The H.323 Revolution

A Technology Review

Contact us at: H323Info@Radvision.com

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Background

The H.323 protocol is best known as the original call signaling protocol that made real-time voice and video over IP possible. Being the first solution to work, H.323 is the most widely deployed protocol in the market, and through its veteran status and wide acceptance provides telecommunication equipment with the benefits of a highly mature and completely interoperable signaling solution.

Recently, other signaling protocols have become available on the market, although none has reached the level of maturity and widespread deployment of H.323. Even though other signaling protocols may have the advantage of being built on a more modern base, the H.323 standard has kept abreast of the latest industry changes and has undergone a metamorphosis from an early-day protocol that was mainly concerned with enabling voice and video calls over limited bandwidth to a high-performance, feature-rich solution that is deployable in large-scale carrier-class networks.

While the capability and performance boost of the H.323 version 4 protocol is impressive, the standard leverages its early multi-media roots through its many annexes and open-system plug-ins. These provide feature-rich and billable services that enhance the user experience and provide financial incentives to Service Providers. The latest features of the H.323 version 4 protocol can be grouped into the following general categories:

- Improved user experience through web-based and advanced supplementary services
- Improved PSTN signaling interworking
- Advanced service capabilities for features such as privacy, security, and ease of billing
- Carrier-class network optimization features to enable required scalability and to ensure service availability

Because of these new improvements, built on an extremely solid and proven foundation, the H.323 version 4 protocol is an excellent solution for any real-time voice or video IP implementation. This paper will describe the latest advances in the H.323 version 4 protocol and how they are being integrated in real-time voice and video over IP networks.

Improved User Experience

Participants of IP voice or videoconferences expect to receive at least the same level of reliable service and user experience that they are used to with other means of communication such as telephones, e-mail, and instant messaging. When communicating via the Internet, people enjoy web-enabled services such as click to dial, address books, presence and instant messaging. When communicating via the telephone, subscribers enjoy call forward, call waiting, hold and many other such supplementary services.

H.323 addresses this market need. Of particular note is Annex K of H.323 version 4, which enables H.323 applications to work in tandem with HTTP. H.450, which is part of the H.323 umbrella, provides an extensive range of useful supplementary services.

Web-Based User Services

One of the major innovations of H.323 version 4 is Annex K, which defines how HTTP can be used for the transport of service control with call or non-call related H.323 signaling. This combination of best-of-breed protocols enables full-featured advanced videoconference services to be accessed from a simple web browser.

With these new features an application can locate a pre-defined (personalized) URL and enable the provision of a personalized service through the web. Examples of web-based services that can be incorporated in applications are:

- Address book navigation
- Click to dial
- Point and click for multi-party conference setup
- Interactive Web/Voice Response
- Web Multimedia Instant Messaging and Presence
- Follow-me services

How it Works

During initial RAS signaling, Annex K of H.323 version 4 allows a central device, such as a gatekeeper (or Softswitch with gatekeeper call control), to initiate a parallel HTTP session to a predefined URL. In the RCF message, the gatekeeper may return a URL. The parallel HTTP session, allows call control and non-call control services to be incorporated into H.323 real-time voice and video over IP applications and devices. The HTTP server maps HTTP events and call control actions in a manner transparent to the endpoints. Further, H.323 version 4 allows various types of text formats, images and sounds[1] to be added and utilized dynamically.

The flow diagram in Figure 1 demonstrates address book navigation using combined H.323 and HTTP protocol services.

[1] As MIME types

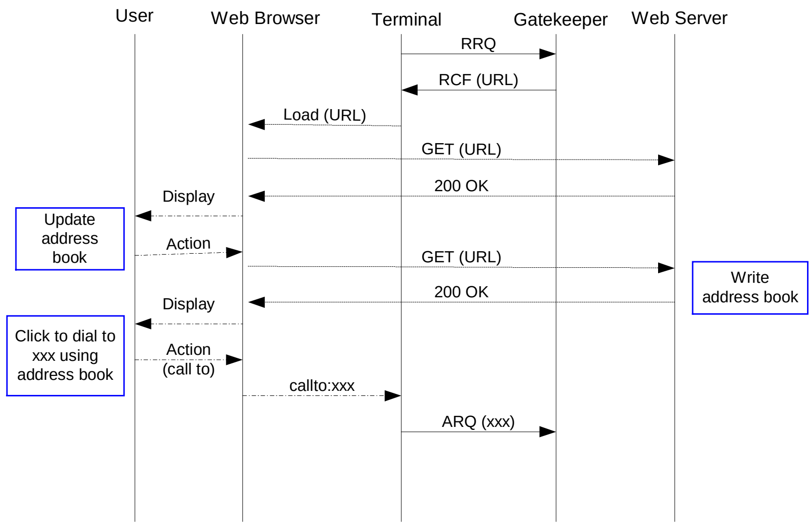

Figure 1: Combining H.323 with HTTP

Supplementary Services

Supplementary services provide a variety of user facilities. Surveys indicate that many vendors have already adopted the new supplementary services while key players regard the addition of the latest advanced supplementary services as product differentiators.

Recommendation H.450, defines supplementary services. The first supplementary services were “transfer”, “forward” and “hold” (specified in H.450.2 to H.450.4). Many vendors have implemented these already. Further, an enhanced suite of services (H.450.5 to H.450.9) was published in H.323 version 3 followed by more advanced services (H.450.10 to H.450.12) in H.323 version 4.

The following list details the supplementary services provided by H.450.5 to H.450.12:

- Park and Pick-up (H.450.5) facilitates call center services and terminal mobility. A user calling a call center is put into park mode until an agent is available for call pick-up or a user can put a call into park mode at endpoint A and pick the call up at endpoint B. For example, a user can start a call at a telephone, put it into park mode and pick the call up at a videoconference terminal.
- Call Waiting Indication (H.450.6) notifies a busy user that another caller is attempting to connect.
- Message Waiting Indication (H.450.7) provides a general-purpose mechanism for notifying a user that a message such as voice mail, fax, or video mail is waiting.
- Naming Indication (H.450.8) enables the exchange of readable descriptive text about the endpoint user, such as the useful information that normally appears on a business card.
- Call Completion Services (H.450.9) signals the caller and the callee when both parties are free to receive a call. Applies to “call completion for busy” or “no answer” scenarios.

- Call Offering (H.450.10) enhances the Call Waiting service by notifying the called user who can then accept the offered call or reject it. The called user can also accept the offered call after putting the existing call on hold or terminating it.
- Call Intrusion (H.450.11) enables the ranking of endpoints. A higher-ranking endpoint can intrude in the call of a lower-ranking endpoint. Intrusion can be the breaking into an existing call or silently monitoring the called user.
- Common Information Additional Network (H.450.12) enables endpoints to exchange information relating to the endpoint category and the supplementary services supported by each. This information can be used by the endpoint or gatekeeper applications to determine what supplementary services to use and how they can best interoperate.

Data Entry via Phone Keypad

The Dual Tone Multi-Frequency (DTMF) method was defined in legacy telephony many years ago by the ITU to enable the use of the phone keypad for data entry. For example, DTMF is used when ordering a theater ticket by phone using a credit card. Each dialed digit is decoded as two unique frequency tones mixed together, which the telephone transmits as a DTMF signal to the remote phone. The remote phone encodes the dual-tone back into an ASCII character, the central office or exchange recognizes these signals and processes the requested service.

From the earliest versions H.323 provides DTMF support. In the latest ratified version 4, H.323 enables an RTP payload for DTMF format and supports RFC2833 compliant operation mode. The DTMF support is defined through H.245 version 7 message exchange.

Leveraging PSTN Networks

PSTN networks have evolved over several decades. Due to the capital invested in it, its large installed base, and its solid reliability, PSTN will continue to be used for many years. In order to leverage past investments in PSTN, Service Providers need to be able to provide seamless services between PSTN and IP. H.323, and version 4 in particular, facilitates greater interoperability with legacy domains.

Smooth Mapping of Supplementary Services

In many cases, the architects of H.323 adopted the same approach as ISDN. This makes interoperability easier. For example, H.323 uses the same call control signaling protocol (Q.931) as ISDN for call establishment and teardown. H.450 supplementary services use the same logic as QSIG.

QSIG is the most common protocol used to communicate between PBXs in a PSTN network. The H.323 supplementary services suite H.450 has been based on QSIG logical procedures. That is, H.323 supplementary services use exactly the same logic as QSIG. So when a supplementary service needs to be provided between a PSTN network and an IP network (via a gateway) the inter-network mapping is straightforward and simple.

Tunneling PSTN Signaling Protocols over H.323

As part of the effort to interoperate with PSTN, H.323 version 4 specifies a generic means of tunneling signaling messages in any H.225.0 call control message. This method allows non-H.323 signaling information to be transported over H.323 networks.

Consequently, protocols such as QSIG (Annex M1), ISUP (Annex M2) and DSS1 (Annex M3) can be transmitted by tunneling over H.323 networks.

Optimized Network Capabilities

Gateways compatible with H.323 version 4 can interface with one or more PSTN switches and associate trunk groups or carrier information into the call signaling path. This means that the right gateway can be chosen for the right task. For example, if an endpoint chooses to send fax data, the call can be routed to a gateway that only knows how to send a fax instead of routing the call to a sophisticated gateway that can handle complex video calls.

The Desired Protocols feature defined in H.323 version 4 allows an endpoint through various H.225.0 messages to indicate what protocols (such as fax, H.320, or T.120) it desires to utilize during the course of a call. If an endpoint has indicated a list of desired protocols to its gatekeeper, the gatekeeper attempts to locate a gateway that can provide support for the desired protocols.

Well-Managed Service Capabilities

As IP voice and video conferencing becomes mainstream, Service Providers require more secure and well-managed service capabilities including robust call session accounting, call services such as caller identification and restriction, caller authorization, and protection against eavesdropping.

Robust Call Session Accounting

By combining the features available in H.323 Version 4, Service Providers can now manage call session accounting. The following features can be used to improve accounting and billing information:

- User Authentication (see *Security and Privacy* below).
- During call setup information from the newly specified H.225.0 fields and optionally, from H.248, provides credit-related capabilities. An endpoint can receive user credit or debit information from a gatekeeper before and after call establishment.
- A gatekeeper may specify a time limitation and an endpoint can limit the user's call duration accordingly. For instance, an endpoint may disengage the call when the time or credit on the user's account is exhausted.
- A gatekeeper can send balance-related announcements to the endpoint and indicate a call duration limit to the endpoint.
- A gatekeeper can indicate the call billing mode, which can be credit mode or debit mode.
- Annex G of H.225.0 (added in H.323 version 2) allows administrative domains to make a request to other domains to provide information about the resource usage of a specific call.

Caller Identification Services

H.323 version 4 provides caller identification services, including:

- Calling party number presentation and restriction
- Connected party number presentation and restriction

- Called (alerting) party number presentation and restriction
- Busy party number presentation and restriction

Security and Privacy

H.235 defines authentication as “the provision of assurance of the claimed identity of an entity”. User authentication is accomplished by adding a signature to RAS and Q.931 messages with an agreed shared secret key known only to the endpoint (user) and the central device.

H.235 Version 2 defines a multimedia encryption security profile. This allows privacy through the prevention of eavesdropping. Eavesdropping on media is prevented through the use of H.245, which presents RTP media streams with well-known encryption algorithms.

Generic Extensibility Framework

The extensible framework provides a common method for feature negotiation that operates over multiple domains and may be managed and configured by different operational entities. The framework allows new features to be readily added to the protocol without affecting the underlying (H.225.0) core specification.

An extended service can be defined on one side without knowing whether the other side has the capability for this feature to operate successfully. If the other side has the capabilities, it can provision the extended service. If the other side does not have the capabilities, the extended service request will simply be ignored without affecting the call.

The generic extensibility framework is an optional feature of H.323 Version 4.

Enabling Scalability

Looking toward the future, as VoIP networks grow, scalability issues and the support of large numbers of users need to be addressed. New features in H.323 Version 4, such as the ability to register many aliases to a gatekeeper, have overcome problems previously encountered in very large-scale systems. Q.931 multiplexing and load balancing facilities reduce resource consumption and boost call handling performance.

Load Balancing and Redundancy

For the purposes of ensuring system availability, redundancy, and scalability, H.323 defines an entity called an Alternate Gatekeeper. A gatekeeper may utilize multiple physical or logical devices as Alternate Gatekeepers. When a gatekeeper becomes overloaded it can load balance by redirecting the registered endpoint to an Alternate Gatekeeper. Figure 2 shows how an endpoint registers with a gatekeeper. The gatekeeper rejects the registration but adds the address of an Alternate Gatekeeper to the rejection message.

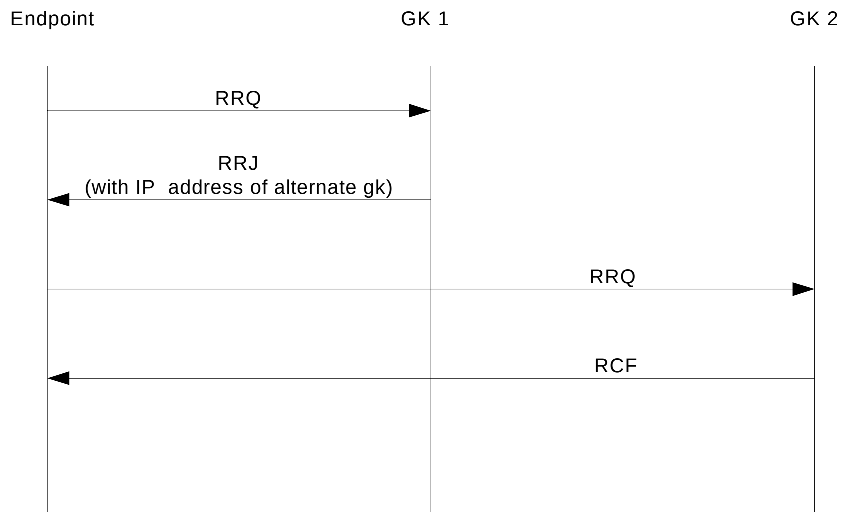

Figure 2: Alternate Gatekeeper

Massive Address Registration to a Gatekeeper

Previous versions of H.323 were not efficient when a gateway or MCU had to register very large numbers of alias addresses to a gatekeeper. This inefficiency was caused by the size limitation of UDP packets. H.323 version 4 has addressed and resolved this problem through “Additive Registration”, an incremental registration process by which an endpoint registers to a gatekeeper using several messages. When Additive Registration is used, the gateway or MCU registers to a gatekeeper by sending an initial list of aliases in an RRQ. Subsequently the gateway or MCU sends additional RRQs until the full list of alias addresses is complete.

TCP Channel Multiplexing

TCP assures reliable signaling over error-prone Internet connections. It also assures connection state auditing for accurate service charges required by Service Providers. However, each TCP connection requires a large amount of operating system resources and CPU time. By using Q.931 multiplexing the same TCP connection can be used for multiple call setups saving TCP resources and improving performance.

The Call Signaling Channel carries H.225.0 call control messages. The Call Reference Value (CRV) associates the message with the call and allows the Call Signaling Channel to carry signaling for many concurrent calls.

More Efficient Setup Time

Since H.323 version 2, the 1.5 Round Trip Time (RTT) for call establishment connect time has been recognizably faster than many other VoIP protocols. Since H.323 messages use ASN.1 PER (Packet Encoding Rules), call setup messages require less bytes than any other VoIP protocol. In H.323 version 4 call setup can perform fast connect and establish an H.245 session, via tunneling, in parallel. Hence call setup can be performed even faster than before.

This means that you can set up an audio call in both directions (as before) while also fully exchanging terminal capabilities with the endpoints (in three messages[2]). You can thus set up two-way audio and video without any further signaling, thereby improving performance and cutting down on signaling time.

Light Connectionless Signaling

Annex E (defined in H.323 version 3) is a generic protocol for performing reliable message exchange over UDP. You can use UDP instead of TCP[3] to carry Q.931 messages. This gives better control of parameters such as time-outs, retransmision, and peer failure detection and also enables automatic multiplexing.

[2] One in each direction plus an ACK.
[3] TCP is a closed protocol provided with the operating system so that you cannot change parameters.
[4] Ratified after H.323 version 4 was ratified (May 2001)

Conclusion

Today H.323 is much more than a legacy IP telephony protocol. The new H.323 version 4 enables full-featured and well-managed real-time voice and video over IP with a rich set of add-on user services. The combination of H.323 and HTTP introduces a new level of user experience and enables the creation of a new breed of services.

H.323 version 4 makes enhanced integration with PSTN networks possible, using similar PSTN logic schemes for both basic and advanced call services.

For Service Providers, H.323 version 4 provides a rich set of services for user authentication, user privacy, billing, service creation as well as scalability capabilities.

H.323 dominates the VoIP industry, with billions of minutes of voice and video traffic every month. H.323 version 4 is the professional answer of choice for addressing market needs—full-featured, well-managed conferencing and advanced service creation, all within an extremely stable, robust, and interoperable environment.

Learn about RADVISION enabling technologies for H.323 at:

http://www.radvision.com/f_products/f2_h323.php3?prod=H.323+Toolkit

Glossary of Terms

Abbreviation	Description
Annex K	HTTP Based Service Control Transport Channel
Annex L	Stimulus-signaling procedures between H.323 terminals and a Feature Server functional entity.
Annex Q	H.281/H.224 based FECC
DTMF	Dual-Tone Multi Frequency
H.235 v2	Security and Encryption for H-Series (H.323 and other H.245 Based) multimedia terminals
H.245	Call control signaling protocol for determining capabilities and for issuing open and close channels commands
H.320	The ITU standard for videoconferencing over digital networks such as ISDN
H.450	Protocol for the support of Supplementary Services in H.323
HTTP	Hypertext Transfer Protocol
ISDN	Integrated Services Digital Network
PER	Packet Encoding Rules
PSTN	Public Switched Telephone Networks
Q.931	Call control signaling protocol for the establishment and tear down of calls
QSIG	A global signaling and control standard for Private Integrated Network Exchange
T.120	An ITU standard for real-time data conferencing (sharing data among multiple users).
TCP	Transmission Control Protocol

Contact RADVISION at:

USA/Americas
575 Corporate Drive
Mahwah, NJ USA 07430
Tel 201.529.4300
Fax 201.529.3516
info@radvision.com

Europe
24 Raul Wallenberg
Tel Aviv, Israel 69719
Tel +972.3.645.5220
Fax +972.3.647.6669
infointernational@radvision.com

United Kingdom/Middle East
Abbey House,
Wellington Way
Weybridge, Surrey
Tel +44.1932.268315
Fax +44.1932.268316
infoUK@radvision.com

Hong Kong
Suite F, 17/F
China Overseas Bldg.
139 Hennessy Road
Wanchai, Hong Kong
Tel +852.2801.4070
Fax +852.2801.4071
apacinfo@radvision.com

China
Beijing Representative Office
Unit 301, Tower B, COFCO Plaza
No. 8 Jianguomennei Avenue
Beijing 100005, China
Tel +86.10.65249484
Fax +86.10.65260794
apacinfo@radvision.com

