Session Initiation Protocol – Why It Matters!

Vonage, VoiceWing, Skype, Packet8 – you've been hearing/reading about Voice over IP services (maybe you have one already). Underlying all of these services (and probably part of your voice-enabled home network router) is a very important call control protocol – SIP, or Session Initiation Protocol. SIP has been designed as a simple and efficient way to establish a "telephone" call across an IP network using many of the existing IP standards such as TCP, UDP, IP and DNS. And, in the future will most likely merge with web and Service Oriented Architectures to enable converged voice/data applications that will further blur the traditional "voice/data divide" and enable businesses and individuals to communicate and collaborate in new ways.

SIP emerged from an older foundation - H.323 and its related standards (H.225, Q.931, H.245, T.120, G.7xx, H.25x, H.235 and H.450) that provided a standard way to establish voice or video calls over a packet network. H.323 however, it required the participation of an H.323 Gatekeeper for call control (gateway location, address translation, bandwidth management, feature implementation, registration), had address space limitations (256 characters), required TCP for reliable message transport and a MCU for video conferencing and had a limit of three sessions per conference. Additionally, H.323 required 18 messages to establish a single call, further limiting system scalability. These strictures, along with its highly defined service standard – H.450 – severely limited the ability of H.323-based systems to deliver higher-level, interoperable application services such as collaboration, instant messaging and voice-enabled web applications. And, were one of the main reasons for the slow growth of IP telephony in the past.

The Promise of SIP and Session-based Multimedia Communications

SIP, however, is designed to function in an Internet/web-based environment. It uses standard Internet protocols – TCP, UDP, IP, DNS – for transport and, because it is UDP capable, (unlike H.323) can support multicast sessions. SIP is a "session oriented" protocol. It establishes, modifies and terminates *multimedia* sessions and can be used to invite new members into an existing session. It is also independent of the type of multimedia session it handles and is therefore capable of sessions ranging from simple telephony to video and audio conferencing, shared whiteboards and even gaming.

One of SIP's key strengths lies in its ability to distinguish between session establishment and session description. SIP locates users, but does not interfere with the session once it is established. This design enables SIP to cooperate with new session description protocols to establish an unlimited number and type of sessions. Additionally, SIP is designed so that the core protocol can interoperate with any other implementation and incorporates negotiation methods for the protocols extensions that will be used in each session. That means that companies providing SIP-based voice services are no longer tied to a single vendor's phone or user interface. And, that communication services can be delivered to almost any device, almost anywhere.

SIP Entity	Function
User Agent	Interface to user
Redirect and Proxy Serv-	Provide addresses and alternate addresses for users
Registrars	SIP servers accepting registrations for users and
Location Servers	Store and return locations for users

Because SIP is a session establishment protocol only, it allows end systems to use their intelligence to negotiate services and eliminates the need for SIP Location Servers to store state information and monitor signaling for session durations. This makes SIP a highly scalable way to deliver multimedia applications to a large number of users. And, because SIP uses standard Internet components – HTTP request/response model, URL's to address SIP resources, SMTP-like routing – it functions well with existing web –based systems and architectures.

Finally, SIP's design enables it to combine simple applications into more complex services. And, these simple applications are location and implementation independent. SIP signaling is used only to coordinate the application servers to arrive at the expected, consolidated application result.

Element	SIP	H.323
Transport	TCP, IP, UDP	TCP
Conferencing	IP Multicast	MCU required
Encoding	Text	Binary
Design	Signaling	Signaling and Services
Functions	Internet and PSTN	PSTN
Server Type	Stateful or Stateless	Stateful
Loop Detection	Stateless	Stateful

What does this mean for you? Two things:

1. The "openness" and scalability of SIP will allow service providers, enterprises, and even you and your home-based business to integrate voice, video and data in new ways. The result will be an ever increasing number of new collaborative communications environments and services from which you can choose, use or offer to others.
2. When purchasing new voice, data or video equipment or software for your home business, make sure that it is "SIP-enabled". A few $$'s more now, will save you from another "upgrade" purchase in the near future.

SIP is on its way to changing the way we communicate. Watch for new services and capabilities based on SIP coming to your home and home network soon!

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