CoSIP

IETF Sessions Initiation Protocol (RFC 2543)

Product overview

CoSIP is a text based Session Initiation Protocol, as defined by IETF in RFC 2543, using which a user can establish, modify, and terminate multimedia sessions or calls within an IP network.

CoSIP is an application layer control protocol on a client-server model based on the Hyper Text Transfer Protocol (RFC 2616). SIP can invite parties to both unicast and multicast sessions and it is independent of the type of session being established.

CoSIP product is a modular and complete implementation of the SIP protocol stack for use in a SIP enabled device. CoSIP includes implementation of SIP, SDP, RTP/RTCP and RSVP protocol suites. SDP protocol (RFC 2327) describes the session for the call while RTP and RTCP are necessary for the transmission of real time data. RSVP handles all the resource management and provides for Quality of Service. CoSIP comes with two main components, namely User Agent and Network Server.

CoSIP runs on top of Transmission Control Protocol (TCP) or the User Datagram Protocol (UDP) as provided by any OS. An OS Abstraction layer is also available for easy portability.

CoSIP also comes with a Sample Application to test the various protocols. The application uses socket calls to interact with TCP/UDP stack. The interface between sample application and modules is through APIs. Broadly, the APIs provided for handling the following functions are:

- Call Initiation
- Media Transfer
- Resource reservations
- Call Termination

CoSIP also provides for detailed call statistics and accounting information, and error detection and recovery procedures, enabling a user to get familiar with the stack quickly.

Product features:

- Transmission of messages using TCP.
- Transmission of messages using UDP.
- Automatic re-transmissions based on timeouts for UDP.
- Supports INVITE message.
- Supports OPTIONS message.
- Supports ACK message.
- Supports REGISTER message.
  Supports BYE message.
- Supports CANCEL message.
- Determination of media types and media parameters used for the call.
- End-to-End Encryption of messages using PGP.
- Understands all kinds of response code.
- Determination of the end system to be used for communication.
- User willingness to accept or reject a call.
- Supports all message headers specified in RFC 2543 bis 02.
- Support for querying the DNS.
- Handling ICMP notifications.
- Provision to change existing session parameters.
- SIP Stack can be used to implement state full SIP Proxy servers, SIP Redirect servers and SIP User Agents.
- Facility to do conference calls.
- Authentication of messages using Digital signatures.
- Use of SIP Stack for SIP Registrars.
- Unsupervised call transfer.
- Translate INVITE message to a telephone-signaling message.

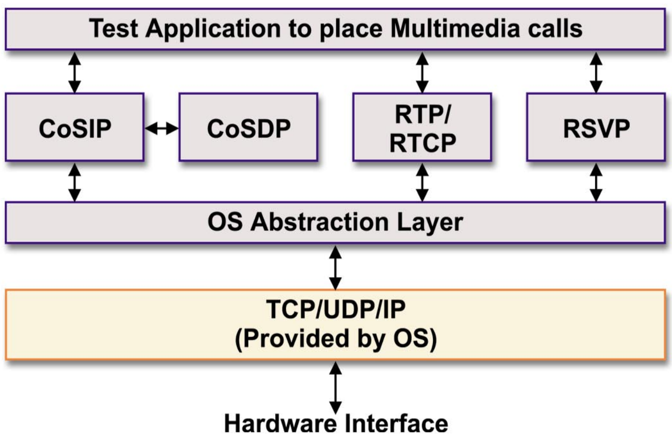

Figure-CoSIP stack architecture

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