to ensure the QoS needed for voice conversation. The purpose of this article is to survey the state-of-the-art technologies in enabling the QoS support for voice communications in the next-generation Internet. In this article, we first review the existing technologies in supporting voice over IP networks, including the basic mechanisms in the IETF **Internet telephony** architecture and ITU-T H.323-related Recommendations. We then discuss the IETF QoS framework, specifically the Intserv and Diffserv framework. Finally, we present

two leading companies' (Cisco and Lucent) solutions to offering IP telephony services as examples to illustrate how real systems are implemented.

Wanjiun Liao and Jen-Chi Liu, "VoIP Mobility in IP/Cellular Network Internetworking," *IEEE Communications Magazine*, vol. 38, no. 4, pp. --, Apr. 2000.

Abstract: This article explores VoIP mobility in the context of IP and cellular networks interworking. ITU-T Rec. H.323 gateways provide the interconnection between IP networks and switched circuit networks. They allow a call originating from an SCN phone to be transmitted over an IP network to an H.323 terminal, or bridged to another SCN phone. While H.323 provides interoperability with other SCN terminals, the major efforts have been focused on IP/wired SCN (PSTN, ISDN, etc.) interworking. In this article we discuss the challenges associated with the interworking between IP networks and cellular networks through H.323 gateways, and propose an innovative approach using the existing call transfer supplementary service to provide VoIP mobility in the H.323 IP telephony networks. The proposed approach uses existing components in the H.323 standard, thereby allowing VoIP mobility service in hybrid IP/cellular networks to be a value-added feature in the existing H.323-compliant **Internet telephony** systems.

Mahbub Hassan, Alfandika Nayandoro and Mohammed Atiquzzaman, "**Internet Telephony**: Services, Technical Challenges, and Products," *IEEE Communications Magazine*, vol. 38, no. 4, pp. --, Apr. 2000.

Abstract: The rapid proliferation of the Internet in the last few years has given rise to a strong interest in carrying telephony over the Internet. Because the Internet supports data communications, a range of other services can be bundled together with **Internet telephony**. The Internet, however, was designed for non-real-time data communications, and hence it poses several technical challenges that must be overcome before the Internet can be successfully used for carrying telephone services. This article discusses new services we can expect from **Internet telephony**, the technical challenges and solutions, and the emerging products that promise to support **Internet telephony**.

Henning Schulzrinne and Elin Wedlund, "Application-Layer Mobility using SIP," *Mobile Computing and Communications Review*, vol. 4, no. 3, pp. 47--57, Jul. 2000.

Abstract: Supporting mobile Internet multimedia applications requires more than just the ability to maintain connectivity across subnet changes. We describe how the Session Initiation Protocol (SIP) can help provide terminal, personal, session and service mobility to applications ranging from **Internet telephony** to presence and instant messaging. We also briefly discuss application-layer mobility for streaming multimedia applications initiated by RTSP.

Keywords: mobility; SIP

Mitrabarun Sarkar, "An Assessment of Pricing Mechanisms for the Internet--A Regulatory Imperative," in *Proc. of MIT Workshop on Internet Economics*, Mar. 1995.

Abstract: This paper argues that however much of an anathema the notion of regulating the Internet may be, there is a strong need to start putting the appropriate regulatory structures in place as the commercialized Internet moves incrementally towards a usage-based pricing system. Various factors such as new bandwidth-hungry applications; the massification of the net; the concerted entry of the telephone, cable, and software companies; and the proliferation of electronic commerce all imply unimaginable potential growth rates for the Internet and a resultant scarcity of bandwidth, thus making it imperative to put a pricing system in place that would effectively ration scarce bandwidth. As has been argued by many, a usage-based pricing system seems to be an innovative way to effectively ration scarce bandwidth. In this context, this paper examines the Precedence and the Smart Market models of Internet pricing. We note that (a) the perceived homogeneity of the Internet's load, and (b) the threat of market-power abuse through artificial creation of a high network load by those who control the bottleneck facilities, remain the fundamental weaknesses of usage-based pricing. However, given that usage- based pricing is inevitable, and that the Smart Market mechanism does present an innovative and a potential solution, it is important to consider the appropriate safeguards that need to be put in place. In this context, the paper argues that a usage based, free market pricing system needs to be combined with some form of regulatory oversight to protect against anti-competitive actions by the firms controlling the bottleneck facilities and to ensure non- discriminatory access to emerging networks.

Keywords: Internet telephony; Internet economics; pricing; regulation

Henning Schulzrinne, "Personal Mobility for Multimedia Services in the Internet," in *European Workshop on Interactive Distributed Multimedia Systems and Services (IDMS)*, (Berlin, Germany), Mar. 1996.

Abstract: Personal mobility is one of the goals of Universal Personal Telecommunications (UPT) being specified for future deployment. Most current efforts focus on telephony, with SS7 signaling. However, many of the same goals can be accomplished for multimedia services, by using existing Internet protocols. We describe a multimedia call/conference setup protocol that provides personal videophone addresses, independent of the workstation a called party might be using at the time. The system is set up to use the existing Internet email address as a videophone address. Location and call handling information is kept at the subscriber's home site for improved access and privacy.

Keywords: SCIP; signaling; **Internet telephony**; packet audio; packet video; conference control

Henning Schulzrinne, "**Internet Telephony --** Towards the Integrated Services Internet," in *Proc. of IEEE Workshop on **Internet Telephony***, (Utrecht, The Netherlands), Feb. 1996.

Abstract: Currently, the Internet is mostly used for non-real time, data services such as electronic mail, news groups or WWW browsing. Increased availability of high-speed modems and ISDN as well as audio-equipped workstations and PCs have made it feasible to use the Internet for telephony, as well as an alternative for circuit-switched multimedia conferencing applications. Besides possible economic advantages, the Internet allows much easier addition of advanced functionalities and user interfaces. However, a large number of technical and infrastructure problems remain to be solved before **Internet telephony** becomes viable on a large scale. We present measurement results on Internet behavior, and algorithms to compensate for the Internet-specific impairments, in particular, large delay variations. Bandwidth control adapts encodings to the available bandwidth. A multicast-based signaling protocol allows to set up connections to the callee's email address, without having to know the callee current network location. An example research application, NeVoT, incorporates these algorithms and protocols.

Keywords: Internet telephony; packet audio; packet video; conference control; NeVoT

Clemens Fricke, Lutz Grüneberg and Prof. Dr. Helmut Pralle, "Click and Meet -- Confman: Telefonieren über das Internet," *DFN Mitteilungen*, vol. 40, Mar. 1996.

> **Abstract:** Desktop-Online-Konferenzsysteme gewinnen immer mehr an Bedeutung. Gegenüber den technisch etablierten, aber in geringem Maße verbreiteten, dedizierten Konferenzräumen bieten sie den Vorteil, daß ihr Einsatz spontan möglich ist und der Endanwender seine gewohnte Arbeitsumgebung nicht verlassen muß. Daß dasübertragene Video-Bild kaum Fernsehqualität erreicht, läßt sich häufig verschmerzen. Der Vorteil durch die lokale Verfügbarkeit der benötigten Arbeitsunterlagen in Form von Daten und Anwendungen macht dies leicht wett. Mit Confman liegt ein derartiges, komfortables Online-Konferenzsystem vor.
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> **Keywords:** teleconferencing; **Internet telephony**; MBONE

Louise Turner and Peter Sommerer, "The Impact of the Internet on the Phone Industry: Facts and Vision," in *Interop*, (Frankfurt), Jun. 1996.

> **Abstract:** Vision for **Internet telephony**. Today, 9.5 million Internet hosts, to grow to 100 mio. by year 2000, compared to 650 million telephones today. 10^{13} bytes/month of Internet traffic in 1996, compared to 10^{15} bytes/month of international phone traffic in 1994. Global phone network capacity 10^{18} bytes/month. 15,000 Internet phone users, 20 mio. if Netscape bundles it. Quality can be improved by reducing the number of hops, with toll-quality in corporate Intranets. A Newbridge Affiliate, Vienna Systems, will be launching its LAN-PSTN equipment for this type of application at the Atlanta Interop in September (American launch) and at the Paris Interop in October (European launch). This type of Phone-to-Phone VOI service is being launched by two competing firms in Canada: Shadowtel and Alphanet. RSVP, dedicated service lines, co-locating telephone access with backbone nodes, bigger routers and new network architectures can be used to improve quality. A giga-router can handle 5 mio. to 20 mio. instead of 500,000 to 1,000,000 packets per second. MPOA is better as it minimizes router hops. Circuit switching cost are 15c/kbit, while packet switching costs are 4c/kbit, resulting in a reduction of transatlantic phone costs to \$0.04/min or less, compared to \$0.40/min for resellers (call back). Large U.S. long distance carriers make around 6\% net profit each year. To remove this profit, 1 in every 17 phone calls needs to be made over VOI, or 5.7 million households in the U.S. need to switch to VOI. 6\% of international voice traffic is 10^{14} bytes a year, increasing current Internet traffic 11 times.
>
> **Keywords: Internet telephony**; economics; Internet; long-distance services; voice over Internet

Colin Low, "The **Internet Telephony** Red Herring," in *Proceedings of Global Internet*, (London, England), pp. 72--80, Nov. 1996.

> **Abstract:** The spectre of low-cost, real-time voice communications over the Internet has polarised Internet service providers and telephone network operators, as the expense of finding solutions to the problem of integrating communications services (in particular, existing wireline and wireless telephony) with WWW content services. This paper argues that solutions not only exist and appear to be commercially viable, but could bring about a transformation of the WWW as a tool for business and personal communications.
>
> **Keywords: Internet telephony**; intelligent networks; SS7; SCP; WWW; AIN; DPE; Corba; PSTN; TSAPI; DNS; computer-telephony integration; CTI; billing; call forwarding; call handling; signaling; Nexus

Andrew Sears, "Directory services for **Internet Telephony**," in *Transparencies of **Internet Telephony** July 1 Meeting*, (Boston, Massachusetts), Jul. 1996.

Keywords: Internet telephony; directory services; ULS; LDAP

Ed Margulies, "Understanding the voice-enabled Internet," New York, NY, 1996.

Keywords: Internet; packet voice; **Internet telephony**; NeVoT

Henning Schulzrinne, "Signaling for **Internet Telephony** Services," in *Proc. of Opensig'96*, (New York, New York), Oct. 1996.

Abstract: Describes signaling services for **Internet telephony**, including 800 and 900 services and intelligent network (AIN) services.

Keywords: SCIP; signaling; **Internet telephony**

Henning Schulzrinne, "Real-Time Services in the Internet?," in *Panel discussion at the International Conference on Network Protocols (ICNP)*, (Columbus, Ohio), Oct. 1996.

Abstract: Discusses issues that stand in the way of widespread deployment of real-time services in the Internet, particular in the consumer realm.

Keywords: Internet real-time services; **Internet telephony**; resource reservation

Peter Waters, Liza Carver and Michael Reede, "The Internet and Telephony: The Impact of Uncontrollable Technology on Traditional Telephony Regulation," Gilbert and Tobin, Sydney, Australia, White paper, Dec. 1996.

Abstract: The clash between low untimed Internet usage charges and timed PSTN telephony challenges the fundamental paradigms of telephony regulation. The carriers have legitimate concerns that Internet telephony threatens their substantial investment in PSTN infrastructure. Equally, ISPs and Internet users also are legitimately concerned that any departure from the Internet's basic philosophy of untimed charges to protect PSTN telephony will imperil the Internet itself. Given the highly decentralised nature of the Internet, traditional regulatory tools are unlikely to be successful in resolving this conflict, and a more fundamental reckoning is required between **Internet telephony** and PSTN telephony charging.

Keywords: Internet telephony; economics; pricing

Andrew L. Sears, "The Effect of **Internet Telephony** on the Long Distance Voice Market," MIT, Cambridge, Massachusetts, ITC working paper, 1996.

Abstract: With the rapid growth of the Internet, one of the questions frequently asked is how the Internet will affect the telecommunications market. While most Internet applications are likely to be seen as another source of revenue to the Interexchange Carriers (IXCs), Internet telephony, or the transportof voice over the Internet, might pose a potential threat. The focus of this paper is to consider how the current long distance voice market mightbe affected by the development of **Internet telephony** through a conceptcalled a 'phone gateway network.' Although a phone gateway network does not yet exist, the basic idea of it would be to provide regular phone-to-phone calling using Internet transport as a substitute circuit switched transmission.The paper uses a simplified cost model to examine the competitiveness of a phone gateway network in the existing long distance market. The paper then examines the expected competative response

of various players in the telecommunications industry, including the strategic use of regulation. The paper concludes noting that the future success of **Internet telephony** will depend on technical and regulatory factors in the future.

Keywords: Internet economics; **Internet telephony**; pricing

Andrew L. Sears, "Innovations in **Internet Telephony**: The Internet as the Competitor to the POTS Network," MIT, Cambridge, Massachusetts, Working paper, Feb. 1996.

Abstract: The focus of this paper is to examine how innovations in **Internet telephony** are bringing the Internet into competition with traditional telephone networks. Differences between the two networks will be examined to consider whether the Internet might be emerging as a challenger network.

Keywords: Internet telephony

S. Forrest, S. A. Hofmeyr, A. Somayaji and T. A. Longstaff, "A Sense of Self for Unix Processes," in *Proceedings of the 1996 IEEE Symposium on Security and Privacy*, (Los Alamitos, CA), pp. 120-128, 1996.

Terrence P. McGarty, "Internet Voice: Regulatory and Legal Implications," , Sep. 1996.

Abstract: This paper presents an overview of the regulatory and legal implications of Internet Voice. The implications are that Internet Voice is at one time a product and at another time a service. The product characterization is protected based upon the Carterphone decision of 1968. The service aspect is more problematical. We argue herein that there are significant advantages of Internet Voice and that these implications are significant in terms of their ability to provide a strong competitor to the existing carriers.

Keywords: packet voice; **Internet telephony**

Colin Low, "Integrating Communication Services," *IEEE Communications Magazine*, vol. 35, no. 6, pp. --, Jun. 1997.

Abstract: The need for communication services which span multiple communication technologies is growing. Communication services are being developed in three areas: in the public switched telephony networks, on the Internet in the form of integrated multimedia including voice-over-Internet, and in private switched telephony networks in the form of enterprise computer-telephony integration applications. This article shows it is plausible to create unified services which span the Internet and public switched telephony networks, and goes on to describe Nexus, an architecture and prototype for integrated communication services.

Keywords: multimedia; voice-over-internet; computer telephony; telephony; intelligent network services; **Internet telephony**; AIN; web; DNS

Henning Schulzrinne, "A comprehensive multimedia control architecture for the Internet," in *Proc. International Workshop on Network and Operating System Support for Digital Audio and Video (NOSSDAV)*, (St. Louis, Missouri), May 1997.

Abstract: The Internet and intranets have been used to deliver continuous media, both stored and interactive, for a number of years. Most of the attention has focused on providing guaranteed quality of service (RSVP) and end-to-end data transport (RTP), with every application using its own control protocol. In this paper, we describe a control architecture that supports most standard

advanced telephony features and allows to integrate stored and interactive multimedia. The protocol re-uses much of the ‘‘infrastructure’’ of HTTP, including its security and proxy mechanisms. The architecture is instantiated by two related, but independent protocols: the Session Initiation Protocol (SIP) for inviting participants to a multimedia session and the Real-Time Stream Protocol (RTSP) to control playback and recording for stored continuous media.

Keywords: SIP; RTSP; signaling; **Internet telephony**; continuous media; video on demand; audio on demand

Henning Schulzrinne, "Feature Interaction in **Internet Telephony**," in *Proc. of Feature Interaction in Telecommunication Networks IV*, (Montreal, Canada), pp. 371, Jun. 1997.

Keywords: feature interaction

Henning Schulzrinne, "Internet Problems and Potential," in *Proc. of Global Information Infrastructure Workshop at IEEE International Conference on Communications (ICC)*, (Montreal, Canada), Jun. 1997.

Abstract: The Internet has created new modes of communication, but is now poised to also serve as the underlying technology for services currently provided by separate networks. However, as much as it offers additional functionality and possibly cost advantages, the provision of telephony and audio/video distribution services requires changes to the current Internet protocols and the Internet access infrastructure.

Keywords: Internet architecture; **Internet telephony**; Internet access

Vint Cerf, "Everything on the Net," in *Voice on the Net (VON)*, (San Jose, California), Apr. 1997.

Annote: Contains maps of MCIs Internet backbone

Keywords: Internet telephony; VAULT; MCI

Michael A. Ramalho, Michael Goldstein, Mike Buckley and Robert Barr, "Patents and speech coders panel," in *Voice on the Net (VON)*, (San Jose, California), Apr. 1997.

Annote: Brief speech coding overview (LPC, CELP, MBE).

Keywords: Internet telephony; speech coding

John MacMillan, "**Internet Telephony** Gateways," in *Voice on the Net (VON)*, (San Jose, California), Apr. 1997.

Keywords: Internet telephony; gateways; PSTN; PBX

Sam Paltridge, "Internet regulatory session," in *Voice on the Net (VON)*, (San Jose, California), Apr. 1997.

Annote: Internet hosts per 1000 inhabitants; name registration policies; local charging practices (flat rate, unmeasured rates, measured rates); tariffs as a function of distance

Keywords: Internet telephony; OECD

Anonymous, "H.323 and Firewalls: The problems and pitfalls of getting H.323 safely through firewalls," Intel Corporation, Developer Note, Apr. 1997.

Abstract: The first part of this document provides an overview of H.323 - what the protocol is, why it's important, and how it works. The second section provides a framework for discussing firewall issues, including a taxonomy for classifying firewalls. The third section discusses the issues of H.323 and proxies - why H.323 is hard for firewalls, and what implications a proxy has on H.323 applications. The fourth section is a short overview of the changes necessary to an H.323 application to support proxies. Finally, the appendices provide additional information, including pointers to other sources, a 'decoder ring' for the ITU-T's 'alphabet soup' of protocols, and a detailed trace from a typical H.323 call.

Keywords: H.323; firewall; proxy; **Internet telephony** signaling

Herbert L. Tinger, "IP Telephony," First Albany, Albany, New York, Research Report, Jun. 1997.

Abstract: Summarizes IP telephony market, applications and providers.

Keywords: Internet telephony; Internet fax; packet audio

François Ménard, "Massively distributed **Internet telephony** gateways -- gateways to/from analog plain telephone service," in *Summer Internet World*, (Chicago, Illinois), Jul. 1997.

Abstract: Describes that each household should run its own PSTN gateway.

Keywords: Internet telephony; gateway; PSTN

Mordy Rothberg, "Global interoperable **Internet telephony -** challenges and opportunities," in *MSAF Conference*, (Washington, DC), Dec. 1997.

Keywords: deregulation; H.323; **Internet telephony**; POTS; price; quality

Simona Novi, "Global interoperable **Internet telephony -** challenges and opportunities," in *MSAF Conference*, (Washington, DC), Dec. 1997.

Keywords: deregulation; H.323; **Internet telephony**; POTS; price; quality

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Upcoming Events Related to Internet Telephony

Date	Place	Event
		Warning: MySQL Connection Failed: Access denied for user: ’iptel@localhost’ (Using password: YES) in **/usr/local/httpd/htdocs/www.iptel.org/info/events/events.php3** on line **28**

Glossary

(Try Acronym Definitions by Communications Standard Review if you do not find an acronym in this list. Other glossaries are also provided by Lucent at http://www.lucent.com/search/glossary/ and whatis.com .)

ALG	Application-Level-Gateway
AIN	Advanced Intelligent Networks
BOF	Birds of a Feather; IETF interest groups
CDR	Call Detail Record
CPL	Call Processing Language
DTMF	Dual Tone Multi-Frequency
ETSI	European Telecommunications Standards Institute
GLP	Gateway Location Protocol
GSM	Global System for Mobile communications
GSTN	Global Switched Telephone Network
gwloc	Gateway Location, also Gateway Discovery (IETF work in progress)
IETF	Internet Engineering Task Force
IN	Intelligent Networks
IP	Internet Protocol
IPDC	IP Device Control (family of protocols, IETF work in progress, see also MGCP)
IPT	Internet Telephony
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ITSP	Internet Telephony Service Providers
ITU	International Telecommunication Union

IVR	Interactive Voice Reponse
IXC	Long Distance Carrier
LAN	Local Area Network
LEC	Local Exchange Carrier
LNP	Local Number Portability
NAT	Network Address Translation
MGCP	merged SGCP and IPDC protocols (IETF work in progress)
NNI	Network-to-Network Interface (of a signaling interface)
non-repudiation	The inability of one entity involved in a communication to deny having participated in all or part of the communication.
OSP	Open Settlement Protocol (ETSI/Tiphon)
POTS	Plain Old Telephone System, Pretty Old Telephone System
PSTN	Public Switched Telephone Network
QoS	Quality of Service
R&D	Research and Development
RTCP	RTP Control Protocol
RTP	Real-Time Transport Protocol
RTSP	Real-Time Streaming Protocol
SAP	Session Announcement Protocol
SDP	Session Description Protocol
SGCP	Simple Gateway Control Protocol (IETF work in progress, see MGCP)
SIP	Session Initiation Protocol
SS7	Signaling System Nr. 7 (NNI telcos' signaling system)
Tiphon	Telecommunications and Internet Protocol Harmonization over Networks (ETSI project)
UNI	User-to-Network Interface (of a signaling interface)
VoIP	Voice over IP
VON	Voice over Net
VRU	Voice Response Unit

ITU-T Glossary

ADPCM	Adaptive Differential PCM (codec)
CELP	Codebook Excited Linear Predictive Coding (codec)
CSA-CELP	Conjugate-Structure-Algebraic-Celp (codec)
G.711	64 kbps PCM half-duplex codec (high quality, high bandwidth, minimum processor load)
G.723.1	6.4/5.3 kbps MP-MLQ codec (low quality, low bandwidth, high processor load due to the compression) *Note: A summary of ITU-T Speech / Audio Codecs is available at the PictureTel's site.*
G.726	40/32/24/16 ADPCM codec (good quality, medium bandwidth, low processor load) *Note: A summary of ITU-T Speech / Audio Codecs is available at the PictureTel's site.*
G.728	16 kbps LD-CELP codec (medium quality, medium bandwidth, very high processor load) *Note: A summary of ITU-T Speech / Audio Codecs is available at the PictureTel's site.*
G.729	8 kbps ACELP codec (medium quality, low bandwidth, high processor load) *Note: A summary of ITU-T Speech / Audio Codecs is available at the PictureTel's site.*
gatekeeper	an H.323 entity on the LAN which provides address translation and controls access to the LAN for H.323 terminals, gateways and MCUs
gateway	an endpoint on the LAN which provides for RT 2-way communications between H.323 Terminal on the LAN and other ITU terminals (ISDN, GSTN, ATM, ...) on WAN or to another H.323 gateway
H.225	protocols (RAS, RTP/RTCP, Q.931 call signaling) and message formats of the H.323 are covered in this standard
H.245	protocol for capability negotiation, messages for opening and closing channels for media streams, etc. (i.e. media signaling)
H.323	an umbrella standard for audio/video conferencing over unreliable networks; architecture and procedures are covered by this standard; H.323 relies on H.225 and H.245
LD-CELP	Low -Delay-CELP (codec)
MC	The Multipoint Controller provides the capability negotiation with all terminals taking part in a multipoint conference.
MCU	The Multipoint Control Unit is an endpoint on the LAN which provides the capability for 3 or more terminals and gateways to participate in a multipoint conference. The MCU consists of a mandatory MC and optional MPs.
MP	The Multipoint Processor provides for centralized processing (mixing, switching, ...) of audio, video, and/or data streams in a multipoint conference.

Q.931	ISDN call signaling protocol (in H.323 scenario this protocol is encapsulated in TCP and sent to the well known port 1720)
PCM	Pulse Code Modulation (codec)
RAS	Registration, Admission, Status - management protocol between terminals and gatekeepers
T.120	data conference protocol
