

SIP – SESSION INITIATION PROTOCOL

Few communication protocols have gained so much attention and expectations as SIP, Session Initiation Protocol. When SIP was created in the days of the IT boom around the millennium it was riding on the wave of Internet and promised uncomplicated global communications regardless of media type, device and location. Whether the device is a PC, cellular phone or a game console it may use SIP for establishment of any kind of media flow, “sessions”. A session can be a phone call or video call but can also be a file transfer or a chat.

Although IP-telephony has been around since its early days back in 1995, SIP is the first IP-telephony protocol that is endorsed and eventually backed up by both the telecom industry and the Internet community. The broad industry support and strong initiatives to achieve interoperability has led to an open de-facto standard for IP-based telephony systems.

H.323 vs. SIP

In the very early days of IP-telephony only proprietary endpoint-to-endpoint protocols existed. In 1996, the International Telecommunications Union (ITU) began the work with H.323 that became an umbrella recommendation for voice, video, data and fax communications over IP-based networks. The H.323 recommendation was from the beginning defining not only basic calls but also supplementary services that made H.323 very useful for enterprise-class communication and was quickly adopted as the de-facto standard for IP-PBX's and associated VoIP terminals.

Roughly by the same time, the SIP protocol was designed at University of Columbia and backed up by the Internet Engineering Task force (IETF) as they started a working group to ratify SIP in an RFC. Those were the days of protocol fighting; the powerful telecommunication industry has already adopted H.323 and had in the beginning of the new millennium a significant installed base, but the emerging Internet industry grew strong and endorsed SIP as the future communication protocol.

The first versions of SIP were lacking a lot of functionality that in practice made it useless for traditional telephony. For example it was designed to be completely stateless, which made it impossible for telecommunication providers to detect and charge for a phone call. The Internet community did not see this as a problem since “everything on the Internet should be free of charge”.

Eventually, SIP evolved and due to its simplicity and similarities to other internet communication protocols such as HTTP and e-mail, SIP started to gain attraction to the expense of H.323. When SIP was selected by 3GPP as an element in IMS (IP Multimedia Subsystem), the “battle” was over and most major vendors began to announce future support for SIP.

SIP Today

Most traditional PBX manufacturers have today two client interfaces, one native to support the vendor's “own” phones and one open interface to support 3rd party phones. Virtually all IP-PBX's that have some kind of open client interface are supporting SIP. On the contrary, many of the PBX's native telephones are communicating with some proprietary extended version of H.323 due to legacy reasons.

When using SIP for IP telephony applications, only basic functionalities such as call setup, transfer and call termination are defined by SIP operation. Other functions like call diversion, call-back etc are not part of SIP and thus the implementation of such functions differs from vendor to vendor. Also the wordings in the SIP RFCs are sometimes vague and different interpretations of the standard leads to differences in procedures between PBX vendors.

Since no formal test specification exists, interoperability events are organized where people from different vendors gather and run informal tests to agree on best practice. Also vendor-specific interoperability programs are common to ensure SIP interoperability between different vendors.

SIP was originally designed as a peer-to-peer protocol but today most deployments are done with the use of a SIP proxy that acts like a traditional PBX by relaying or redirecting messages between the calling parties.

SIP Technology overview

SIP is standardized by IETF in RFC3261 and the current ratified version is 2.0.

SIP is a signalling protocol for establishment of communication sessions between endpoints. It is used to initiate voice, video and instant messaging sessions and can also be used to convey location and presence information.

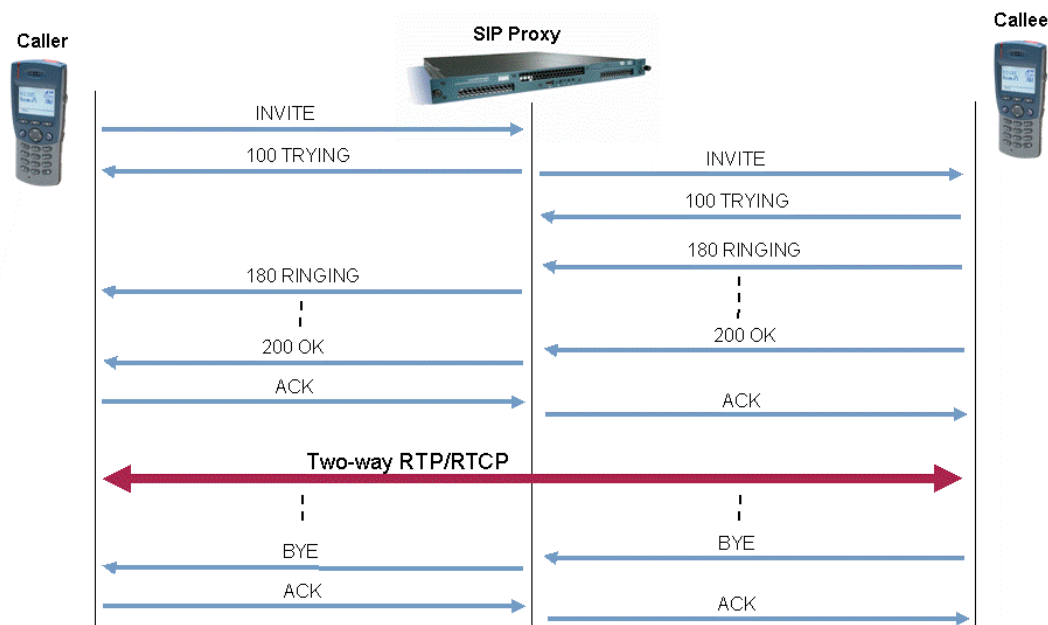
SIP is text-based and can use either UDP or TCP as transport protocol on an IP network.

The SIP signalling is request-response structured and uses the Session Description Protocol (SDP) to describe the media content of the session, e.g. what IP ports to use, the codec being used etc.

In a telephone call, a SIP session is simply a stream of voice packets transmitted in the Real-time Transport Protocol (RTP). Just like in H.323, RTP is the carrier for the actual voice content itself.

The message flow below shows a typical phone call and seems to be quite simple, however procedures such as codec negotiation, RTP address determination and port selection etc are also done within the SIP message exchange. Those procedures are specified by Session Description Protocol (SDP) and SDP messages are embedded in some of the SIP messages.

TYPICAL MESSAGE FLOW IN A SIP VOICE CALL



SIP Development

As discussed above, SIP is here to stay and implementations will converge to a common de facto standard.

There is also another trend in the enterprise PBX business today, open protocols also can mean sources of customer problems and some manufacturers are only allowing a few selected SIP products to be used on their PBX. Those selected SIP products must pass the PBX vendor's interoperability test before they are certified and can be used with support from the PBX supplier. This is of course not liked by larger customers who want to have the freedom of purchasing SIP clients from different sources.

Another market trend is that the traditional PBX is "reduced" to a simple voice switch with only basic call features and the user features are delivered from external business applications. The Microsoft Office Communications Server is an example where applications like e-mail, instant messaging and voice communication are integrated.

Ascom activities

Both our IP DECT system and the i75 are supporting both H.323 and SIP. Internal signalling for IP DECT will remain H.323 for the foreseeable future but more or less all endpoint development effort is concentrated on SIP interoperability.

As described above, SIP has a different development mode than traditional Telco protocols and it's a constant struggle to understand interoperability issues and adjust our products to be "good SIP community citizens".

For a professional mobility supplier like Ascom it's also very important to have a relation with all the different SIP devices that we are claiming interoperability with. New releases of SW have to be verified and changes in functionality tested. We have just recently launched an interoperability process and an official interoperability program is on its way.