ENEA® SIP-BRICKS

**VoIP and IMS SIP stack for any NGN equipment**

Enea® SIP-Bricks is a versatile and portable software product written in ANSI C that implements the Session Initiation Protocols (SIP).

Enea® SIP-Bricks is compliant with the latest IETF standards and includes numerous protocol extensions that are required for use in Next Generation Network (NGN) equipment such as residential gateways, IP phones, soft-switches, the recently 3GPP specified mobile IP Multimedia System (IMS) and the ETSI TISPAN converged architecture.

Enea SIP-Bricks is based on Enea’s Netbricks architecture using object-oriented design and a message-passing mechanism for inter-entity communication. Enea SIP-Bricks is designed to interface UDP, TCP or SCTP through the standard BSD Socket Interface. If SCTP is not available, Enea can also provide its own widely deployed SCTP layer.

SIP is a description and signaling protocol used to describe, establish, modify and terminate multimedia sessions or calls. SIP uses an IP network to transport the signaling messages through UDP, TCP or SCTP. These mono or multicast sessions include Internet telephone calls, multimedia distribution or multimedia conferences.

Interfaces to many commercial operating systems are provided including Microsoft Windows NT/2000/XP®, Linux (RedHat® and other commercial distribution) Embedded Linux (Monta-Vista® and other embedded distributions), Solaris® (32/64bits), VxWorks®, AMX®, Nucleus®, PSOS++, Enea OSE®, RTC®, VRTX®, and others.

Enea’s Netbricks audit built-in logging mechanism also provides a dynamic tool to trace internal operations of the Enea SIP-Bricks package.

Enea SIP-Bricks can be easily combined with other Enea signaling stacks such as Enea ISDN-Bricks, Enea MGCP-Bricks, Enea H248-Bricks, and Enea SS7-Bricks. In IMS environments, Enea can supply combine signaling protocol packages required for specific IMS or TISPAN functions.

Enea SIP-Bricks is designed for the OEM market. Enea can develop custom product based on Enea SIP-Bricks technology according to customers’ specifications.

Enea SIP-Bricks is a complete framework to develop SIP-enabled equipment such as IP phones, simple or complex UAs, gateways, servers and proxies. Enea SIP-Bricks is available in both UA and Proxy-oriented packages.

Enea SIP-Bricks has passed the “PROTOS—Security Testing of Protocol Implementations” c07-SIP. More information on this test can be found at www.ee.oulu.fi/research/ouspg/protos/testing/c07/sip/index.html.
ENEASIP-BRICKS

- Session description, establishment and termination
- Point to point and multi-point calls
- Support of user location, user capabilities, setup, call handling
- Support of user agents for SIP endpoint applications
- Support of B2BUA, registrar and redirect servers
- SIP transaction ‘stateful’ proxy server package
- Support of 3GPP IMS TS 24.229 and ETSI TISPAN ES 283 003
- IP address resolution of domain name through DNS
- Works on TCP (TLS procedures support), UDP and SCTP (future release) transport layers
- Support of forking, strict and loose routing, record routing, in band DTMF, multipart MIME bodies, rport, private headers and many other key extensions
- Transparent transport of non standard headers and SDP parameters
- Complete finite state machine implementation
- Strictly defined message passing API for efficiency and ease of integration
- Common SDP library
- Compliant with Enea’s Netbricks portable architecture
- High level of scalability through configurable optimization (small footprint, large server…)
- Compatible with Enea’s HANS (High Availability Netbricks System – coming soon)
- Widely field proven inter working capability
- Script based testing framework included
- Powerful audit capability through log and traces

Enea SIP-Bricks supports the following methods:
- INVITE
- ACK
- CANCEL
- BYE
- REGISTER
- OPTIONS
- INFO
- PRACK
- SUBSCRIBE
- NOTIFY
- REFER
- UPDATE
- MESSAGE
- PUBLISH

Enea SIP-Bricks Compliance
Enea SIP-Bricks supports the following set of IETF, 3GPP, ETSI and ITU standards. Enea closely follows progress in various standards bodies in order to ensure up-to-date technology. Please contact us to get the latest supported features that may not be reflected in this list at the time of publication.

- RFC 3261 SIP Session Initiation Protocol
- RFC 3263 SIP Session Initiation Protocol: locating SIP servers
- RFC 3264 An Offer/answer model with SDP
- RFC 4566 SDP Session Description Protocol
- RFC 2806 URLs for telephone calls
- RFC 2833 RTP payload for DTMF digits, telephony tones and telephony signals
- RFC 2976 The SIP INFO method
- RFC 3108 conventions for the use of the SDP for ATM bearer connections
- RFC 3204 mime media types for ISUP and QSIG objects
- RFC 3262 reliability of provisional responses in SIP
- RFC 3265 SIP specific event notification
- RFC 3265 SIP specific event notification
- RFC 3310 HTTP digest authentication using AKA
- RFC 3311 SIP UPDATE method
- RFC 3313 media authorization
- RFC 3323 A privacy mechanism for SIP
- RFC 3325 asserted identity within trusted networks
- RFC 3326 reason header field
- RFC 3327 extension header field for registering non-adjacent contacts (Path)
- RFC 3362 Support of T.38 in SIP (including ITU-T.38 Annex D)
- RFC 3372 SIP-T: SIP for telephones
- RFC 3398 ISUP to SIP mapping
- RFC 3420 SipFrag function
- RFC 3428 SIP extension for instant messaging (MESSAGE method)
- RFC 3455 private header (P-Header) extensions to SIP for the 3GPP project
- RFC 3515 SIP REFER method
- RFC 3556 SDP bandwidth modifiers for RTP Control Protocol (RTCP) bandwidth
- RFC 3581 extension to SIP for symmetric response routing
- RFC 3608 SIP extension header field for service route discovery during registration
- RFC 3842 A message summary and message waiting indication event package for SIP
- RFC 3856 SIP extensions for presence
- RFC 3890 a transport independent bandwidth modifier for SDP
- RFC 3891 SIP replaces header
- RFC 3892 The SIP referred-by mechanism
- RFC 3903 SIP extension for event state publication (PUBLISH method)
- RFC 3959 early session disposition type for SDP
- RFC 4028 session timers in SIP
- RFC 4235 INVITE-initiated dialog event package for SIP
- 3GPP TS 24.229 IP multimedia call control protocol based on SIP and SDP
- ETSI TS 2831003 TISPAN IP multimedia call control protocol based on SIP and SDP
- draft-iejzak-sipping-p-em-auth-02: P-Early-Media header
- IETF draft-levy-sip-diversion-08
- draft-ietf-sipping-kpml-07
SIP-Bricks Software Architecture

- **Common Procedures and Services:** OS abstraction layer and common system procedures
- **System Management Entity:** For configuration and event/error reporting
- **BSD Socket Adapter:** Socket adaptation layer to map standard UDP, TCP or SCTP socket interface
- **SCTP:** Optional Stream Control Transmission Protocol layer
- **SIP Protocol:** SIP protocol finite state machine layer
- **SDP Library:** common SDP parser/builder library
- **SIP Session Controller:** SIP session control layer performing call control operations with user profiling for automatic message completion