All You Need to Know About the Difference between VoIP & SIP

**Summary:** With its increasing accessibility and cost-effectiveness, Internet telephony has become a useful technology for businesses. It is reliable and offers a host of calling and conferencing features. While both VoIP and SIP are digital communication channels, it is good for users to understand what exactly they are, how they differ, and which suits their requirements better.

Business communication today goes far beyond voice calls. Although phones are still critical to connect with clients, vendors and stakeholders, the workforce in any organisation also uses emails, instant messaging apps and video conferencing tools for more effective exchange of ideas and data. Such needs are increasingly driving businesses to adopt flexible IP-based transport medium and communication protocols.

If you are considering new phone systems and wish to know which between SIP and VoIP is a better technology for your enterprise, here are some points that will help you make a choice:

**SIP and VoIP: Not the same ideas**

Session Initiation Protocol (SIP) is an overlay protocol riding over an IP network which can be used in any telephony application.

Voice over IP (VoIP) is the name given to the technology that establishes voice conversations over an IP medium. Essentially, VoIP uses SIP as a protocol to establish the communication. This communication can be between device-to-device (CUG), app-to-app (CUG) and from app or device to any PSTN phone.

A critical difference between VoIP and SIP is their scope.

VoIP is a broad term that may be used to refer to any voice call made over the Internet instead of traditional phone lines. It needs data connectivity and not a Public Switched Telephone Network (PSTN) connection to transmit voice packets. VoIP & SIP are used interchangeably with terms such as digital telephony, IP telephony, Internet telephony, broadband telephony, voice over broadband and IP communications. All of these describe the fact that VoIP is used to transmit voice signals to another telephone.

SIP is a powerful communication protocol that enables calling on IP-based phone lines. It is just one of the signalling protocols used to enable VoIP. A SIP trunk is digital and can be plugged into an Internet Protocol private branch exchange (IPBX) to start communication operations.

Technically, SIP defines the messages sent between endpoints, and it manages the establishment, termination and other essential constituents of a call. SIP can be used to transmit data between two or multiple endpoints.

In addition to voice calls, SIP also enables instant messaging/online chats, video conferencing, media sharing and other web-based applications.

If you are connecting an IP-PBX to a PSTN provider to route PSTN calls, this connection is termed as SIP Trunk. It then depends on the law of the land, which defines the transport medium of this connection. In India, if you are connecting to a BSO, the transport medium has to be private in the form of MPLS or P2P link. If you are connecting to a VoIP provider, the transport medium can be public Internet and can only be used for outgoing international calls.
Devices used for communication

SIP and VoIP handsets are different, too, although it should be mentioned here that a SIP phone is also a VoIP phone.

To make or receive a voice call, you must connect your VoIP phone to a powered (active) computer with an Internet connection or an IP-PBX. In contrast, SIP phones – wireless and wired and provided by a basic service operator (BSO) or VoIP providers – can operate without a computer registering directly to provider switches and platforms. VoIP provider also uses SIP as a connectivity protocol for connecting to SIP-enabled PBXs, Routers and Edge devices like Session Border Controllers.

Traffic handling procedures

When SIP is provided by a BSO in India, the last-mile delivery is changed to Ethernet towards Customer Premises Equipment. The transport layer has to be as MPLS or P2P connection to form the network layer. The session layer is created by defining SIP parameters on either side. PSTN calls handled over this connectivity have to conform to the PSTN numbering scheme of that country. Calls handled at the BSO switch then ride on upstream private interconnects to other local, NLD and ILD providers.

Whereas for VOIP trunks, the transport layer can be over the Internet connection to form the network layer and SIP to create the session layer. Call standards are decided by the VOIP provider and usually ride on upstream public interconnects to other providers.

Choosing between VoIP and SIP for your business

While VoIP can be availed via any Internet telephony service, SIP is a specific protocol. As per your business priorities, you can choose VoIP in isolation (deployed through any IP technology) or VoIP with SIP.

Subscribing to VoIP in isolation

The benefits that your enterprise gets on choosing a 'VoIP only' service include:

- Monthly cost savings with predictable and flat-rate billing
- Ease of installation and portability of VoIP phones between work stations
- Availability of features such as caller ID, call waiting and call forwarding even with basic VoIP plans
- Flexibility to use the service without long-term service contracts

On the other hand, the limitations of a VoIP only service are:

- Need for sufficient Internet bandwidth to run the service
- Service support on a best-effort basis
- Ensure security as PBX might get exposed to the public Internet

Subscribing to VoIP with SIP

When you choose the SIP technology to deploy VoIP for your business, you get benefits such as:

- Multimedia-ready communication including voice and video
- Efficient failover mechanisms including call preservation
- Flexibility and cost-effectiveness to scale, enabling the organisation to “pay as you grow” as per changing needs
- Web portals for ease of call monitoring and management
- Low costs of maintenance as there is no special hardware/equipment involved and reduced circuits
- Ability to integrate with the other legacy PBX and ISDN systems for investment protection and to build a hybrid phone system

SIP does not have any particular limitations per se, other than the fact that it also requires sufficient data connectivity bandwidth to support call quality. It is advisable to deploy SIP over a private or a dedicated circuit like Internet leased line (VOIP trunks) to get the most of SIP communications and maintain a high security level for them.

What businesses need to understand is that a comparison between SIP for BSO-based PSTN telephony and VoIP for Internet-based telephony is not fair. SIP is an industry-standard methodology to deploy VoIP, and it is preferred as a deployment method due to the ease of scalability it brings for VoIP communication.

When you opt for SIP, your organisation gets a cost-effective technology for voice calls along with other features such as lesser post-dial delay, incoming call control and caller tunes.

At Tata Tele Business Services, we offer SLA-backed SIP trunk solutions with ease of scaling up and down in multiples of ten channels. Our SIP system is a low-maintenance option to make the most of your IP PBXs along with the benefits of business continuity and simplified call management.

To know more about the service, please download the SIP brochure at https://www.tatateleservices.com/downloads/products/resources/brochure/voice-solutions.pdf.