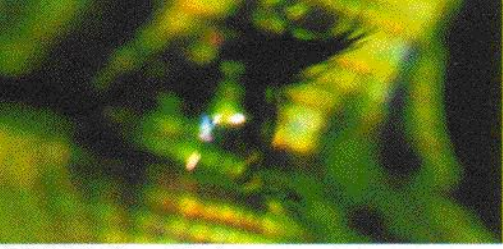

SIP

Session initiation protocol has emerged as the communications convergence technology of choice. **By Rob Buckley**

Convergence of telephony and corporate networks has long been a goal of the technology industry. The 'H.323' array of protocols used in voice-over-IP (VoIP) provides this functionality, but take-up has been slow.

The session initiation protocol (SIP), however, offers similar convergence capabilities and can also unite other communications technology, such as video conferencing and instant messaging. It seems that SIP's time has come.

Third-generation mobile networks will use SIP for call management. Microsoft has incorporated it into the Windows XP operating system and plans to add SIP to the next version of its Exchange email and collaboration server. BT, Cisco and Microsoft are providing a SIP-based service to MSN users in the UK that integrates instant messaging, multimedia and telephony services. Finnish telecommunications company Sonera is running a nationwide VoIP network using both H.323 and SIP. And AOL is using SIP and related protocols to integrate its global instant messaging services with those of other vendors.

With SIP, each user is identified through an address built around elements such as a phone number or host name. When a user wants to call someone, he or she initiates the call with an invite request. The request contains enough information for the recipient to join the session, including the media types and formats that the caller wants to use and a destination for the media data. The request goes to the recipient's SIP server, which determines the recipient's location and routes the request. If the recipient takes the call, his or her SIP software responds to the invitation with details of its capabilities, and establishes a connection using the recipient's preferred method of communication. If the user declines the call, the session can be redirected.

SIP can also 'fork' an incoming call so that several extensions can ring at once, the first extension to answer taking the call. There is also the ability to let callers indicate their various preferences so that someone calling a global company, for example, is immediately put through to someone who speaks their native language.

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SIP v VoIP

There are a number of problems with the established protocol for VoIP, H.323. For example, set-up time can reach up to eight seconds. The delays are even worse on international calls. H.323 does not scale well either.

On the other hand, SIP is able to set up a call within one-tenth of a second. It achieves this by including a client's available features within the invite request, thereby negotiating the features and capabilities of the call within a single transaction. SIP also scales better than H.323 since it only identifies an end user, leaving the actual management of the call to other protocols and applications.

However, two Internet-related issues could limit the success of SIP over the long term.

The first threat comes in the form of network address translators (NATs). They are sometimes used if an enterprise is unable to secure access to a sufficient block of Internet addresses from its Internet service provider (ISP), or if it wants to switch ISPs without having to renumber its network. NATs convey information to and from internal network addresses to Internet sites, re-addressing traffic so it goes to the right destination.

But NATs do not change the information contained in the traffic, only the destinations – and this is where problems for SIP emerge. SIP includes destination data internally, which means addresses do not get altered by the NATs and SIP fails.

A nearly identical problem exists for firewalls. When a user inside the firewall sends traffic to an address outside the firewall, the firewall will drop it unless the administrator sets up a rule to allow it to pass. With SIP, the rule has to change since it works across a whole range of possible addresses and 'ports' (channels).

Nevertheless, on internal networks and the open Internet, SIP has a bright future. As Augustine Kim, the vice president of Sei Yang Network Communications, the Korean equipment manufacturer, recently told a telecoms trade show: "The industry trend is towards SIP." ■

COMPANIES TO WATCH
- Purple Packet, UK: Global SIP consultancy
- Hotsip, Sweden: SIP-based software products
- Net2Phone, US: SIP-based software products

CONTACT: **rbuckley@infoconomy.com**