



## **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.

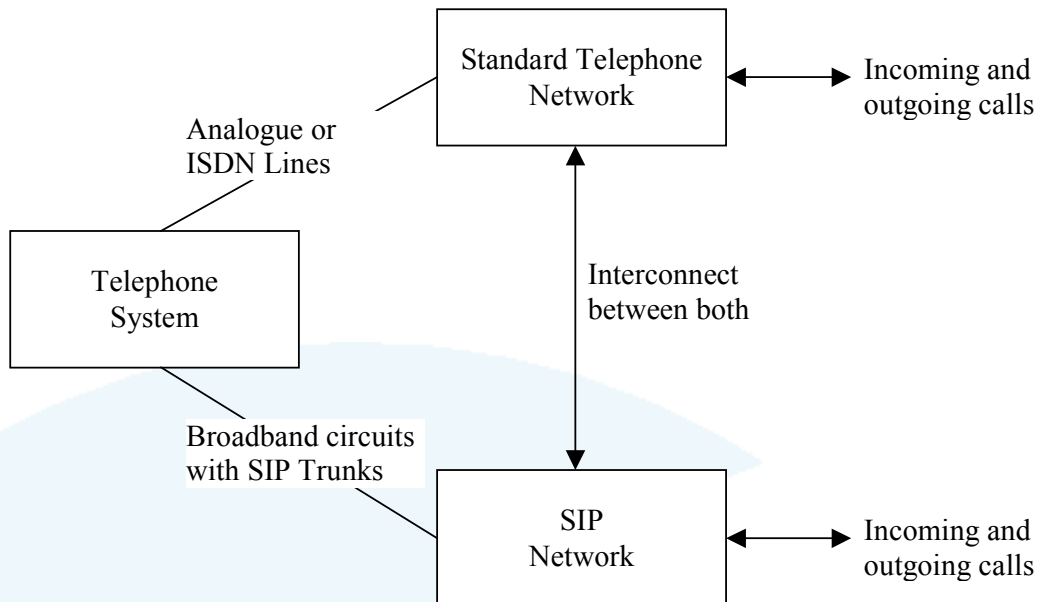
SIP Trunk lines enable businesses to create a single, pure IP connection between enterprises and telephone carriers and make it possible for businesses to make and receive calls over broadband circuits.

## 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.**

# NETWORK DIAGRAM

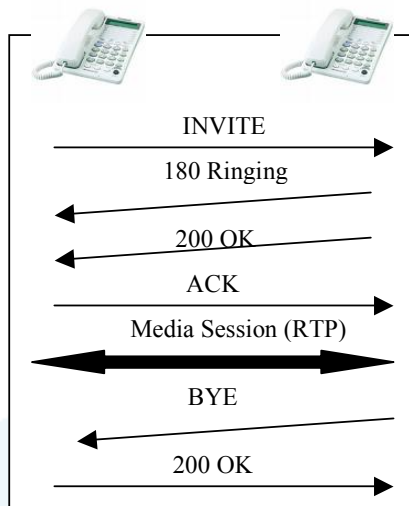
**This diagram explains how the telephone system is connected:-**



**Session Initiation Protocol** is an open, multi-media standard. Its openness allows industry from carriers to premises equipment suppliers to interoperate over IP networks. The connection established between them is termed a SIP Trunk.

# BASIC SIP CALL EXAMPLE

## Basic SIP Call Example



### **INVITE message in detail**

INVITE: [sip.bob@broadway.com/SIP/2.0](mailto: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](mailto: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 CODES

## 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
<b>INVITE header</b>	Inviting user at SIP address bob@broadway.com to a media session.
<b>Via</b>	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, SCTP & TLS. The default SIP ports are 5060 for UDP, TCP, SCTP and 5061 for TLS.
<b>From</b>	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.
<b>To</b>	The Called party. In later responses, the called party should add its own unique tag, similar to the one represented with the From party.
<b>Call-ID</b>	A unique field, used to identify the call.
<b>Max-Forwards</b>	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.
<b>Cseq</b>	Command sequence header. The field value advances with each new message. Responses carry the same Cseq as their corresponding requests.
<b>Contact</b>	The SIP address of the calling party. In short, how the called party should reach the calling party in future messages.
<b>Content-type</b>	SIP messages carry bodies that are transparent to the SIP protocol. The content-Type distinguishes one body type from another. In this example SDP is being carried. The media session is being negotiated using the SDP.
<b>Content-Length</b>	The length of the body (in bytes). Explicitly announced for technical reasons.
<b>SDP</b>	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 User 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](http://www.ietf.org) RFC3261 – The current official specification

[www.cs.columbia.edu/sip/](http://www.cs.columbia.edu/sip/) - Excellent and well organized reference materials.

[www.packetcomm.org](http://www.packetcomm.org) - Packet communications Forum

## ABOUT ACTIMAX

Actimax, established in 1997 has a firm track record of proven results and satisfied customers in the telecommunications market. Our core business is the supply, installation and maintenance of communications systems, and our strength is in the technical background of our team of 38 employees.

With a management team that have worked together for a combined total of over 75 years, we are a hard working company that knows where it is heading. The experience of John Massey Managing Director who's background in the communications industry goes back over 30 years is invaluable in the company being one step ahead of the competition. Actimax invests over 50k per annum on training it's team, including technical engineering courses, management courses, sales and marketing seminars and we also recently offered Learn Direct courses to the whole company, which saw a good up take. The company ethos is to keep our team, happy, motivated and interested, that way they are more likely to stay with us. This has proved to be a benefit with 60% of our team having been with us for over 3 years,

We provide communications hardware from five leading manufacturers along with a whole portfolio of supporting products. We have gained superior accreditation with all of the suppliers we work with, along with awards for Customer Care and Service ( 2004 and 2005 ) , Comms Channel Reseller of the Year 2005, Essex Business of the Year 2005 and Business-to-Business winner 2005.

Our target market are companies, either single or multisites with over 20 employees, and with no upper limit. Our market sector is any company looking to improve their process and procedures along with increase in sales, cutting costs and improving service, and we have been especially successful in Government, Financial, Travel and the Motor Trade.

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