Signaling In H.323 Protocol

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Abstract: Voice over Internet Protocol, referred as VOIP, is one of the most popular and ever-growing technology in the world of high speed communication. Not more than eight years have passed but VOIP has found the best platform to transmit voice over internet and that too without paying much!

For those who know VOIP and its benefits, it's hard to resist. Perhaps that is why vendors are flooding the market with VOIP products and services. With the H.323 and SIP signaling protocols, VOIP provides audio transmission and multimedia conferencing at lightening speed!

Here, I present the key issues that inhibit Voice over IP (VOIP) to be popular with the users. Then I shall discuss H.323 protocol and standards that exist today and are required to make the VOIP products from different vendors to interoperate. It discusses call establishing, its management and termination procedures. My paper also explains the need of a new protocol SIP by comparing both protocols.

Key Words: Voice Transmission, H.323 Call Signaling, RAS, Message Exchanging, Pros and Cons of Technology.

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1. Introduction

Voice over IP (VOIP) uses the Internet Protocol (IP) to transmit voice as packets over an IP network which can be achieved on any data network that uses IP, like Internet, Intranets and Local Area Networks (LAN). This voice signal is first digitized, then compressed and converted to IP packets and then transmitted over the IP network. Signaling protocols are used to set up and tear down calls, carry information required to locate users and negotiate capabilities. One of the main motivations for Internet telephony is the very low cost involved. Some other motivations are demand for multimedia communication and integration of voice and data networks.

VOIP is getting famous as time elapses but there are still some key issues that stem to the fact that IP was actually designed to transfer cluster of data from one location to the other while the other issues concern to the fact that vendors don't conform to the standards.

Quality of voice
For voice communications over IP to become acceptable to the users, the delay needs to be less than a threshold value and the ***IETF (Internet Engineering Task Force)*** is working on this aspect.
Way to overcome?
To ensure good quality of voice, we can use either Echo Cancellation, Packet Prioritization (giving higher priority to voice packets) or Forward Error Correction

Interoperability
In a public network environment, products from different vendors need to operate with each other if voice over IP is to become common among users.
Way to overcome?
To achieve interoperability, standards are being devised and the most common standard for VOIP is the H.323 standard. Other is SIP.

Security
Anyone on internet can capture the packets meant for someone else! Then what is the security?
Way to overcome?
Some security can be provided by using encryption and tunneling. The common tunneling protocol used is *Layer 2 Tunneling protocol* and the common encryption mechanism used is *Secure Sockets Layer (SSL)*.

Integration with Public Switched Telephone Network (PSTN)
While Internet telephony is being introduced, it will need to work in conjunction with PSTN for a few years.

Scalability
Researchers are working to provide the same quality over IP as normal telephone calls and that too at a much lower cost. It's quite possible to see VOIP and its Internet telephony cousins growing at a great rate but still care needs to be taken that VOIP systems remain flexible enough to grow to large user market and allow a mix of private and public services.

2. H.323 Protocol

H.323 Protocol (defined, **ITU-T**, International Telecommunications Union)by was originally developed for multimedia conferencing on LANs, but was later extended to cover Voice over IP. The standard, which went through changes from 1996 to 1998, encompasses both point to point communications and multipoint conferences. The products and applications of different vendors can interoperate if they abide by the H.323 specification and so it adds interoperability to the growing tree!

2.1 Components of H.323

There are 4 basic components of H.323, which shall be covered under this title.

2.1.1 Terminals (or Endpoints/Clients)

Terminals, that provide real time two way communications, have to support H.245, Q.931, Registration Admission Status (RAS) and Real Time Transport Protocol (RTP).

- H.245 is used for allowing the usage of the channels,
- Q.931 is required for call signaling and setting up the call,
- RTP is the Real Time Transport protocol that carries voice packets and
- RAS is Registration Admission Status which is used for interacting with the gatekeeper.

2.1.2 Gateways

A gateway is an endpoint on the network which provides for real-time, two-way communications between H.323 terminals on the IP network and other ITU terminals on a switched based network, or to another H.323 gateway. They perform the function of a "**translator**" i.e. they perform the translation between different transmission formats, e.g from H.225 to H.221.

FIGURE 1: WORKING OF GATEWAY

They are also capable of translating between audio and video codec. The gateway is the interface between the PSTN and the Internet. They take voice from circuit switched PSTN and place it on the public Internet and vice versa.

Gateways are optional in that terminals in a single LAN can communicate with each other directly. When the terminals on a network need to communicate with an endpoint in some other network, then they communicate via gateways using the H.245 and Q.931 protocols

2.1.3 Gatekeepers (alias 'Manager')

Gatekeeper is a central point for all calls within its zone (its range) and provides services to the registered endpoints. Some of the functionalities that gatekeepers provide are listed below:

1. Admissions Control – Grant or deny access on call authorization
2. Call signaling – Provide a Call Signaling Channel to the endpoints and process Call.

3. Call Authorization – Reject Calls due to authorization failure through use of H.225 signaling due to restricted access for some time periods.
4. Call Management – Maintain list of ongoing H.323 calls to inform if terminal is busy, provide information for the Bandwidth Management function.

2.1.4 Multipoint Control Units (MCU)

The MCU is an endpoint on the network that provides the capability for three or more terminals and gateways to participate in a multipoint conference. It consists of a mandatory Multipoint Controller (MC) and optional Multipoint Processors (MP). MC determines the common capabilities of the terminals but it does not perform the multiplexing of audio, video and data rather controls the multiplexing of media streams which is handled by MP.

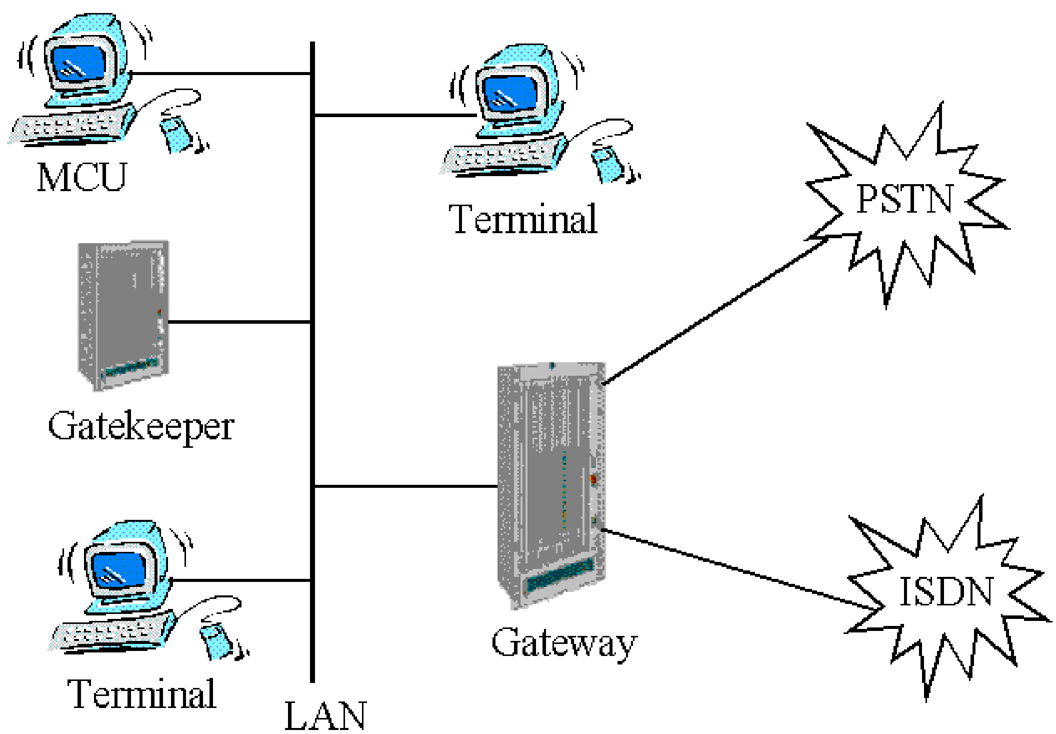

FIGURE 2: INTERACTION BETWEEN COMPONENTS OF H.323

2.2 Control and Signaling In H.323

The H.323 provides three control protocols:

- H.225.0/Q.931 Call Signaling
- H.225.0 RAS and
- H.245 Media Control.

2.2.1 H.225.0/Q.931 Call Signaling

H.225/Q.931 is used in conjunction with H.323 and provides the signaling for call control. For establishing a call from a source to a receiver host, the **H.225 RAS** *(Registration, Admission and Signaling)* channel is used. After the call has been established, **H.245 Media Control** is used to negotiate the media streams.

2.2.2 H. 225.0: RAS

The RAS channel is used for the communication between the endpoints and the gatekeeper. Since the RAS messages are sent over UDP (an unreliable channel), it recommends timeouts and retry counts for messages. The procedures defined by the RAS channel are:

2.2.2.1 Gatekeeper discovery

By this process, an endpoint determines the gatekeeper with which it should register. The endpoint normally multicasts a **Gatekeeper Request (GRQ)** message asking for its gatekeeper. One or more gatekeepers may respond with the **Gatekeeper Confirmation (GCF)** message thereby indicating the willingness to be the gatekeeper for that endpoint. The response includes the **transport address** of the gatekeeper's RAS channel. Gatekeepers who do not want the endpoint to register with it can send a **Gatekeeper Reject (GRJ)** message. If more than one gatekeeper responds with GCF, then the endpoint may choose the gatekeeper and register with it. If no gatekeeper responds within a timeout interval, the endpoint may retransmit the GRQ.

2.2.2.2 Endpoint Registration

By this process, endpoint joins a zone (collection of gatekeeper and endpoints) later informs the gatekeeper of its transport and alias addresses. Endpoint send a **Registration Request (RRQ)** to gatekeeper's RAS channel Transport Address. Endpoint has network address of gatekeeper from Gatekeeper Discovery Process and uses RAS channel **TSAP** identifier (**Transport Layer Service Access Point**-allows multiplexing of various channels, gatekeeper and endpoints have their own TSAP). The gatekeeper responds with either a **Registration Confirmation (RCF)** or a **Registration Reject (RRJ)**. The gatekeeper then ensures that each alias address translates uniquely to a single transport address. An endpoint or a gatekeeper may cancel its registration by sending a **Un-register Request (URQ)** message to the gatekeeper. The gatekeeper shall respond with a **Un-register Confirmation (UCF)** message.

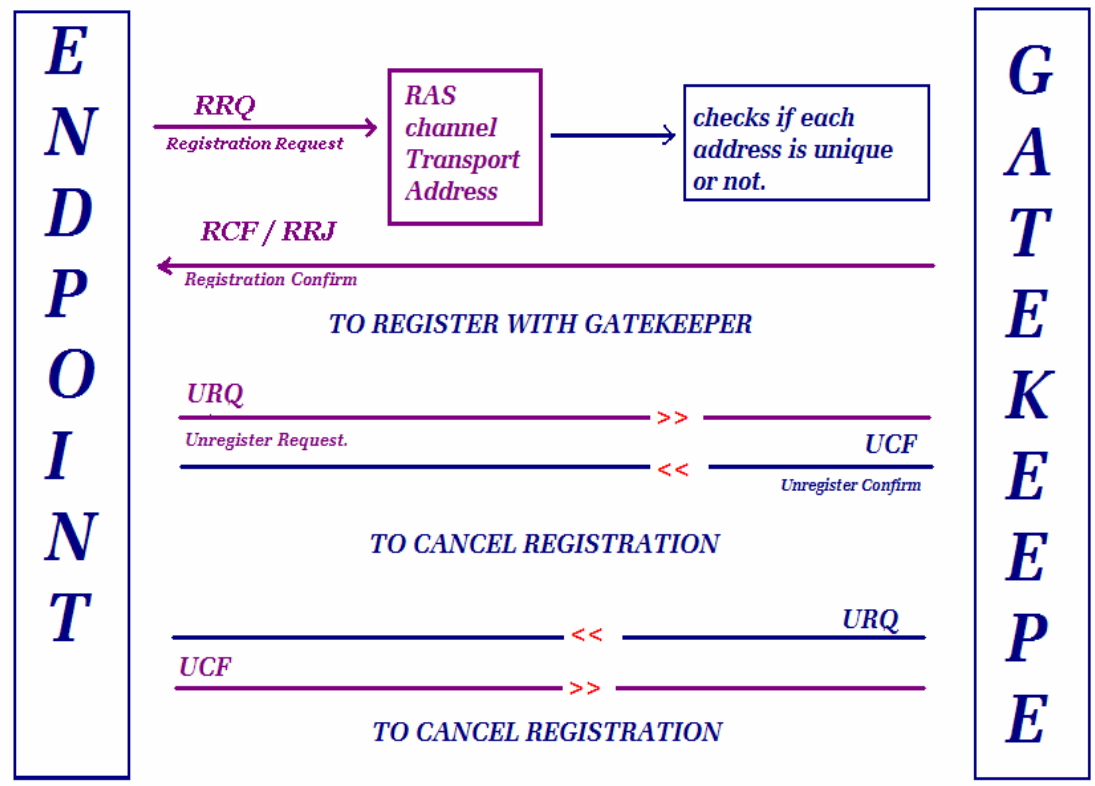

FIGURE 3: END POINT REGISTRATION

2.2.2.3 Endpoint Location

A Gatekeeper, having an address of endpoint, would like to locate endpoint, which it does by sending **Location request (LRQ)** message. The gatekeeper, with whom the requested endpoint is registered, shall respond with the **Location Confirmation (LCF)** message containing the contact information of the endpoint or the endpoint's gatekeeper.

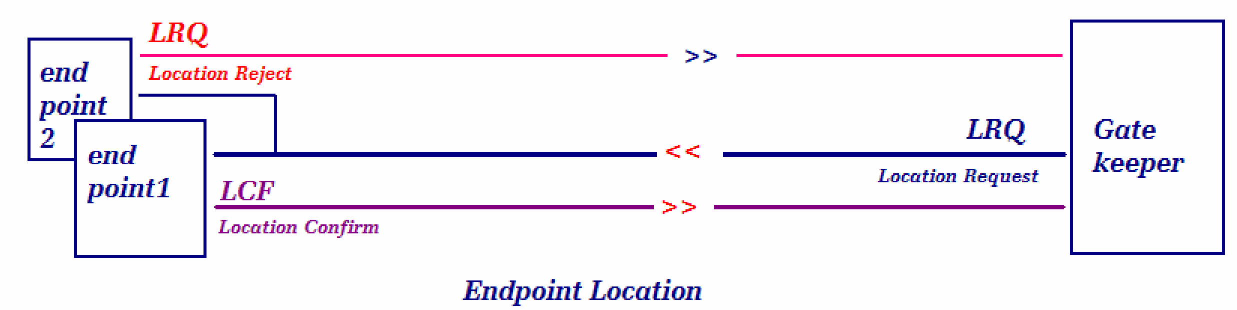

FIGURE 4: ENDPOINT LOCATION

All gatekeepers, with whom the requested endpoint is not registered, shall return **Location Reject (LRJ)** if they received the LRQ on the RAS channel

2.2.2.4 Admissions, Bandwidth Change, Status and Disengage

The RAS channel is also used for the transmission of Admissions, Bandwidth Change, Status and Disengage messages. These messages are exchanged between an endpoint and a gatekeeper and are used to provide admissions control and bandwidth management functions. The

Admissions Request (ARQ) message specifies the requested Call bandwidth. The gatekeeper may reduce the requested call bandwidth in the **Admissions Confirm (ACF)** message. An endpoint or the gatekeeper may attempt to modify the call bandwidth during a call using the **Bandwidth Change Request (BRQ)** message.

2.2.3 H.245 Media and Conference Control

After the call is established, H.245 (Media protocol) is used to negotiate and establish all those media channels that are carried by **RTP (Real Time Transport Protocol)**.

The functionality offered by H.245 is:

1. Determining master and slaves - Multipoint Controller (MC) is held responsible for central control in cases where a call is extended to a conference
2. Capability Exchange – It negotiates the capabilities when a call has been established or when call is going on.
3. Media Channel Control - After exchange of capabilities, endpoints may open and close logical channels of media often termed as logical channels.
4. Conference Control - Provides endpoints with mutual awareness and establishes the media flow model between all the endpoints.

2.3 Call Setup in H.323

The procedure to set up a call involves:

- Discovering a gatekeeper which would take the management of that endpoint
- Registration of the endpoint with its gatekeeper
- Endpoint enters the call setup phase
- The capability exchange takes place between the endpoint and the gatekeeper
- The call is established
- When the endpoint is done, it can terminate the call. The termination can also be initiated by the gatekeeper

2.3.1 H.225.0 Call Signaling

H.225 control messages are carried by Call Signaling Channel. In absence of a gatekeeper, call signaling messages are directly passed between a calling and called endpoint. And this link is established using Call Signaling Transport Address.

Assume that **calling** endpoint **knows** Call Signaling Transport **Address of called** endpoint and hence direct communication is possible. In networks not involving gatekeeper, the initial admission message exchange takes place between the **calling** endpoint and the **gatekeeper** using the gatekeeper's RAS channel transport address. The call signaling is done over TCP.

Call Signaling messages may be passed in two ways. The first way is Gatekeeper Routed Call Signaling, where the call signaling messages are routed through the gatekeeper between the endpoints. The other alternative is Direct Endpoint Call Signaling where the call signaling messages are passed directly between the endpoints.

Admissions messages are exchanged with the gatekeeper over the RAS channel, followed by an exchange of call signaling messages on a Call Signaling Channel which in turn is followed by the establishment of the H.245 Control Channel, which follows the same pattern as Call Signaling.

3. Conclusion

Is H.323 the only Protocol?

H.323, the oldest protocol, is not the only protocol used for audio transmission. It was designed with keeping in mind ATM and ISDN signaling but was not found to be well-suited for controlling the voice over IP systems. Being the oldest of its kind, it has a larger market hold compared to SIP (Session Initiation Protocol) which is backed up by IETF.

Seeing the inherent complexity, overheads, lack of extensibility, SIP has been designed by keeping Internet in mind, so it avoids both complexity and extensibility and has the benefit of reusing most of the header fields, encoding rules, error codes and authentication mechanisms of HTTP. When confined to single LAN, H.323 gives great speed as data is transmitted in pure binary form.

4. Further Reading

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