H.323 and Associated Protocols

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Abstract

H.323 is an ITU-T recommendation for multimedia conferencing over packet-based networks such as LANs and the Internet. H.323 is broad and comprehensive in its scope yet flexible and practical in its applicability. This report describes the components, protocols, and procedures in H.323; the challenges to the widespread adoption of H.323; the outlook for H.323 as a standard for multimedia conferencing; and key H.323 products and services.

See Also: Voice Over IP (Lecture by Prof. Jain) | Internet Telephony | Voice Over IP: Protocols and Standards | Voice Over IP: Products, Services, and Issues | Multimedia Networking Products | Voice Over IP: References | Books on Voice Over IP
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1. Introduction

Recommendation H.323 is a set of protocols for voice, video, and data conferencing over packet-based networks such as the Internet. The current recommendation, known as version 2.0, was ratified by the Study Group 16 (SG16) of the Telecommunications Sector of the International Telecommunication Union (ITU-T). The H.323 protocol stack is designed to operate above the transport layer of the underlying network. As such, H.323 can be used on top of any packet-based network transport like Ethernet, TCP/UDP/IP, ATM, and Frame Relay to provide real-time multimedia communication. H.323 uses the Internet Protocol (IP) for inter-network conferencing.

H.323 is one of several videoconferencing recommendations issued by ITU-T. The other recommendations in the series include H.310 for conferencing over broadband ISDN (B-ISDN), H.320 for conferencing over narrowband ISDN, H.321 for conferencing over ATM, H.322 for conferencing over LANs that provide a guaranteed quality of service, and H.324 for conferencing over public switched telephone networks (PSTN). The H.323 standard is designed to allow clients on H.323 networks to communicate with clients on the other videoconferencing networks.

This report provides a detailed overview of the components, protocols, and procedures defined in H.323. The technical challenges facing the widespread adoption of H.323, the current extensions and developments in H.323, and the future prospects of H.323 as a videoconferencing standard are also discussed.

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2. Scope of H.323

H.323 is a broad and flexible recommendation. As a minimum, H.323 specifies protocols for real-time point-to-point audio communication between two terminals on a packet-based network that do not provide a guaranteed quality of service. The scope of H.323, however, is much broader and encompasses inter-network multipoint conferencing among terminals that support not only audio but also video and data communication.

The scope of Recommendation H.323 can be summarized in the following broad categories:

- **Point-to-point and multipoint conferencing support**: H.323 conferences may be set up between two or more clients without any specialized multipoint control software or hardware. However, when a multipoint control unit (MCU) is used H.323 supports a flexible topology for multipoint conferences. A multipoint conference may be centralized where new participants can join all the others in the conference. This is the so-called hub-and-spoke topology. Or, a multipoint conference may be decentralized where new participants can elect to join one or more participants in the conference but not all. This approach will produce a flexible tree topology.
- **Inter-network interoperability**: H.323 clients are interoperable with switched-circuit network (SCN) conferencing clients such as those based on Recommendations H.320 (ISDN), H.321 (ATM), and H.324 (PSTN/Wireless).
- **Heterogeneous client capabilities**: A H.323 client must support audio communication; video and data support is optional. This heterogeneity and flexibility does not make the clients incompatible. During call set-up capabilities are exchanged and communication established based on the lowest common denominator.
- **Audio and video codecs**: H.323 specifies a required audio and video codec. However, there is no restriction on the use of other codecs and two clients can agree on any codec which is supported by both of them.
- **Management and accounting support**: H.323 calls can be restricted on a network based on the number of calls already in progress, bandwidth limitations, or time restrictions. Using these policies the network manager can manage H.323 traffic. Further, H.323 also provides accounting facilities that can be used for billing purposes.
- **Security**: H.323 provides authentication, integrity, privacy, and non-repudiation support.
- **Supplementary services**: Recommendation H.323 recognizes the huge potential for applications based on IP telephony and multimedia. It provides a basic framework for development of such services. In version 2.0 of H.323, two services -- call transfer and call forwarding -- have been specified.

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3. Why is H.323 Important?

Trend: The explosive growth of the Internet and the almost universal deployment of corporate LANs have made packet-based networks ubiquitous. It is therefore natural for individuals and enterprises to use this resource for audio and video communication to offset some of the tariffs of public switched telephone networks (PSTN). Multimedia over packet-based networks (primarily IP networks) has grown rapidly in the last few years. Industry research put the growth at 37% annually and is expected to reach $39 billion by the year 2002 (http://www.imtc.org/faq.htm). In a similar forecast, Probe Research estimates that by the year 2002 18.5% of all U.S. phone traffic will be carried over data networks. This rapid expansion and potential underlies the importance of an enabling and unifying standard such as H.323.

Standardization: In the 1995-1997 time period, several vendors developed products and services to cater for the emerging IP telephony market. These products and services, however, were based on proprietary protocols that prevented widespread interoperability. H.323 is a standard protocol that has been widely accepted. This will promote greater awareness, availability, and acceptability of multimedia conferencing over packet-based networks.

Internetworking: H.323 bridges multimedia communications between packet-based and switched-circuit networks SCN). Existing clients based on SCN conferencing standards like H.320 (ISDN), H.321 (ATM), and H.324 (PSTN) can inter-operate with H.323 clients. For example, it is possible to call from a H.323 client to a regular telephone on a PSTN. At the corporate level this internetworking capability allows enterprises to migrate voice and video from existing networks to their data network.

Integrated services: H.323 makes possible the development of additional services such as e-mail, voice mail, fax, call center functionality, and videoconferencing in an integrated environment. For example, an e-commerce business can provide a direct voice link from their web site to a sales representative to answer customers' questions. A few services have been standardized in H.450.x (e.g. call transfer, call forwarding). Others will be added later.

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4. H.323 Architecture: Components

Recommendation H.323 specifies components, protocols, and procedures for real-time point-to-point and multipoint multimedia communication over packet-based networks. It also sets interoperability guidelines for communication between H.323-enabled networks and the H.32X-based family of conferencing standards. Figure 1 shows a general layout of a H.323 enabled inter-network with all the necessary components.

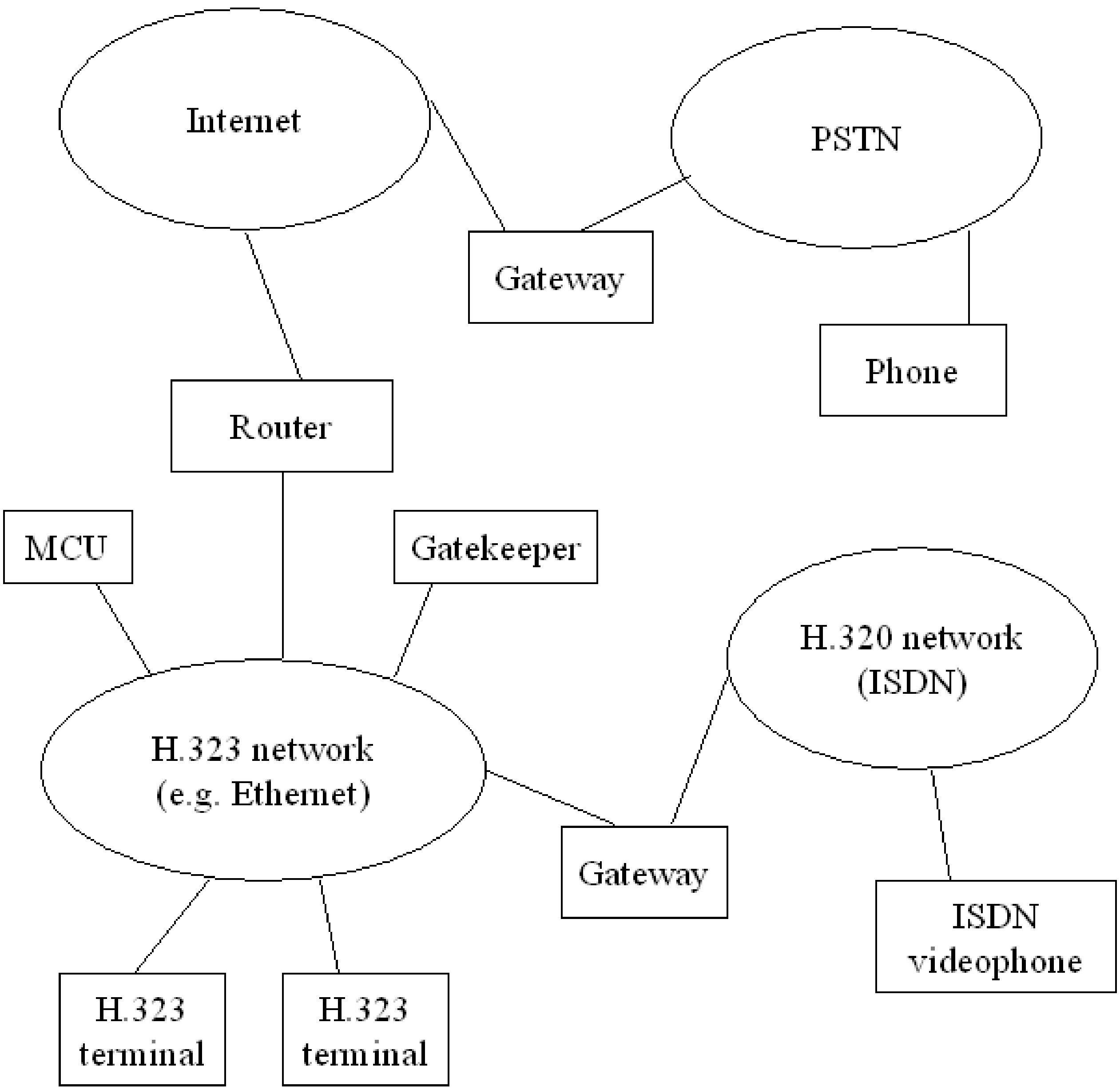

Figure 1. Layout of H.323-enabled inter-network

In a general H.323 implementation four logical entities or components are required. These are terminals, gateways, gatekeepers, and multipoint control units (MCU). Terminals, gateways, and MCUs are collectively known as endpoints. Even though an H.323-enabled network can be established with only terminals the other components are essential to provide greater practical usefulness of the services.

4.1 Terminal

A terminal, or a client, is an endpoint where H.323 data streams and signaling originate and terminate. It may be a multimedia PC with a H.323 compliant stack or a standalone device such as a USB (universal serial bus) IP telephone. A terminal must support audio communication; video and data communication support is optional.

4.2 Gateway

A gateway is an optional component in a H.323-enabled network. However, when communication is required between different networks a gateway is needed at the interface. Through the provision of gateways in H.323 it is possible for H.323 terminals to inter-operate with other H.32X compliant conferencing terminals. For example, it is possible for a H.323 terminal to set up a conference with terminals based on H.320 or H.324 through an appropriate gateway. A gateway provides data format translation, control signaling translation, audio and video codec translation, and call setup and termination functionality on both sides of the network. Depending on the type of network to which translation is required a gateway may support H.310, H.320, H.321, H.322, or H.324 endpoints.

4.3 Gatekeeper

A gatekeeper is a very useful, but optional, component of an H.323-enabled network. Gatekeepers are needed to ensure reliable, commercially feasible communications. A gatekeeper is often referred to as the brain of the H.323 enabled network because of the central management and control services it provides. When a gatekeeper exists all endpoints (terminals, gateways, and MCUs) must be registered with it. Registered endpoints' control messages are routed through the gatekeeper. The gatekeeper and the endpoints it administers form a management zone.

A gatekeeper provides several services to all endpoints in its zone. These services include [Trillium]:

- **Address translation**: A gatekeeper maintains a database for translation between aliases, such as international phone numbers, and network addresses.
- **Admission and access control of endpoints**: This control can be based on bandwidth availability, limitations on the number of simultaneous H.323 calls, or the registration privileges of endpoints.
- **Bandwidth management**: Network administrators can manage bandwidth by specifying limitations on the number of simultaneous calls and by limiting authorization of specific terminals to place calls at specified times.
- **Routing capability**: A gatekeeper can route all calls originating or terminating in its zone. This capability provides numerous advantages. First, accounting information of calls can be maintained for billing and security purposes. Second, a gatekeeper can re-route a call to an appropriate gateway based on bandwidth availability. Third, re-routing can be used to develop advanced services such as mobile addressing, call forwarding, and voice mail diversion.

4.4 Multipoint Control Unit (MCU)

A multipoint control unit enables conferencing between three or more endpoints. It consists of a mandatory multipoint controller (MC) and zero or more multipoint processors (MP). Although the MCU is a separate logical unit it may be combined into a terminal, gateway, or gatekeeper. The MCU is an optional component of an H.323-enabled network.

The multipoint controller provides a centralized location for multipoint call setup. Call and control signaling are routed through the MC so that endpoints capabilities can be determined and communication parameters negotiated. A MC may also be used in a point-to-point call which can later be extended into a multipoint conference. Another useful job of the MC is to determine whether to unicast or multicast the audio and video streams depending on the capability of the underlying network and the topology of the multipoint conference. The multipoint processor handles the mixing, switching, and processing of the audio, video, and data streams among the conference endpoints.

The MCU is required in a centralized multipoint conference where each terminal establishes a point-to-point connection with the MCU. The MCU determines the capabilities of each terminal and sends each a mixed media stream. In the decentralized model of multipoint conferencing, a MC ensures communication compatibility but the media streams are multicast and the mixing is performed at each terminal.

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5. H.323 Architecture: Protocols and Procedures

Recommendation H.323 is an umbrella recommendation that depends on several other standards and recommendations to enable real-time multimedia communications. Table 1 list the key recommendations referenced in H.323.

Table 1. Key standards referenced in H.323

Recommendation #	**Title (issue date)**
Audio Codecs	
G.711	Pulse code modulation of voice frequencies (11/88)
G.722	7 kHz audio coding within 64 kb/s (11/88)

G.723.1	Dual rate speech coders for multimedia communication transmitting at 5.3 and 6.3 kb/s (03/96)
G.728	Coding of speech at 16 kb/s using low-delay code excited linear prediction (09/92)
G.729	Coding of speech at 8 kb/s using conjugate-structure algebraic-code-excite linear-prediction (03/96)
Video Codecs	
H.261	Video codecs for audiovisual services at p x 64 kb/s (03/93)
H.263	Video coding for low bit rate communication (02/98)
Data Conferencing	
T.120	Data protocols for multimedia conferencing (07/96)
Control	
H.245	Control protocol for multimedia communication (09/98)
H.225.0	Call signaling protocols and media stream packetization for packet-based multimedia communication systems (02/98)
Real-time Transport	
RTP/RTCP	RFC1889 (01/96), RFC1890 (01/96), RFC2032 (10/96), RFC2429 (10/98)
Security	
H.235	Security and encryption for H-Series (H.323 and other H.245-based) multimedia terminals (02/98)
Supplementary Services	
H.450.1	Generic functional protocol for the support of supplementary services in H.323 (02/98)
H.450.2 and 450.3	Call transfer and call diversion supplementary services for H.323 (02/98)

This section describes these protocols in detail with particular reference to the H.323 terminal protocol stack. A typical call scenario between two terminals is also described.

5.1 H.323 Terminal Protocol Stack

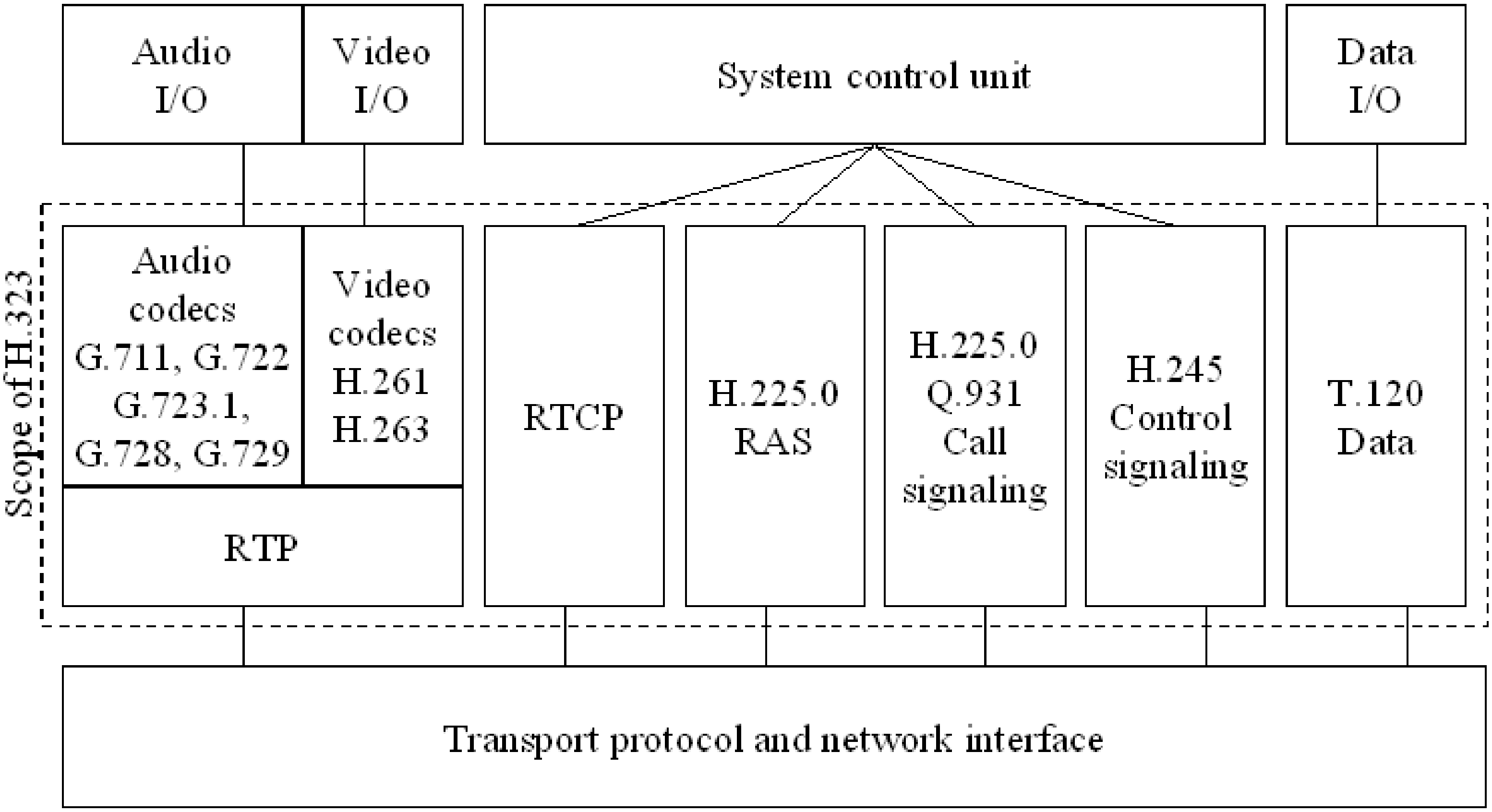

Figure 2. H.323 terminal protocol stack

Some of the key features and concepts of the H.323 protocol stack are discussed below:

- The scope of H.323 does not include the audio/video capture equipment. It is assumed that the audio and video digital streams are available to the H.323 terminal for processing.
- The real-time transport protocol (RTP) and the associated control protocol -- real-time control protocol (RTCP) -- is employed for timely and orderly delivery of audio and video streams. RTP/RTCP is an Internet Engineering Task Force (IETF) recommendation that provides logical framing, sequence numbering, timestamping, payload distinction (e.g. between audio and video and between different codecs), and source identification. It may also provide basic error detection and correction. Note that the RTP layer is above the transport layer of the underlying network.
- The H.323 protocol stack runs on top of the transport and network layers. If the underlying network is IP-based (which is the most common network) then the audio, video, and H.225.0 RAS packets use the unreliable UDP (user datagram protocol) for transport while the data and control (H.245 and H.225.0 call signaling) packets are transported using the reliable TCP (transmission control protocol).

5.2 Audio Codecs

H.323 specifies a series of audio codecs ranging in bit rates from 5.3-64 kb/s. The mandatory codec is G.711 which uses pulse code modulation to produce bit rates of 56 and 64 kb/s. G.711 is a popular codec designed for telephone networks. However, it is less appropriate for communication over the Internet where subscriber loop bandwidths are much smaller. Nowadays, most H.323 terminals support G.723.1 which is much more efficient and produces good quality audio at 5.3 kb/s and 6.3 kb/s. The G.728 and G.729 codecs use advanced linear prediction quantization of digital audio to produce high quality audio at 16 kb/s and 8 kb/s, respectively.

5.3 Video Codecs

Video communication is bandwidth intensive and bursty in nature. Therefore, efficient compression and decompression techniques are essential for good performance. Recommendation H.323 specifies two video codecs: H.261 and H.263. However, H.323 clients are not limited to these codecs only. Other codecs can be used provided both terminals agree on and support it. Video support in H.323 terminals and MCUs is optional.

The H.261 codec produces video transmission for channels with bandwidths p x 64 kb/s where p can range from 1 to 30. The discrete cosine transform (DCT) is used for compression together with quantization and motion compensation. H.261 supports two video formats. The common intermediate format (CIF) has a resolution of 352 x 288 pixels while the quarter common intermediate format (QCIF) has a resolution of 176 x 144 pixels. The CIF format support is optional.

The H.263 codec is designed for low bit rate transmission without loss of quality. It uses the same DCT coding with quantization for compression but this is accompanied by both motion estimation and prediction. Additional coding efficiency parameters have also been defined which can be negotiated between the terminals. The result is better quality at a lower bit rate. The video formats supported by H.263 are: sub-QCIF (128 x 96), QCIF (176 x 144), CIF (352 x 244), 4CIF (702 x 576), and 16CIF (1408 x 1152). The first three are required while the remaining two are optional. Through the QCIF format H.263 is compatible with H.261.

The quality of video transmission strongly depends on compression techniques. Active work is on-going in the development of more efficient codecs like MPEG-4 and MPEG-7. The architecture of H.323 is designed to allow the incorporation of new codecs as they become available.

5.4 Data Conferencing

Real-time data conferencing capability is required for activities such as application sharing, whiteboard sharing, file transfer, fax transmission, and instant messaging. Recommendation T.120 provides this optional capability to H.323.

T.120 is a real-time data communication protocol designed specifically for conferencing needs. Like H.323, Recommendation T.120 is an umbrella for a set of standards that enable the real-time sharing of specific applications data among several clients across different networks. T.120 provides several advantages over regular data transmission such as [DataBeamB]:

- **Multipoint conferencing support**: T.120 supports multipoint data delivery which enables group collaboration activities. The MCU handles the mixing and switching of data in a similar manner to that used for video and audio.
- **Network and platform independence**: T.120 operates on top of the transport layer of the underlying network. As such, it is transparent and independent of the network hardware and software.
- **Interoperability**: T.120 is referenced in all the H.32X conferencing standards. This cross referencing, together with the network and platform independence, ensures a high degree of interoperability at the application level.
- **Multicast support**: T.120 supports multicast of data streams in multicast-capable networks. This support is flexible with mixed unicast and multicast also possible during a conference.
- **Other benefits**: T.120 provides error correction capability on top of the network transport ensuring reliable delivery. In general, T.120 has a scalable and extendible architecture with provisions for the addition of new applications that take advantage of real-time reliable and efficient data delivery among a group of collaborators.

T.120 Protocol Stack

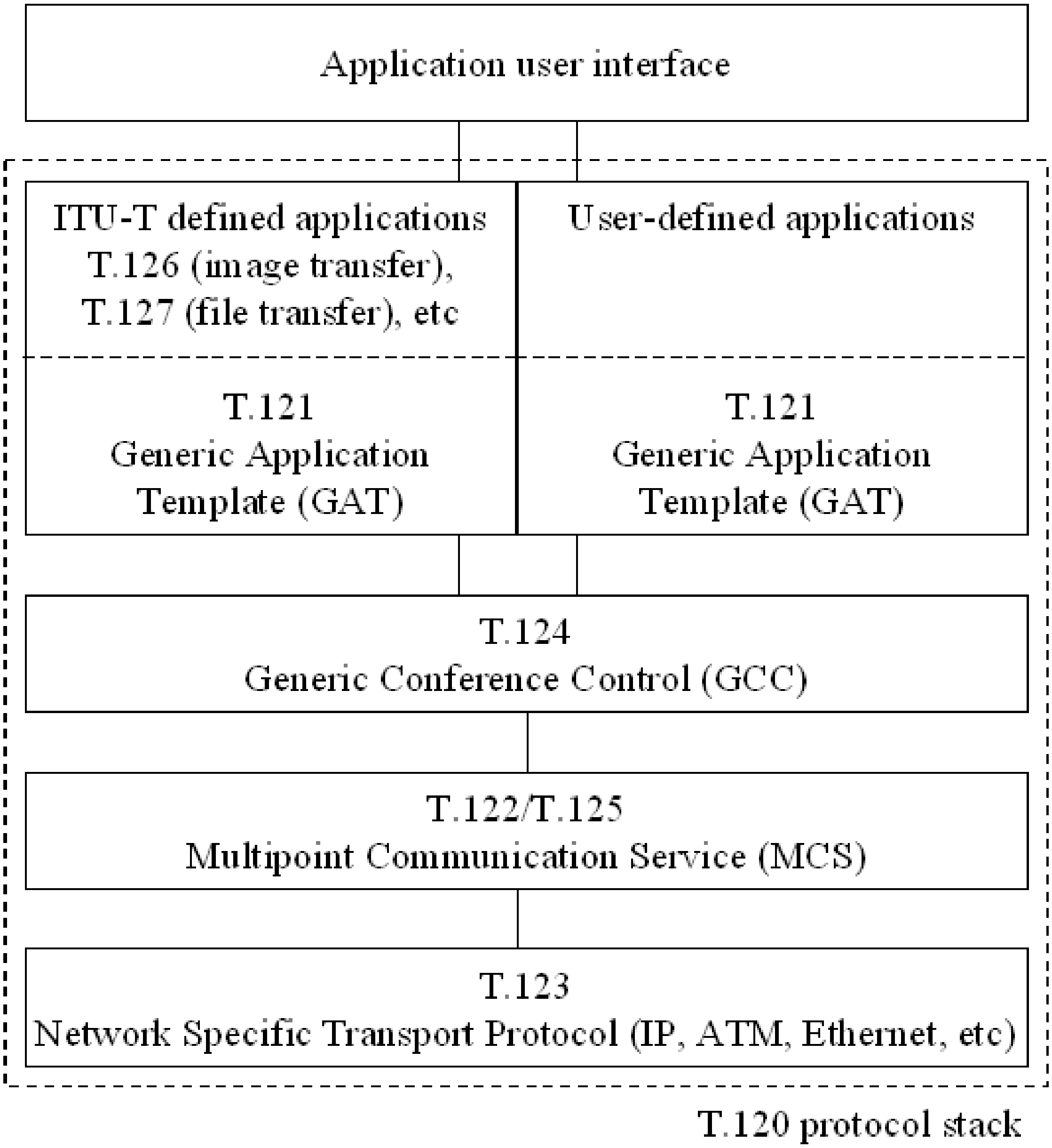

Figure 3. T.120 protocol stack

The T.120 protocol stack, shown in Figure 3, is based on a layered architecture with more specific protocols on higher layers than on lower layers. The three lower layers provide common transport functionality to data conferencing applications defined in the top layer.

T.123 is a transport protocol for data conferencing modeled after the Open Systems Interconnection (OSI) transport layer (see ITU-T X-series recommendations). The implementation of this layer is adapted to the capabilities of the underlying network transport such as ATM or IP. T.123 also provides error detection and correction support to T.120 communications.

T.122 and T.125 define the multipoint communication service layer (MCS). T.122 specifies the services while T.125 defines the implementing protocol. The primary purpose of MCS is the coordination and prioritization of data among multiple conference participants.

T.124 is the generic conference control (GCC) protocol. This protocol facilitates the creation and management of conferences with features such as conference creation, terminal entry and exit from a conference, authentication security, service resource management, and information management.

T.121 specifies a template for the development of data conferencing applications. Following this development template or blueprint ensures compatible applications that effectively utilizes the GCC and MCS. Use of the template is mandatory for the standard applications and is highly recommendation for user-defined applications. The standard ITU-T defined applications include T.126 for still image exchange and annotation (e.g. a whiteboard application) and T.127 for multipoint binary file transfer. Additional applications can be developed by end users.

5.5 Control and Signaling Mechanisms

The flow of information in a H.323-enabled network consists of a mix of audio, video, data, and control packets. Control information is essential for call setup and tear down, capability exchange and negotiation, and administrative purposes. H.323 uses three control protocols: H.245 media control, H.225/Q.931 call signaling, and H.225.0 RAS.

H.225.0 Call Signaling

Call signaling is a basic requirement needed to set up and tear down a call between two endpoints. H.225.0 uses a subset of Q.931 signaling protocol for this purpose. Q.931 was initially developed for signaling in integrated services digital networks (ISDN). H.225.0 adopts Q.931 signaling by incorporating it in its message format. H.225.0 call signaling is sent directly between the endpoints when no gatekeeper exist. When a gatekeeper exists then it may be routed through the gatekeeper. The exchange of messages needed during a basic H.323 call are described in detail in the next section.

H.245 Media Control

The flexibility of H.323 requires that endpoints negotiate to determine compatible settings before audio, video, and/or data communication links can be established. H.245 uses control messages and commands that are exchanged during the call to inform and instruct. The implementation of H.245 control is mandatory in all endpoints.

H.245 provides the following media control functionalities:

- **Capability exchange**: H.323 allows endpoints to have different receive and send capabilities. Each endpoint records its receiving and sending capabilities (e.g. media types, codecs, bit rates, etc) in a message and sends it to the other endpoint(s).
- **Opening and closing of logical channels**: H.323 audio and video logical channels are uni-directional end-to-end links (or multipoint links in the case of multipoint conferencing). Data channels are bi-directional. A separate channel is needed for audio, video, and data communication. H.245 messages control the opening and closing of such channels. H.245 control messages use logical channel 0 which is always open.
- **Flow control messages**: These messages provide feedback to the endpoints when communication problems are encountered.
- **Other commands and messages**: Several other commands and messages may be used during a call like a command to set the codec at the receiving endpoint when the sending endpoint switches its codec.

H.245 control messages may also be routed through a gatekeeper if one exists.

H.225.0 RAS

H.225.0 RAS (registration, admission, status) messages define communications between endpoints and a gatekeeper. H.225.0 RAS is only needed when a gatekeeper exists. Unlike H.225.0 call signaling and H.245, H.225.0 RAS uses unreliable transport for delivery. In an IP network H.225.0 RAS uses UDP.

H.225.0 RAS communications include:

- **Gatekeeper discovery**: Gatekeeper discovery is used by endpoints to find out their gatekeeper. An endpoint who needs to find the transport address of its gatekeeper(s) will multicast a gatekeeper request (GRQ) message. One or more gatekeepers may reply with a GCF message containing the gatekeeper transport address.
- **Endpoint registration**: Once a gatekeeper exists all endpoints must be registered with it. This is necessary because gatekeepers need to know the aliases and transport addresses of all endpoints in its zone to route calls.
- **Endpoint location**: Gatekeepers use this message to locate endpoints with a specific transport address. This process is required, for example, when the gatekeeper updates its alias-transport address database.
- **Other communications**: A gatekeeper performs many other management and control duties such as admission control, status determination, and bandwidth management which are all handled through H.225.0 RAS messages.

5.6 H.323 Call Setup and Tear Down

Table 2 shows the control messages that are exchange between terminals [Toga98] A and B from call setup to call termination. Terminal A initiates the call to B directly without any intermediate gateway or gatekeeper. The shaded messages (messages 4-11) are H.245 messages while the rest are H.225.0 call signaling messages. Some of the messages can be overlapped even though they are shown in sequence in the table.

Table 2. Message exchange between terminals A and B during a call

Message	Terminal A	Terminal B
1	Setup	
2		Alerting
3		Connect
4	termCapSet	
5		termCapAck
4		termCapSet
5	termCapAck	
6	masterSlvDet	
7		masterSlvDetAck
8	masterSlvDetConfirm	
9	openReq	
10		openAck
9		openReq
10	openAck	
11	endSession	
11		endSession
12	Release	

Terminal A initiates the call by sending the *Setup* message to B. Terminal B replies with messages *Alerting* and *Connect* to indicate it is ready for the call.

The call signaling is followed by H.245 capability exchange messages. Each terminal sends a *termCapSet* message to communicate its media settings to the other terminal. The terminals then acknowledge each other's settings by the *termCapSetAck* messages. Next a master and slave terminal is determined by the *masterSlvDet* and *masterSlvDetAck* messages. The master/slave distinction is necessary to avoid conflicts in situations such as the opening of a bi-directional channel for communication. The master (terminal A) leads the opening of a logical channel using the *openReq* message. Terminal B follows by opening a logical channel in the other direction. The terminals can open as many channels as is practically possible.

The call is terminated after the exchange of H.245 *endSession* messages with the H.225.0 *Release* message.

5.7 Supplementary Services

Recommendation H.323 bridges traditional telephone networks with media rich packet-based networks . There is a huge potential for new services and applications that take advantage of the capabilities of both networks. These services can range from value-added traditional telephone services such as call transfer and diversion to new services such as integrated messaging (e-mail, voice mail, fax, instant messaging, etc). H.323 provides a flexible architecture for supplementary services through the H.450.x series of recommendations.

H.450 adopts a hierarchical architecture for the development of new services. A general framework for supplementary services is defined in H.450.1. Several basic services are specified in H.450.2 and above. New services can be developed by end-users by combining zero or more of the basic services. However, all services must use the control mechanisms defined in H.450.1.

H.450.1 provides an essential mechanism for end-to-end control signaling between peer service entities. The H.450.1 protocol is based on the QSIG protocol developed by International Organization for Standardization (ISO) for private ISDN networks. QSIG is the most common services control mechanism employed in call centers and PBXs (private branch exchange). Using QSIG as a basis for H.323 supplementary services provides several advantages such as:

- Interoperability with QSIG based networks
- Existence of several basic services models (from ISDN)
- Extendibility and flexibility of QSIG
- Existence of implementation knowledge base

Two supplementary services have been ratified by ITU-T. These are call transfer (H.450.2) and call diversion (H.450.3). Other services that are being developed include call hold (H.450.4), call park/pickup (H.450.5), call waiting (H.450.6), message waiting (H.450.7), name identification (H.450.8), and call completion on busy subscriber (H.450.9).

5.8 Security

Recommendation H.235 specifies the security requirements for H.323 communications. Four security services are provided: authentication, integrity, privacy, and non-repudiation.

Authentication is provided by admission control of endpoints. This is handled by the gatekeeper that administers the zone. Data integrity and privacy is provided by encryption. Non-repudiation ensures that no endpoint can deny that it participated in a call. This is also provided by gatekeeper services.

To implement these security service H.235 can use existing standards such as IP Security (IPSec) and Transport Layer Security (TLS).

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6. Challenges

The H.323 recommendation for multimedia conferencing was first ratified in late 1996. It therefore has a head start and is also widely supported by vendors. However, it has not made any significant inroads into the traditional telephony market. There are several reasons for this, both social and technical. In the following sections some of the technical challenges facing H.323 are discussed.

6.1 Interoperability and Implementation Issues

H.323 is a large, complex, and flexible standard. Different interpretations and oversight in implementations have led to non-compliant and incompatible devices. This issue is significant and considered to be the primary impediment to the wider deployment of H.323 enabled products and services in the enterprise, small business, and residential markets.

The Problem

The complexity and the flexibility of H.323 make its implementation difficult and prone to errors and omissions. Vendors of H.323 products and services often choose to implement a subset of H.323 that meets their immediate requirements. Furthermore, ITU-T does not provide an implementation guideline that can help ensure compliance and interoperability.

All endpoints must inter-operate before a H.323 multimedia call can be established. Consider, for example, a phone-to-phone call over the Internet. This setup implies that the phones (terminals) are interoperable, the gateways are interoperable, and the phones and their respective gateways are interoperable. If the endpoints are manufactured by different vendors there is a good possibility that problems may occur.

The implementations of codecs is well-developed. The issue that breaks interoperability is the capability exchange and signaling process between endpoints which is often not implemented completely by vendors. For example, the latest version of NetMeeting (Version 3.01) still does not implement gatekeeper bandwidth management requests. An implementation note for compliant H.323 development can be found at http://web2.airmail.net/plong/h323impl.html.

Solution Efforts

The poor interoperability of H.323 endpoints is widely recognized. The International Multimedia Teleconferencing Consortium (IMTC) was set up with the primary goal of ensuring that vendors' products and services are interoperable. IMTC is a non-profit organization with over 150 members that regularly conducts interoperability and compliance testing of products and services.

In 1998, ITXC, Lucent Technologies, and VocalTec established the iNOW! Profile as a set of implementation guideline for gateway-to-gateway and terminal-to-gateway interoperability. Recently, the iNOW! Profile implementation initiative has been turned over to an IMTC activity group (http://www.imtc.org/act_inow.htm). The iNOW! Profile has also been extended to include interoperability between gatekeepers and other endpoint. Compliant products and services are certified by IMTC.

Several software vendors have developed H.323 protocol stacks for the major operating systems. These software modules allow developers of H.323 compliant products and services to work with a higher level application programming interface (API) rather than the lower level implementation details, thus minimizing the risk of interoperability problems. Vendors that offer H.323 protocol stacks include:

- DataBeam
- Trillium
- Hughes Software Systems
- Elemedia
- RADVision
- OpenH323 Project: Open source H.323 protocol stacks for Windows and Linux.

Many hardware and software companies have set up their own interoperability labs to test their and other vendors products and services. NetMeeting, a H.323 compliant PC software from Microsoft, uses DataBeam's H.323 protocol stack.

6.2 Lack of Value Added Services

The vision of H.323 is global interoperability between packet and circuit switched networks. H.323 also promises new and integrated services of value to customers currently using circuit switching technologies exclusively. These goals have not yet been achieved. From a business point of view a technology cannot succeed solely on the basis of lower operational costs. It must also provide additional value and performance in the form of ease of use and improved feature set for it to supplant a well-entrenched technology such as PSTN.

Internet telephony service providers (ITSP) and Internet service providers (ISP) were expected to provide this level of interoperability with better services and value as compared to plain old telephone service (POTS). Several ITSPs exist today (e.g. BizTrans) that do provide good value in certain regions such as North America and Europe. Global interoperability is still a problem. Furthermore, the features and quality of service being offered is often inferior to POTS. Two key factors have hampered ITSPs and ISPs efforts. The first is the lack of interoperability of endpoints (especially gateways) from different vendors. And the second is the poor scalability of H.323 communications.

ITXC, the Internet telephony exchange carrier, has the largest global presence of H.323 gateways. As such, it is well positioned to offer services to ITSPs, ISPs, and traditional telephone companies to route their multimedia traffic over its packet-based network.

Net2Phone, the leader in IP telephony in terms of total call time, has recently inked a deal with America Online (AOL) to bring Internet telephony services to AOL customers [News.com]. It had earlier signed similar service deals with Priceline.com and ICQ.

6.3 Quality and Security Issues

The quality of video on public networks like the Internet is poor. Even audio over the Internet does not compare favorably to the PSTN. However, H.323 is a higher layer protocol and can use any QoS that is built into lower layers. For example, in IP networks H.323 can make use of IntServ/RSVP and DiffServ. Development of more efficient codecs will also play an important role in improving quality. Enterprise IT managers are also reluctant to deploy H.323 enabled voice and data integrated solutions because of the generally less reliable nature of packet-based networks as compared to PSTN.

Security of H.323 communications is also a big issue. Concerns over privacy has prevented businesses from moving over to H.323 for intra- and inter-office communications. Part of the problem was resolved with H.323 Version 2.0 when H.235 was incorporated as a security recommendation. However, many products have not implemented the security recommendation.

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7. Future Prospects

In this section several key developments and trends that are shaping H.323's future are discussed.

7.1 H.323 or SIP?

This is the question IP telephony vendors are asking. The H.323 recommendation has been around since 1996 but has failed to capture the market in the way it was predicted to. The Internet Engineering Task Force (IETF) is working on parallel standards for IP telephony. The Session Initiation Protocol (SIP), an application level protocol defined in RFC2543 for establishing multimedia communications, has gained momentum. SIP proponents cite the following as advantages of SIP over H.323:

- **IP-based**: IP is the dominent protocol both at the edges and in the core of the Internet. As a result, interoperability with ATM and ISDN is not an issue. H.323 carries a lot of extra baggage to make sure that it is interoperable with the other standards in the series. SIP is free of this extra rarely-needed luggage.
- **Less complex**: SIP is a much smaller and less complicated standard that is based on the architecture of existing popular protocols such HTTP and FTP. On the other hand, H.323 is large and complicated. As a result, H.323 products and services are more expensive to develop.
- **Easy to decode/debug**: SIP uses a simple format for commands and messages. These are text strings that are easy to decode, and hence, easy to debug. The entire set of messages is also much smaller than in H.323.
- **Client/server architecture**: SIP messages are exchanged between a client and a server like HTTP messages. This client server operation mode allows security and management features to be implemented easily in SIP when compared to H.323 calls.
- **Easier firewall/proxy design and configuration**: SIP commands can easily be proxied and firewalls can be designed to allow/disallow SIP communications. Getting H.323 through firewalls and proxies is much more complicated.
- **Extendible and scalable**: Because SIP is based on a client/server distributed architecture it is more scalable than H.323 which often requires peer-to-peer communications. Extending SIP is also easier because of its simpler message format and greater experience with similar protocols such as HTTP.

Analysts often make an analogy between H.323 and ATM as standards that provide too much too soon; the market was not ready for them.

Although the share of SIP will increase H.323 will also grow as most of its interoperability problems have been addressed. Moreover, the investment in H.323 by vendors and customers alike will prevent wholesale migration to SIP. SIP and H.323 will coexist in products and services for several years to come.

7.2 Trends in the Enterprise Market

The greatest growth in H.323 usage is likely to occur in enterprises where integration of packet-based and switched-circuit networks will reduce maintenance and operation costs. Integration and development of new services is also an advantage. Furthermore, H.323 provides a migration path for enterprises from switched-circuit to packet-based multimedia conferencing.

7.3 Trends in the Home and Small Business Markets

Small business and home users put more emphasis on quality, value, and features. They are less concerned with interoperability with legacy systems and security. This is the market where H.323 had the least significant impact. Adoption will increase in the coming years as the price-performance ratio improves. This is also the market where SIP can make a significant impact.

7.4 Future Developments in H.323

The H.323 standard is still under development. Version 2.0 was issued in January 1998 and work is continuing on adding new features. The focus has shifted from LANs to larger networks such as the Internet and the challenges associated with it. Work is also being done to add mobile communication capability to H.323 and its interoperability with GSM, a wireless communication standard. Another initiative is the preparation of an implementation guide for H.323 to ease interoperability concerns. Interoperability issues and security are on-going initiatives. For documents on current developments in H.323 see PictureTel's FTP site ftp://standard.pictel.com/avc-site/ [Pictel].

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8. Products and Services

An analysis of H.323 compliant terminals, gateways, and gatekeepers is presented in a separate report available on this Web site (VoIP: Products, Services, and Issues). An extensive listing of vendors and their products is also maintained at IPTelephony.org's Web site. This section lists and briefly describes currently available multipoint control units (MCU) and key Internet telephony service providers (ITSP).

8.1 MCUs

Multipoint control units (MCU) are needed when dozens of endpoints participate in a multimedia conference like in a virtual corporate meeting. There are two different configurations or architectures in which MCUs are available: server-based or rack mounted. Server-based MCUs are built on a server platform (like a PC running Window NT) with appropriate software and add-in hardware boards. Rack-based MCUs use proprietary hardware and software to provide the necessary functionality.

In the following paragraphs key features of five currently available MCUs are described. For a detailed comparative review of these products see [Brown99].

Ezenia! Encounter NetServer

- Server-based solution running under Windows NT 4.0 and using add-in media streams processing card.
- Supports up to 32 H.323 or 48 T.120 endpoints
- Supports H.261, H.263, G.711, G.723.1, G.728, T.120
- H.323 version 1.0 compliant
- Allows mixed endpoint codec capability in the same conference
- Uses SNMP for monitoring
- Browser-based interface for administrative tasks such scheduling, setup, control, and management of conferences
- Built-in gatekeeper but can also work with external gatekeeper
- Ezenia! was formerly known as VideoServer

WhitePoint Software MeetingPoint 4.0

- Server-based solution for Windows NT 4.0 and Sun Solaris. Does not use any special hardware accelerators
- Supports H.261, H.263, G.711, G.723.1, T.120
- H.323 version 2.0/1.0 compliant
- Does not support codec translations for mixed mode conferences
- Built-in gatekeeper for H.323 call management
- Browser and Java based setup, management, and control of conferences

PictureTel 330 NetConference Multipoint Server Software

- Server-based software only solution for Windows NT 4.0
- Supports H.261, H.263, G.711, G.722, G.723, T.120
- H.323 version 2.0/1.0 compliant
- Does not support mixed mode conferences
- Browser and Java based setup, control, and management of conferences
- Can use an external gatekeeper

Lucent Multimedia Communications Exchange (MMCX)

- Rack mounted solution
- Supports H.261, G.711, T.120
- H.323 version 1.0 compliant
- Does not support mixed mode conferences
- Support mixed conferencing between H.323 and H.320 endpoints
- Cannot use an external gatekeeper
- Browser-based conference setup and management. Cannot schedule conferences

RADVision MCU-323

- Rack mounted solution

- Supports H.261, G.711
- H.323 version 1.0 compliant
- Supports conferencing between H.320 and H.323 endpoints
- Does not support mixed mode conferences
- Built-in gatekeeper but can also use external gatekeeper
- Browser-based monitoring, control, and management. No scheduling support

8.2 ITSPs

Internet telephony service providers (ITSP) maintain gateways from PSTN to the Internet and provide low-cost telephony service to traditional phones and fax machines. Depending on the proximity of the gateway to the PSTN where a call is placed the savings can be significant. For example, Net2Phone provides a rate of 3.9c per minute for calls within the U.S. and 8c per minute for calls to most areas in Europe. Note that Net2Phone services are based on proprietary software that is NOT H.323 compliant.

The common services provided by ITSPs are PC-to-phone, phone-to-phone, and PC-to-fax. Flat rate charges are often adopted. Key H.323 compliant ITSPs that have points of presence (PoP) in the U.S. and worldwide are listed below:

- Access Power: PC-to-phone, phone-to-phone
- BizTrans: PC-to-phone, phone-to-phone, fax
- CoolCall.com: Phone-to-phone
- Cyberfax: Fax
- Delta Three:PC-to-phone, phone-to-phone
- Exicom: PC-to-phone, phone-to-phone
- Eurocall: PC-to-phone, phone-to-phone
- Global Exchange Carrier: Phone-to-phone, fax
- GlobalNet Telecommunications: PC-to-phone, phone-to-phone, fax
- Innofone.com: PC-to-phone, phone-to-phone
- TeleMatrix: Phone-to-phone
- WorldWideTalk: PC-to-phone, phone-to-phone

An extensive list of ITSPs is maintained at the following two Web sites:

- IPTelephony.org
- VocalTec

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9. Summary

Recommendation H.323 is a set of protocols issued by ITU-T for audio, video, and data conferencing over packet-based networks such as the Internet. It is a broad and flexible specification that caters to the needs of both enterprises and individuals. The goal of H.323 is the widespread interoperability of multimedia communication devices irrespective of the type of network it is connected to. H.323 is facing a serious challenge from IETF's SIP especially in the lucrative small business IP telephony market. However, H.323 will continue to play a significant role in the videoconferencing market. Work is on-going in improving the H.323 standard.

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Both these sites have extensive links to tutorials, standards, news, vendors, and products

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List of Acronyms

CIF: Common intermediate format

Codec: Compression/decompression

DCT: Discrete cosine transform

GCC: Generic conference control

IETF: Internet Engineering Task Force

IP: Internet protocol

ISDN: Integrated services digital network

ISO: International Organization for Standardization

ITSP: Internet telephony service provider

ITU-T: International Telecommunication Union - Telecommunications Sector

MCU: Multipoint control unit

MCS: Multipoint communication service

OSI: Open systems interconnection

PBX: Private branch exchange

POTS: Plain old telephone service

PSTN: Public switched telephone network

QCIF: Quarter common intermediate format

RAS: Registration, admission, status

RFC: Request for comments

RTCP: Real-time transport control protocol

RTP: Real-time transport protocol

SCN: Switched circuit network

SIP: Session initiation protocol

TCP: Transmission control protocol

UDP: User datagram protocol

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Note: This paper is available on-line at http://www.cis.ohio-state.edu/~jain/cis788-99/h323/index.html