Everything You Wanted To Know About SIP

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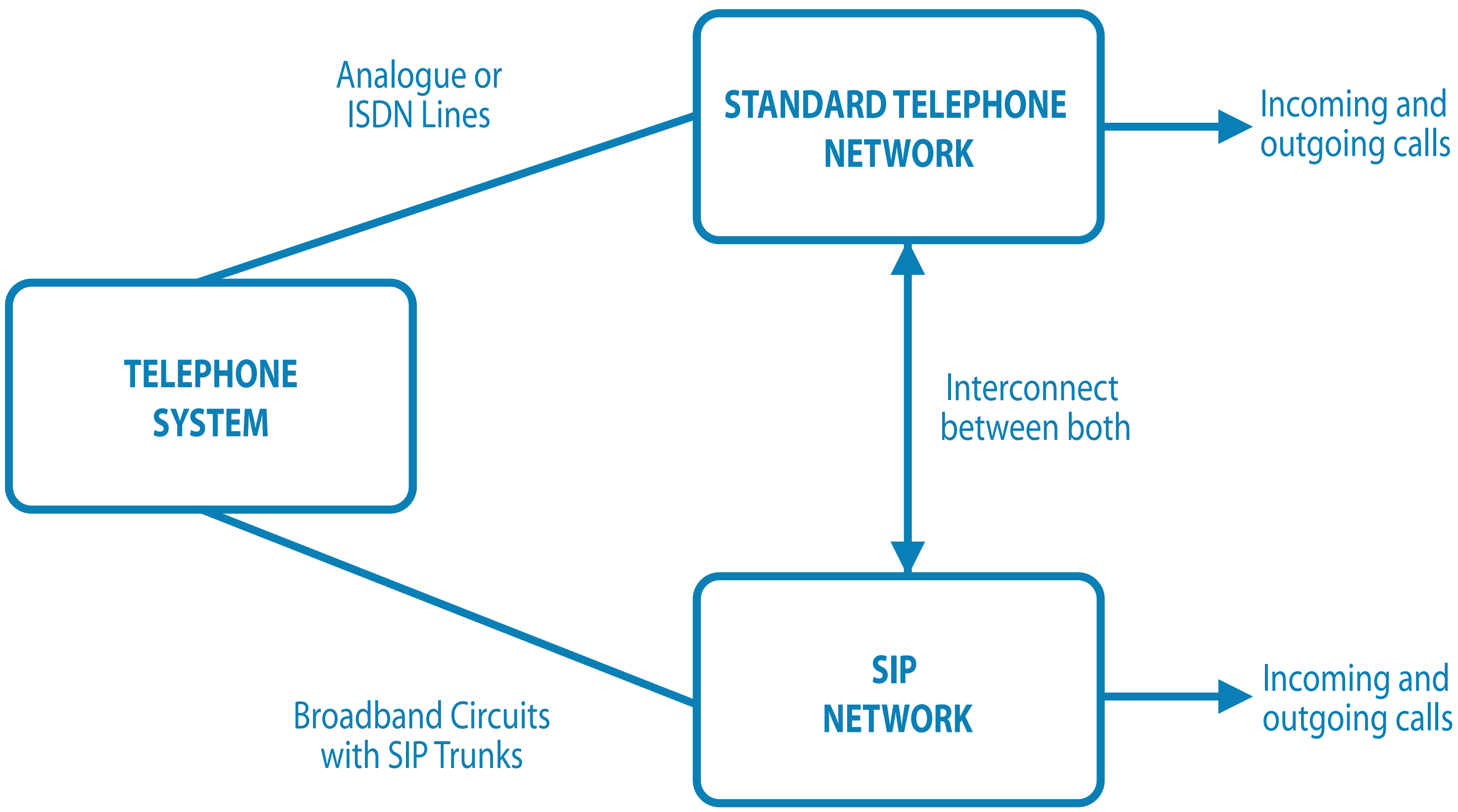
WHAT IS SIP?

SIP STANDS FOR SESSION INITIATION PROTOCOL

Session Initiation Protocol is an open, multi-media signalling standard. Its openness allows enterprises and carriers to interoperate over IP connections. The logical channel established between them is termed a SIP Trunk.

SIP Trunk lines enable businesses to create a single, pure IP connection between enterprises and telephone carriers and make it possible for businesses to make and receive calls over broadband circuits.

This diagram explains how the telephone system is connected:-



KEY BUSINESS BENEFITS

• You can port your existing BT numbers when moving from one exchange to another. (I.e. relocation of offices). This can provide business the flexibility to port multiple BT DDI ranges and numbers, along with new numbers, all of which are owned by the customer. **This is an important factor to enable businesses to retain its existing customers. It saves timely and costly address/phone number notifications.**

• Companies can have virtual incoming and outgoing 'presence' virtually anywhere across the UK or even internationally. This means that a company can have telephone numbers for incoming and outgoing calls on local exchanges without having a physical presence there.
Companies using SIP trunks can appear to have 'virtual offices' by using local telephone numbers.

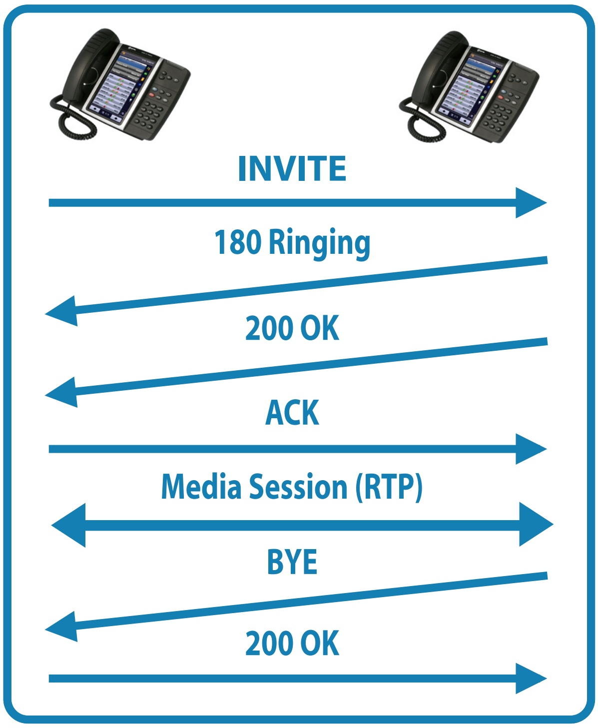
• Companies can reduce communication costs.
Calls made over SIP trunks are generally much cheaper than traditional telephone services. Also, calls between two SIP users are free of charge.

• A massive amount of press coverage on the benefits of VoIP from services such as Skype, which are all proprietary and targeted at the consumer market, establishes a proven technology.
Means that this is no longer a lagging technology and creating awareness in the business community.

• SIP Trunks represent a business class VoIP service, using the SIP protocol as an open signalling standard to link customer premises equipment to the SIP provider.
SIP is now a trusted and mature protocol, offering business level reliability. Giving the customer confidence of an acknowledged and accepted standard.

BASIC SIP CALL EXAMPLE

Basic SIP Call Example



INVITE message in detail
INVITE: sip.bob@broadway.com/SIP/2.0
Via: SIP / 2.0 / UDP
Audiocodes.com:5060;branch=z9hG4bK74bf9
From: Alice <sip:alice@audiocodes.com>
tag=1c289323
To: Bob <sip:bob@audiocodes.com>
Call-ID: abfd43978343@audiocodes.com
Max-Forwards : 70
Cseq: 1 INVITE
Contact: Alice <sip:alice@audiocodes.com>
Content-type: application/sdp
Content-length: 142
SDP Content.
List Alice's supported coders, IP address & Port for voice media
V=0
O=Alice 7439443843 7439443843 IN IP4 audiocodes.com
S= -
C=IN IP4 198.64.138.249
T=0 0
M=audio 10000 RTP / AVP 0
A= rtpmap: 0 PCMU / 8000

METHODS & RESPONSE CODE

SIP Methods

REGISTER: Registers a user with a Proxy/Registrar
INVITE: Session setup request or media negotiation. Used also to hold and retrieve calls
CANCEL: Used to cancel an INVITE transaction
ACK: Acknowledgement for an INVITE transaction completion
BYE: Terminating a session
OPTIONS: Used as a query for remote's status and capabilities
INFO: Mid-call signalling information exchange.
SUBSCRIBE: Request notification of call events
NOTIFY: Event notification after an explicit/implicit subscription
REFER: Call transfer request

SIP Response Codes:

100: Trying – Request has been received by a proxy/gateway
180: Ringing – The called party received the INVITE request, the phone is ringing
181: Call is being forwarded
182: Queued – Invite has been received and will be processed in a queue.
183: Session Progress – Used to convey report of incoming early media
200: OK – Successful transaction completion
302: Moved temporarily – Forward call to given contact
305: Use Proxy – Repeat same call step using a given proxy
400: Bad request: General error
401: Unauthorized – The server requires client authentication.
404: Not found – The user does not exist at the specified domain
408: Request timeout
486: Busy here
5xx: Server Failure
6xx: Global failure

SIP FIELDS

Field	Meaning
INVITE header	Inviting user at SIP address bob@broadway.com to a media session.
Via	The response to the INVITE message should be returned to the specified address, using the specified protocol (UDP). The SIP protocol is carried over UDP, TCP, STCP & TLS. The default SIP ports are 5060 for UDP, TCP, SCTP and 5061 for TLS.
From	The calling party. The caller can choose to remain anonymous by filling the address field with 'anonymous'. A unique tag is created by the initiating party to help identify the addressee in future messages.
To	The Called party. In later responses, the called party should add its own unique tag, similar to the one represented with the From party.
Call-ID	A unique field, used to identify the call.
Max-Forwards	The Max-Forwards value is an integer in the range 0-255 indicating the remaining number of times this request message is allowed to be forwarded.
Cseq	Command sequence header. The field value advances with each new message. Reponses carry the same Cseq as their corresponding requests.
Contact	The SIP address of the calling party. In short, how the called party should reach the calling party in future messages.
Content-type	SIP messages cay bodies that are transparent to the SIP protocol. The content-Type distinguishes one body type form another. In this example SDP is being carried. The media session is being negotiated using the SDP.
Content-Length	The length of the body (in bytes). Explicitly announced for technical reasons..
SDP	Session Description Protocol Used to announce the order capabilities, media IP address & ports.

SIP PRODUCTS

TP-260/SIP
PCI form-factor digital gateway, available in 1, 2, 4 and 8 digital T1/E1 spans.

IPM-260/SIP
PCI form-factor media server platform, available in 30-240 media recourses, 8 E1/T1 spans.

Mediant 2000
1 to 16 span digital media gateway for enterprise & carrier applications.

Ipmedia 2000
Media server platform providing conferencing, transcoding and tone detection recourses.

MediaPack Series
Analog media gateways with 2-4 port (FXS) or 4-8 port (FXO) connectivity.

AC494 Voice over Packet SoC
System on a chip family for IP phone and CPE developers

SIP TERMINOLOGY

SIP: Session Initiation Protocol (RFC 3261) – application layer control (signalling) protocol for creating, modifying and terminating sessions with one or more participants.

SIP Methods: SIP protocol commands or messages (e.g. INVITE, BYE)

SIP Response Codes: Responses to SIP methods indicating success, failure or other information. (e.g. 200 – Ok)

SIP U**ser Agent (UA):** An endpoint device that can issue or respond to SIP protocol methods.

SIP User Agent Client (UAC): A SIP endpoint device issuing the request (e.g. Phone, PC, PDA)

SIP Server (UAS): An application or embedded software that can accept and respond to SIP methods.

SIP Gateway: A network element that can convert SIP methods and response codes to another protocol.

SIP Proxy Server: An intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients.

SDP: Session Description Protocol (RFC 2327) – Text based protocol describing multi-media sessions.

Softswitch: Software application that coordinates VoIP call switching between endpoints, commonly duplicating.

Helpful URLs

www.ietf.org RFC3261 – The current official specification
www.cs.columbia.edu/sip/ - Excellent and well organized reference materials.
www.packetcomm.org - Packet communications Forum

ABOUT ACTIMAX

We are a National supplier of IT and Telecoms Infrastructure. We aim to help improve our clients business processes by implementing change, which improves service levels and efficiency while reducing cost.

How we do it

Our strategy for providing solutions for our customers is split into two components. The first of these is infra-structure and we provide Mitel and Alcatel telephony solutions together with Alcatel Data Switches, WiFi Solutions and Cisco routers, to provide a complete infrastructure for our clients.

This infrastructure is then connected to the network by a variety of methods including, ISDN, SIP Trunks and Wide Area Network connections. We also provide as part of our infrastructure offering, Call Recording and Mobile Telephony.

Actimax are able to provide an end-to-end offering for any customer infrastructure requirements.

The second component of our product solution is providing elements of the solution, which we would regard as value added services. These would include Collaboration and Unified Communications, Video Conferencing and Managed Services such as Least Cost Routing, Maintenance, Fraud Detection and various hosted services to provide the customer with value added services to enable them to manage their business on an ongoing business.

Our History

Actimax have won a large number of awards for technical innovation. In 2009 we were awarded the contract for Salvation Army to deploy a voice infrastructure for their business. This followed successes in 2008 with being awarded the contract to be the supplier for the telephone system for the Wimbledon Championships and five Royal Palaces in 2006, including Hampton Court, Tower of London and Kensington Palace.

We have a large number of high profile customers including The Church of England, The Delfont Mackintosh Theatre chain, The National Union of Teachers and The Restaurant Group.

We are Mitel Premier Resellers and direct Alcatel Voice and Data Partners.

We aim to achieve excellence in the solutions we provide and also in our technical accreditations and customer support. We believe this is paramount in looking after our customers.

We have very high staff retention, with around two thirds of our staff having been with the business for more than 5 of our 12 years.