H.323 Series Primer

White Paper

A discussion of the H.323 standard and its advantages, architecture, and applications

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Introduction

H.323 is the International Telecommunication Union (ITU) "series H" recommendation for real-time multimedia communications and conferencing over packet-based networks such as the Internet or corporate intranets. The H.323 recommendation defines how H.323-compliant components set up calls, exchange compressed audio and video, participate in multiple-unit conferences, and operate with non-H.323 endpoints.

H.323 provides the framework for creating interoperable multimedia communications and conferencing solutions for packet-based networks. This framework enables solution providers to create H.323 standards-based products that interoperate with other H.323-compliant solutions. It also enables users to choose the H.323-based solutions that suit their requirements, without having to be concerned about compatibility. These advantages are making H.323 the globally accepted standard for audio/video/data communications between user terminals, network equipment, and assorted services on packet-based networks.

This White Paper examines H.323 and its evolution, features, and applications. This White Paper also provides technical information about H.323 components and architecture. At the end of this White Paper is a glossary of terms. If you find a term in blue that you do not understand, click it to go to the term's definition in the glossary; clicking the term in the glossary returns you to your original location in this PDF file. Following the glossary is a bibliography that lists additional information sources about H.323.

Overview

H.323 is a comprehensive recommendation that:

- Defines how audio and videoconferencing systems communicate over packet-switched networks that do not guarantee Quality of Service (QoS).
- Addresses call control and management for point-to-point and multipoint conferences.
- Handles QoS issues by employing a centralized gatekeeper to manage media traffic, bandwidth, and user participation. (Gatekeepers and other H.323 components are described in detail later in this Paper.)
- Defines gateway functions that enable calls to connect from a LAN to the public-switched telephone network (PSTN) and H.32*x*-compliant terminals.

H.323 is referred to as an "umbrella" recommendation because it contains references to other ITU recommendations, including:

- H.225.0 packet and synchronization
- H.245 control
- H.261 and H.263 video codecs

- G.711, G.722, G.728, G.729, and G.723 audio codecs
- The T.120 series of multimedia communications protocols.

H.323 is a member of a larger family of recommendations known as H.32*x*. Each H.32*x* recommendation pertains to a specific network and a corresponding set of ITU standards for:

- Call setup and processing
- Exchange of compressed audio and video data
- Channel aggregation
- Media stream multiplexing
- Data conferencing

For example:

- H.320 addresses Integrated Services Digital Network (ISDN) communications.
- H321 addresses Asynchronous Transfer Mode (ATM) communications.
- H.324 addresses communications over the public switched telephone network (PSTN).

Of all the H.32*x* recommendations, H.323 is particularly noteworthy its broad acceptance. Companies from computer telephony and VoIP vendors to personal computer manufacturers, communications systems vendors, and operating system developers are embracing H.323. This global support is revolutionizing IP telephony and teleconferencing interoperability. Worldwide adoption of H.323 is likely to grow as more and more VoIP networks rely on H.323 gateways to adapt traditional telephony to IP.

Why H.323 is Necessary

The purpose of H.323 is to establish a standard that allows solution providers to develop interoperable products for real-time audio and video communications over packet-based networks. Creating such a standard fosters interoperability in an otherwise heterogeneous, multivendor environment.

Standards offer solution providers an arena in which their solutions can flourish. They also make it easy for consumers to purchase a product they know is going to work with every other product that complies with the same standard. Imagine the chaos that would arise if there was no standard for such commonly used products as telephones, radios, televisions, and facsimile machines. More recently, the absence of standards for quadraphonic sound, video cassette recorders, and 56 Kbps modems led to divergent, incompatible technologies; this chasm was resolved by either eliminating a competing technology (as with quadraphonic sound and Betamax) or introducing a new standard that rendered the competing technologies interoperable (as with the ITU V.90 standard, which enabled the two 56 Kbps modem technologies, k56flex and x2, to communicate with each other).

Unlike the products mentioned above, attempting to standardize IP-based audio and videoconferencing is more formidable, because it requires the interworking of many sophisticated technologies. To put this challenge in perspective, consider that a simple telephone call requires a telephone set, wire, jacks, a switch in the central office, software, batteries, and transmission equipment that must all work in concert to provide reliable service. When it comes to interactive IP-based videoconferencing — where the goal is to see, hear, and exchange data with remotely located colleagues and friends as easily as if you were face-to-face in the same room — the number and degree of interworking technologies increases dramatically. As a result, achieving reliable, efficient, and effective audio-video communications across networks is predicated on the adoption of a standard that addresses interoperability among participating technologies. That standard is H.323.

The Origin of H.323

The ITU issued the first draft of H.323 in October, 1996. H.323 Draft 1 addressed visual telephone systems and equipment communications over Ethernet, Fast Ethernet, Fiber Distributed Data Interface (FDDI), and Token Ring Local-Area Networks (LANs) with no guaranteed QoS. Within the context of the original H.323 standard, LANs included internetworks consisting of multiple LANs interconnected by bridges, routers, and switches.

Around the time when H.323 Draft 1 was being implemented, IP telephony was gaining popularity. The idea of placing voice and video calls from a personal computer to others around the world for the cost of an Internet connection appealed to business and residential users alike. To keep pace with this popularity, solution providers developed proprietary IP-based telephones that could not communicate with each other. It soon became clear that a standard was needed to promote interoperability among these islands of incompatible solutions.

Initially, the quality of Internet-based calls was inferior to those placed over the PSTN. As vendors explored various ways to simplify and improve the quality of Internet-based calls, the following avenues were explored:

- Dividing voice communications between the Internet and the PSTN.
- Conducting voice communications between a telephone attached to the PSTN and a PC-based voice client.
- Placing voice calls over the Internet using a personal computer.

The last solution became the most popular and became the driving force behind Draft 2 of the H.323 standard.

H.323 Draft 2

Draft 1 of the H.323 standard provided a starting point for establishing a universal IP voice and multimedia communication standard for large, connected networks. Draft 2, approved in February 1998, builds on the initial draft by adding new features and enhancements suited for packet-based multimedia communication systems and IP telephony. The following sections describe these features and enhancements.

New Features

Draft 2 of the H.323 standard offers the following new features, many of which address limitations in H.323 Draft 1.

- **V.Chat Arbitration**
  A chat protocol that allows for Internet-based virtual chat networks.

- **Fast Start**
  Eliminates the delay between when a call is answered and when participants can hear each other. This results in instant audio connections similar to regular telephone calls, instead of the standard lengthy H.323 startup procedures.

- **Overlapped Sending**
  Affords faster connections by allowing the calling party to provide a partial address to the gatekeeper while the gatekeeper performs process routing. If the gatekeeper requires additional address information, it prompts the calling party until it has sufficient information to process the call.

- **Security (H.235)**
  Supports authentication, integrity, privacy/confidentiality, and nonrepudiation mechanisms.
  - Authentication ensures that all parties are authorized to participate in the conference.
  - Integrity validates data in packets.
  - Privacy/confidentiality protects encryption and decryption controls from hackers.
  - Nonrepudiation verifies whether callers actually participated in a conference.

- **Supplementary Services (H.450 series)**
  Defines the signaling protocol between endpoints for:
  - Supplementary Services (H.450.1)
  - Call Transfer (H.450.2)
  - Call Diversion (H.450.3)
  - Call Hold (H.450.4)
  - Call Park and Pickup (H.450.5)

- Call Waiting (H.450.6)
- Call Message Waiting Indication (H.450.7).

H.450.1, H.450.2, and H.450.3 are loosely based on the Q-signals (QSIG) protocol that PBXs and switch vendors use. Currently, only call-transfer and call-forward services have been defined. Both services are peer-to-peer and require no assistance from a PBX or other centralized network device. A number of standardized supplementary services are planned for future development.

Enhancements

Draft 2 of the H.323 Recommendation adds the following enhancements to Draft 1.

- **Additional Alias Address Types**
  Supports friendly URL-style addresses and RFC 822-formatted e-mail addresses, with which network-based users are more familiar and find easier to remember and use. H.323 Draft 2 also supports public and private party numbers, and transport-style addresses. Each endpoint can register under a number of these aliases.

- **Call Identification**
  Creates a unique global identifier that permits all messages associated with a call to be interoperable between the Registration, Admission, and Status (RAS) protocol used in IP networks as well as the Q.931 signaling protocol used in circuit-switched telephony networks. These unique call identifiers make it easier to track calls as they traverse multiple network nodes and topologies.

- **Conference List**
  Allows a Multipoint Control Unit (MCU) hosting multiple conferences to specify a calling endpoint by sending a Q.931 Facility Message, containing a list of conferences, to that endpoint.

- **Conference Requests**
  Introduces a new H.245 conference request packet, which can be sent to control a conference.

- **Dynamic Replacement of Channels**
  Provides a seamless way to change from one codec to another without requiring two media decoders.

- **Empty Capability Set**
  Allows the gatekeeper to re-route connections from an endpoint that does not support Supplementary Services.

- **Endpoint Redundancy**
  Allows an endpoint to have a secondary, more generic network interface or H.323-defined interface (such as a terminal) as a backup.

- **Endpoint Type Prefixes and Data Rules**
  Lets gateways indicate the data rates for all protocols that a device supports and the prefixes associated with the protocols.

- **Expanded Topology Model**
  Allows an H.323 conference model to include thousands of participants. The tightly controlled H.323 “panel” is encircled by a large number of “listeners.” Members can join the panel to participate in the conference and leave when desired. This scenario is akin to a panel on a stage in an auditorium, where participants from the audience join the panel on the stage while others leave and return to the audience. This movement of participants follows the Q.931 Invite and Join signaling standard defined in H.225.0.

- **Gatekeeper Redundancy**
  Permits a gatekeeper to specify an alternate set of gatekeepers that an endpoint can contact if the primary contact fails.

- **H.263**
  Adds video capabilities and structures to improve H.263-compliant video streams and allow for more robust media possibilities. In addition to layered codecs, which divide the video stream into additive layers of detail, H.263 provides additional Quality of Service parameters such as RSVP, which can be signaled when opening media streams. H.263 also adds the ability to pass media to other transports, such as ATM, while keeping the rest of the call and media control on IP.

- **Keep Alive**
  Allows an endpoint to register with a gatekeeper, so the gatekeeper can ascertain whether a user is online or offline and take the appropriate action. This feature also informs the gatekeeper of all aliases being used, such as nicknames, phone numbers, and LAN addresses.

- **Layered Codecs**
  Divides the video stream into layers of detail. This enables Quality of Service parameters such as RSVP to be signaled when opening media streams. It also allows certain media streams to be passed to other transports such as Asynchronous Transfer Mode (ATM), while keeping the rest of the call and media control on IP.

- **Pre-Granted Admission Speeds**
  Accelerates call processing by pre-verifying endpoints.

- **QoS (Quality of Service)**
  Allows endpoints to set QoS parameters, such as RSVP, for media streams.

- **RAS Retry Timers and Counters**
  Provides more robust RAS transaction processing by assuring that messages arrive within a specified time period and are resent as needed. This feature also notifies users if the delivery fails. It also adds a Request in Progress (RIP) message, which enables timers and counters to be reset.

- **Reliable Information Request Response (IRR)**
  Enhances the message-delivery process. This feature includes two new messages, IACK and INAK, that indicate whether messages were delivered.

- **Request in Progress (RIP) Message**
  Allows the receiver of a request to ask for a new timeout value when it cannot process the request before the current timeout value expires.

- **Resource Availability**
  Allows the gateway to notify a gatekeeper of its current call capacity, empowering the gatekeeper to make intelligent decisions when routing calls.

- **Time to Live**
  Specifies how long a gatekeeper should keep a registration active.

- **T.120/H.323 Integration**
  Requires endpoints that support both T.120 and H.323 standards to initiate the call with H.323.

- **Tunneling**
  Allows H.245 Protocol Data Units (PDUs) to be transmitted through the Q.931 call-signaling channel.

- **User Input Indication (PDU)**
  Provides broader Dual Tone Multifrequency (DTMF) signaling capabilities, such as length of tones.

H.323 Benefits

The following sections identify key benefits of H.323.

Audio/Video Codex Standardization

H.323 defines standards for compressing and decompressing audio and video data streams. H.323 also establishes common call setup and control protocols. Defining these standards ensures that solutions from different vendors offer a degree of common support. It also ensures that users can purchase solutions without concern for compatibility.

Unrestrained Independence

H.323 is not constrained by hardware, software, operating systems. This means H.323-compliant platforms can appear, such as video-enabled PCs, IP-enabled telephones, and cable television boxes.

Moreover, H.323 runs on top of networks. This enables H.323 solutions to take full advantage of the embedded services afforded by a network. It also makes it easier and more economical for vendors to offer solutions that keep pace with the rapidly evolving network technologies.

Bandwidth Management

Video and audio traffic consume vast amounts of bandwidth. This drain on network resources can wreak havoc with other business-critical applications that are struggling to function efficiently over the network. To address this issue, H.323 provides bandwidth management functions that limit the number of simultaneous H.323 connections on a network and constrain the amount of bandwidth available to H.323 applications. In this way, H.323 ensures that mission-critical data is not disrupted while, at the same time, preventing network congestion.

Endpoint Adaptability

H.323 allows endpoints with varying degrees of capabilities to participate in the same conference. For example, terminals equipped with video and data functions can participate in a conference with terminals that support only audio capabilities. Similarly, H.323 multimedia terminals can share data, video, and voice with other H.323 terminals, while sharing the data segment of the same video conference with T.120-compliant data-only terminals.

H.323 Applications

Many applications both in the corporate and home-user environment can take advantage of H.323 technology. The following sections describe a few key applications.

Video Conferencing

One obvious application is video conferencing between multiple users on a network. H.323 provides the framework for conducting real-time face-to-face video conferences over the Internet.

IP Telephony

Another obvious application that can benefit from H.323 is IP telephony. As described earlier, H.323 is well-suited for IP telephony for the following reasons.

- H.323 uses the Q.931 signaling protocol to establish, maintain, and release a connection. This protocol allows for relatively easy bridging to circuit-based telephones and the PST.

- H.323 supports the G.711 voice codec standard, providing easy connections to the legacy network of telephones. Furthermore, the uncompressed 64kb/sec stream can easily be translated between digital and analog media.

- One of the addressing formats that H.323 supports, E.164, is an ITU Recommendation that permits the use of standard telephone numbers (digits 0-9 and symbols * and #). As a result, standard telephones can be used to “dial” addresses that eventually map to IP addresses for H.323 endpoints.

- H.323 supports gatekeepers, which allow for integrated directory and routing functions as part of the call setup. These functions are critical for real-time voice and video applications, when dynamic points of connectivity require resources to be balanced. Moreover, the call permission and bandwidth control operations that gatekeepers provide enable load monitoring, provisioning, and, in the end, commercial-grade IP telephony service.

VoIP Deployment

H.323 directly benefits businesses deploying VoIP by offering a stable and backward-compatible standard upon which future generations of products can rely. The Recommendation's adoption by hundreds of vendors worldwide also means that businesses deploying VoIP have multiple sources for any given solution.

In addition, competitive pressure in technology guarantees continued innovation, which eventually translates into lower prices for goods and services and spurs further growth in the industry. In this way, consumers, vendors, software designers, and service providers all benefit from H.323.

Multimedia Call Centers

Another compelling application for H.323 is multimedia call centers. An H.323 call center provides a well-integrated environment for Web access and other data/voice business applications. Call centers are ideal for banks, utilities, high-technology companies, and other businesses that offer customer service. They are also well-suited for shops with retail outlets distributed throughout a city, state, country, or scattered around the world.

A call center can consist of just an H.323 terminal or an MCU, or it can offer a full-featured endpoint with a gatekeeper, a gateway, and MCU capability. The front end of a legacy call center can serve as a gateway that allows installed systems to operate with minimal disruption.

Remote Access

H.323 is ideally suited for remote-access applications. Remote access enables remote personnel, telecommuters, and after-hours workers to attend company meetings, access the company's LAN, read and send e-mail, and communicate with individuals at the company as if they were connected directly to the company's network. Remote access also encourages companies to offer work-at-home programs to their employees.

Remote Learning

H.323 is also well-suited for remote learning applications, where students attend class even though they may live miles away from a school, college, or university. H.323's multimedia framework is perfectly suited for transmitting real-time audio-video feeds from a classroom or laboratory. Furthermore, H.323's interactive capabilities allow students to speak with teachers and peers while downloading class work and lectures.

H.323 Building Blocks

H.323 defines four major components, and the way they interact with each other and with SCNs (for example, with H.320 conferencing systems).

- Terminals
- Gateways
- Multipoint control units (MCUs)
- Gatekeepers

In an H.323-compliant conferencing system:

- Terminals, gateways, and MCUs are considered endpoints because they generate and/or end H.323 sessions.
- Gatekeepers are considered network components because they cannot be called; however, they can be addressed to perform functions such as address translation and access control.

The following sections describe these components.

Terminals

H.323 terminals are client endpoints on the network and provide real-time communications. The most common H.323 terminals are PC-based applications, such as Microsoft's NetMeeting.

H.323 specifies the operating modes required for audio, video, and data terminals. The following table lists the required and optional functions for H.323 terminals.

Required H.323 Terminal Functions	Optional H.323 Terminal Functions
■ Voice communications	■ Video and/or data communications
■ Support for the Q90 protocol, for call signaling and setup	■ Support for data capability
■ Support for the H.245 protocol, to negotiate channel usage and RAS gatekeeper interface	■ T.120 data conferencing protocols
■ Registration Admission/Status (RAS), to communicate with a gatekeeper	■ MCU capabilities
■ RTP/RTCP support, for audio and video packet sequencing	

Gateways

A gateway operates as an endpoint on the network and provides real-time, two-way communication between the following components:

- H.323 terminals on the packet-based network and ITU terminals on a switched-circuit network. The terminals on the switched-circuit network (SCN) include those that comply with the following ITU Recommendations:

    - H.310 (B-ISDN)
    - H.320 (ISDN)
    - H.321 (ATM)
    - H.322 (GQoS-LAN)
    - H.324 (GSTN)
    - H.324M (Mobile)
    - POTS

- H.323 terminals on the packet-based network and another H.323 gateway.

A gateway is responsible for completing calls in both directions between the network endpoint and the SCN endpoint a transparently. To complete calls successfully, gateways are responsible for:

- Performing call setup and clearing on the network side and the SCN side.
- Translating transmission formats (for example, between H.225.0 of an H.323 endpoint and H.221 of an H.320 endpoint).
- Translating communication procedures (for example, between H.245 of an H.323 endpoint and H.242 of an H.320 endpoint).
- Translating video, audio, and data formats.

Gateways are optional components in an H.323 conference. They are not required if connections to other networks are not needed, since endpoints may directly communicate with other endpoints on the same LAN.

Multipoint Control Units (MCUs)

A Multipoint Control Units (MCU) is an endpoint on the network that allows three or more terminals or gateways to participate in a multipoint conference. An MCU can also connect two terminals in a point-to-point conference that may evolve into a multipoint conference.

An MCU manages negotiations and conference resources between endpoints to determine which audio or video codec to use. In some cases, MCUs may also determine which audio and video streams to multicast.

An MCU consists of two parts: a mandatory Multipoint Controller (MC) and optional Multipoint Processors (MP).

- The MC provides the call control needed to communicate with terminals. This capability is required for all multipoint conferences.
- The MP allows mixing, switching, and processing of audio, video, and data streams under the control of the MC. The MP may process single or multiple media streams, depending on the type of conference supported.

MC and MP capabilities can coexist in an integrated component or be part of other H.323 components. In the simplest case, an MCU can consist of just an MC.

Gatekeepers

A gatekeeper is an optional component in an H.323 system.

- If a gatekeeper is present, all endpoints have to register with it and use its services.
- If no gatekeeper is present, H.323 endpoints can interact with each other directly in a point-to-point or multipoint conference.

The gatekeeper monitors all H.323 calls within its area (or zone) on the network and performs the following required and optional functions.

Functions	Description
Required Functions	
■ **Address Translation**	Translates an alias address (such as an e-mail address) from a packet-based network to a destination client's transport address. Gatekeepers use a translation table containing Registration messages.
■ **Admission Control**	Limits H.323 access to a network or specific conference using Request, Admission, and Status (RAS) messages, as defined in Recommendation H.225.0. Network access can be based on criteria such as call authorization, available bandwidth, or a null function that admits all requests.
■ **Bandwidth Allocation**	Manages bandwidth allocation to H.323 endpoints using RAS messages. Bandwidth allocation can be based on bandwidth management or a null function that accepts all requests for bandwidth changes.
■ **Zone Management**	Defines the extent of the components, including endpoints, gateways, and MCUs, over which the gatekeeper has control.
Optional Functions	
■ **Bandwidth Management**	Gatekeepers can reject calls from a terminal if it ascertains there is insufficient bandwidth to support the call. They can also reject terminal requests for additional bandwidth during an active call. The criteria used to determine whether additional bandwidth is available is beyond the scope of the H.323 Recommendation.

Functions	Description
■ **Call Authorization**	Gatekeepers can reject calls from a terminal in accordance with the Q.931 recommendation based on such criteria as restricted access to or from specific gateways or terminals, or during specific times. The criteria for determining if authorization passes or fails is beyond the scope of the H.323 Recommendation.
■ **Call Control Signaling**	In a point-to-point conference, gatekeepers can process Q.931call-control signals or have endpoints send their Q.931 signals directly to each other.
■ **Call Management**	Gatekeepers can retain a list of ongoing H.323 calls to indicate that a called terminal is busy or to provide information for the bandwidth-management functions mentioned above.
■ **Secure Access**	Gatekeepers can employ authentication mechanisms to allow secure access to a conference.
■ **Administrative Point of Control**	Gatekeepers can act as a network's nerve center, allowing administrators to control H.323 traffic on and off the network, route calls to the appropriate gateway based on a proprietary routing arrangement, and even bill for calls placed through the network. Additionally, Q.931 and H.245 messages can be routed through a gatekeeper, and statistical information about calls in progress can be collected.
■ **Telephone-Service Activities**	Gatekeepers can perform telephony-service activities such as call transfer and call forwarding. They can also provide optional functions such as call accounting and call authorization.

H.323 Technical Information

The following sections provide technical information about an H.323 call scenario and the H.323 architecture.

H.323 Call Scenario (Example 1)

This section describes the steps for establishing and tearing down a call between two H.323 endpoints without a gatekeeper.

Call Signaling Phase

1. One endpoint (endpoint 1) sends another endpoint (endpoint 2) a setup message that contains the destination address.

2. Endpoint 2 sends a Q.931 alerting message.

3. If endpoint 2 accepts the call, it sends a connect message.

Call Control Phase

1. Each terminal sends a terminalCapabilitySet message to exchange information about its capabilities and logical-channel information (for example, codec preferences, supported media, and multiplex parameters). Note that H.323 allows various video codecs (for example, H.261, H.263, and H.263+) and audio codecs (for example, G.711, G.723.1). Data collaboration is also permitted using Protocol T.120.

2. Each terminal responds with a terminalCapabilitySetAck message. (Terminal capabilities can be resent any time during the call.)

3. The terminals exchange random numbers in the H.245 masterSlaveDetermination message to determine which will be the master and which the slave. This determination avoids conflicts between two endpoints that try to be the MC for a conference or try to open bi-directional channels at the same time.

Call Establishment Phase

1. The call is established.

2. Logical channels between the two terminals are open. Audio and video channels are unidirectional, while data is bi-directional.

3. Audio/video/data transmission begins. The terminals use the Real Time Protocol (RTP) defined by the Internet Engineering Task Force (IETF) as the transport protocol for audio and video data.

Call Termination Phase

1. At the end of the connection, one endpoint (endpoint 1) sends an endSession message to the other (endpoint 2).

2. When endpoint 2 receives the endSession message, it replies with its own endSession message to endpoint 1.

3. Endpoint 1 sends a Q.931 ReleaseComplete message and the call terminated.

H.323 Call Scenario (Example 2)

This section describes the steps for establishing and tearing down a call between an H.323 endpoint and a gatekeeper.

Call Signaling Phase

1. Both endpoints multicast a GatekeeperDiscovery message (GRQ) to locate a gatekeeper that can participate in the call.

2. A gatekeeper either accepts the request and issues a GatekeeperConfirm (GCF) message, or denies the request and issues a GatekeeperReject (GRJ) message.

3. When a gatekeeper is found, both endpoints send a RegistrationRequest (RRQ) message to register their alias names with the gatekeeper. This allows endpoints to call each other using e-mail or other user-friendly addresses, instead of transport addresses.

4. The gatekeeper sends either a RegistrationConfirm (RCF) message to acknowledge the endpoints' request or a RegistrationReject (RRJ) message to deny their request.

5. An endpoint or gatekeeper can use a LocationRequest (LRQ) message to request the location of another endpoint by alias name. The gatekeeper replies with a LocationConfirm (LCF) message that contains the resolved address that corresponds to the alias name.

Call-Establishment Phase

1. When a user places a call from an endpoint, the endpoint uses an AdmissionRequest (ARQ) message to request admission from the gatekeeper.

2. The gatekeeper either accepts (ACF) or denies the request (ARJ).

3. If the gatekeeper accepts the call, the Q.931 call-signaling sequence starts, with the endpoint sending a Q.931 Setup message to the remote destination.

4. The recipient of the Setup message sends an ARQ to request admission from its gatekeeper.

5. When the call is accepted, the Q.931 call-signaling sequence completes and H.245 message negotiation begins. The AdmissionRequest (ARQ) message

includes the initial bandwidth the endpoint requires for the duration of the conference. If an endpoint needs more bandwidth during H.245 logical channel negotiation, it sends a BandwidthRequest (BRQ) message to the gatekeeper. The gatekeeper replies with a BandwidthConfirm (BCF) message if it accepts the request or with a BandwidthReject (BRJ) message if it denies the request.

Call Termination Phase

1. When the call terminates, both endpoints send a DisengageRequest (DRQ) message to inform the gatekeeper that the call is ending.

2. The gatekeeper replies with a confirm (DCF) or reject (DRJ) message.

   Alternatively, endpoints can unregister from the gatekeeper by sending an UnregisterRequest (URQ) message. The gatekeeper replies with an UnregisterConfirm (UCF) or UnregisterReject (URJ) message.

H.323 Architecture

The following sections provide technical information about the H.323 architecture.

Transport Layer

In an H.323 system, the transport layer carries control and media packets across the network. H.323 does not define specific requirements for the underlying transport protocol, but supports reliable and unreliable packet modes. User Datagram Protocol (UDP) over Internet Protocol (IP) is an example of a commonly used transport protocol.

Video Codec

The video codec encodes the video from a camera or other video source for transmission. It also decodes any video code it receives for output to a video display. H.261 [3] with quarter common intermediate format (QCIF) resolution is the compulsory video codec for H.323 terminals. However, other video codecs (such as H.263, which delivers superior video quality and more options than H.261) and video resolutions (such as CIF and SQIF) can be used.

During the terminal-capability exchange sequence, the sending and receiving terminals use the H.245 protocol to exchange video codecs, resolution, bit rate, and algorithm options. After the H.323 terminals obtain the necessary information from this sequence, they open channels using the capabilities supported by all participating components. However, it is the receiving terminal that specifies what data the transmitting terminal can send.

Audio Codec

The audio codec encodes the audio signal from a microphone for transmission. It also decodes any audio code it receives for output to a loudspeaker. G.711 0 is the compulsory codec for H.323 terminals. However, H.323 terminals can optionally use the following recommendations to encode and decode speech:

- G.722
- G.723.1
- G.728
- G.729
- MPEG1 audio

Note that G.711 is a high bit rate codec (64Kb/s or 56Kb/s) and, therefore, cannot be transmitted over low bit rate (< 56 Kbps) links. A preferable alternative is G.723.1, which offers a reasonably low rate (5.3Kb/s and 6.4 Kb/s).

As with the video codec, H.323 terminals open logical channels using the capabilities supported by all participating components, ascertained during the H.245 terminal-capability exchange sequence.

Data Channel

The data channel supports applications such as electronic whiteboards, still image transfer, file exchange, database access, and audiographics conferencing. T.120 is the standardized data application for real-time audiographics conferencing. However, other applications (such as chatting and fax) and protocols can be used by way of H.245 negotiation.

System Control Unit

The System Control Unit provides the required signaling and flow control to ensure that H.323 terminals operate properly. The System Control Unit relies on the following ITU standards for performing these activities:

- H.245 — addresses the media control protocol for capability exchange, channel negotiation, and media-mode switching.
- H.225 — addresses the call-signaling protocol for admission control and establishing connections between terminals. H.225 derives the connection-establishment protocol from the Q.931 specification.

  The H.225 layer formats transmitted audio, video, data, and control streams into messages that can be sent to the network interface. It also uses the Real Time Transport Protocol (RTP) and Real Time Control Protocol (RTCP) to receive audio, video, data, and control streams from the network interface.

  - The RTP conducts the appropriate framing, sequence numbering, time-stamping, payload distinction, source identification, and error detection and correction activities for each media type.

- The RTCP provides reporting and status that sending and receiving terminals can use to correlate media-stream performance.

Deploying H.323

Given the intricacies involved with H.323, deploying an end-to-end H.323 solution, building a top-to-bottom H.323 solution requires wide-ranging expertise. Therefore, it is unlikely that a single vendor can provide a complete solution. Therefore, when considering the deployment of a comprehensive H.323 solution, select vendors that specialize in particular H.323 components and guarantee interoperability with all the other components you need to complete your H.323 system. Since some H.323 functions are optional, we recommend that you also determine which components are important to you and your business.

Another important factor to consider when deploying any new technology, including H.323, is network-level issues such as policy management and security. Policy management is achieved with gatekeepers, which provide call admission, authentication, and zone management. QoS protocols such as RSVP can also assist with policy management. Securing media streams is critical for a successful H.323 deployment, especially in unsecured environments such as the Internet.

Selecting H.323 Components

The following sections provide guidelines to consider when considering the purchase of H.323 components.

Endpoints

When selecting endpoints, observe the following guidelines:

- An endpoint is H.323-compliant if it runs over a packet network, such as IP, and supports only audio. If video and data sharing are important to you, look for endpoints that offer these capabilities.

- If you require enhanced audio, consider endpoints that support the following codecs, depending on the network bandwidth available.

Codec	Description
G.722	A wideband kHz audio codec that provides high-fidelity sound. G.722 consumes 48-to-56Kbps of bandwidth and is ideal for group systems using 384Kbps ISDN. G.722 ensures that sufficient bandwidth is available for delivering high-quality video.
G.723	A codec that requires only 5.3-to-6.3Kbps of bandwidth. G.723 is ideal for modem-based video conferencing environments, where bandwidth is at a premium. However, quality is on a par with that of

Codec	Description
	standard telephones.
G.728	A codec that delivers telephone-quality audio and consumes only 16Kbps of bandwidth. G.728 requires significant amounts of computer resources for compression/decompression. It usually requires a DSP for processing, or a fast microprocessor when used in host-based environments.
G.729	A codec that consumes small amounts of bandwidth while delivering near-telephone quality. G.729 is primarily used for voice-over IP applications. It is compatible with public switched circuit networks, making it ideal for use in voice gateways. Currently, very few video-conferencing endpoints support G.729

- If you require video, note that H.261 is required for basic interoperability with H.323 and H.320 endpoints. However, H.263 and H.263+ deliver better video quality and use less bandwidth than H.261. The efficiency gained from H.263 and H.263+ is especially significant for connections over POTS or BRI lines.

- For data conferencing, the T.120 data conferencing standard is an optional component of H.323. Microsoft NetMeeting has become the de-facto T.120-compliant application for data sharing. When evaluating an H.323 endpoint, check whether it supports NetMeeting.

- Avoid endpoints that claim H.323 compliance while supporting proprietary codecs to achieve higher-quality conferencing. Because these codec are non-standard, these codecs are useful only when connected to endpoints that offer the same capability, typically from the same vendor. They will not interoperate in this mode with other standards-based video conferencing products and therefore limit their advantages.

- Select endpoints that support a comprehensive suite of H.323 bandwidth-scalable features such as H.263 video and G.723 audio. Some endpoints take shortcuts with their H.323 compliance, implementing only the ISDN-based, H.261 video codec, and G.711 audio codec.

- Choose endpoints that support H.320 and H.323 in a single product. These products allow you to place ISDN or LAN calls simply by referencing the appropriate phone number or IP address. Avoid endpoints that claim H.320 and H.323 compliance, but do not integrate the two transports into a single product. Typically, these vendors require you to buy two separate systems or use two different applications on the same hardware.

- To simplify endpoint manageability, choose endpoints that support auto-discovery of gatekeepers when the conferencing application starts. Otherwise, administrators will have to manually configure each H.323 endpoint with the IP address of the gatekeeper managing its calls.

- If you plan to deploy H.323 alongside your current H.320 network, select an endpoint that supports both H.320 and H.323. A system that supports both standards permits video conferencing with older ISDN-based systems while easing the migration to IP-based video conferencing systems.

MCUs

H.323 MCUs are available as hardware and software solutions. Both are functionally identical, but differ in price and performance.

- Hardware-based MCUs use a PC-like chassis containing cards equipped with DSPs that implement the MCU functionality. One card may or may not support multiple users, depending on the implementation. Therefore, the number of cards that a chassis supports determines the number of simultaneous users. Hardware-based MCUs can be cascaded to support more users, though most H.323 MCUs do not support cascading.

- Software-based MCUs usually cost less per seat. Additionally, the number of simultaneous users they support is constrained by the performance of the server.

When considering an MCU, determine:

- The maximum number of simultaneous calls it can handle.

- The type of simultaneous calls it can handle (for example, voice, data, audio, and video, or just audio, video, and data).

- Which audio and video codecs it supports. The MCU-supported codecs should be the ones that your H.323 endpoint and gateway support. If an MCU does not support some audio codecs, it performs transcoding. Transcoding introduces latency and reduces quality. It also consumes substantial amounts of MCU resources, which decreases the number of simultaneous users or conferences that can be supported.

- Which audio codecs your MCU can mix in full-duplex mode. If an MCU cannot mix certain audio from an endpoint, attendees will encounter audio degradation.

- Whether the MCU provides T.120 data support in addition to audio and video. Your MCU should offer T.120 data support for all participants, even ones that join a conference call through an H.323-to-H.320 gateway.

- Whether the MCU provides a call-logging facility for monitoring MCU usage and MCU diagnostics for troubleshooting.

- Whether the MCU provides SNMP and remote-configuration capabilities using an Internet Browser, across a LAN, and/or by dial-up modem. These features are especially useful for large organizations with centralized help desks that may want to remotely manage in-house MCUs.

Most MCUs include a bundled H.323 gatekeeper. If you want to use a different gatekeeper on your network, make sure the bundled gatekeeper can be disabled. Also, check whether you will lose any MCU functionality if you decide to use an alternate gatekeeper.

Gatekeepers

An H.323 gatekeeper is a software application. Gatekeepers are either sold as standalone products or bundled with H.323 hardware such as an H.323-to-H.320 gateway, an H.323 MCU, or a network router.

Typically, standalone gatekeepers cost less than bundled ones because you do not have to incur the cost of the bundled hardware. Before you purchase a gatekeeper, however, be sure your H.323 endpoints will register with it and that it is compatible with the gateway or multipoint control unit you are considering.

Bundled gatekeepers typically are compatible with the hardware that they come with. Together, gatekeepers are optimized for use with their bundled hardware and may offer capabilities that are not available with standalone gatekeepers

Select an H.323 gatekeeper that responds to Multicast or Unicast discovery. These discovery methods relieve administrators from having to manually configure each video-conferencing system to reference gatekeepers by IP address. Multicast discovery is the most efficient way for endpoints to contact gatekeepers. If your organization prefers not to enable Multicasting, a Unicast-based discovery message can be sent to the gatekeeper (so long as the gatekeeper supports Unicast) using IP addresses.

Gatekeepers manage zones of H.323 equipment. The procedures for mapping a gatekeeper to a zone range from simple to complex procedures. Therefore, review the procedures for configuring a gatekeeper and revising zones when considering the purchase of a gatekeeper.

Some gatekeepers offer the ability to log calls. If this feature is important to your organization, be sure the gatekeeper can also differentiate between different service levels. For example, a 384Kbps ISDN call is more expensive than a 128Kbps ISDN call. Logging such calls enables you to charge the expense to the department or individual who used the service.

Also, make sure the information that is logged is helpful to your organization. At the very least, the gatekeeper should log:

- Call start date and time
- Call end date and time
- Call type (audio, video, and/or data)
- Bandwidth utilization (for example, 128Kbps, 384Kbps, and so on)
- Endpoint determination

If a gatekeeper's logging facility is not suited to your organization, investigate whether there are compatible billing packages that work with the gatekeeper. If the logged data will be shared with and analyzed by other departments, be sure the data can be exported to a spreadsheet or database application.

Exercise care when using gatekeepers and gateways from different vendors, because the products may not be interoperable. However, if a gatekeeper from one vendor does work with the gateway from another vendor, you may lose valuable features that are only available when using gatekeepers and gateways from the same vendor.

For example, assume you use a gateway from Vendor A that supports 128Kbps and 384Kbps ISDN calls and voice calls and a gatekeeper from Vendor B that does not support 128Kbps ISDN calls. When used together, the gatekeeper routes only one type of request to the gateway (128Kbps ISDN calls, for example) while ignoring all other calls (384Kbps ISDN calls and voice calls, in this example).

Glossary

The following glossary defines the technical terms in this White Paper. A ***bold italicized*** word is itself defined in this glossary.

Asynchronous Transfer Mode	A network technology based on transferring data in packets of a fixed size. The packet used with ATM is relatively small compared to units used with older technologies. The small, constant packet size allows ATM equipment to transmit video, audio, and data over the same network without having one of these data types hog the line.
Codec	Abbreviation for compression/decompression. Codec is a technology for compressing and decompressing data. Codecs can be implemented in hardware, software, or a combination of both.
Conference	An area where participants can meet to discuss a topic of common interest.
Fiber Distributed Data Interface	A set of ANSI protocols for transmitting digital data over fiber optic cable. FDDI networks are token-passing networks, and support data rates up to 100 Mbps. FDDI networks are typically used as backbones for wide-area networks.
Gateway	A combination of hardware and software that connects two different types of networks. Gateways between different e-mail systems, for example, allow users on those e-mail systems to exchange messages.
Integrated Services Digital Network	An international communications standard for sending voice, video, and data over digital telephone or normal telephone lines. ISDN supports data transfer rates of 64 Kbps. Most ISDN lines offered by telephone companies offer two lines, called B channels. One line can be used for voice and the other for data, or both lines can be used for data to boost the data rate to 128 Kbps.
IP	Abbreviation for Internet Protocol, pronounced as two separate letters. IP specifies the packet format and addressing scheme for sending data over a network. Most network combine IP with a higher-level protocol called Transport Control Protocol (TCP), which establishes a virtual connection between a destination and a source. IP by itself is similar to the postal system; it lets you address a package and drop it in the

	system, but there's no direct link between you and the recipient.
ITU	A United Nations-based agency responsible for adopting international treaties, regulations, and standards governing telecommunications.
Multipoint	A conference among three or more participants.
Point-to-point	A conference between only two participants.
Public Switched Telephone Network	The international telephone system based on copper wires that carrying analog voice data. This is in contrast to newer telephone networks base on digital technologies, such as ***Integrated Services Digital Network*** and ***Fiber Distributed Data Interface***.
Quality of Service	A networking term that specifies a guaranteed throughput level.
Quarter common intermediate format	A ***videoconferencing*** format that specifies data rates of 30 frames per second (fps), with each frame containing 144 lines and 176 pixels per line.
Real Time Transport Protocol	An Internet protocol for transmitting real-time data such as audio and video. RTP does not guarantee real-time delivery of data, but provides mechanisms for sending and receiving applications to support streaming data. Typically, RTP runs on top of the ***User Datagram Protocol***; however, the specification is general enough to support other transport protocols.
Tunneling	A technology that allows one network to send its data via another network's connections. Tunneling works by encapsulating a network protocol within packets carried by the second network. For example, Microsoft's PPTP technology allows organizations to use the Internet to transmit data across a virtual private network by embedding its own network protocol within the TCP/IP packets carried by the Internet.
User Datagram Protocol	A connectionless protocol that, like TCP, runs on top of IP networks. Unlike TCP/IP, UDP/IP provides very few error recovery services, offering instead a direct way to send and receive datagrams over an IP network.
Video-conferencing	Conducting a ***conference*** between two or more participants at different sites using computer networks to transmit audio and video data. Participants use video cameras, microphones, and speakers to communicate with each other. As the participants speak to one another, their voices are carried over the network and delivered to the speakers at the other locations; similarly, images that appear in front of a video camera at one location display in a window or screen on the other participants' monitors. In this way, videoconferencing provides participants with a "virtual" conference room where they can communicate as if they were sitting next to each other.
VoIP	An abbreviation for Voice over IP, a category of hardware and software that allows people to use the Internet as the transmission medium for

telephone calls. For users who have free, or fixed-price Internet access, Internet telephony software essentially provides free telephone calls anywhere in the world.

Zone An area on the network consisting of all the endpoints, gateways, and MCUs that are or will be registered with a gatekeeper.

Bibliography

The following sources provide additional information about H.323. To visit a Web site shown in this section, click it.

International Multimedia Teleconferencing Consortium, Inc. (IMTC)

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