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Voice over IP Protocols

An Overview

What is in this Module

Module Title:
Voice over IP Protocol – An Overview

Objectives:

This module provides an introductory overview of the voice over IP protocols: SIP, H.323 and MGCP. At the end of this module, you will:

- **Understand the basics of SIP and its architecture.**
- **Understand H.323 and how it compares to SIP.**
- **Understand MGCP.**

Target Audience:

Marketing or business development professional who would like an introductory yet technical overview of the voice over IP protocols.

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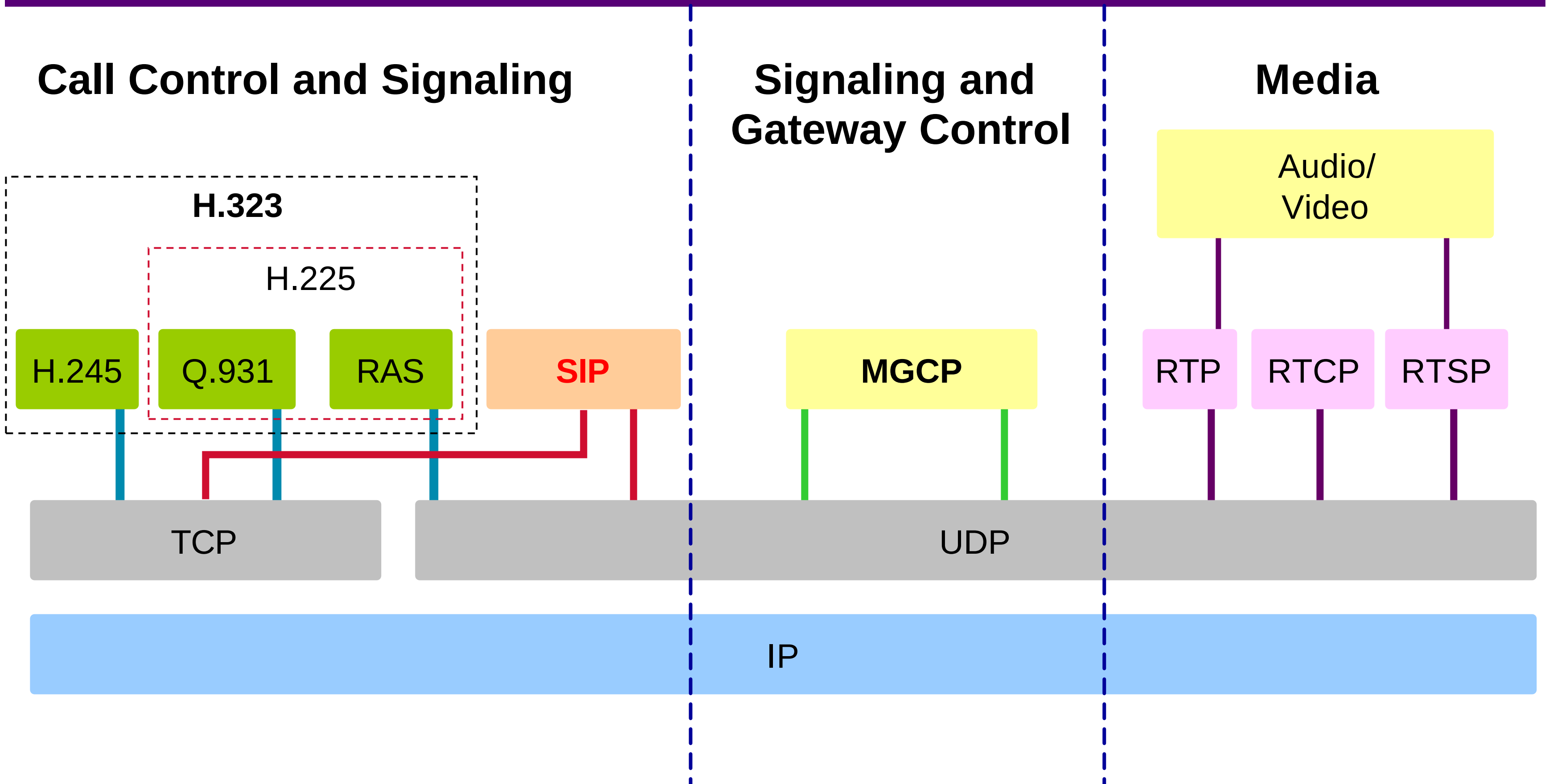
Voice over IP Protocols

Pictorial Overview

SIP, H.323 and MGCP



H.323 Version 1 and 2 supports H.245 over TCP, Q.931 over TCP and RAS over UDP.

H.323 Version 3 and 4 supports H.245 over UDP/TCP and Q.931 over UDP/TCP and RAS over UDP.

SIP supports TCP and UDP.

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Session Initiation Protocol

What is SIP?

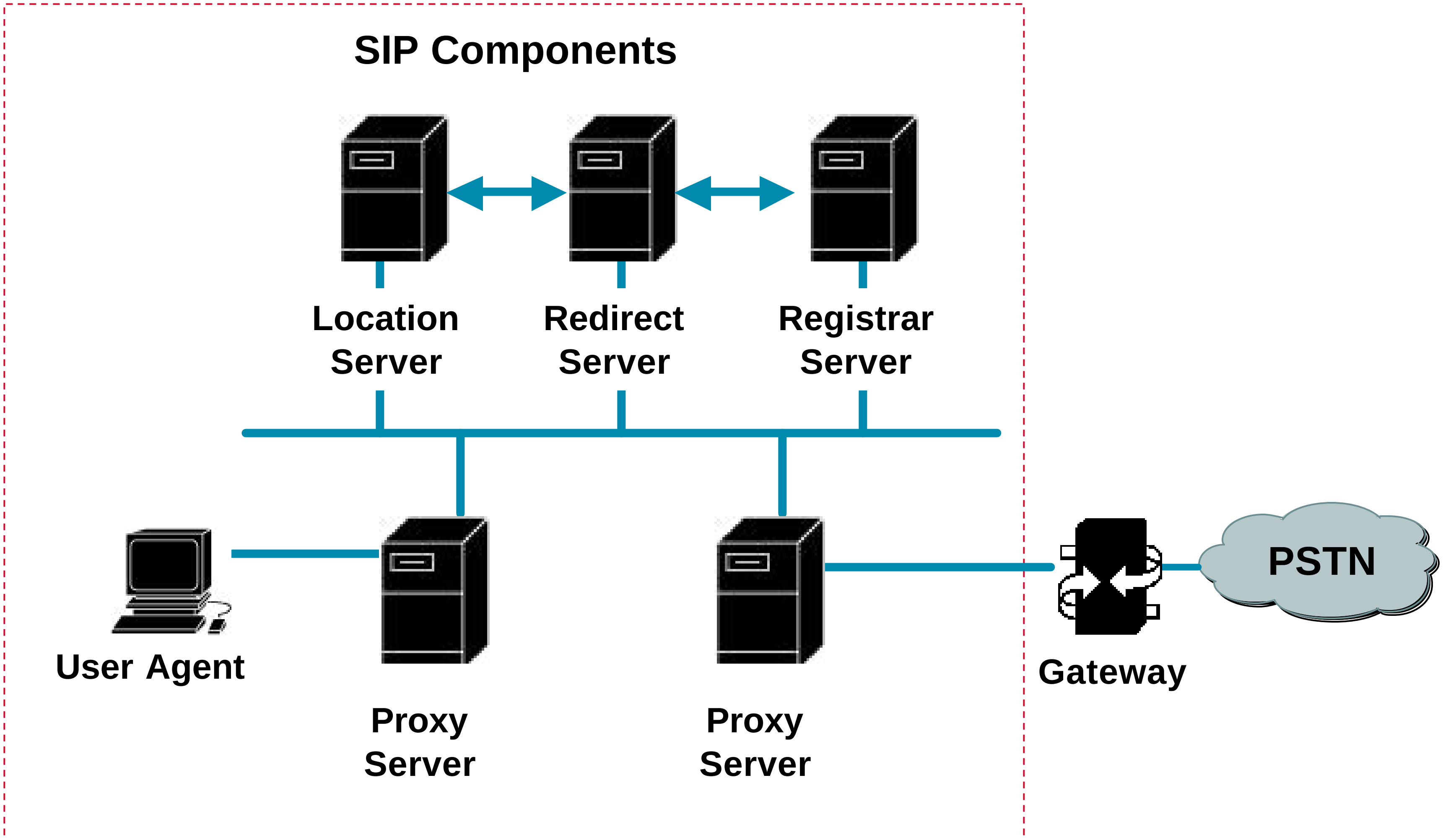
“Session Initiation Protocol - An application layer signaling protocol that defines initiation, modification and termination of interactive, multimedia communication sessions between users.”

IETF RFC 2543 Session Initiation Protocol

SIP Framework

- Session initiation.
- Multiple users.
- Interactive multimedia applications.

SIP Distributed Architecture



User Agents

An application that initiates, receives and terminates calls.

- **User Agent Clients (UAC) – An entity that initiates a call.**
- **User Agent Server (UAS) – An entity that receives a call.**

➢ **Both UAC and UAS can terminate a call.**

Proxy Server

- An intermediary program that acts as both a server and a client to make requests on behalf of other clients.
- Requests are serviced internally or by passing them on, possibly after translation, to other servers.
- Interprets, rewrites or translates a request message before forwarding it.

Location Server

- **A location server is used by a SIP redirect or proxy server to obtain information about a called party's possible location(s).**

Redirect Server

- A server that accepts a SIP request, maps the address into zero or more new addresses and returns these addresses to the client.
- Unlike a proxy server, the redirect server does not initiate its own SIP request.
- Unlike a user agent server, the redirect server does not accept or terminate calls.

Registrar Server

- **A server that accepts REGISTER requests.**
- **The register server may support authentication.**
- **A registrar server is typically co-located with a proxy or redirect server and may offer location services.**

SIP Messages – Methods and Responses

SIP components communicate by exchanging SIP messages:

<u>SIP Methods:</u>

- **INVITE – Initiates a call by inviting user to participate in session.**
- **ACK - Confirms that the client has received a final response to an INVITE request.**
- **BYE - Indicates termination of the call.**
- **CANCEL - Cancels a pending request.**
- **REGISTER – Registers the user agent.**
- **OPTIONS – Used to query the capabilities of a server.**
- **INFO – Used to carry out-of-bound information, such as DTMF digits.**

<u>SIP Responses:</u>

- **1xx - Informational Messages.**
- **2xx - Successful Responses.**
- **3xx - Redirection Responses.**
- **4xx - Request Failure Responses.**
- **5xx - Server Failure Responses.**
- **6xx - Global Failures Responses.**

SIP Headers

- SIP borrows much of the syntax and semantics from HTTP.
- A SIP messages looks like an HTTP message – message formatting, header and MIME support.
- An example SIP header:

```
-----------------------------------------------------------------
                        SIP Header
-----------------------------------------------------------------
INVITE sip:5120@192.168.36.180 SIP/2.0
Via: SIP/2.0/UDP 192.168.6.21:5060
From: sip:5121@192.168.6.21
To: <sip:5120@192.168.36.180>
Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21
CSeq: 100 INVITE
Expires: 180
User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Accept: application/sdp
Contact: sip:5121@192.168.6.21:5060
Content-Type: application/sdp
```

SIP Addressing

- The SIP address is identified by a SIP URL, in the format: user@host.
- Examples of SIP URLs:
  - sip:hostname@vovida.org
  - sip:hostname@192.168.10.1
  - sip:14083831088@vovida.org

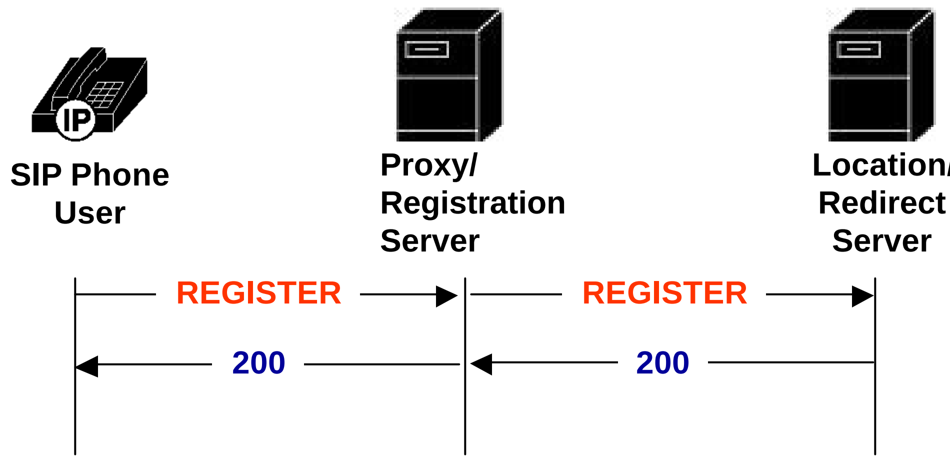
Process for Establishing Communication

Establishing communication using SIP usually occurs in six steps:

1. **Registering, initiating and locating the user.**
2. **Determine the media to use – involves delivering a description of the session that the user is invited to.**
3. **Determine the willingness of the called party to communicate – the called party must send a response message to indicate willingness to communicate – accept or reject.**
4. **Call setup.**
5. **Call modification or handling – example, call transfer (optional).**
6. **Call termination.**

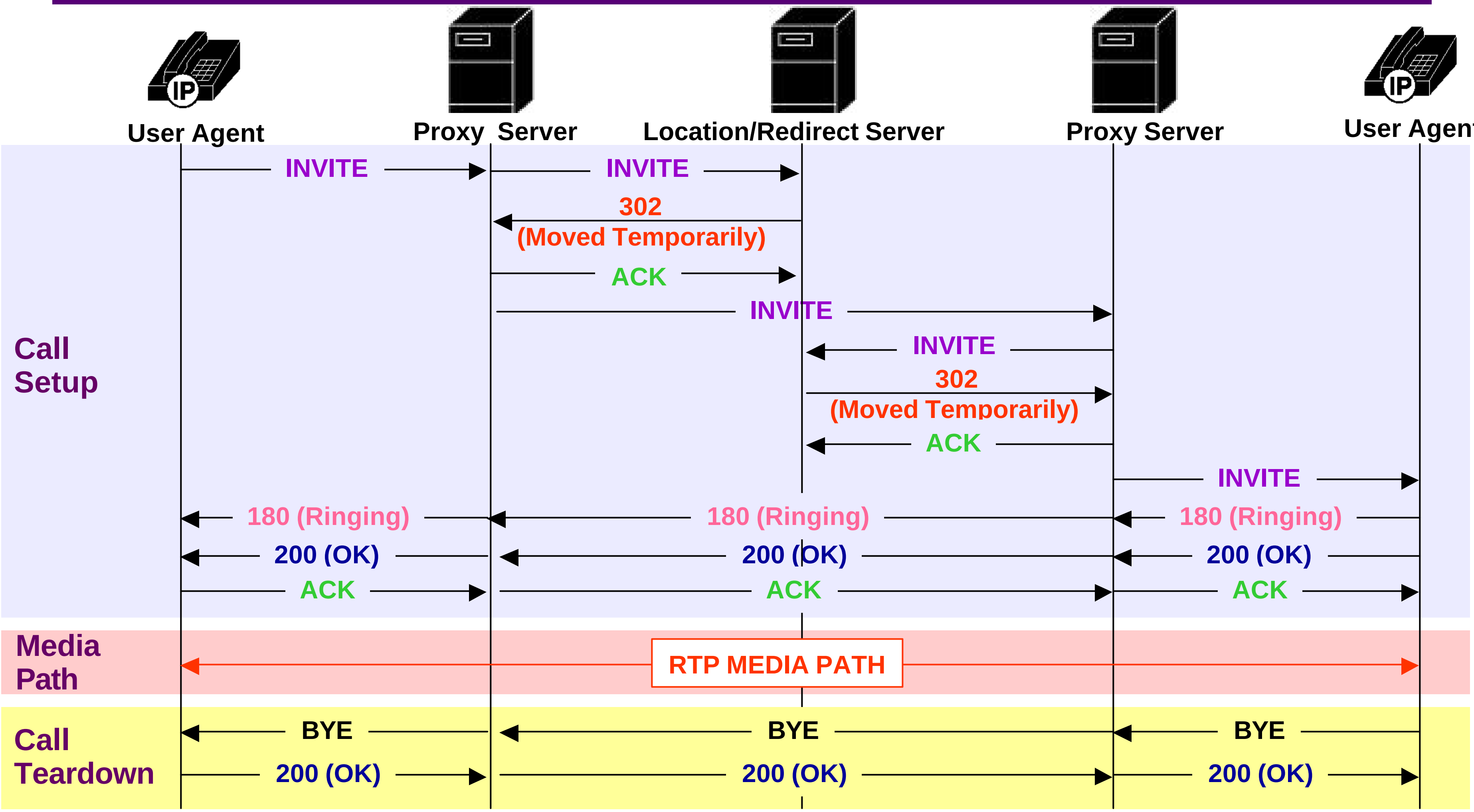
Registration

- **Each time a user turns on the SIP user client (SIP IP Phone, PC, or other SIP device), the client registers with the proxy/registration server.**
- **Registration can also occur when the SIP user client needs to inform the proxy/registration server of its location.**
- **The registration information is periodically refreshed and each user client must re-register with the proxy/registration server.**
- **Typically the proxy/registration server will forward this information to be saved in the location/redirect server.**



SIP Messages:
REGISTER – Registers the address listed in the To header field.
200 – OK.

Simplified SIP Call Setup and Teardown



SIP – Design Framework

SIP was designed for:

- **Integration with existing IETF protocols.**
- **Scalability and simplicity.**
- **Mobility.**
- **Easy feature and service creation.**

Integration with IETF Protocols (1)

Other IETF protocol standards can be used to build a SIP based application. SIP can works with existing IETF protocols, for example:

- **RSVP - to reserve network resources.**
- **RTP Real Time Protocol -to transport real time data and provide QOS feedback.**
- **RTSP Real Time Streaming Protocol - for controlling delivery of streaming media.**
- **SAP Session Advertisement Protocol - for advertising multimedia session via multicast.**

Integration with IETF Protocols (2)

- SDP Session Description Protocol – for describing multimedia sessions.
- MIME – Multipurpose Internet Mail Extension – defacto standard for describing content on the Internet.
- HTTP – Hypertext Transfer Protocol - HTTP is the standard protocol used for serving web pages over the Internet.
- COPS – Common Open Policy Service.
- OSP – Open Settlement Protocol.

Scalability

- The SIP architecture is scalable, flexible and distributed.
  - Functionality such as proxying, redirection, location, or registration can reside in different physical servers.
  - Distributed functionality allows new processes to be added without affecting other components.

Simplicity

SIP is designed to be:

- **“Fast and simple in the core.”**
- **“Smarter with less volume at the edge.”**
- **Text based for easy implementation and debugging.**

Mobility

- SIP supports user mobility by proxying and redirecting requests to a user’s current location.
- The user can be using a PC at work, PC at home, wireless phone, IP phone, or regular phone.
- The user must register their current location.
- The proxy server will forward calls to the user’s current location.
- Example mobility applications include presence and call forking.

Feature Creation

A SIP based system can support rapid feature and service creations.

For example, features and services can be created using:

- **Call Processing Language (CPL).**
- **Common Gateway Interface (CGI).**

Feature Creation (2)

SIP can support these features and applications:

- **Basic call features (call waiting, call forwarding, call blocking etc.).**
- **Unified messaging.**
- **Call forking.**
- **Click to talk.**
- **Presence.**
- **Instant messaging.**
- **Find me / Follow me.**

References

For more information on SIP refer to:

IETF

- http://www.ietf.org/html.charters/sip-charter.html

Henning Schulzrinne's SIP page

- http://www.cs.columbia.edu/~hgs/sip/

H.323

What is H.323?

“Describes terminals and other entities that provide multimedia communications services over Packet Based Networks (PBN) which may not provide a guaranteed Quality of Service. H.323 entities may provide real-time audio, video and/or data communications.”

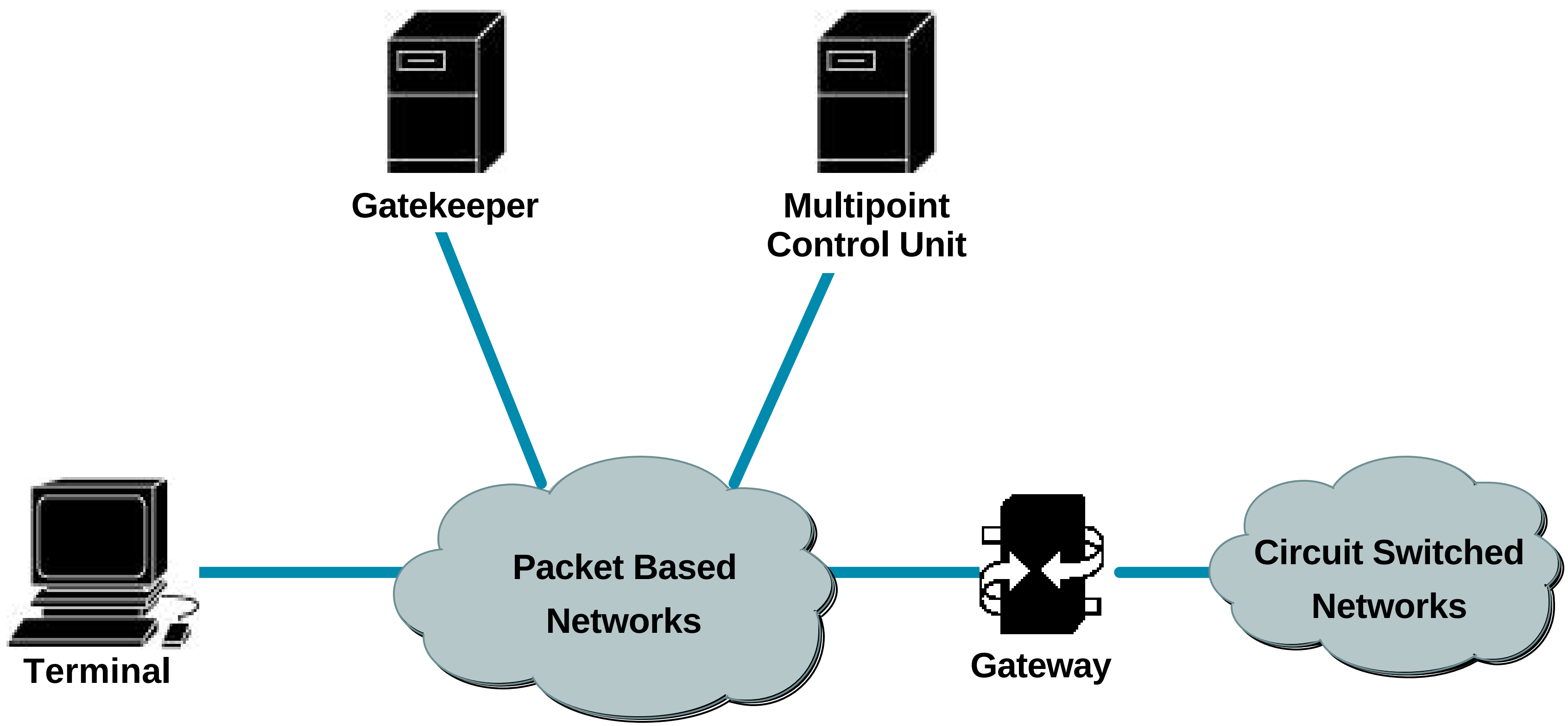
ITU-T Recommendation H.323 Version 4

H.323 Framework

H.323 defines:

- **Call establishment and teardown.**
- **Audio visual or multimedia conferencing.**

H.323 Components



H.323 Terminals

H.323 terminals are client endpoints that must support:

- **H.225 call control signaling.**
- **H.245 control channel signaling.**
- **RTP/RTCP protocols for media packets.**
- **Audio codecs.**

➢ **Video codecs support is optional.**

H.323 Gateway

A gateway provides translation:

- For example, a gateway can provide translation between entities in a packet switched network (example, IP network) and circuit switched network (example, PSTN network).
- Gateways can also provide transmission formats translation, communication procedures translation, H.323 and non-H.323 endpoints translations or codec translation.

H.323 Gatekeepers

Gatekeepers provide these functions:

- **Address translation.**
- **Admission control.**
- **Bandwidth control.**
- **Zone management.**
- **Call control signaling (optional).**
- **Call authorization (optional).**
- **Bandwidth management (optional).**
- **Call management (optional).**

Gatekeepers are optional but if present in a H.323 system, all H.323 endpoints must register with the gatekeeper and receive permission before making a call.

H.323 Multipoint Control Unit

MCU provide support for conferences of three or more endpoints.

An MCU consist of:

- **Multipoint Controller (MC) – provides control functions.**
- **Multipoint Processor (MP) – receives and processes audio, video and/or data streams.**

H.323 is an "Umbrella" Specification

Media

H.261 and H.263 – Video codecs.

G.711, G.723, G.729 – Audio codecs.

RTP/RTCP – Media.

Data/Fax

T.120 – Data conferencing.

T.38 – Fax.

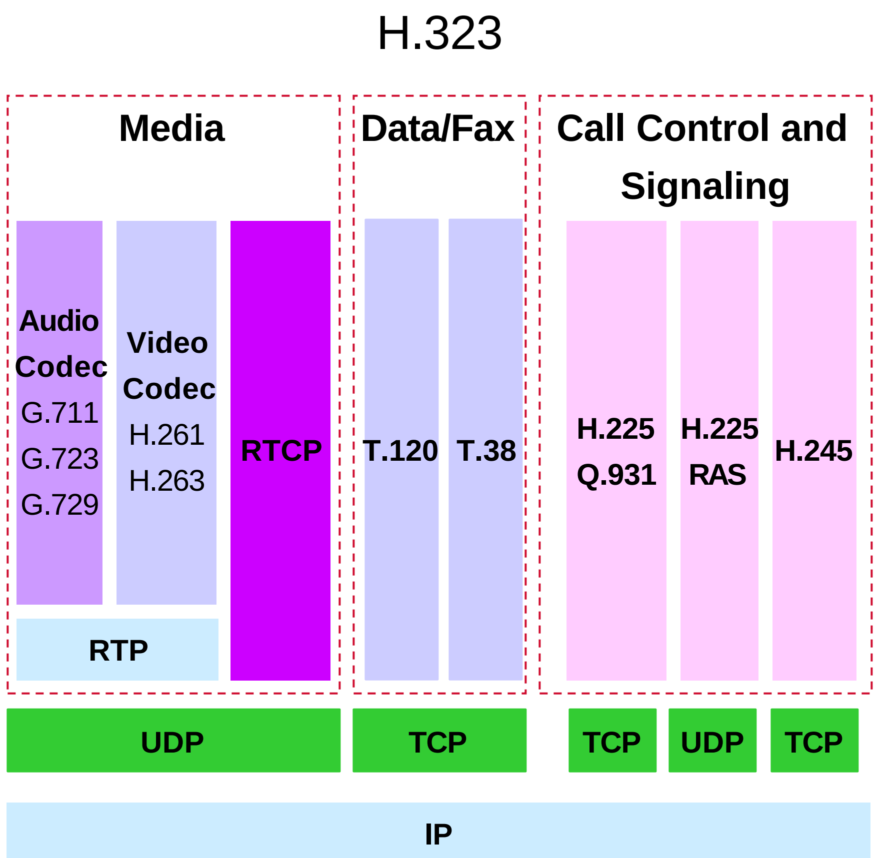
Call Control and Signaling

H.245 - Capabilities advertisement, media channel establishment, and conference control.

H.225

Q.931 - call signaling and call setup.

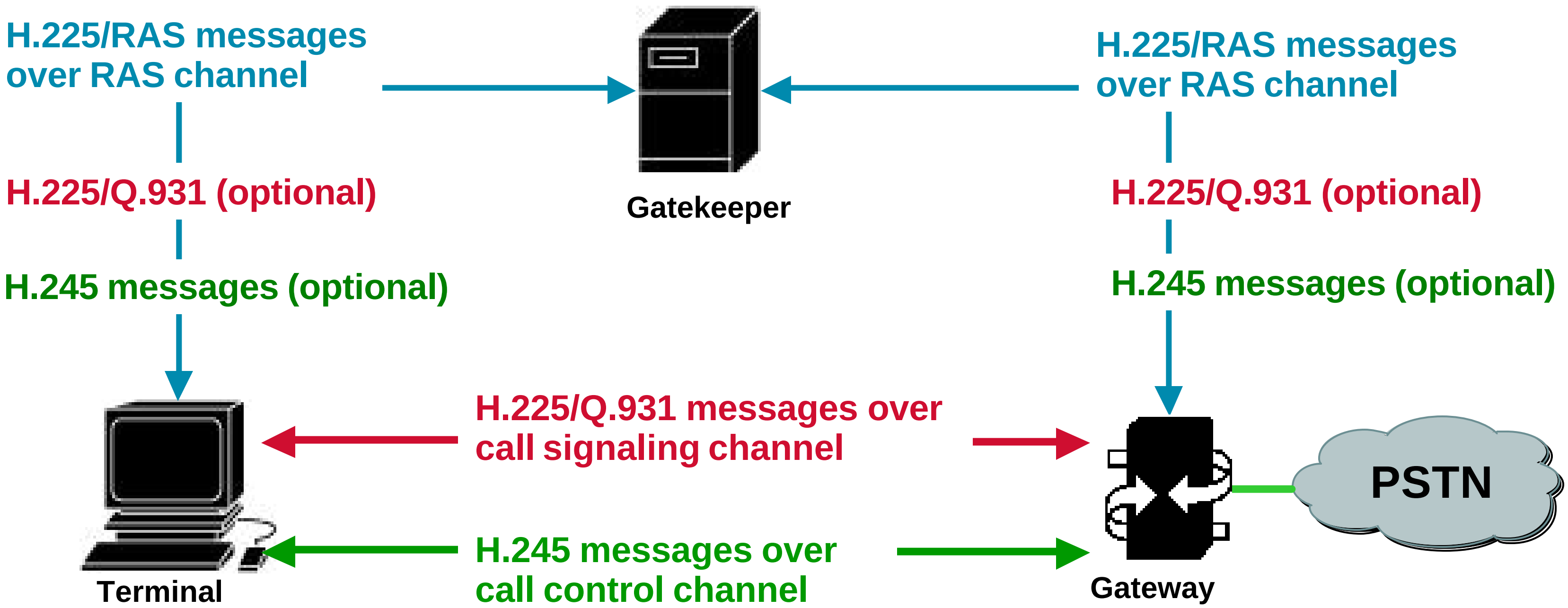
RAS - registration and other admission control with a gatekeeper.



Other ITU H. Recommendation that work with H.323

Protocol	Description
H.235	Specifies security and encryption for H.323 and H.245 based terminals.
H.450.N	H.450.1 specifies framework for supplementary services. H.450.N recommendation specifies supplementary services such as call transfer, call diversion, call hold, call park, call waiting, message waiting indication, name identification, call completion, call offer, and call intrusion.
H.246	Specifies internetworking of H Series terminals with circuit switched terminals.

H.323 Components and Signaling



H.245 – A protocol for capabilities advertisement, media channel establishment and conference control.

H.225 - Call Control.

- Q.931 – A protocol for call control and call setup.

- RAS – Registration, admission and status protocol used for communicating between an H.323 endpoint and a gatekeeper.

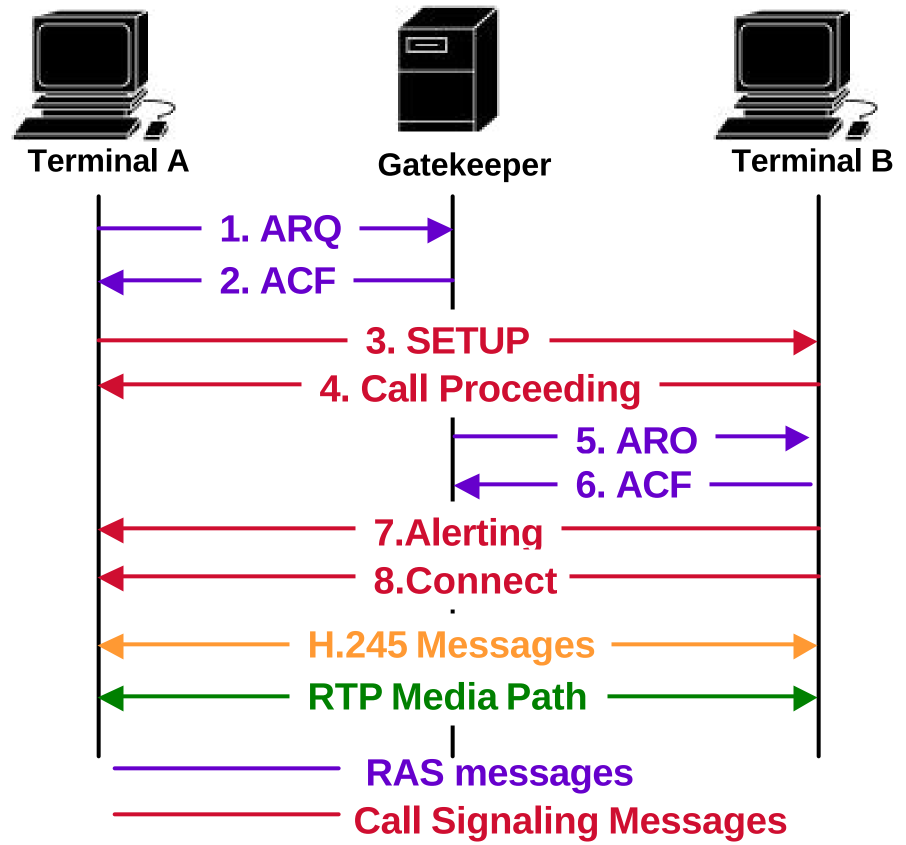
Process for Establishing Communication

Establishing communication using H.323 may occurs in five steps:

1. **Call setup.**
2. **Initial communication and capabilities exchange.**
3. **Audio/video communication establishment.**
4. **Call services.**
5. **Call termination.**

Simplified H.323 Call Setup

- **Both endpoints have previously registered with the gatekeeper.**
- **Terminal A initiate the call to the gatekeeper. (RAS messages are exchanged).**
- **The gatekeeper provides information for Terminal A to contact Terminal B.**
- **Terminal A sends a SETUP message to Terminal B.**
- **Terminal B responds with a Call Proceeding message and also contacts the gatekeeper for permission.**
- **Terminal B sends a Alerting and Connect message.**
- **Terminal B and A exchange H.245 messages to determine master slave, terminal capabilities, and open logical channels.**



Note: This diagram only illustrates a simple point-to-point call setup where call signaling is not routed to the gatekeeper. Refer to the H.323 recommendation for more call setup scenarios.

Versions of H.323

Version	Date	Reference for key feature summary
H.323 Version 1	May 1996	New release. Refer to the specification. http://www.packetizer.com/iptel/h323/
H.323 Version 2	January 1998	http://www.packetizer.com/iptel/h323/whatsnew_v2.html
H.323 Version 3	September 1999	http://www.packetizer.com/iptel/h323/whatsnew_v3.html
H.323 Version 4	November 2000	http://www.packetizer.com/iptel/h323/whatsnew_v4.html

References

For more information on H.323 refer to:

ITU-T

- http://www.itu.int/itudoc/itu-t/rec/index.html

Packetizer

- http://www.packetizer.com/iptel/h323/

Open H.323

- http://www.openH323.org

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Comparing

SIP and H.323

Comparing SIP and H.323 - Similarities

Functionally, SIP and H.323 are similar. Both SIP and H.323 provide:

- **Call control, call setup and teardown.**
- **Basic call features such as call waiting, call hold, call transfer, call forwarding, call return, call identification, or call park.**
- **Capabilities exchange.**

Comparing SIP and H.323 - Strengths

- **H.323 – Defines sophisticated multimedia conferencing. H.323 multimedia conferencing can support applications such as whiteboarding, data collaboration, or video conferencing.**
- **SIP – Supports flexible and intuitive feature creation with SIP using SIP-CGI (SIP-Common Gateway Interface) and CPL (Call Processing Language).**
- **SIP – Third party call control is currently only available in SIP. Work is in progress to add this functionality to H.323.**

Table 1 - SIP and H.323

	SIP	H.323
Standards Body	IETF.	ITU.
Relationship	Peer-to-Peer.	Peer-to-Peer.
Origins	Internet based and web centric. Borrows syntax and messages from HTTP.	Telephony based. Borrows call signaling protocol from ISDN Q.SIG.
Client	Intelligent user agents.	Intelligent H.323 terminals.
Core servers	SIP proxy, redirect, location, and registration servers.	H.323 Gatekeeper.
Current Deployment	Interoperability testing between various vendor's products is ongoing at SIP bakeoffs. SIP is gaining interest.	Widespread.
Interoperability	IMTC sponsors interoperability events among SIP, H.323, and MGCP. For more information, visit: http://www.imtc.org/	

Table 2 - SIP and H.323

	SIP	H.323
Capabilities Exchange	SIP uses SDP protocol for capabilities exchange. SIP does not provide as extensive capabilities exchange as H.323.	Supported by H.245 protocol. H.245 provides structure for detailed and precise information on terminal capabilities.
Control Channel Encoding Type	Text based UTF-8 encoding.	Binary ASN.1 PER encoding.
Server Processing	Stateless or stateful.	Version 1 or 2 – Stateful. Version 3 or 4 – Stateless or stateful.
Quality of Service	SIP relies on other protocols such as RSVP, COPS, OSP to implement or enforce quality of service.	Bandwidth management/control and admission control is managed by the H.323 gatekeeper. The H323 specification recommends using RSVP for resource reservation.

Table 3 - SIP and H.323

	SIP	H.323
Security	**Registration -** User agent registers with a proxy server. **Authentication -** User agent authentication uses HTTP digest or basic authentication. **Encryption -** The SIP RFC defines three methods of encryption for data privacy.	**Registration -** If a gatekeeper is present, endpoints register and request admission with the gatekeeper. **Authentication and Encryption -** H.235 provides recommendations for authentication and encryption in H.323 systems.
Endpoint Location and Call Routing	Uses SIP URL for addressing. Redirect or location servers provide routing information.	Uses E.164 or H323ID alias and a address mapping mechanism if gatekeepers are present in the H.323 system. Gatekeeper provides routing information.

Table 4 – SIP and H.323

	SIP	H.323
Features	Basic call features.	Basic call features.
Conferencing	Basic conferencing without conference or floor control.	Comprehensive audiovisual conferencing support. Data conferencing or collaboration defined by T.120 specification.
Service or Feature Creation	Supports flexible and intuitive feature creation with SIP using SIP-CGI and CPL. Some example features include presence, unified messaging, or find me/follow me.	H.450.1 defines a framework for supplementary service creation.

Note: Basic call features include: call hold, call waiting, call transfer, call forwarding, caller identification, and call park.

Reference

This section cites a document that provides a comprehensive comparison on H.323 and SIP:

Dalgic, Ismail. Fang, Hanlin. "Comparison of H.323 and SIP for IP Telephony Signaling" in Proc. of Photonics East, (Boston, Massachusetts), SPIE, Sept. 1999.

http://www.cs.columbia.edu/~hgs/papers/others/Dalg9909_Comparison.pdf

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MGCP

Media Gateway Control Protocol

What is MGCP?

"

Media Gateway Control Protocol - A protocol for controlling telephony gateways from external call control elements called media gateway controllers or call agents.

"

IETF RFC 2705 Media Gateway Control Protocol

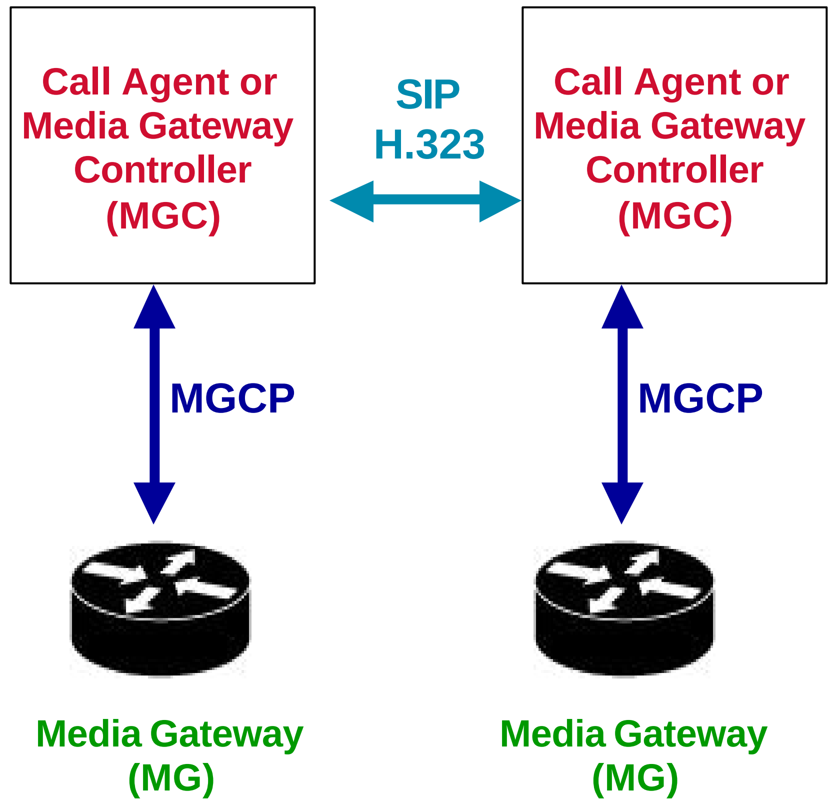
Components

Call agent or media gateway controller

- **Provides call signaling, control and processing intelligence to the gateway.**
- **Sends and receives commands to/from the gateway.**

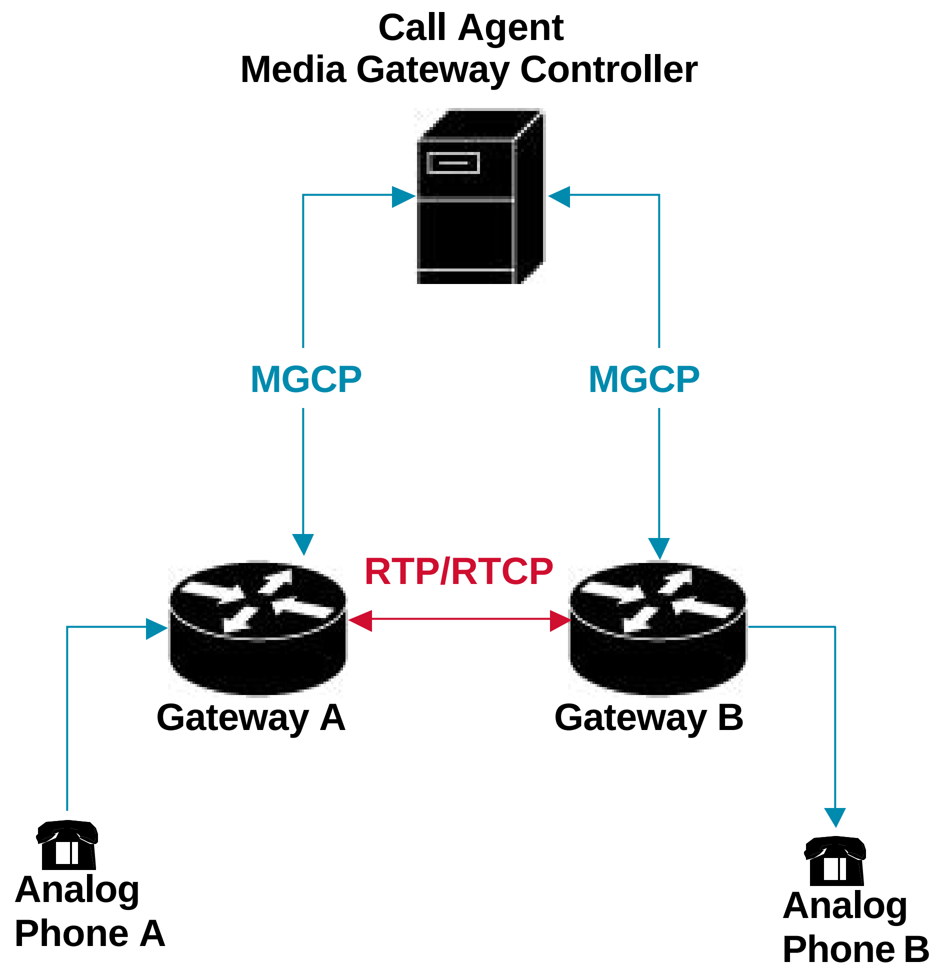
Gateway

- **Provides translations between circuit switched networks and packet switched networks.**
- **Sends notification to the call agent about endpoint events.**
- **Execute commands from the call agents.**



Simplified Call Flow

- When Phone A goes offhook Gateway A sends a signal to the call agent.
- Gateway A generates dial tone and collects the dialed digits.
- The digits are forwarded to the call agent.
- The call agent determines how to route the call.
- The call agent sends commands to Gateway B.
- Gateway B rings phone B.
- The call agent sends commands to both gateways to establish RTP/RTCP sessions.



MGCP Commands

Call Agent Commands:

- EndpointConfiguration
- NotificationRequest
- CreateConnection
- ModifyConnection
- DeleteConnection
- AuditEndpoint
- AuditConnection

Gateway Commands:

- Notify
- DeleteConnection
- RestartInProgress

Characteristics of MGCP

MGCP:

- **A master/slave protocol.**
  - **Assumes limited intelligence at the edge (endpoints) and intelligence at the core (call agent).**
  - **Used between call agents and media gateways.**
  - **Differs from SIP and H.323 which are peer-to-peer protocols.**
- **Interoperates with SIP and H.323.**

MGCP, SIP and H.323

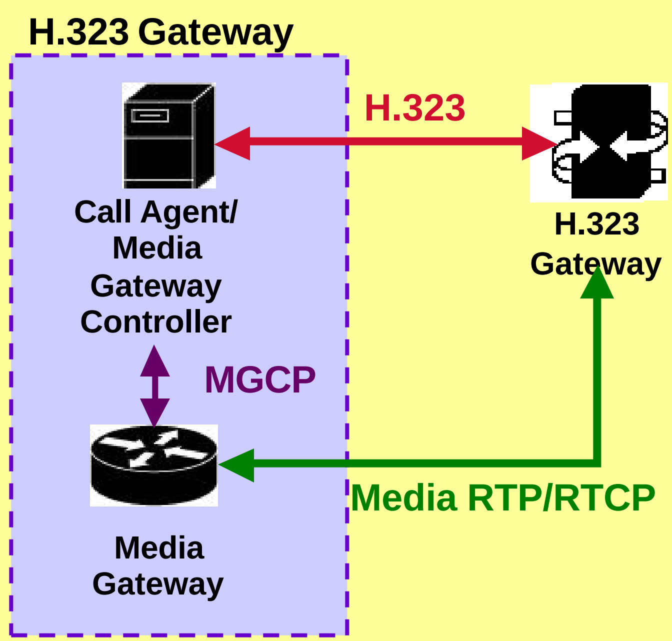
- MGCP divides call setup/control and media establishment functions.
- MGCP does not replace SIP or H.323. SIP and H.323 provide symmetrical or peer-to-peer call setup/control.
- MGCP interoperates with H.323 and SIP. For example,
  - A call agent accepts SIP or H.323 call setup requests.
  - The call agent uses MGCP to control the media gateway.
  - The media gateway establishes media sessions with other H.323 or SIP endpoints.

In this example, an H.323 gateway is "decomposed" into:

–A call agent that provides signaling.

–A gateway that handles media.

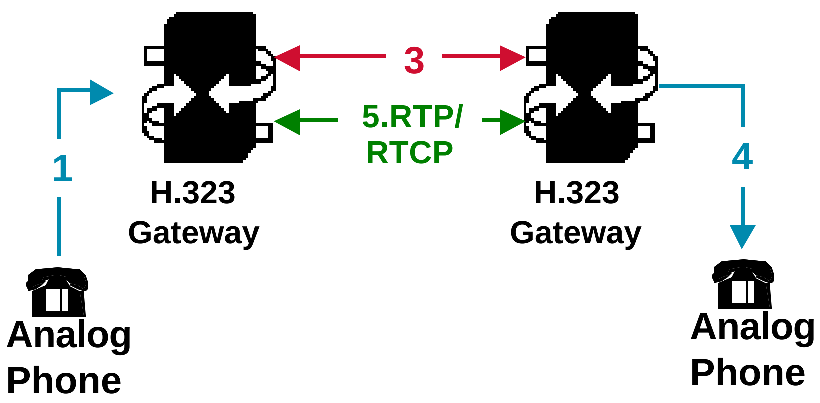
➢MGCP protocol is used to control the gateway.



Example Comparison

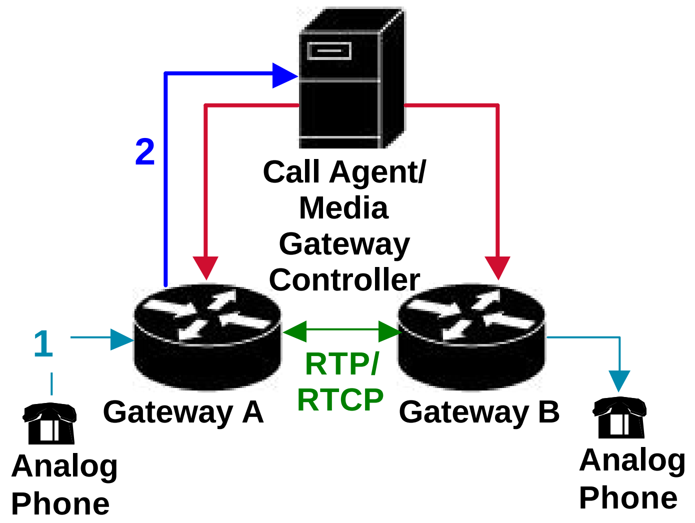
H.323

1. A user picks up analog phone and dials a number.
2. The gateway determines how to route the call.
3. The two gateways exchange capabilities information.
4. The terminating gateway rings the phone.
5. The two gateways establish RTP/RTCP session with each other.



MGCP

1. A user picks up analog phone and dials a number.
2. The gateway notifies call agent of the phone (endpoint) event.
3. The Call agent determines capabilities, routing information, and issues a command to the gateways to establish RTP/RTCP session with other end.



What is Megaco?

A protocol that is evolving from MGCP and developed jointly by ITU and IETF:

- **Megaco - IETF.**
- **H.248 or H.GCP - ITU.**

For more information refer to:

- **IETF - http://www.ietf.org/html.charters/megaco-charter.html**
- **Packetizer - http://www.packetizer.com/iptel/h248/**

References

For more information on MGCP refer to:

IETF

- http://www.ietf.org/rfc/rfc2705.txt?number=2705

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Summary

Summary

- **SIP and H.323 are comparable protocols that provide call setup, call teardown, call control, capabilities exchange, and supplementary features.**
- **MGCP is a protocol for controlling media gateways from call agents. In a VoIP system, MGCP can be used with SIP or H.323. SIP or H.323 will provide the call control functionality and MGCP can be used to manage media establishment in media gateways.**

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Additional References

General VoIP Reference

Pulver – IP Telephony News

- http://www.pulver.com

Internet Telephony

- http://www.internettelephony.com

An overview poster of the SIP, MGCP, and H323 protocols.

- http://www.protocols.com/voip/posvoip.pdf

End of Module

This is the end of the VoIP Protocol Overview training module.

For additional training and documentation visit us at www.vovida.org.