H.323 Protocol Suite

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Abstract

Multimedia communication has affected various areas of people's life. Correspondingly numerous standards, communication technology and networks of multimedia from different vendors are evolving rapidly. The H.323 protocol suite specified by ITU is a main technology for real-time communication of audio, video, and data over packet switched networks. It also specifies the interoperability between the packet switched networks and circuit switched networks. In this paper, the history H.323 protocol suite and the architecture of H.323 system are reviewed first. Then the signaling and connection procedures of H.323 systems are presented. Finally it is discussed that the new features in H.323 version 4 and features that are under developing or to be specified for the future release of H.323.

1   Introduction

At the present time, numerous multimedia applications and services are available. These applications and services include video and audio, synthesized video, audio and text, as well as interactivity. This multimedia information can be used for videoconferencing, telephony, video games, home shopping, video on demand, audio on demand and the like. However, this rapidly advancing multimedia technology is continuously spawning new product and applications, and their emergence has significant impact on a large number of people from all walks of life. This important and constantly evolving area comprises a number of technologies, which include multimedia computers, compression and multimedia networks as well as the transport mechanisms for these networks. The standards and technology for multimedia and multimedia communication are evolving at a prodigious pace. Videoconferencing provides for audiovisual communication as well as document sharing, including text, table and images. The video and audio information must be compressed prior to entering a communication network and decompressed when leaving it. Hardware or software codec can be used for compression and decompression of video and audio information. The multimedia communication can be established with different equipment in the way of point-to-point communication or multipoint communication.

To provide interoperability for equipment from multiple vendors, standards have been established for POTS, ISDN, PSTN, and computer networks. For example,

H.320 specifies the standards for ISDN videoconferencing; H.310, H.321, and H.322 specify the visual terminal for networks that guarantee quality of service (QoS); H.324 specifies videoconferencing over POTS modem connections; and T.120 standards provide the specifications for real-time data and audiographics conferencing [2]. Following the H.323 protocol suite for audiovisual communication will be discussed in detail.

2   What is H.323 suite

H.323 is a standard developed by the ITU. It specifies packet-based multimedia communications systems across networks, which might not provide any Qos guarantees. H.323 suite is family of standards that includes many other ITU standards as shown in Table 1 [11].

Table 1: H.323 standards

Network	Non-guaranteed bandwidth packet-switched networks (e.g. IP)
Video	H.261, H.263
Audio	G.711, G.722, G.728, G.723, G. 729
Call Signalling and media packetisation	H.225.0
Call Control	H.245
Multipoint	H.323
Data	T.120

The H.323 standard is a principal technology for the transmission of real-time audio, video, and data communication over packet-based networks. It provides both multipoint and point-to-point sessions. H.323 defines the components, protocols, and procedures providing multimedia communication over packet-based networks, which include Inter-Networks (including the Internet), Local Area Networks, Enterprise Area Networks, Metropolitan Area Networks, and Intra-Networks [12]. Packet based networks also include point-to-point connections or dial up connections over the GSTN or ISDN which can use an underlying packet based transport. H.323 can be used in a variety of mechanisms, which include audio and video (video telephony); audio only (IP telephony); audio, video and data; video and data; multipoint-multimedia communications.

The H.323 standard is part of the H.32X family of recommendations specified by ITU-T. The other recommendations of the family define multimedia

communication service over different networks are shown in Table 2 [11].

Table 2: H.32X recommendations

ITU standard	Network
H.320	ISDNs
H.321, H.310	B-ISDNs
H.324	SCNs
H.323	Non-guaranteed bandwidth packet switched networks
H.322	LANs that provide guaranteed Qos

Interoperability with other multimedia networks is one of the primary goals in the development of the H.323 standard.

3   H.323 Version Suites

Since the first version of H.323 was approved in 1996, it has had 4 versions till the approval of H.323 Version 4 on November 17, 2000 [7].

3.1   H.323 Version 1

"Visual telephone systems and equipment for local area networks which provide a non-guaranteed quality of service", was published in 1996 [4] and was designed for local area networks. The first thing what companies tried to do was use H.323 in WAN, large private VoIP networks, and the Internet. It worked very well.

Recognizing the fact that H.323 was much more than a LAN protocol, the name was changed to H.323 Version2 in 1998 [10].

3.2   H.323 Version 2

H.323 "Packet-based multimedia communications systems" was approved in January of 1998. It brought in H.235 Security (Authentication of participant, Integrity of data, Encryption, and digital signature), Fast Connect, Supplementary Services (H.450.1 Signaling protocol, H.450.2 Call Transfer, and H.450.3 Call Diversion), Integration of data conferencing with T.120, and Scalability features (Alternate Gatekeepers, Time to Live, and Pre-granted ARQs) [9][10].

3.3   H.323 Version 3

H.323 version 3 was approved on September 30, 1999. It introduced a few modest improvements, mostly geared for better PSTN integration and scalability. However, H.323 has progressed substantially, mostly in the form of new Annexes to H.323 and H.225.0 that add considerable value to the overall H.323 system architecture [8].

3.4   H.323 Version 4

Many new enhancements have been introduced into the protocol H.323 Version 4, which was approved November 17, 2000. It contains enhancements in a number of important areas, including reliability, scalability, and flexibility. New features help facilitate more scalable Gateway and MCU solutions to meet the growing market requirements [7][10].

4   H.323 Architecture

The H.323 standard specifies the elements, protocols, and procedures providing multimedia communication over packet-based networks (see Figure 1 [10])
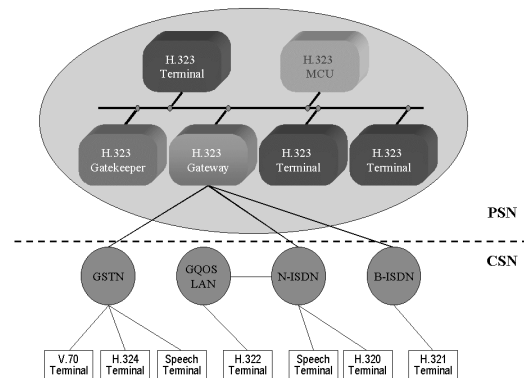

Figure 1: H.323 architecture

An H.323 system provides point-to-point or multipoint multimedia communication services. It has four main elements including terminals, gateways, multipoint control units (MCUs), and gatekeeper [12]. Terminals, gateways, and MCUs are also called endpoints.

4.1   Terminals

Terminals include Video I/O equipment, Audio I/O equipment, User Data Applications, and System Control User Interfaces. Terminals can be used for real-time bidirectional multimedia communications. An H.323 terminal can either be a personal PC or a stand-alone device, running an H.323 and multimedia applications. It supports audio, video and data communications. An H.323 terminal plays a key role in IP-telephony due to is its basic service of audio communications. Interworking with other multimedia network is the primary goal of H.323. The H.323 terminals are also compatible with the terminals on the networks given in Table 2 [4][13].

4.2   Gateways

Gateways connect H.323 networks to other networks, including the PSTN, ISDN, H.320 systems, etc. The connectivity of dissimilar networks is achieved by translating protocols for call setup and release, converting media format between different networks [4][12]. An example of Gateway, which connects H.323 system to PSTN, is given in Figure 2 [3].
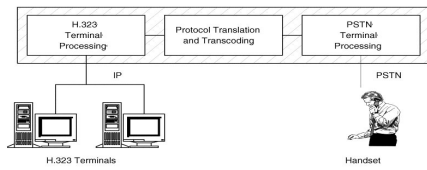
Figure 2: H.323/PSTN Gateway

4.3   MCUs

MCUs are responsible for managing multipoint conferences of three or more H.323 terminals. A two-terminal point-to-point conference can be expanded to a multipoint conference. The MCU consists of a mandatory multipoint controller (MC) and optional multipoint processor (MP). The MC supports the negotiation of capabilities with all terminals in order to insure a common level of communications. It can also control the resources in the multicast operation. The MP is the central processor of the voice, video, and data streams for a multipoint conference [13].

The MCU may (or may not) control three types of multipoint conference (Figure 3 [4]):

•   Centralized multipoint conference

All participating terminals communicate with the MCU point-to-point. The MC manages the conference, and the MP receives, processes, and sends the voice, video, or data streams to and from the participating terminals.

•   Decentralized multipoint conference

The MCU is not involved in this operation. Rather the terminals communicate directly with each other through their own MCs. If necessary, the terminals assume the terminals take the responsibility for summing the received audio streams and selecting the received video signals for display.
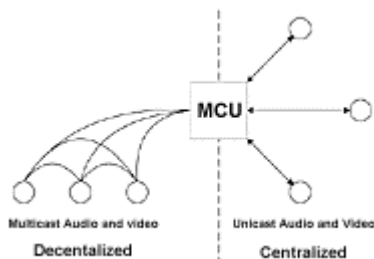

Figure 3: Multipoint conference

•   Hybrid multipoint conference

This conference is a mix of the centralized and decentralized modes. The MCU keeps the operations transparent to the terminals (see Figure 4 [4]).
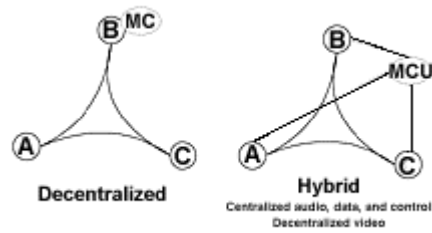

Figure 4: Hybrid multipoint conference

4.4   Gatekeepers

Gatekeepers are used for admission control and address resolution. A gatekeeper may allow calls to be placed directly between endpoints or it may route the call signaling through itself. A gatekeeper is also responsible for the services of band control, accounting, and billing. A single gatekeeper manages a collection of Terminals, Gateways, and MCUs forming a zone. A zone is logical association of these components and may span multiple LANs [4] (Figure 5 [3]).
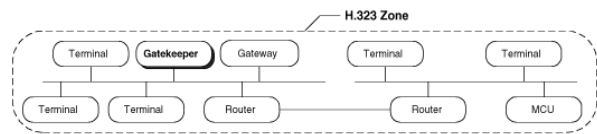

Figure 5: H.323 zone

5   The H.323 Protocol Stack

The H.323 suite consists of a set of standards. H.323 cites the use of the others shown in Figure 6 [13].

For audio applications, the minimum requirement is the support of recommendation G.711 (64 kbps channel). Other voice codec standards cited by H.323 are G.722 (48, 56, and 64 kbps channels), G.723 (5.3 and 6.3 kbps channels), G.728 (16 kbps channel), G.729 (8 kbps channel) [4].

The H.245, control protocol for multimedia communication, is used during an initial handshake between the machines to determine the audio encoding algorithm, terminal capabilities, and media channels. The terminals should be capable of sending and receiving different audio streams. After H.245 has completed the agreements on the terminals' capabilities and media channels, the H.225, call signaling and setup protocol, is used to format the audio stream.

H.261 is video coding standard. It was designed for data-rates which are multiples of 64kpbs. H.261 supports two resolutions, QCIF (Quarter CIF) and CIF (Common Intermediate format). If video is supported, the H.323 terminals must code and decode the video streams in accordance with H.261 QCIF. Options are available, but they must use the H.261 or H.263 specifications. The coding algorithm of H.263 is similar to that used by

H.261, however with some improvements and changes to improve performance and error recovery. H.263 supports five resolutions, QCIF, CIF, SQCIF (Sub-QCIF), 4CIF, and 16CIF.

Data support is through T.120, and the various control, signaling, and maintenance operations which are provided by H.245, Q.931, and the Gatekeeper specification.

The audio and video packets must be encapsulated into the Real-time Transport Protocol (RTP) and carried on a UDP socket pair between the sender and the receiver. The Real-Time Control Protocol (RTCP) is used to assess the quality of the sessions and connections as well as to provide feedback information among the communication parties. The data and support packets can operate over TCP or UDP [4][13].
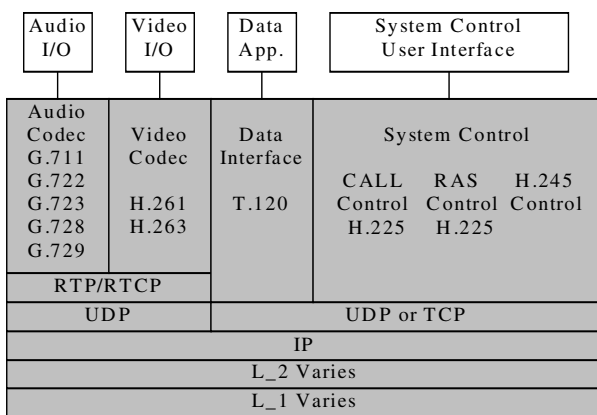
Audio I/O	Video I/O	Data App.	System Control User Interface		
Audio Codec G.711 G.722 G.723 G.728 G.729	Video Codec H.261 H.263	Data Interface T.120	System Control CALL Control H.225	RAS Control H.225	H.245 Control
RTP/RTCP					
UDP		UDP or TCP			
IP					
L_2 Varies					
L_1 Varies					

Figure 6: H.323 protocol stack

6   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 [4].

6.1   Addresses

In H.323 system, each entity has at least one Network Address (e.g. IP 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). For each Network address, each H.323 entity may have several Transport layer Service Access Point (TSAP) identifiers. These TSAP Identifiers allow multiplexing of several channels sharing the same Network address. An endpoint may also have one or more alias addresses associated with it. An alias address may represent the endpoint or it may represent conferences that the endpoint is hosting. The alias addresses provide an alternate method of addressing the endpoint [4].

6.2   Registration, Admission and Status (RAS)

The RAS channel is used between H.323 endpoints and gatekeepers for gatekeeper discovery, endpoint registration, endpoint location, and admission control. The RAS messages are carried on a RAS channel that is unreliable. Hence, RAS message exchange may be associated with timeouts and retry counts.

- Gatekeeper discovery

Gatekeeper discovery is the process an endpoint uses to determine which Gatekeeper to register with. The gatekeeper discovery can be done statically or dynamically. In static discovery, the endpoint knows the transport address of its gatekeeper a priori. In the dynamic method of gatekeeper discovery, the endpoint multicast GRQ message on the gatekeeper's discovery multicast address. One or more gatekeepers may respond with GCF message [4].
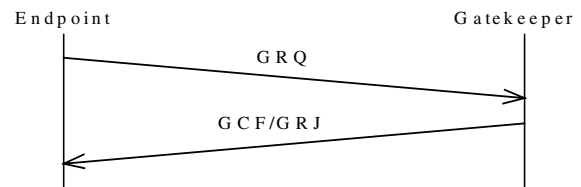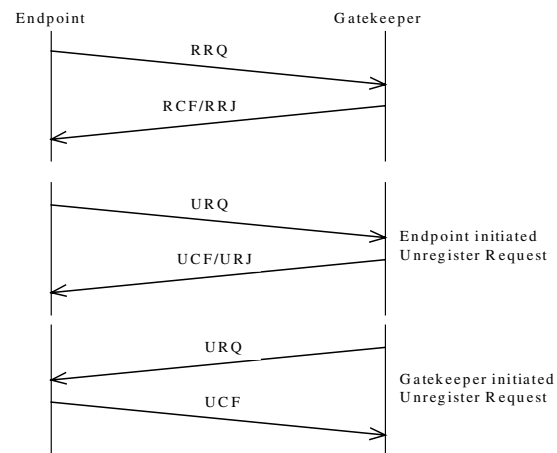

Figure 7: H.323 - Gatekeeper discovery

- Endpoint registration

Endpoint registration is the process by which an endpoint joins a Zone, and informs the Gatekeeper of its Transport Address and alias address. All endpoints register with a gatekeeper as part of their configuration process. Registration occurs before any calls are

attempted and occurs periodically as necessary [4] (see Figure 8).

Figure 8: H.323 Endpoint registration

- Endpoint location

Endpoint location is a process by which the transport address of an endpoint is determined and given its alias name or E.164 address [4].

- Other Controls

The RAS channel is also used for other controls, such as admission control, to restrict the entry of an endpoint into a zone; bandwidth change, to modify the call bandwidth during a call; and disengagement control, to disassociate an endpoint from a gatekeeper and its zone [4].

6.3 H.225 Call Signaling and H.245 Control Signaling

- H.225 Call signaling

The H.225 call signaling is used to set up connections between H.323 endpoints, over which the real-time data can be transported. The call signaling channel is a reliable channel, which is used to carry H.225 (adopted a subset of Q.931 messages and elements) call control messages. For example, H.225 protocol messages are carried over TCP in an IP based H.323 network [4].

In networks that do not contain a Gatekeeper, call signaling messages are passed directly between the calling and called endpoints. It is called direct call signaling. In networks that do contain a Gatekeeper, the H.225 messages are exchanged either directly between the endpoints or between the endpoints after being routed through the gatekeeper. It is called gatekeeper-routed signaling. The method chosen is decided by the gatekeeper during RAS-admission message exchange.

Gatekeeper-Routed Call Signaling
The admission messages are exchanged between endpoints and the gatekeeper on RAS channels. The gatekeeper receives the call-signaling messages on the call-signaling channel from one endpoint and routes them to the other endpoint on the call-signaling channel of the other endpoint (see Figure 9)[4].
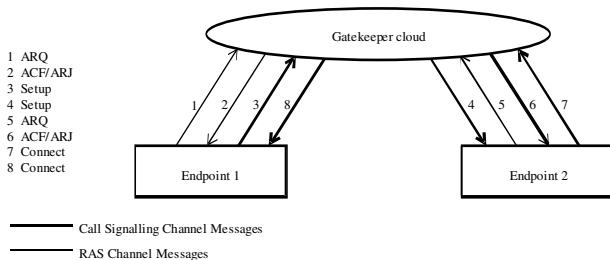

1 ARQ
2 ACF/ARJ
3 Setup
4 Setup
5 ARQ
6 ACF/ARJ
7 Connect
8 Connect

Call Signalling Channel Messages
RAS Channel Messages

Figure 9: H.323-Gatekeeper routed call signaling

Direct Call Signaling
During the admission confirmation, the gatekeeper indicates that the endpoints can exchange call-signaling messages directly. The endpoints exchange the call signaling on the call-signaling channel (see Figure 10) [4].
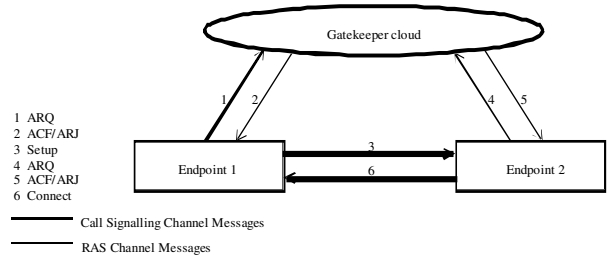

1 ARQ
2 ACF/ARJ
3 Setup
4 ARQ
5 ACF/ARJ
6 Connect

Call Signalling Channel Messages
RAS Channel Messages

Figure 10: H.323-Direct endpoint call signaling

- H.245 Control Signaling

When Gatekeeper routed call signaling is used, there are two methods to route the H.245 channel. In the first method, the H.245 control channel is established directly between the endpoints (see figure 11). In the second method, the H.245 control channel is routed between the endpoints through the Gatekeeper (see Figure 12). This method allows the Gatekeeper to redirect the endpoints through the Gatekeeper. This 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.

When direct endpoint call signaling is used, the H.245 control channel can only be connected directly between the endpoints [4].
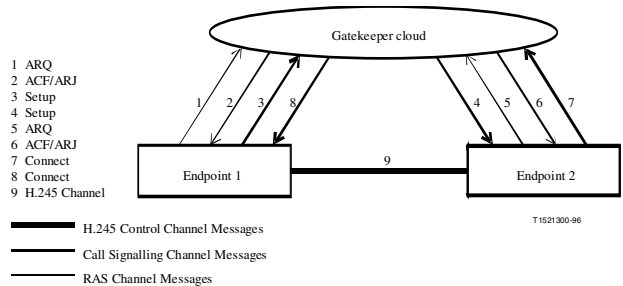

1 ARQ
2 ACF/ARJ
3 Setup
4 Setup
5 ARQ
6 ACF/ARJ
7 Connect
8 Connect
9 H.245 Channel

T1521300-96

H.245 Control Channel Messages
Call Signalling Channel Messages
RAS Channel Messages

Figure 11: H.323 – H.245 control channel connection between endpoints

1 ARQ
2 ACF/ARJ
3 Setup
4 Setup
5 ARQ
6 ACF/ARJ
7 Connect
8 Connect
9 H.245 Channel
10 H.245 Channel

T1521310-96

H.245 Control Channel Messages
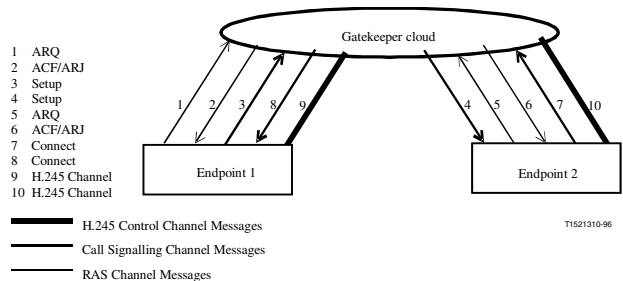Call Signalling Channel Messages
RAS Channel Messages

Figure 12: H.323 – Gatekeeper routed H.245 control

7 Connection Procedures

The connection procedures of the H.323 systems communication are made in the steps of Call setup,

Initial communication and capability exchange, Establishment of audiovisual communication, Call services, and Call termination. This section uses an example network, which contains two endpoints connecting to a gatekeeper to illustrate the whole connection steps.

7.1 Step A: Call setup

Call setup can be in all following cases:
- all combinations of Direct Routed Call signaling (DRC)/Gatekeeper Routed Call signaling (GRC), same or different Gatekeepers;
- Fast connect procedures;
- call forwarding using facility (restarts the procedure);
- and setting up conferences [6].

Figure 13 illustrates the call setup process with the example of both endpoints registered to the same Gatekeeper. It assumes direct call signaling [12].
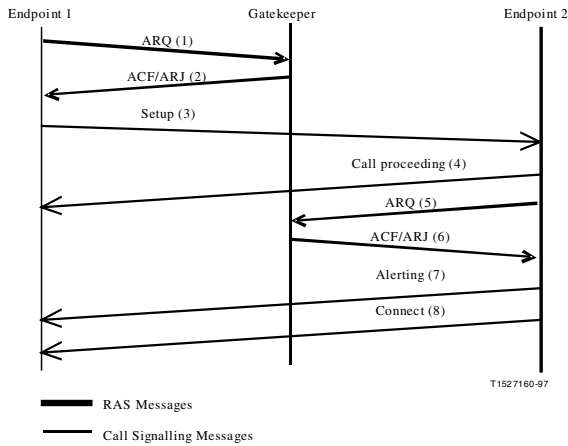

Figure 13: Call Setup

7.2 Step B: Initial communication and capability exchange

This step includes the procedures of Capability exchange, Master/Slave determination, and H.245 tunneling [4].

Once both sides have exchanged call setup messages from step A, the endpoints shall establish the H.245 Control Channel. The procedures of H.245 are used over the H.245 Control Channel for the capability exchange and to open the media channels.

The H.245 Master-slave determination procedures are used to resolve conflicts between two endpoints which can both be the MC for a conference, or between two endpoints which are attempting to open a bidirectional channel. Figure 14 is an example H.323 control signaling flows.
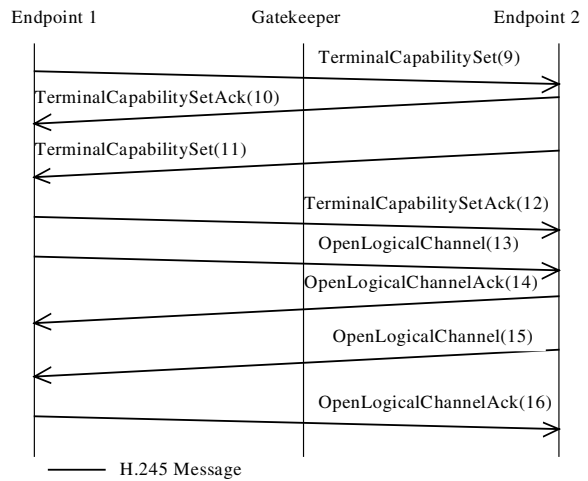

Figure 14: H.323 Control Signaling Flows

7.3 Step C: Establishment of audiovisual communication

Following the exchange of capabilities, master-slave determination, and opening of the logical channels for the various information streams, the audio and video streams, which are transmitted in the logical channels setup in H.245, are transported over dynamic Transport layer Service Access Point (TSAP) Identifiers using an unreliable protocol. Data communications, which are transmitted in the logical channels setup in H.245, are transported using a reliable protocol. Figure 15 is an example of illustrating the H.323 media stream and media control flows [4][11].
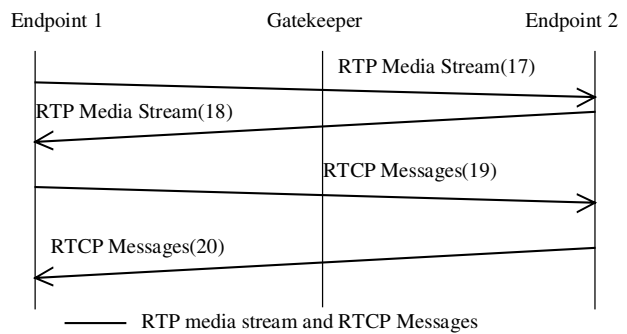

Figure 15: Media Stream and Media Control Flows

7.4 Step D: Call services

Call services include Bandwidth change, Status Information request for management, Conference expansion, multicast cascading, and H.450 Supplementary Services [4].
- Bandwidth changes

Call bandwidth is initially established and approved by the Gatekeeper during the admission exchange. At any

time during a conference, the endpoints or Gatekeeper may request an increase or decrease in the call bandwidth. An example of Bandwidth changes is given in Figure 16.
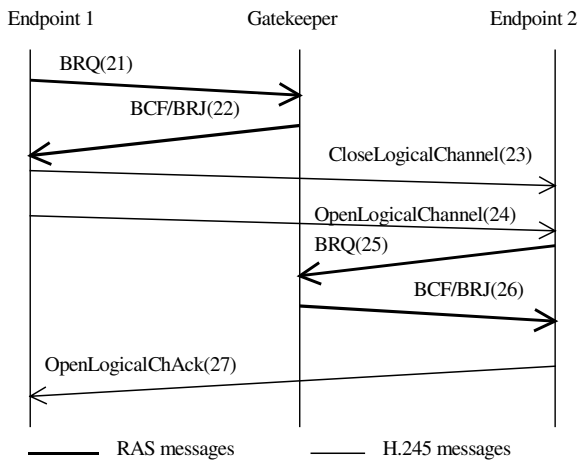

Figure 16: H.323 – Bandwidth Change

• Status
Status is procedures of gatekeeper determining the work status, on/off or failure, of the endpoints. The Gatekeeper may use the H.225 Information Request (IRQ) /Information Request Response (IRR) messages to poll the endpoints periodically.

• Conference expansion
Conference expansion is the procedure for expanding a point-to-point conference involving an MC to a multipoint conference. First, a point-to-point conference is created between two endpoints. At least one endpoint or the gatekeeper must contain an MC. Once the conference has been created, the conference may be expanded to multipoint conference by any endpoint in the conference inviting another endpoint into the conference through the MC, or an endpoint joins an existing conference by calling an endpoint in the conference. Figure 17, 18 [4] illustrate the H.245 Control Channel topology for the Direct Call Signaling model, and the Gatekeeper routed Call Signaling model.
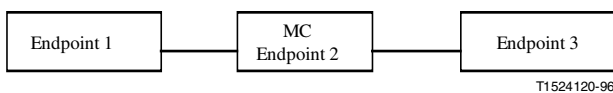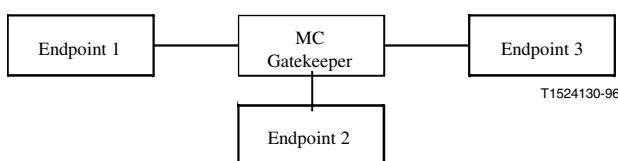

Figure 17: Direct Call Signaling model

Figure 18: Gatekeeper routed Call Signaling model

• Multicast cascading
Multicast cascading is the case when a call is established between the entities containing the MCs, and the H.245 Control Channel is opened, the active MC (Master/Slave procedure) may active the MC in a connected entity. Once the cascade conference is established, either the master or slave MCs may invite other endpoints into the conference. There is only one master MC in a conference. A slave MC can only be cascaded to a master MC.

• H.450 Supplementary services
The H.450 supplementary services are optional to H.323 systems. These services include call forward, call hold, call waiting, message waiting indication, and name identification etc.

7.5 Step E: Call termination

Call termination can be made by any endpoint when video, audio, or data transmissions are at end. Correspondingly all logical channels for video, audio, or data are closed. Terminating a call may not terminate a conference. It can be done by MC that the terminating of a conference. Figure 19 [12] illustrates the call release procedure.
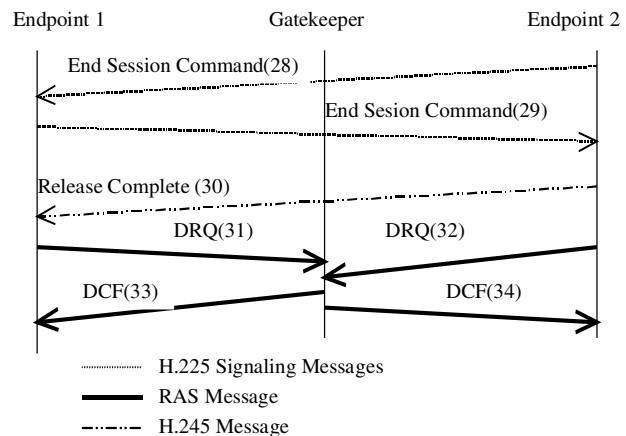

Figure 19: H.323 Call Release

8 New Feature of H.323 Version 4

H.323. Version 4 was approved on November 17, 2000. It contains enhancements in a number of important areas including scalability, reliability, flexibility, services, must have features, and generic extensibility framework [7][10][1].

8.1 Scalability, Reliability, and Flexibility

The H.323 Version 4 enhances the scalability of H.323 systems in the areas including Gateway Decomposition with H.248, Additive Registrations, Alternate Gatekeepers, and Endpoint Capacity Reporting.

• Gateway Decomposition

Traditional Gateways were designed that both media and call control were handled in the same box. Recognizing the need to build larger, more scalable gateway solutions for carrier solutions, the ITU-T worked jointly with the IETF produced the Recommendation H.248, which describes the protocol between the Media Gateway Controller (MGC) and the Media Gateway (MG). H.323 version 4 supports the decomposition of Gateway into Media Gateway Controller (MGC) and Media Gateway (MG).

The decomposed Gateway separates the MGC function and the MG function. Multiple MGs may exist to allow the decomposed Gateway to scale to support much more capacity than a composite Gateway. The communication between the MGC and MGs is done through H.248 (see Figure 20 [11]).
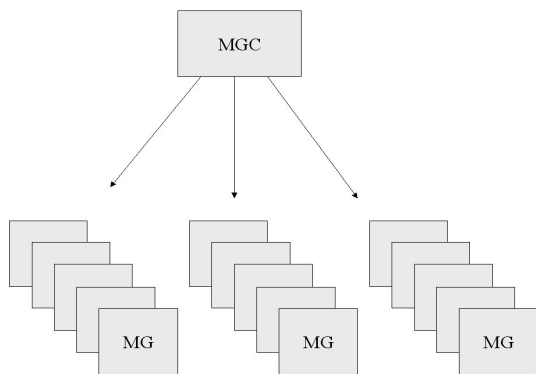

Figure 20: Decomposition Gateway

- Alternate Gatekeepers

The architecture of alternate Gatekeepers is shown in Figure 21 [11]. By using Alternate Gatekeepers, endpoints can continue functioning when the communication between the endpoints and one or more Gatekeepers. It increases the reliability and never loses calls.
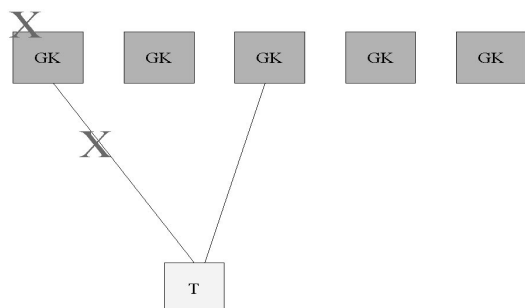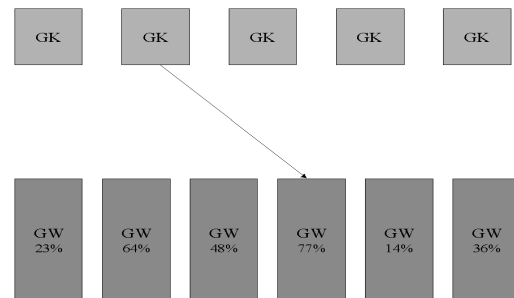

Figure 21: Alternate Gatekeepers

- Endpoint Capacity reporting

H.323 endpoints report capacity to Gatekeepers. By utilize endpoint capacity reporting, Gatekeepers may select an endpoint that is best capable of handling the call. It is very useful for large scale deployments of Gateways, and extremely increases the availability (see Figure 22 [11]).

* GK selects the GW with the most capacity.
* *H.323 terminals report capacity in absolute terms, not in percentages.*

Figure 22: Endpoint Dispatcher

8.2 Services

One of the most important features of a VoIP protocol is its ability to provide services to the service provider and end users. H.323 has a rich set of mechanisms to provide supplementary services. Version 4 introduces a few more supplementary services to strengthen the protocol in this regard. These services mainly include HTTP-based Service Control, Stimulus-based Control, and Call completion [1][10][7].

- HTTP-based Service Control

H.323 version 4 specifies a means of providing HTTP-based control for H.323 devices. With HTTP-based control, service providers have the ability to display web pages to the user with meaningful content that ties into the H.323 systems. In essence, it is a third party call control mechanism that utilizes a separate HTTP connection for control.

- Stimulus-based Control

H.323 version4 provides a new "stimulus-based" control mechanism. With this mechanism, an H.323 device may communicate with a feature server to provide the user with various services. The H.323 endpoint may possess some intelligence, but some intelligence may reside only in the feature server or multiple feature servers. The features may be numerous. Any new features may be added to the feature servers without the delay by standard procedure.

- Call completion

This is a new H.450 supplementary service, which provides a standard means of allowing calls to complete when the user is either busy or there is no answer.

8.3 "Must Have" Features

The features included are listed below [7][10][5]:

- Usage Information Reporting

To help providing accurate billing information, the Gatekeeper can request the endpoint to provide usage information reporting to the Gatekeeper at various times during the call, including at the beginning of the call, during the call, and at the end of the call.

- Caller Identification

H.323 Version 4 contains complete information for providing caller identification services with H.323.

- Tones and Announcements

H.323 version 4 details the procedure for indicating the presence of in-band tones and announcements. Such tones and announcements are often heard when the destination number is incorrect or unreachable.

In addition to in-band tones and announcements, the Gatekeeper may signal an endpoint to play specific announcements at various times: pre-call, mid-call, or end-call.

- Alias Mapping

When routing calls, a telephone number in the IP-world may not be sufficient for proper routing into the SCN. In addition, it might be that a service provider would like to use the same Gateways to provide Virtual Voice Private Networks, but need some intelligence in a device to perform proper mapping. With Version 4, a Gateway, for example, can indicate that it can perform alias mapping at either the ingress or egress side of a call. This will reduce the number of malformed numbers, as well as provide a means for providing Voice Virtual Private Network (VVPN) services.

- Better Bandwidth Management (multicast)

Prior to H.323 Version 4, and endpoint could request much more bandwidth than it actually needs, and thus, cause wasting network resources. With Version 4, it is mandatory that an endpoint made bandwidth requests with a lower value if, indeed, the endpoint is using less bandwidth than it had initially indicated in the ARQ.

In addition, managing bandwidth for multicast sessions has been nearly impossible since, unless the Gatekeeper routed the H.245 signaling and carefully monitored the media channels that were opened, it could not determine whether two endpoints that request bandwidth are actually requesting bandwidth for a multicast session or unicast session. This becomes a much bigger issue when many people are participating in a multipoint multicast conference. With Version 4, specific details about the media channels are conveyed to the Gatekeeper in (Information Request Response) IRR messages (if the Gatekeeper requests them), so that the Gatekeeper can better control bandwidth utilization.

- Fax Enhancements

Version 4 of H.323 allows an endpoint to be able to initiate a voice call and then switch to fax at some point. It allows an IP-based fax device to operate in a similar manner as today's PSTN fax devices. Version 4 also

enhanced to utilize TCP for carrying fax data. Previously, UDP was the only real option for carrying fax data.

- Tunneling other protocols

H.323 is often used to inter-work between two circuit networks. To provide better inter-working, Version 4 provides a mechanism whereby QSIG (Signaling between the Q reference points) and ISUP may be tunneled without translation essentially. H.323 may act as a transparent tunnel for those non-H.323 signaling protocols (see Figure 23 [5]).
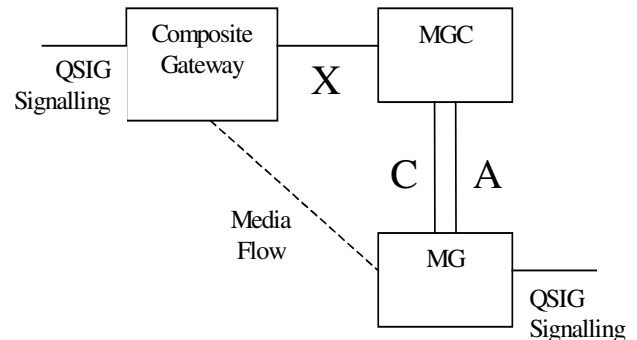

Figure 23: H.323 – QSIG tunneling example

- H.323 specific URL

Version 4 introduced URL scheme "h323". The H.323 URL allows entities to access users and services in a consistent manner. The form of the H.323 URL is "h323:user@host", where "user" is a user or service and "host" might be the Gatekeeper that can translate the URL into a call signaling address.

- Call Credit-related capabilities

H.323 v4 provides the means of communicating available funds or for the Gateway to control early call termination based on available funds for the prepaid IP telephony. H.323 v4 adds these features to the RAS protocol.

- Multiplexing audio and video

One weakness with the current usage of RTP is difficulty in synchronizing the separate audio and video streams. Version 4 now includes an optional procedure, which allows both video and audio to be multiplexed in a single stream. This will assist endpoints in synchronizing video and audio.

- DTMF Relay via RTP

H.323 version 4 allows an endpoint to utilize RFC 2833 "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals" to send and receive DTMF digits.

8.4 Further Features are under developing on H.323

ITU-T is working or is going to work on some of further enhanced features of H.323, which include Generic Extensibility Framework, Protocol Inter-working, Mobility, and Robustness [1][10].

• Generic Extensibility Framework

The Generic Extensibility Framework (GEF) introduces new means by which H.323 may be further enhanced or extended with optional features, which does not require changes to the current ASN.1 syntax

• Inter-working or integrating with other protocols.

The inter-working or integrating with newly developed protocols may need to be developed. These protocols include SIP, H.248/Megco, and Bearer Independent Call Control (BICC).

SIP is gaining in popularity as a VoIP protocol. H.248/Megaco may find its way into many "media gateway" devices, ranging from residential gateways to large-scale service provider gateways. The Bearer Independent Call Control (BICC) protocol will compete with both H.323 and SIP for a place in the service provider network. Making H.323 work with is also important.

• Mobility

Mobility includes terminal mobility, user mobility, and service mobility. To implement the mobility of H.323, the functions of mobility management need to be defined, which include Home Location Function (HLF), Visitor Location Function (VLF), Authentication Function (AuF), and Inter-working Function (IWF) (see Figure 24 [1])
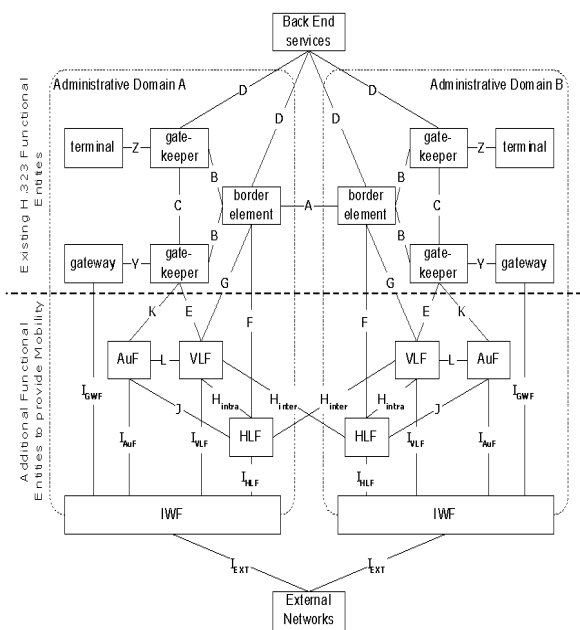

Figure 24: H.323 - Mobility

• Robustness

Robustness is under developing, it requires refining the architecture for recovery from crashes. Currently two architectures are proposed: small scale systems and large scale systems.

For small scale systems, the architecture requires each element responsible to detect failure of the others. If one element is fail in the system, the others can go to the backup element. Some state information of the failure element is then need to be provided. For large scale system, the architecture is very complex and need to be specified.

9   Comments on H.323

H.323 is a very complex system with all of kinds of features for multimedia communications, but not every part of H.323 has to be implemented when building a powerful and useful system. Multimedia over IP, in itself, has a certain amount of complexity associated with it. It results in that the system for implementing the inter-working between different multimedia systems with various features and services is inevitably complex. What the complexity does exist in H.323 system for a reason, the reason may become even more evident as video, audio, and data conferencing become more prevalent [10].

H.323 allows the use of multiple codecs. In the systems, there is a good reason for using each of the codecs.

Gatekeepers are optional in H.323 system. They provide consistent means for H.323 endpoints to perform address resolution, and may perform inter-working between simple H.323 (set devices) and more protocol-complete H.323 entities. Gatekeepers can act as a platform from which powerful new IP-based services can be built and provided.

H.323 is scalable. Service providers can deploy H.323 networks in small scale or large scale depending on the expected features and services.

H.323 is a proven technology used in large networks. It has excellent integration with PSTN.

Multimedia conferencing shows the real potential for H.323 used in multimedia communication

Many equipment manufacturers, software vendors, and service providers have built products and services supporting H.323. It greatly supports the success of H.323.

With the constantly coming of new technologies, for example BICC, H.323 has big pressure on keeping its place in the service provider network.

10   Conclusions

As just presented, H.323 is organized around four major facilities: (a) terminals, (b) Gateways (which can perform protocol conversion), (c) Gatekeeper (bandwidth manager), and (d) multipoint control units (MCUs), responsible for multicasting. The H.323 standard is a principal technology for the transmission of real-time audio, video, and data communication over packet-based networks. It provides both multipoint and point-to-point sessions. One of the primary goals of developing H.323 standards is to provide the interoperability between packet switched networks and

other multimedia networks. H.323 is a rich and complex specification. Especially the version 4 is a powerful system for multimedia communication. It contains enhancements in a number of important areas, including, scalability, reliability, flexibility, supplementary services, and new features. Future release will be more powerful. Especially the inter-working or integrating with other newly developed protocols will strengthen its position in the multimedia communication area. Mobility will greatly increase flexibility for using H.323 system in the fields of terminal mobility, user mobility, and service mobility. Of course, mobility will also greatly increase the complexity of the H.323 system.

Even though H.323 is powerful system for multimedia communication, if has faced great compression from some newly developed protocols, such as SIP, H.248/Megco, and BICC. Reducing the complexity of H.323, and simplifying its usage will evidently improve its leading position in fast changing multimedia communication world.

11 Acronyms

ACF/ARJ – Admission Confirm/Reject
ARQ – Admission Request
AuF – Authentication Function
BCF/BRJ – Bandwidth Confirm/Reject
BICC – Bearer Independent Call Control
B-ISDN – Broadband ISDN
BRQ – Bandwidth Request
CIF – Common Intermediate Format
DCF/DRJ – Disengage Confirm/Reject
DRC – Direct Routed Call signaling
DRQ – Disengage Request
DTMF – Dual-Tone Multi-Frequency
GCF/GRJ – Gatekeeper Confirm/Reject
GEF – Generic Extensibility Framework
GK – Gatekeeper
GQOS – Guaranteed Quality of Service
GRQ – Gatekeeper Request
GSTN – General Switched Telephone Network
GRC – Gatekeeper Routed Call signaling
GRQ – Gatekeeper Request
HLF – Home Location Function
IRR – Information Request Response
IRQ – Information Request
ISDN – Integrated Services Digital Network
ISUP – ISDN User Part
ITU – International Telecommunication Union
IWF – Inter-working Function
MC – Multi-point Controller
MCU – Multi-point Control Unit
MG – Media Gateway
MGC – Media Gateway Controller
MP – Multi-point Processor
N-ISDN – Narrow-band ISDN
PISN – Private Integrated Services Network
POTS – Plain Old Telephone Service
PSN – Packet Switched Network
PSTN – Public Switching Telephone Network
QCIF – Quarter Common Intermediate Format
QoS – Quality of Service
QSIG - Signaling between the Q reference points
RAS – Registration/Admission Status
RCF/RRJ – Registration Confirm/Reject
RRQ – Registration Request
RTCP – Real Time Control Protocol
RTP – Real-time Transport Protocol
SCN – Switched Circuit Network
SIP – Session Initiation Protocol
SQCIF – Sub Quarter Common Intermediate Format
TCP – Transmission Control Protocol
TSAP – Transport Service Access Point
UCF/URJ – Unregistration Confirm/Reject
UDP – User Datagram Protocol
URQ – Unregistration Request
VLF – Visitor Location Function
VoIP – Voice over Internet Protocol
VVPN – Voice Virtual Private Network

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