



SIP Toolkit

For developing SIP applications

The award-winning SIP Toolkit is a powerful and highly versatile set of tools designed to dramatically reduce development efforts of SIP applications. It includes all the components developers require including a large set of quick start sample applications that demonstrate efficient API usage, a GUI test application and detailed documentation. The SIP Toolkit is an IETF, 3GPP and TISPAN standards compliant, high performance SIP implementation and provides multiple API layers for full user control and flexibility.

The SIP Toolkit is part of RADVISION's standard-compliant SIP Developer Suite, designed to dramatically accelerate development of SIP applications.

Products developed with the SIP Toolkit include:

- Softswitches
- CSCF
- Gateways
- MRFC
- IM-MGW
- Application Servers
- BGCF
- Access Concentrators
- Conference Bridges
- Interactive Voice Response
- SIP-Enabled Firewall/NAT
- SIP Multimedia Servers
- 3G Cellular Phones
- IP Phones
- 3G-SEG
- Connected PDAs
- Video Terminals
- Soft Phones
- Voice Enabled and e-Commerce Solutions
- Voice/Video Messaging IAD
- Session Border Controllers

SIP Basics

The Session Initiation Protocol (SIP) is the industry dominant signaling protocol for real-time communication applications such as voice over IP (VoIP) and Instant Messaging (IM). Based on ubiquitous and accepted Internet protocols such as SMTP and HTTP, SIP is text encoded and well suited for the Internet and other IP environments. SIP provides the mechanisms to implement a broad range of features including call control services, next-generation service creation, interoperability with existing telephony systems, and mobility.

SIP signaling functionality is divided into the following entities:

- User Agents for SIP endpoint functionality
- SIP Proxy for routing SIP messages to their appropriate destinations
- SIP Redirect Servers for re-directing clients to contact an alternate set of URIs
- SIP Registrar for managing user location information
- SIP Back-to-Back User Agent (B2BUA) for routing and connecting calls with stronger control
- SIP Presence Server - Handles presence subscription requests from watchers and notifies them about changes in presence status

Product Specifications

The SIP Toolkit provides all necessary SIP, SDP and RTP services, such as encoding, sending, parsing and receiving SIP messages over UDP, TCP and TLS, managing SIP calls and transactions, and providing reliability. The SIP Toolkit complies with the latest IETF and 3GPP standards. Coded in ANSI C and cross-platform compatible, the SIP Toolkit is available for all common operating systems.

The Toolkit features an open, object-oriented architecture, which makes it programmer-friendly and highly flexible. It provides multiple layers of APIs including:

- High-level APIs that hide the complexity of the protocol and enable rapid development of applications.
- Mid- and low-level APIs that expose the internals of the protocol and allow for more power and customization based on application requirements.

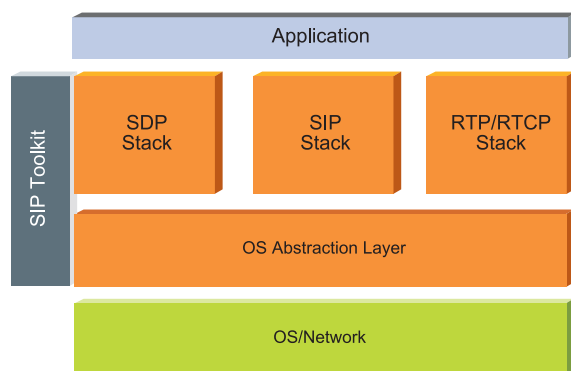
SIP Toolkit Architecture

SIP Stack

The SIP Stack is an internally multi-threaded (configurable) library containing all SIP-specific functionality of the Toolkit including message encoding and decoding, transaction and call management and SIP extensions.

SDP (Session Description Protocol) Stack

The SDP Stack is a library for SDP message processing. The SDP Stack was written in compliance with RFC 2327 and it enables parsing/encoding of any SDP message field.



The SDP Stack also provides an SDP Message Layer for creating, browsing and editing SDP message parts.

RTP/RTCP Stack*

The RTP/RTCP Stack is a library for sending and receiving RTP and RTCP packets.

*RADVISION also offers a standalone Advanced RTP/RTCP (RFC3550/3551 compliant) Toolkit providing IPv6 and other advanced functionality like secured RTP (SRTP as defined in RFC 3711).

SIP Toolkit APIs

The SIP Toolkit is standards based and enhanced with intuitive object oriented APIs to provide optimal control over SIP Stack activities. Highlights of the Toolkit's APIs include:

- SIP Message Layer for creating, browsing, editing and comparing SIP messages and message parts
- Dialog Control Engine for rapid SIP application development

- Subscription Control Engine
- Transaction Control Engine
- Connection Management
- SIP Stack Manager Layer for setting system configuration, memory allocation, logging and other resources.

Enhanced Features

TLS

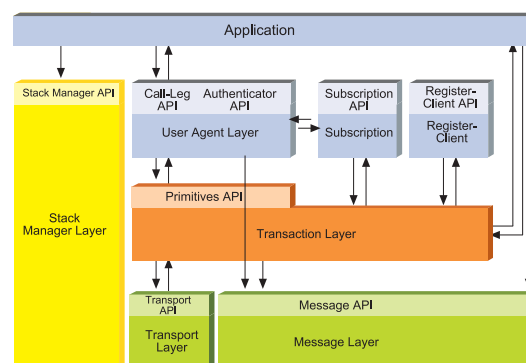
TLS is a security protocol which is typically layered on top of connection-oriented transports such as TCP. TLS allows client/server applications to communicate over TCP in a way that is designed to prevent eavesdropping, tampering, or message forgery. TLS provides a solution for many of the security issues SIP applications face and ties in well with the existing SSL/TLS infrastructure that serves HTTP applications. SIP uses SIP: URL scheme for TLS addresses.

Persistent Connection

In many cases, a single TCP connection may be reused for different messages/transactions/dialogs. Opening and closing TCP connections often is not desirable because of the extra messaging overhead of the TCP handshake (and even more so in TLS connections).

ENUM

When SIP users dial a phone number, the SIP network has to convert it to SIP addresses for routing purpose. This advanced feature is done automatically by SIP Toolkit as defined in RFC 3764.



General URL Scheme Support

SIP defines and uses different URL schemes such as SIP, IM, TEL. In addition, some implementations define proprietary URL schemes. A general framework in the SIP Stack provides support for sending and receiving any type of URL scheme. In Particular, SIP, SIPS, TEL, PRES and IM URIs are implement and all the other URI schemes are supported.

SDP key management extension

Before the initiation of secured media streams (SRTP), cryptographic data has to be exchanged. This is done by key management framework. In particular MIKEY (RFC 3830) protocol is implemented for SRTP support.

REFER (Transfer) Extension Support

REFER is a SIP method defined by RFC 3515. The REFER method indicates that the recipient of the REFER request should contact a third party using the contact information provided in the REFER request. RFC 3515 provides a mechanism allowing the party that is sending the REFER to be notified of the outcome of the referenced request with a NOTIFY request. This implementation uses subscription objects for REFER implementation.

Connection reuse

Connection reuse is a networking feature that results double network efficiency. The working principle is using one network connection (TCP socket) for SIP requests in two directions.

High Availability (HA) Support

The SIP Toolkit provides the necessary building blocks for implementing Highly Available systems that can recover from machine failure without losing call state. The SIP stack enables creation of one or more replicated/ redundant network entities (for standby or parallel operation) and seamless switching between them.

SIP-T (Interworking with ISUP/QSIG) Support

SIP-T is an IETF umbrella specification that utilizes different SIP extensions and advanced capabilities (such as PRACK, 183, INFO, Multi-Part MIME, Server Features) in order to interwork SIP with SS7/ISUP or QSIG networks. The SIP Toolkit provides the entire feature set and simplifies enabling SIP-T functionality in any application rapidly.

Lean footprint

Latest SIP Toolkit release introduces reduced memory footprint achieved several techniques like:
Optimized SIP parsing engine, SDP optimization and other techniques.

PRACK (RFC 3262- Reliable Provisional Responses) Extension Support

PRACK (PRovisional ACKnowledgment) is an IETF SIP extension for sending provisional responses reliably. PRACK is useful opening one way media sessions before call establishment and QoS negotiation before completing the INVITE transaction.

IPv6 Support

The SIP Stack fully supports both IPv4 and IPv6 as underlying protocols and can be used seamlessly with either type of network simultaneously.

Advanced DNS Queries

RFC 3263 (Locating SIP Servers) defines procedures for using advanced SRV and NAPTR DNS queries to determine the transport protocol, IP address and port at which a specific SIP server is available. These procedures can be used to dynamically update server location and to implement redundancy among servers for fault tolerance or load balancing.

SUBSCRIBE-NOTIFY (SIP Events)

RFC 3265 (SIP Specific Event Notification) is a SIP extension that allows for subscription and event notifications using SIP. SIP Events an important infrastructure for services such as Presence and Message Waiting Indication.

Enhanced Parser Functionality

Enables the application to accept or fix bad message syntax. This provides ability to handle corrupted message parts and interwork with proprietary implementations.

Standards Supported (partial list)

- IETF RFC 3261 (SIP: Session Initiation Protocol)
- IETF RFC 3262 (Reliability of Provisional Responses in Session Initiation Protocol (SIP))
- IETF RFC 3263 (Locating SIP Servers)
- IETF RFC 3264 (An Offer/Answer Model with Session Description Protocol (SDP))
- IETF RFC 3265 (SIP Specific Event Notification)
- IETF RFC 3266 (Support for IPv6 in Session Description Protocol (SDP))
- IETF RFC 2327 (SDP-'D0 Session Description Protocol)
- IETF RFC 1889 and 1890 (RTP/RTCP)
- Numerous Internet Drafts for Various SIP Extensions
- Dozens of IETF and 3GPP SIP and SDP extensions

Operating Systems (partial list)

- Windows 2000/2003/XP
- Windows Mobile
- Windows Vista
- Linux Redhat / SUSE
- VxWorks
- Solaris
- Nucleus
- pSOS
- Embedded Linux (Monta Vista)
- 64 bit operating systems

The SIP Toolkit is delivered with:

- Source Code
- Sample Programs
- GUI Test Application (soft phone UA) with full signaling capabilities
- Complete Documentation

The RADVISION SIP Family also includes:

- **SIP Development Suite**

A powerful and highly versatile set of tools, Add-Ons and testing tools that enables developers to combine the necessary components for building an ideal development environment for an application's specific needs.

- **SIP Server Platform**

A comprehensive SIP server development solution with complete standards-based functionality of Proxy, Redirect and Registrar servers, as well as supporting modules like accounting, OMAP and High Availability. SIP server platform is perfect choice for SIP application server developers.

- **ProLab™ Testing Suite**

A versatile VoIP testing solution, based on RADVISION®D5s award-winning SIP Toolkit, that is suitable for use in different stages of the product development cycle.

- **Java SIP Toolkit**

A powerful and highly versatile set of tools to simplify and reduce development time of Java-based SIP applications.

- **Multimedia Terminal Framework**

A complete set of building blocks for developing SIP-based IP phone applications in RTOS or embedded environments.

- **RTP/RTCP Toolkit**

A standalone RTP/RTCP stack providing IPv4/IPv6, security and advanced functionality.

- **Professional Services**

A full range of design, integration and deployment consulting services.

Partial List of SIP Toolkit Features

- Connection reuse
- ENUM
- Service-route/Path headers for mobile registration
- Internally Multi-Threaded
- Multi-Instance
- TLS
- Persistent Connection
- Digest Authentication Support
- MESSAGE Support
- UPDATE Support
- REFER (Transfer) Extension Support
- High Availability (HA) Support
- SIP-T (Interworking with ISUP/QSIG) Support
- Multi-Part MIME Bodies Support
- PRACK (RFC 3262 - Reliable Provisional Responses)
- Extension Support (Manual and Automatic operation)
- 183 response
- Server-Features (Require-Supported) Mechanism
- INFO Extension Support
- Multi-Homed Hosts
- IPv6 Support
- Loose Routing
- Advanced DNS Queries (Locating SIP Servers using SRV and NAPTR for outgoing requests*)
- SUBSCRIBE-NOTIFY (SIP Events)
- Session Timer
- Enhanced Parser
- General URL Scheme Support (e.g.- TEL, IM)
- rPort
- In-Band DTMF
- Merging disabling
- Transmitter object
- IP Address Black/White list enabled
- Dynamic local address setting
- IP TOS setting
- A-Synchronous DNS and dynamic NDS server setting
- Dynamic Via header control
- GRUU (Globally Routable User Agent URIs) Support

About RADVISION

RADVISION (NASDAQ: RVSN) is the industry's leading provider of market-proven products and technologies for unified visual communications over IP and 3G networks. With its complete set of standards-based video networking infrastructure and developer toolkits for voice, video, data and wireless communications, RADVISION is driving the unified communications evolution by combining the power of video, voice, data and wireless - for high definition video conferencing systems, innovative converged mobile services, and highly scalable video-enabled desktop platforms on IP, 3G and emerging next-generation IMS networks. For more information about RADVISION, visit www.radvision.com

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