VoIP SIP Phone
Session Initiation Protocol

http://www.vocal.com

VOCAL Technologies, Ltd. modem software libraries include a complete range of ETSI / ITU / IEEE compliant modulations, optimized for execution on ANSI C and leading DSP architectures (ADI, ARM, DSP Group, LSI Logic ZSP, MIPS and TI). This software is modular and can be executed as a single task under a variety of operating systems or it can execute standalone with its own kernel.

The Session Initiation Protocol (SIP) is an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences. SIP is text-encoded and highly extensible and may be extended to accommodate features and services such as call control services, mobility, interoperability with existing telephony systems.

SIP phone invitations are used to create sessions carry session descriptions that allow participants to agree on a set of compatible media types. SIP makes use of elements called proxy servers to help route requests to the user's current location, authenticate and authorize users for services, implement provider call-routing policies, and provide features to users. SIP also provides a registration function that allows users to upload their current locations for use by proxy servers. SIP runs on top of several different transport protocols

SIP is being developed by the SIP Working Group, within the Internet Engineering Task Force (IETF). The protocol is published as IETF RFC 3261 (http://www.ietf.org/rfc/rfc3261.txt) (SIP v2) and currently has the status of a proposed standard.

SIP phone entities:

A SIP network is composed of four types of logical entities.

- User Agent (UA): is the endpoint entity. User Agents initiate and terminate sessions by exchanging requests and responses. User Agent can be a UA Client (UAC) a client application that initiates SIP requests or a UA Server (UAS) a server application that contacts the user when a SIP request is received and that returns a response on behalf of the user.
- Proxy Server is an intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients. A Proxy interprets, and, if necessary, rewrites a request message before forwarding it.
- Redirect Server is a server that accepts a SIP request, maps the SIP address of the called party into zero or more new addresses and returns them to the client. Unlike Proxy servers, Redirect Servers do not pass the request on to other servers.
- Registar is a server that accepts Register requests for the purpose of updating a location database with the contact information of the user specified in the request.

SIP phone Messages:

Types of SIP messages: requests (client to the server) and responses (server to client) SIP messages are composed of three parts:

- Start line. The Start Line conveys the message type and the protocol version. The Start Line may be either a Request-line (requests) or a Status-line (responses).
- Header. Used to convey message attributes and to modify message meaning.
- Body. Used to describe the session to be initiated (for example, in a multimedia session this may include audio and video codec types, sampling rates etc.). Alternatively it may be used to contain opaque textual or binary. Possible body types include: Session Description Protocol (SDP) and multipurpose Internet Mail Extensions (MIME).

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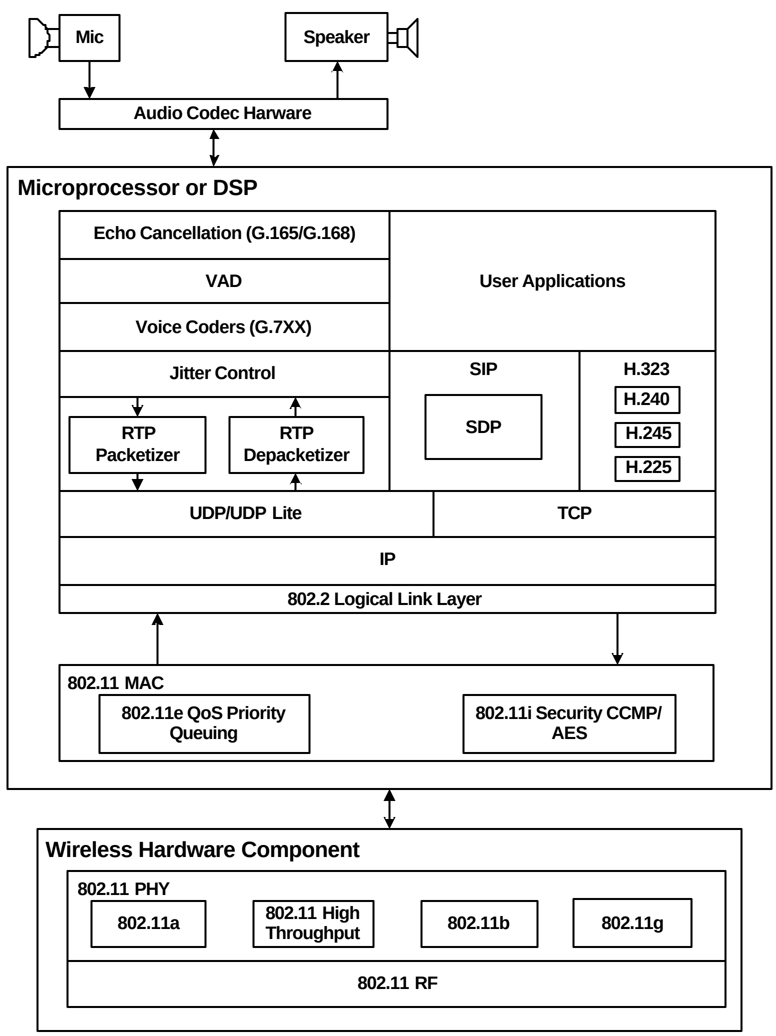
SIP phone Features:

- RFC 3261 "SIP: Session Initiation Protocol" Version 2
- RFC 3262 "Reliability of Provisional Responses in SIP"
- RFC 3263 "Locating SIP Servers"
- RFC 3264 "An Offer Answer Model with Sessions Description Protocol"
- Provides low-level APIs for full control over SIP messages
- Provides high-level APIs to other applications.
- Provides APIs for tracing, logging and error handling.
- Available on Windows and Linux OS.
- Domain Name System (DNS)
- Dynamic Host Configuration (DHCP)
- Internet Control Message Protocol (ICMP)
- Internet Protocol (IP)
- Real-Time Protocol (RTP)
- Transmission Control Protocol (TCP)
- User Datagram Protocol (UDP)
- Supports Simple Traversal of UDP Through Network Address Translators (STUN) RFC 3489

Reference Design:

- TI 5501 DSP
- External Flash
- SDRAM
- Codec
- RF Micro Devices RMD0316 Radio Front End



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