IP Telephony Inter-Gateway Protocols

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Introduction

When the IP telephony equipment manufacturers began to move their technologies from the laboratory into the real world, it became clear that the technical challenge of building a scalable network of end devices and gateways was greater than expected. The technologies needed to encode and transmit voice and fax traffic had been perfected, but the art of call control and address management for large corporate or service provider platforms still needed to evolve.

As a result, a number of protocols have been defined that allow IP telephony systems to inter-communicate. This article examines three common IP telephony protocols used in systems to date: H.323, MGCP, and SIP. These protocols lead an intertwined existence often being combined in many applications.

H.323

The International Telecommunications Union (ITU) adopted the oldest of the three inter-gateway protocols, H.323 in 1996, which was followed by an updated Version 2, in January 1998. H.323 is considered an umbrella standard that encompasses a number of subordinate standards. Because of this, the ITU can define the standard by re-using a number of other data and telecommunications standards such as Q.931, G.711, G.723.1, etc., which are already in existence.

Originally proposed by Intel and PictureTel, H.323 defined a flexible means for multi-media teleconferencing equipment to communicate and provide application-sharing features over an IP protocol stack. The designers proposed that the standard could be used on a variety of devices including videophones, desktop PCs, and large multi-port gateways. As a result, the standard is comprehensive and it offers many different media types and compression techniques to be used in various devices. Version 2 of the standard addresses a number of efficiency and scalability issues that the original standard did not cover.. The result is an H.323 standard that is much better suited to real-world deployment. H.323 continues to be enhanced to better address issues such as scalability and support for additional applications such as IP fax.

H.323 Strengths

- The key strength of H.323 is its maturity, which has allowed a number of software vendors to develop robust implementations.
- The standard's maturity also has allowed the various vendors to eliminate interoperability issues, which has permitted a widely compatible range of H.323 capable devices to be introduced into the market.
- Because the H.323 standard includes an adaptation of the Q.931 protocol for call control, many developers with experience in existing ISDN telephony are familiar with the call control model. In fact, the events and parameters often can be directly passed from H.323 into applications that previously operated with ISDN.

H.323 Weaknesses

- The original H.323 Version 1 recommendation suffered from slow call setup because many messages were interchanged between end devices before the voice path was established. The fast call setup features allowed in Version 2 have overcome this problem.
- Because of the complexity of the standard, many products that require basic "quick and dirty" inter-gateway call control find H.323 too complex or expensive.
- When defining H.323, the designers worked from the prospective of an end device, not a device that would reside within the existing PSTN. As a result, H.323 cannot integrate with SS7 or leverage the powerful capabilities that SS7 has to offer.
- H.323's scalability also has proven to be an issue in very large applications. Designers using deployed gateways that included thousands of ports found the centralized state management to be a bottleneck.
- The cost of implementation has been an issue when the end device needs to be very low cost. The complexity of the standard requires reasonable processing capability at the end device, which has prevented implementation on devices such as set-top cable boxes and hand-held wireless devices.

H.323 Sweet Spot

Based on the market reaction to H.323, it looks as if the sweet spot for H.323 is in the 1- to 200-port system, typically located at or near the end points. H.323 works well in environments where sufficient processing capability is available to implement the call control and media processing. H.323 has gained its strongest support as an IP telephony solution for enterprises.

MGCP

Media Gateway Control Protocol (MGCP) is a protocol that provides the means to interconnect a large number of IP telephony gateways, allowing them to work together as one. MGCP assumes that a Call Agent (CA) performs the intelligence of all call control operations and that Media Gateway Controller (MGC) carries out all media processing and conversion.

The MGCP specification was developed by various companies including Telecordia and Lucent and was published as an “informative” request for comment (RFC 2705) by the Internet Engineering Task Force (IETF). It is the result of a merger between the Simple Gateway Control Protocol (SGCP) and Internet Protocol Device Control (IPDC) protocols, but MGCP is NOT a recognized standard. The Megaco working group of the IETF and the ITUA are working together to develop a recommendation based on MGCP under the name H.248 (formerly H.gcp) This core document and its related specifications are targeted for completion in February 2000 and will be published as IETF standards track RFCs.

MGCP can be used with an array of H.323 gateways and SS7 gateways when the H.323 gateway provides the media conversion and the SS7 gateway translates the call control information. In this case, MGCP relays all call control information from the end-point device to the network. Using this mechanism empowers the developer to leverage the capabilities provided in the SS7 network and allows much larger IP telephony systems to be deployed than by using H.323 alone.

To negotiate the media path and capabilities for individual calls, MGCP depends on Session Description Protocol (SDP), which is part of the MGCP specification. SDP allows the negotiation of the RTP port and IP address for the end points, the voice coding method (G.711, G.723.1…), the packetization period, and other connection type parameters.

MGCP Strengths

- Because MGCP was defined to solve a specific problem with very large deployed systems, it is particularly suited to large deployed application.
- Use of MGCP allows for good integration into the SS7 network, which gives greater control and throughput in handling calls.
- MGCP splits the media handling and signaling functions, thus providing a simpler implementation, which can be developed by multiple vendors.

MGCP Weaknesses

- MGCP is too complex for smaller applications.
- MGCP will compete with the H.248/Megaco standards that will be endorsed by the IETF and the ITU in early 2000. Thus, carriers that require media gateway control may elect to use either MGCP or H.248. Therefore, H.248 implementations may ultimately replace earlier MGCP versions.

MGCP Sweet Spot

Clearly the home for MGCP is in the carrier space – delivering thousands of lines of IP telephony.

SIP

The Session Initiated Protocol (SIP) provides a means to communicate call control information from end devices or proxy servers to each other or to gateway devices. This protocol is the result of the MMUSIC working group of the IETF. Therefore, SIP is similar to many of the existing Internet protocols, including the popular HTTP protocol.

SIP is considered a "lightweight" protocol because it uses simple text commands that are easily created and parsed by end devices. SIP uses only six directives to manage the call control information. This simplicity is key to the SIP protocol being selected for very low cost applications.

SIP does not define a media transport mechanism, which allows it to be used in applications where the media transport might be proprietary, thus providing increased efficiency and possible reduced cost.

SIP also allows the call control messages to be carried over any datagram protocol, making it useful for environments that are not TCP/IP-based (Novell or other proprietary protocols).

SIP Strengths

- The expandable nature of the protocol allows future capabilities to be easily defined and quickly implemented.
- It is simple and easy to embed into inexpensive end-user devices.
- The protocol was designed to ensure interoperability and enable different devices to communicate.
- Non-telephony developers find the protocol easier to understand.

SIP Weaknesses

- SIP is very new, so most application are in the prototype stage.
- The protocol has a narrow scope and thus has limited applications by itself; however,

it gains flexibility when used with other protocols.

- SIP is only one small piece of a complete solution. Numerous other software components are required to build a complete IP/Telephony product.

SIP Sweet Spot

The low cost end devices are natural applications for SIP. Devices such as wireless phones, set-top cable boxes, Ethernet phones, and other devices with limited computing and memory resources are suited to this protocol.

Because SIP is a stand-alone call control protocol, it is currently being looked at as the leading candidate to substitute for the call control portions of the previously discussed MGCP.

Summary

Each of these protocols addresses different aspects of the technology needed to develop IP telephony systems. Numerous systems being developed today include one or more of these protocols, often working together. The following table describes the capabilities of the various protocols and how they interact with one another.

Capability	Protocol		
	H.323	**MGCP**	**SIP**
Complexity	High	High	Low
Cost	High	Moderate	Low
Maturity	Good	Poor	Poor
Scope of Definition	Full	Partial	Limited
Interoperability	Good	Some	Some
Similar to ISDN	Yes	No	No
SS7 Compatibility	Poor(?)	Good	Poor

All of these protocols continue to evolve as they are used to build complex IP telephony systems. This evolution means that interoperability will be a continuing challenge as various vendors attempt to build systems that can work together. The advent of the new standard protocol for Media Gateway Control (H.248/Megaco), derived from the IETF and the ITU, is an “x” factor because it is likely to be a strong competitor to MGCP in the carrier marketplace.

Keep your eyes on the industry “bake-offs” and certification labs for news about compatibility.