

## Sip Made Simple

All you need to know and more about the Session Initiation Protocol, sauce' in Unified Communications and Next Generation Networks.

In an ideal world, no-one except the propeller-heads in the basement would need to know anything more about SIP than what the letters stand for: Session Initiation Protocol. But as it is key to most of the significant sea changes taking place in the convergence networking business today and for the next five years or more, even the most non-technical person in the channel has to have some idea of what SIP is and how it works. The customers will be asking. You need to be able to answer. This primer will help.

As always, let's start with a bit of history, courtesy of Ubiquity, the SIP software company recently acquired by Avaya: SIP emerged in the mid-1990s from the research of Henning Schulzrinne, Associate Professor of the Department of Computer Science at Columbia University, and his

research team. Schulzrinne's intent was to define a standard for Multi-party Multimedia Session Control (MMUSIC). In 1996 he submitted a draft to the Internet Engineering Task Force (IETF) that contained the key elements of SIP. In 1999, Shulzrinne removed extraneous components regarding media content in a new submission, and the IETF issued the first SIP specification, RFC 2543. While some vendors expressed concerned that protocols such as H.323 and MGCP could jeopardize their investments in SIP services, the IETF continued its work and issued SIP specification RFC 3261 in 2001. With the standard in place, vendors and service providers began to launch SIP-based products and services. Today, the enthusiasm for the protocol is growing and increasing numbers of players are entering the SIP marketplace with promising new offerings. SIP is on the path to become one of the most significant protocols since the Web's HTTP and e-mail's SMTP.

### But what is SIP?

“The Session Initiation Protocol (SIP) is an Internet Engineering Task Force (IETF) standard protocol for initiating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality.



Henning Schulzrinne: the father of SIP

Like HTTP or SMTP, SIP works in the Application layer of the Open Systems Interconnection (OSI) 7 layer communications model. The Application layer is the level responsible for ensuring that communication is possible. SIP can establish multimedia sessions or Internet telephony calls, and modify or terminate them. The protocol can also invite participants to unicast or multicast sessions that do not necessarily involve the initiator. Because the SIP supports name mapping and redirection services, it makes it possible for users to initiate and receive communications and services from any location, and for networks to identify the users wherever they are.

SIP is a request-response protocol, dealing with requests from clients and responses from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol, such as UDP, SCTP, or TCP. SIP determines the end system to be used for the session, the communication media and media parameters, and the called party's desire to engage in the communication. Once these are assured, SIP establishes call parameters at either end of the communication, and handles call transfer and termination.”

We asked Niall Gallagher, Director of Product Management at Mitel, to put it more simply for us. Gallagher explained that “SIP is the protocol that will set up, manage and tear down communications between any SIP enabled devices such as a SIP phone, a SIP PDA and a SIP application such as Unified Messaging or the QuickConference application we are using right now. Instead of having multiple personal identifiers such as an e-mail address, a cell phone number, an Instant Messaging handle and a business telephone number, the SIP enabled person will have a SIP identifier, known as a Universal Resource Identifier (or URI for short) such as sip:john.smith@anywhere.co.uk.“

He went on to add, “Should you wish to communicate with John Smith you would select him on whichever SIP device you wished to use, then the call will be placed over a mix of your own SIP enabled network, your service provider’s SIP based IP Multimedia Subsystem next generation network and the private SIP network at anywhere.co.uk, which would seek him out using the Presence application - a key element in SIP, which together with Instant Messaging have their own protocol known as SIMPLE, which stands for ‘SIP for Instant Messaging and Presence Leveraging Extensions’ - and establish which of John Smith’s communications devices would be the best to use. That done, the call will go through, using whatever communications media is appropriate: voice to voice, speech to text or vice-versa, video to video or any mix you want.”

So SIP really is the ‘secret sauce’ that makes Unified Communications and Next Generation Networks come together – and finally breaks the stranglehold of proprietary call control protocols that have been the norm in the PABX market since the mid 1970’s. This will stimulate competition, drive down prices for handsets and stoke the fires of application and services innovation, which is no bad thing for an industry in transition, as ours is.



Source: Nortel

### Okay, I have a handle on SIP, but what is SIP trunking?

SIP trunking is the use of the basic SIP protocol as defined in RFC 3261 to link IP telephone switches to one another. It is also used to link IP PABX's to applications such as Unified Messaging and audio Conferencing. One of the most interesting uses of SIP trunking is its potential to offer over a Broadband network what the PSTN offered for analogue phones, but at much higher speeds and with the possibility of far more services. This in turn will enable the 'federation' of disparate organisations; for example, a manufacturer will be able to communicate using many different media and applications with its suppliers, distributors and customers. To use your communication with John Smith as an example, he may work for one of your suppliers in a different country but you will still be able to avail of all the different communications media - voice, text, video – just as easily as if he were a colleague seated two floors below.

To get a service provider's point of view on SIP trunking, we spoke with Paul Bryce, Business Development Director at Node4 Ltd, a SIP Trunking service provider. He offered us this insight: "SIP, from my perspective, is the missing piece of the communications jigsaw. We have had IP-PBX, Hosted IP Telephony, and VoIP for some time now. These technologies have many great benefits but they all need to be communicating with the same type or make of device, such as our own Cisco Unified CallManager service that uses Cisco IP phones. Now because SIP allows any device to talk to another, this is going to have an impact on the communications market from a variety of perspectives." Bryce also had these words of wisdom for the channel community: "Those Data or Voice Resellers not yet selling Voice Termination, such as

ISDN and CPS, should be aware that SIP now opens up a whole new market for them. In effect, overnight the reseller can be opening up new revenue streams by providing customers with voice breakout. For many of these resellers this will be a value add to their existing customer base.”

“But”, he warned; “Those resellers already selling ISDN & CPS must have an offering in the SIP area because end users are going to be spoiled for choice as they now have different options for call breakout. They will have more choice on their supplier, more flexibility on where calls are routed, improved efficiency of communications, and lower costs. To borrow the old Lottery catch phrase ‘you’ve got to be in it to win it’”.

### **This brings us to the burning question about ISDN. Is it going to die?**

The short answer is yes, but not immediately. It is true that BT has already notified consumers of the demise of Home Highway and has told business users that pre- ETSI ISDN – i.e. Business Highway, ISDN 2 and ISDN 30 - will be discontinued this summer.

Affected customers can of course switch to ISDN 2e and 30e services which will continue to be available for many years to come but there will be a cost associated with the move. And BT would much prefer it if customers switch to their Total Broadband service as this will make it much easier for the carrier to up sell to the SIP based 21CN as it rolls out nationwide.

There are ISDN users which will find it hard to switch: utility companies that require a deterministic networking service for telemetry are worried that if they have to switch to a nondeterministic network based on IP, they may lose control over vital systems such as power grids and gas lines. Broadcasters too are concerned: they use ISDN to send news reports to the studio for editing. There is a concern that the voice quality simply won’t be the same over IP. Service providers such as Thus dispute this, pointing out that they have been carrying radio broadcast over their IP ‘next generation network’ for customers such as GCap Media, whose radio station portfolio includes well known stations such as Classic FM and Capital FM, for many years now, without any quality problems.

But it would seem that the writing is on the wall for ISDN and also its older brother, PSTN. As and when BT and other carriers can offer a universal, nationwide high-speed multimedia broadband network, why would they continue to support multiple old-world networks? The message for the channel is clear: if you have customers using ISDN services call on them today to map out a migration plan. If you don’t you can be sure your competitors will.

### **So does this mean the service providers are going to take over the convergence market, leaving slim pickings for the channel?**

Not at all. For the systems integrators and VAR’s that have grasped the IP convergence nettle with both hands, there are margin-rich product sales and recurring revenue service provision contracts to be signed up today. The manufacturers are lining up with hardware and software designed for SIP based networks – and they are proving interoperability with SIP trunking service providers, to boot.

Of course there are many examples of SIP compliant IP PABX's and SIP based applications; far too many to mention them all here. It is hard to choose any one in particular but Avaya's recent announcement caught our eye because it encapsulates all elements of a SIP offering: hardware, software and services.

The Guildford based manufacturer has announced that VoiceFlex SIP trunking services have been rated compliant with Avaya one-X Quick Edition, the company's peer-to-peer SIP (P2PSIP) phone system for locations with 20 users or less. The company says that as a result small businesses will now be able to reduce their telephone costs, providing a competitive alternative to ISDN lines.

The combination of one-X Quick Edition with VoiceFlex's SIP service is fully scalable allowing Avaya's Business Partners to provide real business benefits at a cost-efficient price point; SIP trunks and telephone numbers will be provided in minutes via the VoiceFlex on line portal and will be charged as a monthly 'per voice channel' rental fee.

"SIP services signal an alternative to hosted telephony, especially when combined with the Avaya Quick Edition offering. It is a simple case of 'hosted telephony or hosted cash' and the reseller community will quickly see the market and margin potential being offered here. This is not a dramatic change of sales model but simply a migration from traditional lines and minutes to the 21st Century alternative." said Jim Robertson, commercial director, MTV, an Avaya distributor for the UK market.

So there you have it. SIP is a technological advance that is going to open up new opportunities for the convergence networking business. There is margin in SIP for those who know what they are talking about and aren't afraid to open their mouths. All you need to do is get out there and do it!

### **SIP: A short glossary.**

**Gateway:** A device that translates SIP into some other realtime communication protocol. Usually, this is a SIP-to-ISUP gateway used to connect SIP networks to the PSTN.

**IETF (Internet Engineering Task Force):** The open-standards body tasked with developing and maintaining Internet protocols.

**IMS (Internet-protocol multimedia subsystem):** A component of the 3GPP's architecture for providing the next version of GSM wireless phone services. The IMS brings traditional and emerging Internet services to the cellular world.

**P2PSIP (Peer-to-peer SIP):** SIP systems with little or no "in the network" infrastructure. The rendezvous function is provided by the endpoints through technologies such as distributed hash tables rather than using a centralized proxy/registrar. The IETF is considering forming a working group to standardize peer-to-peer SIP.

**RTP (Realtime Transport Protocol):** The protocol used to carry media between endpoints. RTP packets contain a time stamp, some sequence data, and a chunk of encoded media. RTP receivers use the information to reconstruct the media stream, accurately accounting for packet delay, jitter, and loss.

SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions): An IETF working group creating extensions to SIP and companion protocols to facilitate building instant messaging and presence systems. SIMPLE defines the SIP Events “presence” event package, the Rich Presence Information Data (RPID) format, the XML Configuration Access Protocol (XCAP) for establishing user lists and permissions, and the Message Session Relay Protocol (MSRP).

SIP (Session Initiation Protocol): An IETF protocol used to establish realtime communications sessions. SIP allows endpoints wishing to communicate to find each other (rendezvous) and negotiate how they want to exchange media.

SIP URI: A uniform resource identifier with the scheme “sip:”. An example is sip:RjS@example.net. SIP URIs identify a particular resource at a domain (example.net). SIP systems use the domain component along with Domain Name Server (DNS) to determine where to send SIP messages.

Transport protocol: The underlying Internet protocol used to carry SIP or RTP messages. Defined transport protocols for SIP include UDP, TCP, TLS, and SCTP. RTP is carried over UDP. Work is under way for both protocols to define transporting them over DTLS.