SIP 0101 MGCP
MEGACO 0101 H.323

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The Visionary's Guide to the IP Universe

H.323 MGCP
MEGACO gateways SIP
conferencing bridges
gatekeepers

Will Take You Beyond...

...The Basics

This pocket guide introduces you to the BASIC concepts behind SIP, MGCP, H.248/MEGACO, and H.323, and is just one part of RADVision's new **Visionary's Guide to the IP Universe.** This guide describes the various entities that are defined by the different protocols as well as the network architecture philosophies behind each protocol. Please accept this handbook with our compliments.

The second part of The Visionary's Guide to the IP Universe that RADVision will be sending you will take you beyond the basics presented here. It will show you how the puzzle pieces described here fit to together in a global IP-centric communications network in a somewhat surprising way!! To make sure you get the complete picture from RADVision, sign up as an "IP Visionary" via email (info@radvision.com), or on our website www.radvision.com.

The Basics...

Session Initiation Protocol (SIP)

What is SIP?

The Session Initiation Protocol, or SIP, is a new IETF signaling protocol for establishing real-time calls and conferences over Internet Protocol networks. SIP sessions involve one or more participants and can use unicast or multicast communication. Each session may include different types of data such as audio and video although currently most of the SIP extensions address audio communication. As a traditional text-based Internet protocol, it resembles the hypertext transfer protocol (HTTP) and simple mail transfer protocol (SMTP).

SIP is independent of the packet layer. The protocol is an open standard and is scalable. It has been designed to be a general-purpose protocol. However, extensions to SIP are needed to make the protocol truly functional in terms of interoperability. Among SIP basic features, the protocol enables personal mobility by providing the capability to reach a called party at a single, location-independent address. The services that SIP provide include:

- User Location: determination of the end system to be used for communication
- Call Setup: ringing and establishing call parameters at both called and calling party
- User Availability: determination of the willingness of the called party to engage in communications
- User Capabilities: determination of the media and media parameters to be used

- Call Handling: the transfer and termination of calls

Relationship Between SIP and H.323

Both SIP and H.323 define mechanisms for call routing, call signaling, capabilities exchange, media control, and supplementary services. SIP is a new protocol but promises scalability, flexibility and ease of implementation when building complex systems. H.323 is an established protocol that has been widely used because of its manageability, reliability and interoperability with PSTN. There is a general consensus among standards organizations, companies and technology experts that standardized procedures need to be specified to allow seamless interworking between the two protocols. Bodies such as TIPHON (ETSI), aHIT (IMTC) and IETF are working to address this topic.

As a champion for interworking between multiple IP communications protocols, RADVision is working closely with industry standards bodies and consortia on addressing issues needed to achieve true end-to-end connectivity between the two protocols. In order to achieve an interoperable system, the following needs to be addressed:

- Topology Definition: The architecture needs to be defined in terms of the respective entities. Call scenarios and interfaces must then be developed based on the defined topology.
- Supported Services: Advanced services such as conference calls and supplementary services should be incorporated.
- Supported "Data Capabilities:" Mapping H.245 channels to Session Description Protocol (SDP) is essential for issues relating to audio and multimedia support.

- Protocol Versions: The protocol versions of both SIP and H.323 need to be synchronized in order to define the scope of the interworking issues.

SIP Architecture & Entities

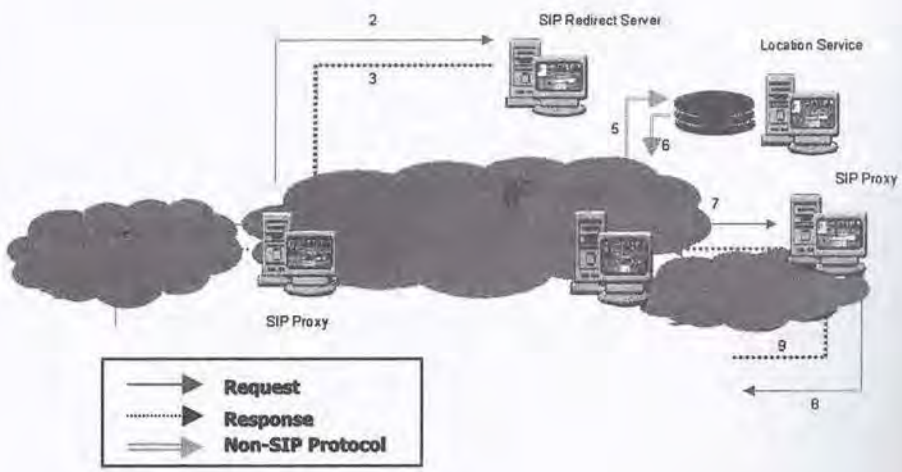
SIP is a client-server protocol that can be transported over UDP for simplicity and speed. The main entities in SIP are the *User Agent*, the *SIP Proxy Server*, the *SIP Redirect Server* and the *Registrar*.

The User Agents, or SIP endpoints, function as clients (UACs) when initiating requests, and as servers (UASs) when responding to requests. The User Agent also stores and manages call states. User Agents communicate with other User Agents directly or via intermediate servers.

SIP intermediate servers have the capability to behave as proxy or redirect servers. SIP Proxy Servers act as a server on one side (receiving requests) and as a client on the other side (possibly sending requests) for the purpose of making requests on behalf of User Agents. A proxy interprets and, if necessary, rewrites a request message before forwarding it.

The functionality of SIP Redirect Servers is similar to the role played by the H.323 gatekeeper when using a direct call model (see H.323 Basics chapter). SIP Redirect Servers respond to client requests and inform them of the requested server's address. Numerous hops can take place until reaching the final destination. When inserted as a front end, a SIP Redirect Server can be a helpful tool to achieve load balancing by distributing calls among secondary servers. SIP's tremendous flexibility allows the servers to contact external location servers in order to determine user or routing policies and therefore, does not bind the user into only one scheme to locate users. In order to maintain scalability, the SIP servers can either maintain state information or forward requests in a stateless fashion. A Redirect Server can also function as a primitive call distribution system.

The third entity comprising SIP is the SIP Registrar. The User Agent sends a registration message to the SIP Registrar and the Registrar stores the registration information in a location service via a non-SIP protocol. Once the information is stored, the Registrar sends the appropriate response back to the user agent. The Registrar can work together with SIP Redirect Servers in redirecting calls to the current location(s) of a caller.



Session Description Protocol

The Session Description Protocol, or SDP, is a protocol for describing audio, video and multimedia sessions. Both SIP and MGCP (Media Gateway Control Protocol) use SDP for media description and carrying out the negotiation for codec identification for the purpose of session announcement, session invitation, etc.

The Basics...

Media Gateway Control Protocol (MGCP)

What is MGCP?

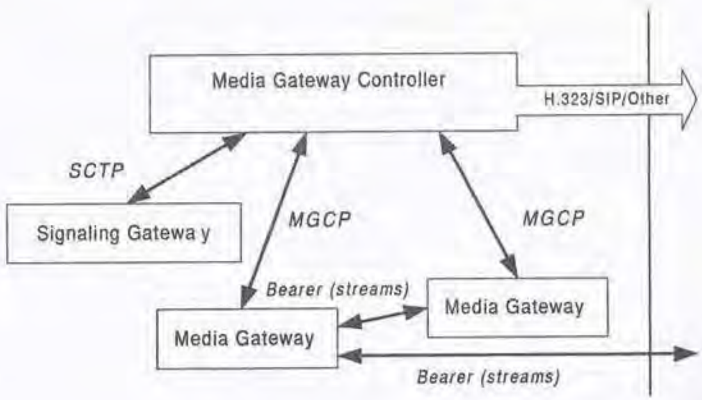
The Media Gateway Control Protocol, or MGCP, was designed to address the requirements of production IP telephony networks that are built using "decomposed" VoIP gateways. MGCP-based VoIP solutions separate call control (signaling) intelligence and media handling. MGCP functions as an internal protocol between the separate components of a decomposed MGCP-compliant VoIP gateway. More specifically, MGCP is a protocol used by external call control elements called Media Gateway Controllers (MGCs) for controlling Media Gateways (MGs). Decomposed MGCP-compliant VoIP gateways appear to the outside as a single VoIP gateway. Example VoIP gateways include:

- Trunking gateways that interface the public telephone network and VoIP network
- Residential gateways that provide traditional analog (RJ11) interfaces to VoIP networks
- Access gateways that provide traditional analog (RJ11) or digital PBX interfaces to VoIP networks

The evolution of the MGCP specification was largely influenced by "political" conflicts between proponents of alternative proposals for decomposed gateway architectures. In particular, MGCP owes its origin to the confluence of the SGCP (Simple Gateway Control Protocol) and IPDC (Internet Protocol Device Control) protocols.

Relationship Between MGCP, SIP and H.323

MGCP is a complementary protocol to both SIP and H.323. It was designed specifically as an internal protocol between MGCs and MGs for decomposed gateway architectures. In the MGCP model, an MGC handles call processing by interfacing with the IP network via communications with an IP signaling device such as an H.323 gatekeeper or SIP Server and with the circuit-switch network via an optional signaling gateway. Using an H.323 analogy, the MGC implements the "signaling" layers of H.323 and presents itself as an "H.323 Gatekeeper" or as one or more "H.323 Endpoints". Within the MGCP approach, MGs focus on the audio signal translation function, performing conversion between the audio signals carried on telephone circuits and data packets carried over the Internet or other packet networks.



MGCP Model

MGCP-compliant VoIP gateways provide a highly scalable solution for global IP communications networks and will co-exist in networks with other H.323 and SIP compliant entities.

MGCP Architecture & Entities

MGCP is a master/slave protocol with a tight coupling between the MG (endpoint) and MGC (server). Like SIP and H.323 it relies on a variety of other existing protocols such as SDP for describing the media aspects of a call, and RTP/RTCP (used by MGs) for handling the real-time transport of media streams.

In an MGCP architecture, the MGC server or "call agent" is mandatory and manages the calls and conferences and supports the services provided. The MG endpoint is unaware of the calls and conferences and does not maintain call states. MGs are expected to execute commands sent by the MGC call agents. MGCP assumes that call agents will synchronize with each other sending coherent commands to MGs under their control. MGCP does not define a mechanism for synchronizing call agents.

MGCP main entities are *Endpoints* and *Connections.* MGCP assumes a connection model where the basic constructs of Endpoints and Connections are used for establishing voice paths between call participants. Endpoints are sources or sinks of data and can be physical or virtual. Physical endpoint creation requires hardware installation while software is sufficient for creating a virtual endpoint. An interface on a gateway that terminates a trunk connected to a PSTN switch is an example of a physical endpoint. An audio source in an audio-content server is an example of a virtual endpoint.

Connections may be either point-to-point or multipoint. A point-to-point connection associates two endpoints. Once this association is established for both endpoints, data transfer between these endpoints can begin. A multipoint connection is established by connecting the endpoint to a multipoint session.

Connections can be established over several types of bearer networks: audio packet transmission using RTP and UDP over a TCP/IP network, audio packet transmission using AAL2 or another adaptation layer over an ATM network, and transmission of packets over an internal connection. For both point-to-point and multipoint connections the endpoints can be in separate gateways or in the same gateway.

The control primitives for MGCP operations are *Signals* sent from the MGC to the MG, and *Events* sent from the MG to the MGC. The concepts of Signals and Events are used for establishing and tearing down calls. Operations are performed by applying Signals TO, and detecting Events FROM endpoints. An MGC initiates transactions to manage/configure MG endpoints using MGCP commands. MGs send responses to MGC transaction requests using either a notification or restart command. Signals and Events needed to support a specific telephony function or type of endpoint are grouped into Event/Signal *PACKAGES*. Example PACKAGES defined in the MGCP specification include:

- Generic Media Package
- DTMF Package
- Trunk Package
- Line Package
- Handset Package
- RTP Package
- Announcement Server Package
- Others...

MEdia GAteway COntrol Protocol (MEGACO/H.248)

What is MEGACO/H.248?

MEGACO/H.248 is the official industry standard protocol for interfacing between external call agents called Media Gateway Controllers (MGCs) and Media Gateways (MGs). The standard is the result of a unique collaborative effort between the IETF and ITU standards organizations. Derived from MGCP (which, it turn, was derived from the combination of SGCP and IPDC), MEGACO draws heavily from MGCP plus introduces several enhancements. Even though MGCP was deployed first, MEGACO/H.248 is expected to win wide industry acceptance as the official standard for decomposed gateway architectures sanctioned by both the IETF and ITU. MGCP is currently being maintained under the auspices of the PacketCable™ and the Softswitch Consortium™. There are no plans for the MGCP specification to be enhanced by any international standards body. MEGACO offers these key enhancements as compared to MGCP:

- Supports multimedia and multipoint conferencing enhanced services
- Improved syntax for more efficient semantic message processing
- TCP and UDP transport options
- Allows either Text or Binary encoding
- Formalized extension process for enhanced functionality
- Expanded definition of PACKAGES

MEGACO Architecture & Entities

MEGACO has the same architecture as MGCP. MEGACO commands are similar to MGCP commands. However, a main difference between the two implementations is that with MEGACO, commands apply to *Terminations* relative to a *Context*, rather than to individual Connections, as is the case with MGCP. Connections are achieved by placing two or more Terminations into a common Context. It is the concept of a Context that facilitates support of multimedia and conferencing calls. The Context can be viewed as a mixing bridge that supports multiple media streams for enhanced multimedia services.

MEGACO Packages include more detail that MGCP Packages. MEGACO packages define additional *Properties* and *Statistics* along with Event and Signal information that may occur on Terminations. With MEGACO, the primary mechanism for extension is by means of Packages. To accommodate expanded functionality, MEGACO specifies rules for defining new packages.

Below is an "at-a-glance" comparison between MGCP and MEGACO.

MGCP

- **Endpoints, Connections**
- **1 transaction = 1 command**
- **1 type of response**
- **Commands :**
  EndpointConfiguration
  NotificationRequest
  Notifications
  CreateConnection
  ModifyConnection
  DeleteConnection
  Audit Endpoint
  Audit Connection
  Restart in progress
- **Grammar :**
  Text encoded (BNF).

MEGACO

- **Terminations, Context**
- **1 transaction = N actions**
  1 action = N commands
- **2 types of response**
  (new Transaction Pending)
- **Commands :**
  Add
  Modify
  Substract
  Move
  AuditValue
  AuditCapabilities
  Notify
  ServiceChange
- **Grammar :**
  Text encoded (ABNF) +
  Binary form (ASN1).

The Basics...

H.323

What is H.323?

Saving the first for last, H.323 is the most widely deployed protocol suite for real time IP communications. Version 1 of the specification was finalized and approved by the ITU in 1996. Since that time, the specification has been continuously enhanced to address the evolving needs of next generation IP communications networks. Originally conceived as a framework for using packet-based networks for real-time multimedia communications, many of the enhancements incorporated into Versions 2, 3 and 4 address specific voice-only IP telephony requirements. Today, the H.323 umbrella specification is the most comprehensive industry standard for real-time voice and video over packet networks including the Internet and other IP-based networks that do not provide a guaranteed Quality of Service (QoS).

The scope of communications under H.323 includes both point-to-point and multi-point sessions which must include control and audio signals, and may include video and data as well. H.245 together with the H.225 suite of protocols (Q.931, RAS and RTP/RTCP) provide the mechanisms for H.323 call control functions. Additional modules and annexes provide enhanced functionality such as security, supplementary services, native ATM support, fax support, etc. H.323 relies on the H.26x family of protocols for video codecs and the G.7xx family of audio codecs.

H.323 Architecture & Entities

H.323 is based on a peer-to-peer architecture with decentralized intelligence, services and control. H.323 defines four entities: *Terminals*, *Gateways*, *MCUs* (multipoint conferencing units) and *Gatekeepers*.

- Terminals are client endpoints that provide real-time, two-way communications; they may communicate with other terminals, a gateway or an MCU.
- Gateways provide the interface between circuit switch and packet networks; they perform the function of "translator" between different transmission formats, communication procedures as well as various codecs, with the purpose of reducing bandwidth of the audio/video flow if the circuit switch bandwidth is limited.
- MCUs allow three or more endpoints to participate in a multipoint conference; an MCU consists of a mandatory Multipoint Controller (MC) that determines common capabilities of endpoints and an optional Multipoint Processor (MP) for multiplexing media streams.
- Last, but not least, the Gatekeeper is an essential component of an H.323 network; it provides critical network management and call control functions for an H.323 "zone." (In practice, the H.323 standard is implemented in production networks through "zones". A zone is essentially the set of endpoints over which one and only one gatekeeper has jurisdiction.) Gatekeepers perform four mandatory functions: Address Translation, Admissions Control, Bandwidth Control and Zone Management. Gatekeepers can also perform four optional functions: Call Control Signaling, Call Authorization, Bandwidth Management and Call Management.

To accommodate the requirements of large-scale IP telephony networks, the role of the gatekeeper has evolved from its original definition as an optional network component. Today, the gatekeeper serves a the "central intelligence agent" and underpinning technology for implementing IP/PBX functionality, softswitches, and the mechanism for delivering compelling new IP services and applications.

H.323 Call Scenarios and Call Models

H.323 uses the concept of channels to structure the information exchange between communication entities. A channel is a transport-layer connection, which can be either unidirectional or bi-directional.

The following table presents the steps involved in a typical H.323 call scenario with a gatekeeper. For each step the type of channel that is used is shown.

TYPICAL H.323 CALL SCENARIO

Step 1:	Terminal/Gatekeeper binding ***Registration & Discovery***	RAS channel (transported over UDP)
Step 2:	IP and alias binding ***Address Translation***	RAS channel (transported over UDP)
Step 3:	Inject policies ***Call Admission***	RAS channel (transported over UDP)

Step 4:	***Call Setup***	Q.931 channel (transported over TCP or UDP with Annex E)
Step 5:	Call Control ***Capabilities Exchange*** ***Master/Slave Determination*** ***Open Logical channel***	H.245 (transported over TCP or UDP with Annex E)
Step 6:	***Media Transfer***	RTP/RTCP (transported over UDP)

H.323 supports two call models. With the *Direct* call model, the gatekeeper allows all call setup and call control signaling to be done directly between endpoints, with no further invention of the gatekeeper (in the same way that RTP/RTCP channels are opened directly between endpoints). In the *Routed* call model, all call setup and control signaling is routed through the gatekeeper. The Routed call model allows the gatekeeper to monitor the state of and control all calls within its zone. This is the preferred call model for realizing the full potential of the gatekeeper as a centralized intelligence agent for delivering next generation IP-centric enhanced services and providing robust network management.

To further accommodate the needs of production IP telephony networks, H.323 supports provisions to allow for faster call setup using either FAST START procedures or EARLY H.245. These methods allow media streams to be transported almost immediately upon initiation of a call.

As alluded to earlier, H.323 is the most mature and market-tested IP communications protocol. This summer at the ITU meeting, several new enhancements for Version 4 were scheduled for ratification this November. These enhancements will make H.323 an even more powerful and flexible enabling technology for large-scale deployments of next-generation communication networks.

FOR MORE INFORMATION ON SIP, MGCP, MEGACO/H.248 AND H.323 PLEASE VISIT THESE WEBSITES: www.itu.int, www.ietf.org and www.softswitch.org.