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H.323 in Telecommunications

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Summary

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The purpose of this work is to find out, what the H.323 standard really is, what does it contain, what you can do with it, and who has already used it? In the end of this work there are some plans, what could be done in future.

This study is done in Sonera Ltd.

Keywords: H.323, VoIP, IP telephony, Gatekeeper, Gateway, MCU

Contents

Abbreviations

ACD	Automatic Call Distribution
ACF	Admission Confirmation
APDU	Active Protocol Data Unit
ARJ	Admission Reject
ARQ	Admission Request
ASN.1	Abstract Syntax Notation One
ATM	Asynchronous Transfer Mode
BCF	Bandwidth Change Confirmation
BRJ	Bandwidth Change Reject
BRQ	Bandwidth Change Request
BSC	Base Station Controller
BTS	Base Station
CID	Conference Identifier
CRV	Call Reference Value
DHCP	Dynamic Host Configuration Protocol
DSP	Digital Signal Processing
ETSI	European Telecommunications Standards Institute
GSM	Global System for Mobile communication
GSTN	General Switched Telephone Network
HTTP	Hyper Text Transfer Protocol
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ITU	International Telecommunications Union
IVR	Interactive Voice Response
LAN	Local Area Network
MC	Multipoint Controller
MCU	Multipoint Control Unit
MP	Multipoint Processor
MPEG	Motion Picture Experts Group
MSC	Mobile Switching Center
PBN	Packet Based Network

PBX	Private Branch Exchange
PER	Packet Encoding Rules
QOS	Quality of Service
RAS	Registration, Admission, Status
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
SCM	Selected Communications Mode
SCN	Switched Circuit Network
SME	Small and Medium Enterprise
SNMP	Simple Network Management Protocol
SP	Service Provider
TSAP	Transport Layer Service Access Point
UDP	User Datagram Protocol
VLR	Visitor Location Register
VoIP	Voice over Internet Protocol

1 What is IP Telephony?

The more descriptive words should be ‘Voice over Internet Protocol’ -telephony. So the pure IP Telephone (the telephones are connected direct to Internet via a LAN) is nothing more than “normal” phone, but the voice does not go through switched circuit network (SCN). The information goes in packet based network (PBN), for example above the Internet Protocol (IP). IP telephony can be implemented in a variety of ways. Within the IP network, the terminal devices must comply with the same communications standard, such as H.323. The devices can be carried over the user’s own IP network, over a service provider’s private network, over the public Internet, or a combination of these. The traffic might break out from the IP network into the SCN to terminate the call, or it might be carried end-to-end over IP infrastructure.

However, there are many difficulties to carry voice over PBN:

- loss of packets
- corruption of packets
- delay (latency and jitter)
- processing time in terminal

Some of those problems are partly solved at present products, but there is much work to be done before the phone calls can be managed in PBN like SCN.

1.1 The Rise of IP Telephony

Five years ago IP telephony was regarded by many telecommunications companies to be far too unreliable for mass market developing. But over the past few years, reliability and quality have quickly improved, and IP telephony is now one of the fastest growing areas in telecommunications.[1] One main reason for this is the huge development of computer technology, which gives the service provider (SP), or operator a possibility to develop such thing as IP telephony. And of course the use of Internet - which has grown explosively - has inspired many computer and telecommunication companies to find out the potentiality of VoIP solution. There is also one very big advantage in VoIP from clients’ side: making calls is practically free. That is a serious problem to operators. Why should they offer something that doesn’t make them money? There are a few reasonable answers: “Other operators emphasis on VoIP anyway so we have to keep our competitiveness”, “Maybe VoIP makes it possible to create new value added services”. Figure 1 describes how 3Com thought 1997 Internet Phone Projections will grow.[2]

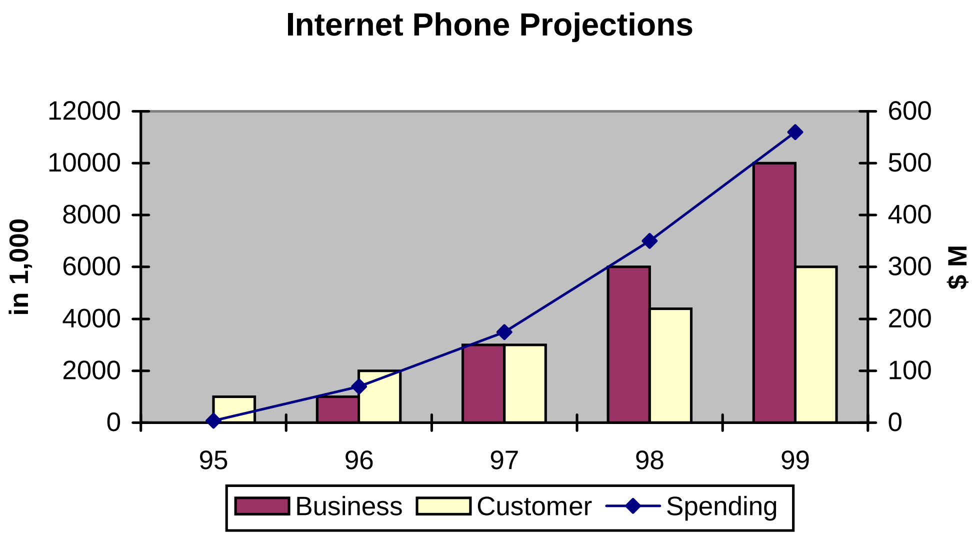

Figure 1 Internet Phone Projections

Nowadays there are several operators offering IP telephony service mainly between continents. In those calls you do not care about a little worse quality while the price of the call is considerably lower. Another interesting branch is corporate network. Combining cellular and PC-based telephony into a single system is a big challenge and at the same time very attractive service possibility.

People need to communicate more and more no matter whether it is computer, telephone, letter etc. The main tendency is to reach your friend somehow and quickly wherever you or your friend is. Making this easy is one of our missions. It means that different networks should understand each other so the message would go transparently from sender to receiver.

2 Why H.323?

Internet protocol was not developed to support real time application, such as calling. So we need a tool to transmit data while talking through PBN and that tool is H.323. There are many other proprietary protocols in VoIP, but because H.323 is ITU's (International Telecommunications Union) specification it is the most significant. The H.323 specification was approved in 1996 by the ITU's Study Group 15. The standard is broad in scope and includes both stand-alone devices, such as videotelephones, and embedded personal computer technology as well as point-to-point and multipoint conferences.[3]

The H.323 standard provides a foundation for audio, video and data communications across IP-based network. Support for audio is mandatory, while data and video are optional, but if supported, the ability to use a specified common mode of operation is required, so that all terminals supporting that media type can interwork.

2.1 Key benefits of H.323

Previously different vendors had a problem to get interwork their codecs. H.323 establishes standards for compression and decompression of audio and video data streams, ensuring that equipment from different vendors will have some area of common support.[3]

Users do not know what equipment the other participants use. Especially in a conference where many people (more than two) call from different places the compatibility is a big problem. Besides the information, H.323 establishes methods for receiving clients to communicate capabilities to the sender. The standard also establishes common call setup and control protocols (Q.931 and H.245).[3]

People use their computers in a wide range of network architectures. That might be a problem but H.323 is designed to run on top of common network architectures. The power of the standard turns out when network technology evolves and bandwidth management techniques improve. Big throughput makes it possible to utilize new multimedia connections such as videoconferencing.[3]

H.323 is flexible to support different hardware and software capabilities. Besides platforms can be tailored for individual user so you can use a system without having all details on it. This means that you do not need to have exactly same equipment than your calling partner. For example, a terminal with audio-only capabilities can participate in a conference with terminals that have also video and data capabilities. Of course, audio-only terminal can't see the picture or read the data.[4]

Bandwidth management is one of the biggest benefits in H.323 because video and audio traffic is so bandwidth intensive. In corporate networks, managers can limit the number of simultaneous H.323 connections or cut down the bandwidth of those users who use it at present.[3]

H.323 can support conferences of three or more endpoints without requiring a specialized multipoint control unit. Though there can be a separate multipoint control unit (MCU) which can take care of controlling. Multipoint capabilities can be included in other components of an H.323 system.[3] H.323 uses common codec technology from different videoconferencing standards to minimize transcoding delays and to provide optimum performance. This feature makes the conferences from remote site possible.

Additionally, H.323 supports multicast transport in multipoint conferences. It means that one person can send one packet to several destinations on the network without replication.

Software that is developed to the H.323 standard is scaleable in hardware. This implies that the more powerful the PC, the better the performance of the software. This allows application developers to provide faster software simply by having the users upgrade hardware.

The H.323 standard itself provides new business opportunities for value-added services such as billing, call tracking, and automatic call distribution. This opportunity is similar to the opportunities for businesses provided by the traditional telecommunication companies. The H.323 standard also defines how an H.323 system will inter-operate with other legacy systems based on regular telephone or ISDN lines. This Gateway capability gives vendors a way to support their legacy applications into the existing customer base.

2.2 Weaknesses of H.323

H.323 doesn't provide a guaranteed Quality of Service. That is the biggest problem because this standard is designed to carry interactive voice, data and picture.

It's difficult to get H.323 through firewalls. That is, because H.323

- is very complex and inflexible
- uses dynamic ports
- includes multiple UDP streams

Much of the complexity of H.323 comes from the multiple protocol components it consists of. The components are tightly intertwined and cannot be separately used or exchanged. Also the message format is a complex issue. The H.323 protocol is based on ASN.1 and PER (packet encoding rules) and uses a binary representation. Generally, this requires large and expensive code-generators to parse. The ASN.1 encoding used in H.323 makes firewalls and proxies very complex. The H.323 protocol suite consists of several components almost without clear separation.[7]

Call setup requires about 6 to 7 round-trip times, depending on whether a Gatekeeper is being used or not. This includes setup of the Q.931 and H.245 connections. Using fast connect in version 2 of H.323, the call setup delays are significantly minimized. Fast connect reduces the number of round-trips and allows the media channels to be operational before the Connect message is sent and the telephone rings.

3 H.323 Architecture

The H.323 recommendation covers the technical requirements for audio and video communications services in LANs (Local Area Network). H.323 references the T.120 specification for data conferencing and enables conferences, which include a data capability. The scope of H.323 does not include the LAN itself or the transport layer that may be used to connect various LANs. Only elements needed for interaction with the SCN are within the scope of H.323. Figure 2 outlines an H.323 system and its components.[4]

H.323 defines four major components for a network-based communications system: **Terminals, Gateways, Gatekeepers** and **Multipoint Control Units.** Next here is told about those components and their relations. In the end of this section is some cases how calling is handled in H.323 environment.

3.1 Terminals

An H.323 terminal is an endpoint on the network, which provides for real-time, two-way communications with another H.323 terminal, Gateway or multipoint control unit. This communication consists of control, indications, audio, moving color video pictures, and/or data between two terminals. All terminals must support voice communications; video and data are optional.[3]

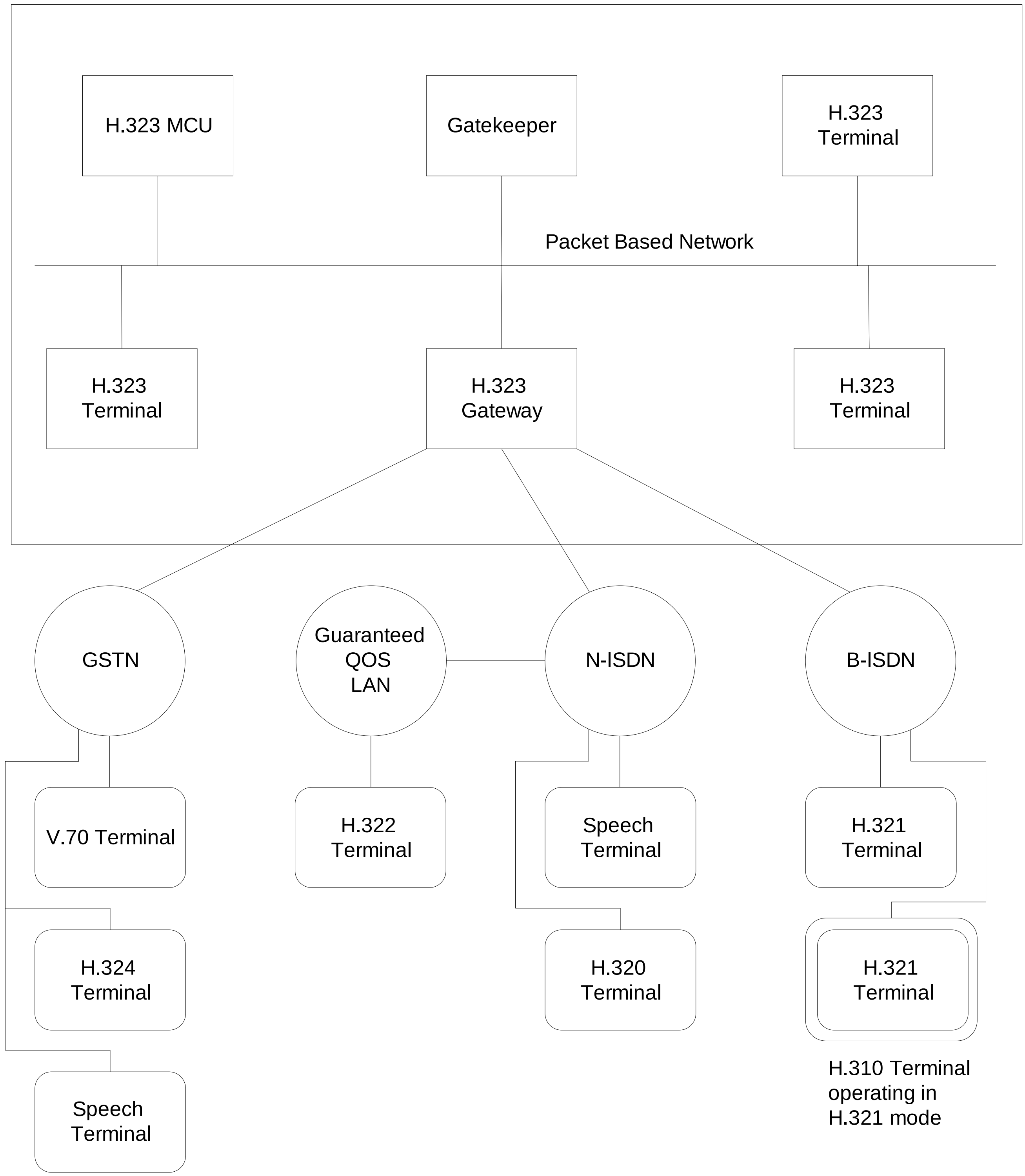

Figure 2. Interoperability of H.323 terminals

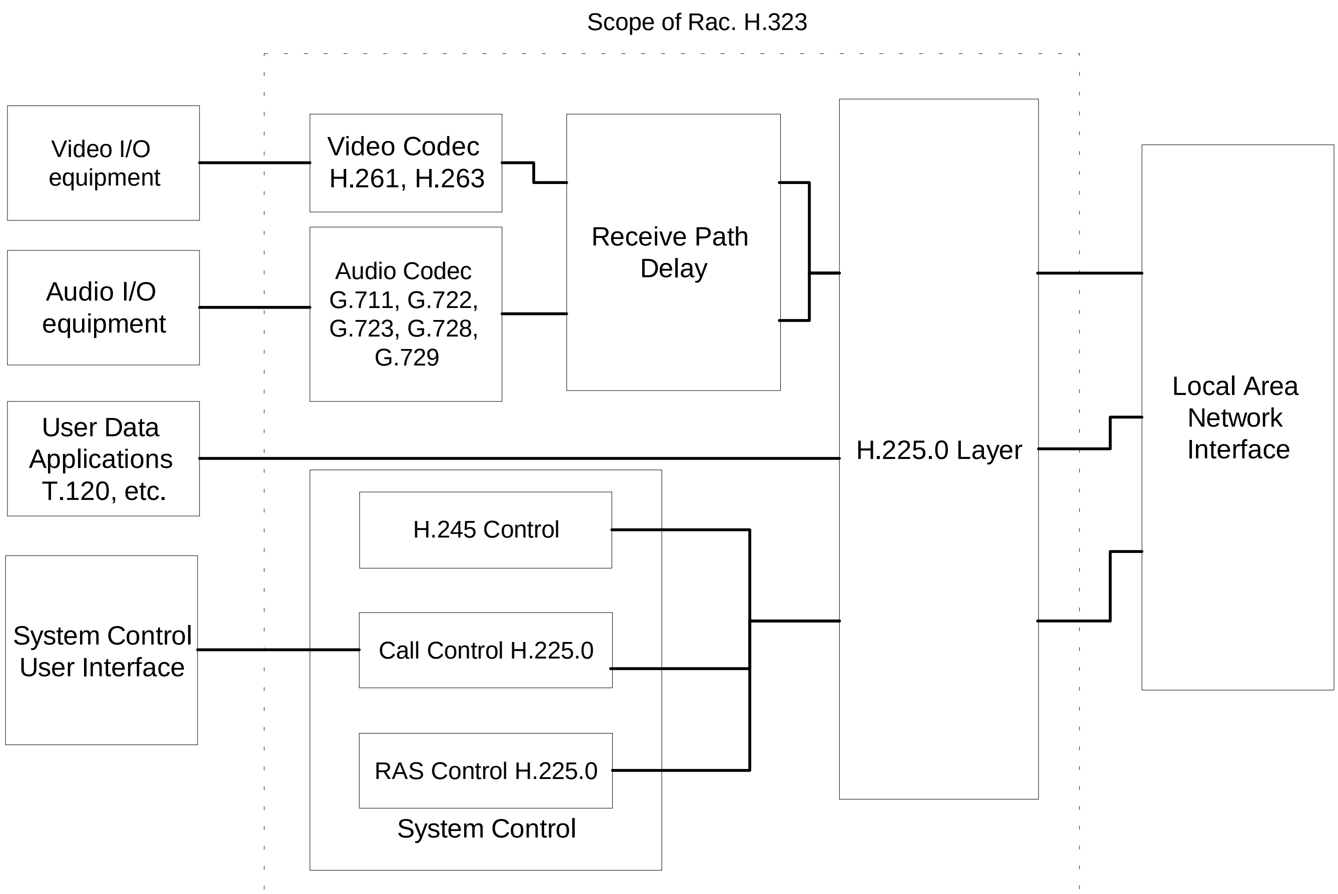

Figure 3. H.323 Terminal Equipment

3.1.1 Terminal characteristics

An example of an H.323 terminal is shown in Figure 3. The figure shows the user equipment interfaces, video codec, audio codec, telematic equipment, H.225 layer, system control functions and the interface to the PBN. In every H.323 terminal must have a system control unit, H.225 layer, network interface and an audio codec unit. With those equipment you can have a voice connection, which is only mandatory connection way in every terminal. The video codec unit and data applications are optional.[4]

3.1.2 Audio

Speech support is mandatory in H.323 terminal. This means that every terminal shall have an audio codec, which shall be capable of encoding and decoding speech. Audio signals contain digitized and compressed speech. All the following compression algorithms supported by H.323 are ITU standards.[4]

G.711

G.722

G.728

G.729

MPEG 1 audio

G.723.1

GSM

The different ITU recommendations for digitizing and compressing speech signals reflect different tradeoffs between speech quality, bit rates, computer power and signal delay.

G.711 is the only mandatory codec. G.711 generally transmits voice at 56 or 64 kbps, well within the bandwidth limits likely on a LAN, although it was designed originally for continuous bit rate networks.[3]

G.722 works on 7 kHz bandwidth at 64, 56 or 48 kbps.[6]

G.728 transmits voice at 16 kbps, G.729 at 8 kbps and G.723.1 at 5,3 or 6,3 kbps.[4]

MPEG 1 audio specifies a family of three audio coding schemes, simply called Layer-1,-2,-3, with increasing encoder complexity and performance. All layers are allowed to work with similar bit rates:[5]

Layer-1: from 32 kbps to 448 kbps

Layer-2: from 32 kbps to 384 kbps

Layer-3: from 32 kbps to 320 kbps

GSM codec uses 5,6-13 kbps throughput.[6]

According to the standards, H.323 terminals will be capable of transmitting and receiving A-law and •-law. Terminal should be capable of asymmetric operation for all audio capabilities it has declared within the same capability. In other words it should be able to send G.728 and receive G.711 if it is capable of both.

Over low bit rate (<56 kbps) links or segments an endpoint should have an audio codec capable of encoding and decoding speech according to recommendation G.723.1, or an audio-only endpoint to recommendation G.729. If there is one or more, low bit rate segments in end-to-end connection, every participants must have this low bit rate codec.

3.1.3 Video

Video codec is an optional in H.323 terminal, but if there exists one the mandatory codec is H.261. H.261 is quite old video codec version, so it can't handle the requirements of newest picture formats. Therefore newer codecs have been developed, i.e., H263 and H.263+. Other video codecs, and other picture formats, may also be used via H.245 negotiation.[4]

H.323 terminal may optionally send (and receive) more than one video channel at the same time. This feature enables to display multiple participants in a distributed multipoint videoconference. Terminal can operate in an asymmetric mode also in video bit rates, frame rates and picture resolutions (if there exist more than one picture).[4]

3.1.4 Data

One or more data channels are optional. Depending on the requirements of the data application, the data channel may be unidirectional or bidirectional. In H.323 environment recommendation T.120 is the default basis of data interoperability.[4]
It provides interoperability at the application, network, and transport level.[3]
T.120 data channel is established during an H.323 call as an inherent part of the call. Closing data channel doesn't shut down whole H.323 connection, but only data

stream channel. On the other hand, when H.323 call or conference is terminated, then the associated T.120 conference must also be terminated.[4]

3.1.5 Control [3]

The call control functions are the heart of the H.323 terminal. These functions include signaling for call setup, capability exchange, signaling of commands and indications, and messages to open and describe the content of logical channels. All audio, video, and control signals pass through a control layer that formats the data streams into messages for output to the network interface. The reverse process takes place for incoming streams. The Q.931, RAS (Registration, Admission and Status) and RTP/RTCP (Real Time Protocol / Real Time Control Protocol) protocols perform these functions.

Overall system control is provided by three separate signaling functions: the **H.245 Control Channel**, the **Q.931 Call Signaling Channel**, and the **RAS Channel**.

The H.245 Control Channel is a reliable channel that carries control messages governing operation of the H.323 entity. H.245 signaling is established between two endpoints: a terminal and an MC, or a terminal and a Gatekeeper. H.245 messages fall into four categories: request, response, command, and indication.[4] Capabilities exchange is one of the fundamental capabilities in the ITU recommendation; H.245 provides for separate receive and transmit capabilities as well as for methods to describe these details to other H.323 terminals. There is only one H.245 Control Channel per call.

The Call Signaling Channel uses Q.931 to establish a connection between two terminals.

The RAS signaling function performs registration, admission, bandwidth changes, status, and disengage procedures between endpoints and Gatekeepers. RAS is not used if a Gatekeeper is not present.

3.2 Gateways

The Gateway is an optional element in H.323 environment but it enables endpoints from different networks to conference. So endpoints do not need a Gateway when the call that is established is in the same network. Beside translation, Gateway performs call setup and clearing on both the LAN side and the SCN side. In general, the purpose of the Gateway is to reflect the characteristics of a LAN endpoint to an SCN endpoint and vice versa.[3] Endpoint can send data through one Gateway and receive data through another Gateway.[4]

The Gateway has the characteristics of an H.323 Terminal or MCU on the network, and of the SCN terminal or MCU on the SCN. The choice of terminal or MCU is left to the manufacturer. Four examples of an H.323 Gateway are shown in Figure 4. The SCN terminal and MCU function has the characteristics described in the appropriate recommendation (H.310, H.320, H.321, H.322, H.324, V.70, GSTN or ISDN speech only terminals) as you can see in Figure 2.

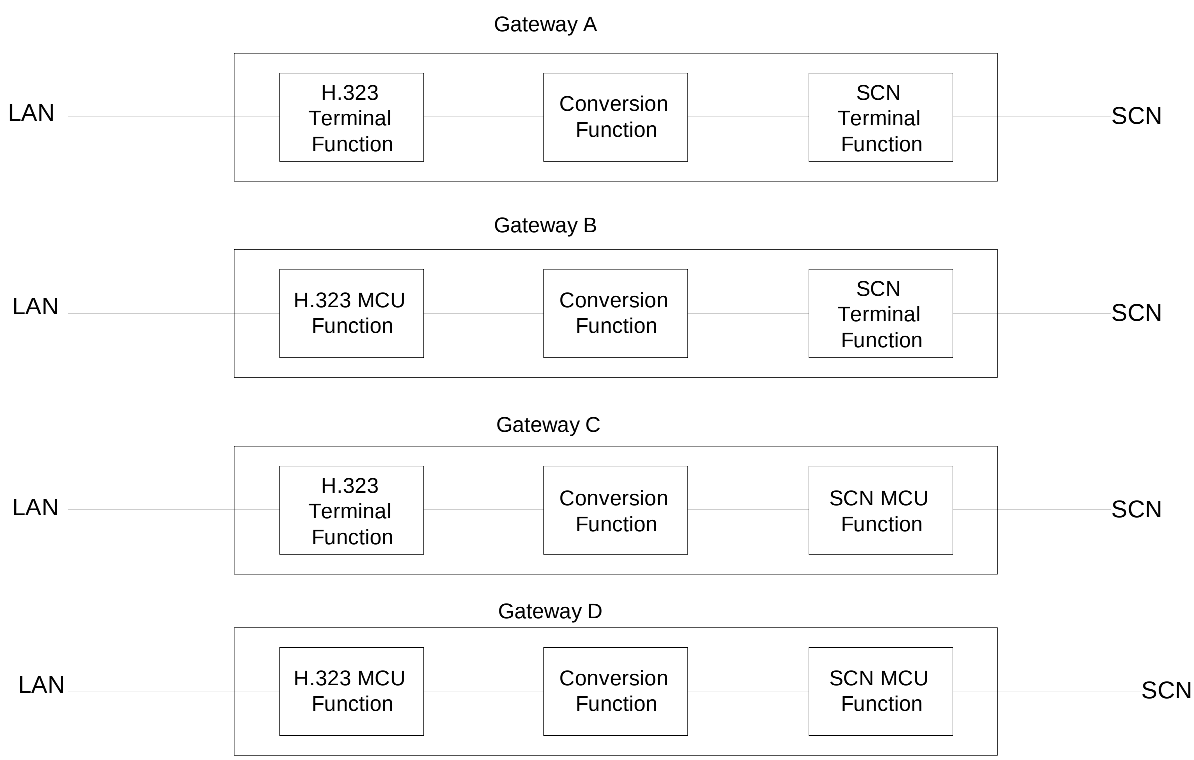

Figure 4. H.323 gateway configurations

The conversion function provides the necessary conversion of transmission format, control, audio, video, and/or data streams between different terminal recommendations.[4] Terminals communicate with Gateways using the H.245 and Q.931 protocols. The actual number of H.323 terminals that can communicate through the Gateway is up to designer. Similarly, the number of SCN connections, the number of simultaneous independent conferences supported, the audio/video/data conversion functions, and inclusion of multipoint functions are left to the manufacturer.[3]

3.3 Gatekeepers

The Gatekeeper, which is optional in an H.323 system, provides call control services to the H.323 endpoints. However, if a Gatekeeper is present, it is mandatory terminals to make use of the services offered by Gatekeeper.[3] There can be more than one Gatekeeper and they could communicate with each other in an unspecified fashion. The Gatekeeper is logically separate from endpoints, but its physical implementation may coexist with terminal, MCU, Gateway, MC, or other non-H.323 network device.[4]

When the Gatekeeper exists in an H.323 environment, it has to provide the following services:

- Address translation: Translation of Alias Address to Transport Address using a table that is updated with Registration messages. Other methods of updating the translation table are also allowed.[3] The alias address may be i.e. name or e-mail address.
- Admissions Control: The Gatekeeper will authorize network access using ARQ/ACF/ARJ H.225 messages (admission request, confirm, reject). This may be based on call authorization, bandwidth, or some other criteria which is left to the manufacturer. It is also possible for function to admit all requests.[4]
- Bandwidth Control: The Gatekeeper supports BRQ/BRJ/BCF messages. This may be based on bandwidth management. It may also be a null function, which accepts all requests for bandwidth changes.[4]

- Zone Management: The Gatekeeper provides the above functions for terminals, MCUs and Gateways, which have registered with it (see next column).[4]

The Gatekeeper may also perform other optional functions:

- Call Control Signaling: In point to point conference, the Gatekeeper may process Q.931 control signals. Alternatively, the Gatekeeper may send the endpoints Q.931 signals directly to each other.[3]
- Call Authorization: The Gatekeeper may reject a call from a terminal based on the H.225.0 specification. The reasons for rejection may include restricted access to/from particular terminals or Gateways, and restricted access during certain periods of time.[4]
- Bandwidth Management: The Gatekeeper may reject calls (through the use of the H.225.0 signaling) from a terminal if it determines that sufficient bandwidth is not available. This function also operates during an active call if a terminal requests additional bandwidth.[3] For example, in Microsoft's NetMeeting uses this functionality. There the voice has priority one and the video comes later, so if the network can't offer enough capacity, video drops out.
- Call Management: For example, the Gatekeeper may maintain a list of ongoing H.323 calls. This information may be necessary to indicate that a called terminal is busy, and to provide information for the Bandwidth Management function.[4]
- Gatekeeper management information data structure.[4]
- Bandwidth reservation for terminals not capable of this function.[4]
- Directory services.[4]

Gatekeeper can also play a role in multipoint connections. To support multipoint conferences, users would employ a Gatekeeper to receive H.245 Control Channels from two terminals in a point-to-point conference. When the conference switches to multipoint, the Gatekeeper can redirect the H.245 Control Channel to multipoint controller (MC). The Gatekeeper need not process the H.245 signaling – it only needs to pass it between the terminals or the terminals and the MC.[4]

Networks, which contain Gateways should also contain a Gatekeeper in order to translate incoming E.164 (or partyNumber) addresses into Transport Addresses.
The collection of all Terminals, Gateways and Multipoint Control Units managed by a single Gatekeeper is known as an H.323 Zone (Figure 5).[3]

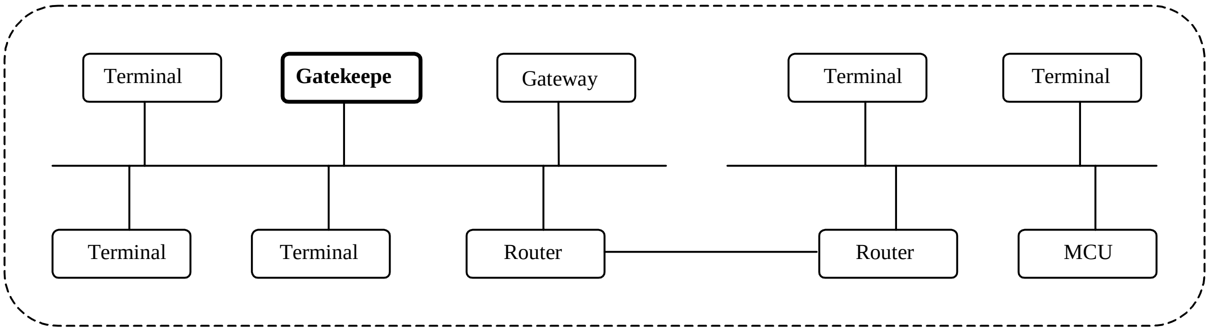

Figure 5. H.323 Zone

3.4 Multipoint Control Units (MCU)

The MCU is an endpoint on the network, which provides the capability for three or more terminals and Gateways to participate in a multipoint conference. It may also connect two terminals in a point-to-point conference, which may later develop into a multipoint conference. The MCU consists of two parts: a mandatory Multipoint Controller and optional Multipoint Processor (can be several in one Zone).[4]

3.4.1 Multipoint Controller [4]

The MC:

- provides control functions to support multipoint conference
- carries out the capabilities exchange with each endpoint in a multipoint conference
- sends a capability set to the endpoints in the conference indicating the operating modes in which they may transmit
- may revise the capability set that is sent to the terminal for some reason (i.e. one terminal join or leave the conference)

In this manner, the MC determines the Selected Communication Mode for the conference. The SCM may be common for all endpoints in the conference or some endpoints may have some extra feature.

The MC may be located within a Gatekeeper, Gateway, terminal or MCU. Whether MC is callable, depends where it locates. The MC does not deal directly with any of the media streams. This is left to the MP.[3]

3.4.2 Multipoint Processor [4]

The MP receives audio, video and/or data streams from the endpoints involved in a centralized or hybrid multipoint conference. It processes these media streams and returns them to the endpoints. The MP may process one or more media stream types. Used algorithm types are described previous in this document.

The MP may provide algorithm and format conversion, allowing terminals to participate in a conference at different SCMs. The MP is not callable, but the MCU, which it is part of, is callable. The MP terminates and sources the media channels.

3.4.3 Multipoint Conferences

Multipoint conference capabilities are handled in a variety of methods and configurations under H.323. The Recommendation uses the concepts of centralized and decentralized conferences, as described in Figure 6.[3]

A typical MCU that supports centralized multipoint conferences consists of an MC and an audio, video and data MP. On the other hand the MCU that supports decentralized multipoint conferences consists of an MC and a data MP supporting Recommendation T.120. It relies on decentralized audio and video processing.[4]

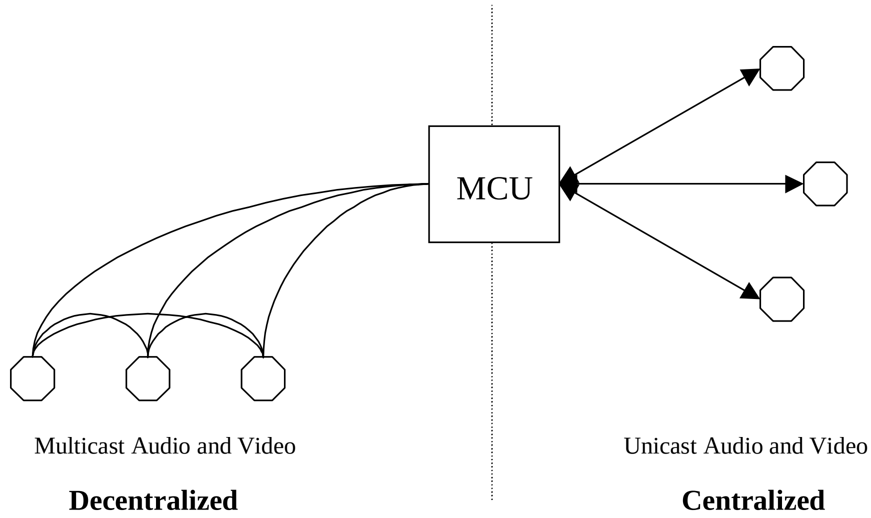

Figure 6. Decentralized and Centralized Conferences

Centralized multipoint conferences require the existence of an MCU to facilitate a multipoint conference. All endpoints have to have centralized multipoint capability. In this mode of operation they communicate with the MC of the MCU in a point-to-point manner on the control channel and with the MP on the audio, video and data channels. The MP may also provide conversion between different codecs and bit rates and may use multicast to distribute processed video. The MC centrally manages the conference using H.245 control functions that also define the capabilities for each terminal.[3]

Decentralized multipoint conferences make use of multicast technology. Participating H.323 terminals multicast audio and video to other participating terminals without sending the data an MCU. Note that control of multipoint data is still centrally processed by the MCU, and H.245 Control Channel information is still transmitted in a point-to-point mode to an MC.

Receiving terminals are responsible for processing the multiple incoming audio and video streams. Terminals use H.245 Control Channels to indicate to an MC how many simultaneous video and audio streams they can decode. The number of simultaneous capabilities of one terminal does not limit the number of video or audio streams, which are multicast in a conference.[3]

Hybrid multipoint conferences use a combination of centralized and decentralized features. H.245 signals and either an audio or video stream is processed through point-to-point messages to the MCU. The remaining signal (audio or video) is transmitted to participating H.323 terminals through multicast.
H.323 also supports mixed multipoint conferences in which some terminals are in a centralized conference and others are in a decentralized conference. In this situation the MCU provides a bridge between the two types (see Figure 6). The terminal is no aware of the mixed nature of the conference, only of the mode of conference, in which it sends and receives.[3]

4 Call signaling

Call signaling is the messages and procedures used to establish a call, request changes in bandwidth of the call, get status of the endpoints in the call, and disconnect the call. Call signaling uses messages defined in Recommendation H.225.0.[4]

4.1 Addresses

4.1.1 Network address

Each H.323 entity has to have at least one network address. This address uniquely identifies the H.323 entity on the network. Some entities may share a network address, i.e. a terminal and a co-located MC. An endpoint may use different network address for different channels within the same call.[4]

4.1.2 TSAP identifier

For each network address, each H.323 entity may have several TSAP (Transport Layer Service Access Point) identifiers. These TSAP Identifiers allow multiplexing of several channels sharing the same network address.[4]

4.1.3 Alias address

An endpoint may also have one or more alias addresses associated with it. The alias addresses provide an alternate method of addressing the endpoint. These address include E.164 or *partyNumber* addresses (network access number, telephone number, etc.), H.323 Ids (alphanumeric strings representing names, e-mail like addresses, etc.), and any others defined in Recommendation H.225.0. Alias addresses must be unique within a Zone. Gatekeepers, MCs, and MPs must not have alias addresses.[4]

4.2 Registration, Admissions and Status (RAS) channel

The RAS channel is used to carry messages used in the Gatekeeper discovery and endpoint registration processes, which associate an endpoint's alias address with its Call Signaling Channel Transport Address. The RAS channel is an unreliable channel. Since the RAS messages are transmitted on an unreliable channel, H.225.0 recommends timeouts and retry counts for various messages.[4]

4.2.1 Gatekeeper discovery

Gatekeeper discovery is the process an endpoint uses to determine which Gatekeeper to register with. This may be done manually or automatically. The automatic method allows the endpoint-Gatekeeper association to change over time. The endpoint may not know who its Gatekeeper is, or may need to identify another Gatekeeper due to a failure. This may be done through auto discovery.[4]

4.2.2 Endpoint registration

Registration is the process by which an endpoint joins a Zone and informs the Gatekeeper of its Transport Address and alias addresses. All endpoints have to register with the Gatekeeper identified through the discovery process. Registration must occur before any calls are attempted and may occur periodically as necessary. An endpoint's registration with a Gatekeeper may have a finite life. An endpoint, which is not registered with a Gatekeeper, is called an unregistered endpoint. This type of endpoint does not request admission permission from a Gatekeeper and so cannot participate in admission control. Bandwidth control. Address translation and other functions performed by the Gatekeeper.[4]

4.2.3 Other RAS channels tasks[4]

- Endpoint location
- Admission, bandwidth change, status and disengage
- Access tokens; can provide privacy by shielding an endpoint's Transport Address and Alias Address information from calling party; ensures that calls are routed properly through H.323 entities

4.3 Call signaling channel

The Call Signaling channel must be used to carry H.225.0 call control messages through reliable channel. In networks that do not contain a Gatekeeper call signaling messages are passed directly between the calling and called endpoints using the Call Signaling Transport Addresses. In networks that contain a Gatekeeper the initial admission message exchange takes place between the calling endpoint and the Gatekeeper using the Gatekeeper's RAS Channel Transport Address. Recommendation H.225.0 specifies the mandatory Q.931 messages that are used for call signaling in this Recommendation.[4]

4.3.1 Call signaling channel routing

Call signaling messages may be passed in two ways. The first method is Gatekeeper Routed Call Signaling (see Figure 7). In this method call signaling messages are routed through the Gatekeeper between the endpoints. The second method is Direct Endpoint Call Signaling (see figure 8). In this method the call signaling messages are passed directly between the endpoints. The choice of which method is used, is made by the Gatekeeper.[4]

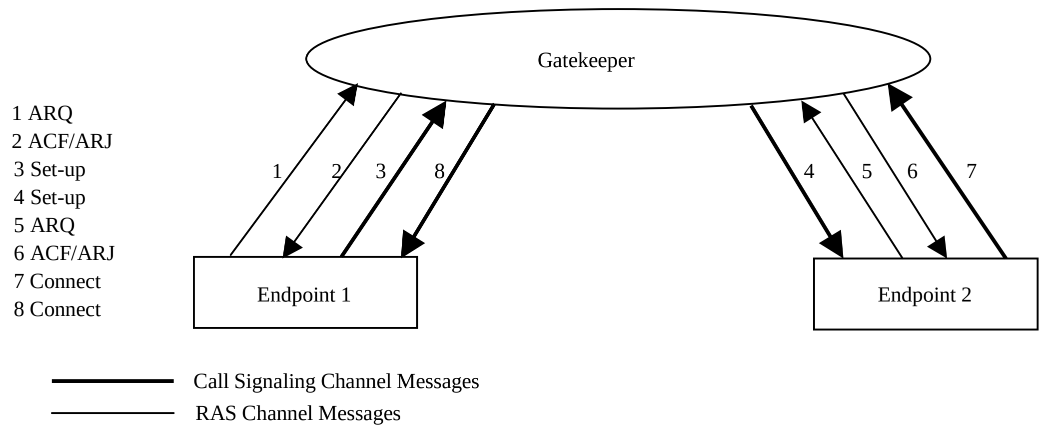

Figure 7. H.323 - Gatekeeper routed call signaling

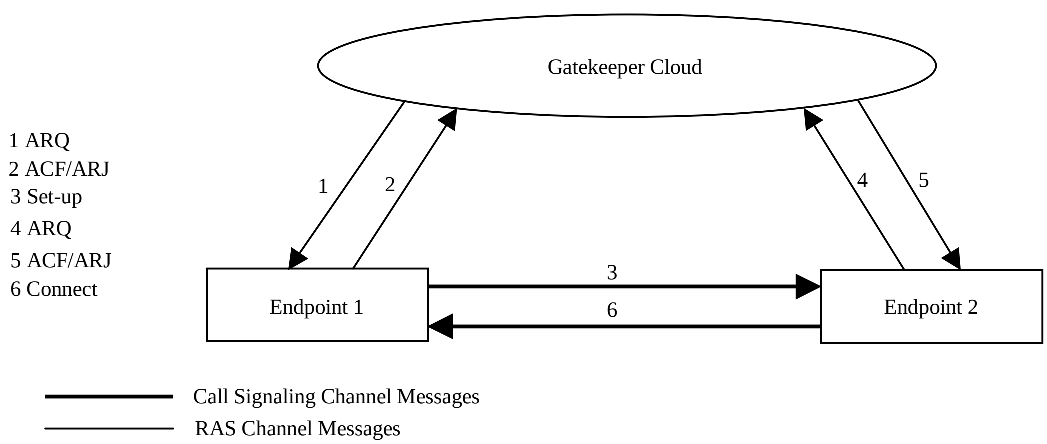

Figure 8. H.323 - Direct endpoint call signaling

Both methods use the same kinds of connections for the same purposes, and the same messages. Admissions messages are exchanged on RAS channels with the Gatekeeper, followed by an exchange of call signaling messages on a Call Signaling channel. The Gatekeeper Clouds in Figures 7 and 8 contain one or more Gatekeepers, which may or may not communicate with each other. The endpoints may be connected to the same Gatekeeper or to different Gatekeepers.[4]

4.3.2 Control channel routing

When Gatekeeper Routed call signaling is used, there are two methods to route the H.245 Control Channel: directly between the endpoints or between the endpoints through the Gatekeeper. When Direct Endpoint call signaling is used, the H.245 Control Channel can only be connected directly between the endpoints. The later method allows the Gatekeeper to redirect the H.245 Control Channel to an MC when an ad hoc multipoint conference switches from a point-to-point conference to a multipoint conference. This choice is made by the Gatekeeper.[4]

4.4 Call reference value

All call signaling and RAS messages contain a Call Reference Value (CRV). There is one CRV for the call signaling channel and an independent CRV for the RAS channel. The CRV is not the same as the Call ID or the Conference ID (CID). The CRV associates call signaling or RAS messages between two entities within the same call, while the Call ID associates all messages between all entities within the same call, and the CID associates all messages between all entities within all calls in the same conference.[4]

4.5 Call ID

The call ID is globally unique non-zero value created by the calling endpoint and passed in various H.225.0 messages. The Call ID identifies the call with which the message is associated. Unlike CRV, the Call ID does not change within a call.[4]

4.6 Conference ID and Conference Goal

The Conference ID (CID) is globally unique non-zero value created by the calling endpoint and passed in various H.225.0 messages. The CID identifies the conference with which the message is associated. Therefore, messages from all endpoints within same conference will have the same CID.

The **conferenceGoal** indicates the intention of the call. Choices are: Create – to create a new conference, Join – to join an existing conference, Invite – to invite a new endpoint into an existing conference, Capability Negotiation – negotiate capabilities for a later H.323 conference, and Call Independent Supplementary Service – transport of supplementary services APDUs.[4]

5 Call signaling procedures

Call signaling is very wide and specific area, so here I deal only with the main ideas. Call signaling is needed in all five phases during a call.

5.1 Call set-up

Call set-up takes place using the call control messages defined in Recommendation H.225.0. Requests for bandwidth reservation should take place at the earliest possible phase. There is no explicit synchronization or locking between two endpoints during the call set-up procedure. This means that endpoint A can send a Setup message to endpoint B at exactly the same time that endpoint B sends a Setup message to endpoint A. The Connect message should be sent only if it is certain that the H.245 capability exchange will conclude successfully and a minimum level of communications can take place. This is to maintain the consistency of the meaning of the Connect message between PBN and SCN.[4]

We get different cases depending on which endpoint, if any, has a Gatekeeper: the calling terminal, the called terminal, both to different Gatekeepers or both to the same Gatekeeper.[7] In the scenario in Figure 9 neither endpoint is registered to a Gatekeeper in other words two endpoints communicate directly. Endpoint 1 (calling endpoint) sends the Set-up (1) message to the well-known Call Signaling Channel TSAP

Identifier of endpoint 2. Endpoint 2 responds with the Connect (4) message, which contains an H.245 Control Channel Transport Address for use in H.245 signaling.[4]

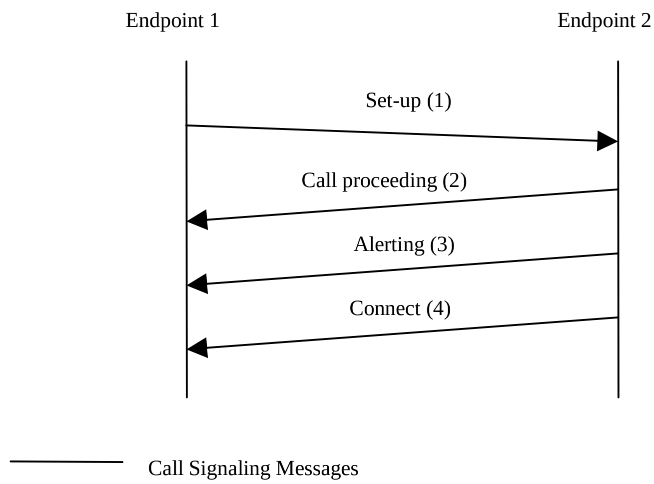

Figure 9. Basic call set-up, no Gatekeepers

When an external terminal calls a network endpoint via the Gateway (or a network endpoint calls an external terminal via the Gateway), call set-up between the Gateway and the network endpoint proceeds the same as the endpoint-to-endpoint call set-up.[4]

5.2 Initial communication and capability exchange

The H.245 channel is set up by sending *terminal capability set* messages. After the capability exchange, the endpoints perform a master-slave determination to determinate, which of the terminals is active MC. Also the master-slave determination is a part of the H.245 specification. To conserve resources, synchronize call signaling and control and reduce call setup time and alternative method exists: The H.245 messages can be sent within the Q.931 channel, a process known as tunneling. To conclude, there are three ways to set up a H.323 call: a separate H.245 channel, fast connect and tunneling. However, it is not always possible to use fast connect or tunneling and a terminal can at any time switch back to a separate H.245 channel.[7]

5.3 Establishment of audiovisual communication

After the call has been accepted by the callee (endpoint 2), the media channels are set up using the H.245 protocol. The feature negotiation is performed. This requires several round-trips and results in a significant delay before the negotiation is completed. In version 2 of the standard, this delay problem is partly solved. Using the *Fast Connect* procedure the media channels can be established with only one round-trip and the media transmission can start immediately after sending a fast start message.[7]

5.4 Call services

5.4.1 Bandwidth changes

At any time during a conference, the endpoints or Gatekeeper may request an increase or decrease in the call bandwidth. An endpoint may change the bit rate of a logical channel without requesting a bandwidth change from Gatekeeper if the aggregate bit rate of all transmitted and received channels does not exceed the current call bandwidth. An endpoint wishing to change its call bandwidth sends a Bandwidth Change Request (BRQ) message to the Gatekeeper. The Gatekeeper determines if the request is acceptable. If the Gatekeeper determines that the request is not acceptable, it returns a Bandwidth Change Reject (BRJ) message to endpoint. If the Gatekeeper determines that the request is acceptable, it returns a Bandwidth Change Confirm (BCF) message.[4]

5.4.2 Status

The Gatekeeper may use the Information Request (IRQ)/Information Request Response (IRR) message sequence to poll what is the status of the endpoint. Also during the duration of a call, an endpoint or Gatekeeper may periodically request call status from another endpoint.[4]

5.4.3 Ad Hoc Conference Expansion

An Ad Hoc Multipoint conference is one that can be expanded from a point-to-point conference involving an MC to a multipoint conference. First, a point-to-point conference is created between two endpoints (endpoint 1 and endpoint 2). At least one endpoint, or the Gatekeeper, must contain an MC. Once a point-to-point conference has been created, the conference may be expanded to multipoint conference in two different ways. The first way is when any endpoint in the conference invites another endpoint (endpoint 3) into the conference by calling that endpoint through the MC. The second way is for an endpoint (endpoint 3) to join an existing conference by calling an endpoint in the conference.[4]

5.4.4 Third party initiated pause and re-routing

To allow Gatekeepers to re-route connections from endpoints that do not support supplementary services, endpoints shall support the reception of empty capability sets. This feature also allows network elements such as PBXs, call centers, and IVR systems to re-route connections independent of supplementary services and facilitates pre-connect announcements.[4]

On reception of an empty capability set an endpoint shall enter a paused state. If media stream or data logical channels were previously opened by the endpoint, they should be closed. In this state an endpoint shall accept the opening of logical channels from the remote end based on the usual rules and continue to receive media from open logical channels from the remote end.[4]

5.5 Call termination

Finally there is only call termination, i.e. end all the procedures related to the call. Either endpoint can do that by the following procedure:

- It should discontinue transmission of video at the end of a complete picture, and then close all logical channels for video.

- It should discontinue transmission of data and audio and then close all logical channels for both data and audio.
- It shall transmit the H.245 endSessionCommand message in the H.245 Control Channel, indicating to the far end that it wishes to disconnect the call and then discontinue H.245 message transmission.
- It shall then wait to receive the endSessionCommand message from the other endpoint and then shall close the H.245 Control Channel.
- If the Call Signaling Channel is open, a Release Complete message shall be sent and the channel closed.
- It shall clear the call without a Gatekeeper, with a Gatekeeper, or by Gatekeeper.
- Terminating a call may not terminate a conference; a conference may be explicitly terminated using an H.245 message (dropConference).[4]

6 Existing H.323 products

The H.323 protocol suite is quickly maturing, mostly because of the influence of companies that are leveraging it for new business opportunities. Companies such as Intel, Cisco, and Microsoft are investing heavily in the development and marketing of H.323 standards and products. It is unlikely these companies are investing so much to gain control of a niche market.

6.1 Protocol stack

All H.323 entities (terminals, Gateways, Gatekeepers, MCUs) need an H.323 protocol stack to establish a session over an IP network (LAN/intranet/Internet).

6.1.1 RADVision H.323 Protocol Stack

The RADVision H.323 Protocol Stack is platform independent. Developers can license either source code or object code for Windows and UNIX operating systems as well as for VxWorks and pSOS real time operating systems. Stack pricing includes a licensing fee, annual maintenance and a one-time prepaid royalty.

6.2 Gatekeepers

6.2.1 VocalTec's Gatekeeper

VocalTec Gatekeeper (VGK) is the intelligent hub for IP telephony networks as well as the platform for advanced services. VGK is compatible with H.323-compliant IP telephony elements. VGK facilitates authenticated call initiation in order to provide the following services in an IP telephony environment: Phone-to-Phone, PC-to-Phone, PC-to-PC, Web-to-Phone, Fax-to-Fax, Store-and-forward, Real-time Fax and Internet Phone Call Waiting.

System Requirements:

- Windows NT Server 4.0
- Service Pack 3 for Windows NT Server 4.0
- Pentium II 266 MHz 512 cache CPU
- 256 MB RAM SDRAM 100 MHz ECC
- 168 PIN DIMM
- 4 GB SCSI Hard disk and SCSI controller (two hard drives recommended)
- PCI 10/100 Network card and PCI display controller
- Fixed IP address
- DNS name

VGK includes lots of enhanced features, and that is why it won Network Magazine's 1999 Product of the Year Award in the category of telephony integration. Let us mention here such features as network security and centralized accounting and billing. VGK authenticates user ID/passwords and authenticates users who want to access the IP telephony system. VGK also provides a centralized open interface for all Call Detail Records (CDR), enabling credit and debit mode billing.[8]

6.2.2 RadVision's Gatekeeper

RADVision's Gatekeeper software is built around the company's H.323 Protocol Stack and provides the mechanism for:

- Call control and call routing
- Telephony services such as directory services and PBX functions (e.g. call forward)
- GK cascading - Implementation of a distributed GK functionality for increased reliability and scaling
- ACD (Automatic Call Distribution) implementation
- Controlling H.323 bandwidth usage to provide Quality of Service (QoS), and protect other critical network applications from H.323 traffic
- Total network usage control
- System administration and security policies

The Gatekeeper can co-reside with any H.323 entity (terminal/Gateway/MCU) or run as a single H.323 application on the host. It can be configured and controlled remotely using http, SNMP and other protocols.[9]

6.3 Gateways

Gateway implementations are divided to two categories. Some of the Gateway implementations are based on a suite of software programs, which translates the traffic between a packet based network and telephone network. In other Gateway implementations translation between networks is made by hardware. Maximum capacity of the software-based Gateway is 30 simultaneous calls, which is the same as the hardware-based Gateway's minimum capacity (30 simultaneous calls = one ISDN/E1 interface). This means, that software-based Gateways can not compete with hardware-based Gateways in long run, although speed of the computer processors is growing.

6.3.1 Natural Microsystems's Gateway: Fusion

Fusion integrates hardware and software within a standard PCI or CompactPCI computing environment. Fusion's modular architecture allows it to support existing protocols such as the ITU's H.323 specification. Building on its basic configuration of up to a dual E1 (60 ports) of IP telephony in a single PCI slot, Fusion's scalable architecture supports the highest port capacity of any solution on the market.
Fusion uses an intelligent hardware and software architecture that integrates PSTN interfaces, telephony protocols, comprehensive IVR functionality, full-duplex echo cancellation, speech encoding, fax processing, and, optionally, LAN interfaces and data protocols into a compact, flexible package. Additionally, Fusion supports the broadest choice of standard vocoding algorithms, including G.723.1, G.729A, G.711, MS GSM, and ETSI GSM.[10]

6.3.2 Digis Gateway: NetBlazer 8500

This Gateway handles the conversion of voice signals and it can scale from 30 to 60 ports. If more than 60 ports are needed for one Point-of-Presence (POP), or if additional fault-tolerance and redundancy are desired, several Gateways can be cascaded, form a single POP. Also NetBlazer supports H.323 and vocoding standards G.723.1, G.729A and G.711. Management is handled via Web-based tools and SNMP.[11]

6.4 Terminals

Different kinds of IP telephony solutions and services require different kinds of terminals. The requirements of the terminal depend on whether the user is doing only voice conversation or also conferencing. In the first case a suitable choice would be IP-telephones while in the latter case multimedia PC with client software.

6.4.1 IP-telephones

The Selsius-Phone is an IP-based telephone that can be installed anywhere in a corporate LAN/WAN IP network. The phones are DHCP supported and do not have to be located with the Selsius-CallManager. DHCP (Dynamic Host Configuration Protocol) support for Selsius-Phones makes phone setup virtually automatic. DHCP also lets you move phones and plug them in anywhere on the IP network (local or remote ports) with no configuration. DHCP is a standard in the data network environment for configuration and management of various IP devices.[12]

6.4.2 PC programs

There are special programs, which act as an Internet telephone (e.g. Microsoft NetMeeting, VocalTec Internet Phone). Some voice conversation applications might even run in a PC with 486 processor and 16 MB RAM. However, as a general recommendation the user should have at least a Pentium 100 MHz with 32 MB RAM. For the network connection a network card or modem is required and the modem should support at least 28800 bps. For video conferencing a video camera is needed. There are plug-and-play video-capturing cameras on the market, e.g. Logitech QuickCam and Intel Create & Share. Some other systems require a video-capturing card. For voice conferencing a full-duplex soundcard, microphone and loudspeaker (or a telephone handset) is needed.

Client software are coming better every year. From Microsoft there came just a new version of Netmeeting (v3). The client interface of that new version is plainer and easier to use. Also the DSP and compression techniques have advanced. However, most of the client software are not interoperable with each other, since the software use different and even proprietary voice compression algorithms/codecs. The H.323 compliant promises guarantees nothing, because, as previously mentioned, H.323 supports several codecs.

6.5 MCU

An H.323 multipoint control unit (MCU) could be implemented either hardware or software. The number of simultaneous conferences and endpoints supported by an MCU varies by implementation. A hardware-based MCU can generally support more simultaneous users by dedicating a digital signal processor on a card to each user, often called a port. A software-based MCU typically runs on a network server, and is performance-bound by the speed of that server. A software-based MCU, however, generally costs much less than a hardware-based MCU. VideoServer is currently shipping a hardware-based H.323 MCU (Encounter NetServer), while WhitePine Software is shipping a software-based MCU (MeetingPoint). Other vendors with H.323 MCUs include Accord, Lucent, Outreach, and PictureTel.

6.5.1 VideoServer’s MCU: Encounter NetServer

Encounter NetServer supports up to 48 conference participants and resides in an Intel processor-based high performance server. The MCS performs both the multipoint controller (MC) and multipoint processor (MP) functions as defined under the H.323 standard. Scheduling and managing conferences is made very easy via www-based user interface. NetServer supports many endpoint types: H.323 audio/video/data, T.120 data over IP, H.323 Internet Telephony and POTS (phone).[13]

6.5.2 Whitepine’s MCU: MeetingPoint

MeetingPoint works with many different conferencing systems, such as Microsoft’s Netmeeting, White Pine's CU-SeeMe and Intel ProShare or TeamStation. A built-in Gatekeeper makes it possible to set bandwidth limits per conference and per user. The Gatekeeper also

- controls who can access the conference by providing authentication services
- does H.323 call management
- alternatively, MeetingPoint supports third-party Gatekeepers

Conference Administration Web Pages provide a browser-based interface for conference administration and user access to conferences. Web-based graphical user interface also manages multiple simultaneous servers and conferences.[14]

7 Current Market Situation

In the last three years many companies providing IP telephony services have emerged. Currently there are about 100 companies offering IP telephony services around the world. Many of these are small companies providing IP telephony services for a limited area. There are a few big global players, which on their own can provide global services. The service providers are usually divided into three groups according to their activities: incumbent telcos, next generation telcos and Internet service providers (ISPs). In 1997, IDT, Delta Three, and OzEmail Interline led the IP telephony service market with a combined market share of 63%.[15] According to Datapro/Gartner Group market share was couple of months ago as follows: InterVoice (8%, US$149M); Lucent (8%, US$134M); Periphonics (5%, US$93M); IBM (5%, US$90M); Voicetek (4%, US$68M); Brite Voice (3%, US$53M); Artisoft (2%, US$37M); others (65%, US$1130M).

In the Finnish market there are currently two companies providing public IP telephony services, RSL.COM and Telia. Sonera has presented IP telephony service, IP Communicator, which provides a complete corporate communication solution based on an IP network.

7.1 IP Markets Forecasts

In a recent survey conducted by Forrester Research of 52 Fortune 1000 Firms, more than 40% of telecom managers plan to move some voice/fax traffic to IP network by 1999. The voice telephony carrier services market is expected to grow to US$1 trillion worldwide by 2000. Sales of IP Gateways alone should approach US$1.81 billion by the end of 2001, a compounded five year annual growth rate of 229 percent. Furthermore, by 2001, it is project that 96% of revenues in the IP telephony market will come from the Gateway segment.[16]

OVUM made country forecasts of IP telecom service revenues in January of 1999. The sample of that is presented in Figure 10.[17]

Value of market ($1.000.000)	1999	2000	2001	2002	2003	2004	2005
Denmark	5	26	55	93	141	198	260
Finland	**5**	**24**	**51**	**87**	**132**	**188**	**249**
France	42	219	468	794	1200	1679	2200
Germany	52	285	612	1037	1564	2183	2855
Italy	38	174	366	622	951	1349	1786
UK	48	230	487	830	1262	1781	2352
China	35	298	778	1498	2512	3690	4602
Japan	94	502	1076	1829	2776	3912	5184
Russia	15	132	338	640	1060	1558	1938
Canada	83	285	532	890	1345	1941	2639
US	812	2815	5293	8858	13478	19329	26219

Figure 10. Country forecasts of IP telecom service revenues

OVUM made IP-based services market development scenario in the same research.

1998-2000:
Depending on the market that the service provider is targeting: either protecting revenue from erosion by new operators or gaining cost-conscious users, i.e. students.

2001-2003:
Remains limited offering to niche markets. Services develop faster in countries with low-medium levels of competition. All service providers recognize need to offer additional value-added services to attract and retain customers. Large businesses make some use of IP voice for corporate voice, especially internally, PSTN breakout use still limited.

2004-2005:
Multi-tier voice service based on quality and price. Large businesses and SMEs make use of IP voice for corporate traffic. Network services grow as PSTN/IP/web integration provides greater value.

8 VoIP and GSM convergence

Beside the common IP telephony, there have been many innovative development areas in VoIP. One of the biggest areas is VoIP and GSM convergence. For example, Omnitel Pronto Italia was the first wireless operator to offer Internet telephony back in 1997. Last week Omnitel reported its co-operation with Lucent. The purpose is to find out a unified, packet-based core transport network, which should lower network management costs and allow Omnitel to manage the dramatic traffic increase in its GSM network, both in terms of subscribers and minutes of conversation.[18]

Two big mobile vendors, Ericsson and Nokia, have researched last few years the possibility to integrate GSM techniques and IP (mostly H.323). Ericsson named their project as GSM on the Net, while Nokia works with GSM Intranet Office (GIO).

8.1 GSM on the Net, by Ericsson[19]

Ericsson introduces GSM on the Net as follows: “An entirely new concept for business communications, offering voice, data and multimedia services over corporate Intranets. The voice service can be either fixed or mobile (in the latter case using GSM access).” GSM on the Net is based on H.323, so all nodes in the system are connected to the corporate Intranet, and communication between system nodes and terminals is handled over the IP. The system is presented in Figure 11.

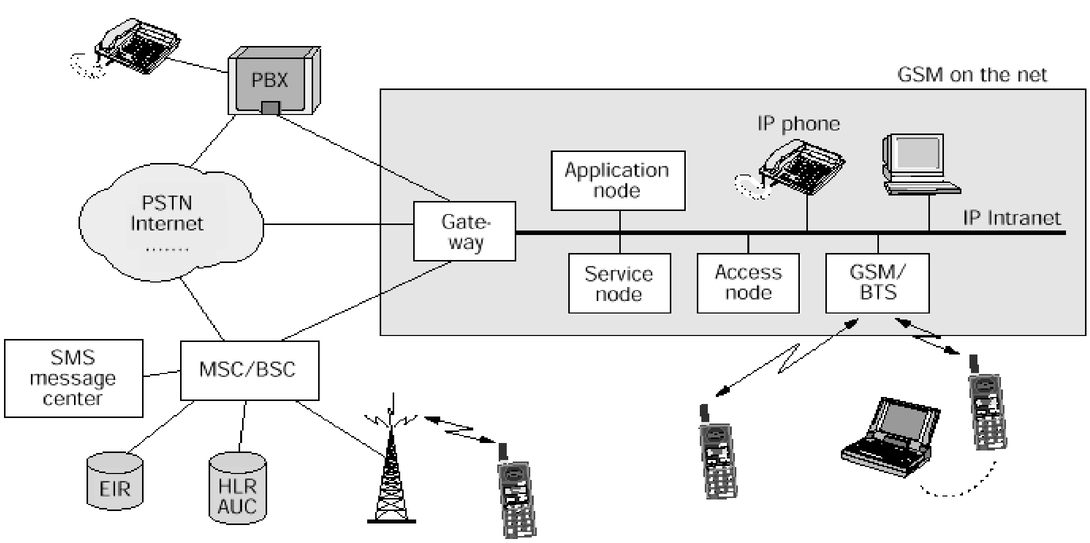

Figure 11. A GSM on the Net system

The service node, i.e. Gatekeeper in an H.323 environment, is responsible for call setup, user administration, security and other system-related services. This node enables user mobility. The functionality of the access node enables GSM terminals to access the system. In other words, access node could be expressed as the MSC (mobile switching center), the VLR (visitor location register), and the BSC (base station controller). Application node is mentioned for other application to interact with GSM on the Net. For example, following applications could be created.

- web-initiated dialing, i.e. users surfing on the net can click on a “phone” icon on the screen to retrieve from a directory the phone number of the person or organization they want to call
- directory-assisted dialing, i.e. make a call from the company’s directory instead of having to enter the digits manually
- unified messaging, i.e. all mail services (voice mail, e-mail, fax mail, video mail) can be conveniently provided in a single mailbox

The Gateway in this system is a normal H.323 Gateway, which contains some extra features mentioned for the GSM system.

Presently this system is used by few pilot-users (companies). I inquired from Ericsson, when the GSM on the Net is available in public, and they answered: “Soon. Probably during this year.”

8.2 GSM Intranet Office - GIO, by Nokia[20]

In GIO the radio access part is based on GSM technology. The concept enables seamless H.323 based voice over IP system integration into a GSM based mobile network. Phone calls can be made via the company's Intranet to various mobile phones, fixed phones or PCs throughout an office by using any standard GSM mobile. In addition, the same mobile phones can be used outside the office premises as well, where the calls are routed as normal using the operator's GSM network. By connecting a dedicated base station (BTS), like the Nokia InSite Base Station, to the corporate LAN, all Intra-office phone calls can be routed locally through the corporate Intranet.

The Nokia GSM Intranet Office solution also includes a management system to monitor the performance of the IP and GSM networks. It stores information about all network element configurations and software versions. Data from all the network elements can be forwarded to other management systems, such as the Nokia NMS2000, for further processing and analysis.

It is expected that also Nokia introduce its VoIP-GSM –product (GIO) in market during year 2000.

9 Conclusion

One of the most vital issues in IP telephony is quality of service, which in the case of voice calls means quality of the perceived voice and delay in two-way conversation. To fulfill these requirements, codecs of the end user equipment and network transmission must be efficient. Nowadays IP networks offer very limited QoS capabilities and especially this is the case in Internet. New techniques can be used to overcome this. Soon the public network and especially the Internet will offer better quality of service options by utilizing RSVP (Resource Reservation Protocol), Multi-protocol over ATM and IPv6. Today H.323 enables some QoS features in restricted network, like Intranet.

While surfing in Internet in VoIP related pages, it is very common to meet a phrase like “full H.323 compatibility”, “product XXX supports H.323 standard” etc. With its large scale of products, it is quite certain that H.323 will be the main architecture for Internet telephony for several years. All commercially available products are based (at least) on H.323, except for a large number of nonstandard systems running their own protocols. H.323 is a complete and comprising protocol, including advanced management and capability exchange functions. Also the compatibility with the existing telecommunication mechanisms through other H.32X protocols is a very competent feature. In my opinion H.323 will be the only good VoIP protocol, although version 2 includes some serious problems.

In the future IP telephony terminals will become easy to use multimedia terminals with capability to handle voice, video, still images, text etc. The future IP telephone equipment will also be capable for mobility (through GPRS/UMTS). But it is clear though, that IP telephony will not replace circuit switched telephony in the short or medium term. Instead it will be a supplement to circuit switched telephony.

10 References

[1] Lakelin, Philip; IP Telephony – Opportinity or Threat; Telecommunications, November - 1998

[2] Available at www.3com.com/technology/reseach/topics/lan_telephony.html; Last checked 9th of March

[3] A Primer on the H.323 Series Standard, DataBeam

[4] ITU, Study group 16 – Contribution 54; ITU-T recommendation H.323; October 1997

[5] Available at www.mpeg1.de/mpegfaq/mpe632.html; Last checked 11th of March

[6] Available at keskus.hut.fi/opetus/s38130/s98/ip_tel/ip_tel.html

[7] Nicklas Beijar; Signaling Protocols for Internet Telephony, H.323 and SIP; Helsinki University of Technology, Laboratory of Telecommunications Technology

[8] Available at www.vocaltec.com; Last checked 1st of June

[9] Available at www.radvision.com; Last checked 1st of June

[10] Available at www.naturalmicrosystems.com/nmss/Nmsweb.nsf/product/fusion; Last checked 3rd of June

[11] Available at www.digieurope.com/voip/components.cfm; Last checked 4th of June

[12] Available at www.selsius.com/Products/default.htm; Last checked 3rd of June

[13] Available at www.videoserver.com/html/ps_ip_conf_solutions1.html; Last checked 7th of June

[14] Available at www.wpine.com/Products/MeetingPoint/index.html; Last checked 7th of June

[15] Frost & Sullivan; World Markets for IP telephony Equipment and Services; 1998

[16] Available at www.telogy.com/our-products/golden-gateway/voip-market-overview.html; Last checked 7th of June

[17] OVUM; IP: The Impact on Telco Services and Revenues, Volume 2: The Telcos Fight Back;
January 1999

[18] Available at www.ilocus.com; Last checked 16th of June

[19] Available at www.ericsson.se/review/search.taf?_function=detail&content_uid2=10&_UserReference=B8677A8D7118B2A83765DCF0; Last checked 15th of June

[20] Available at www.nokia.com/corporate/ipmobility; Last checked at 15th of June