



SIP Developer Suite

For developing SIP applications

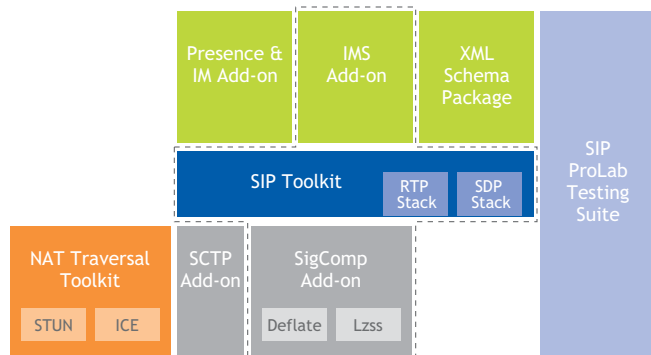
The award-winning SIP Developer Suite is a powerful and highly versatile set of tools designed to dramatically accelerate development of SIP applications. It is comprised of a suite of Toolkits, Add-Ons and testing tools that enables developers to combine the necessary components for building an ideal development environment for an application's specific needs. The SIP Developer Suite complies with IETF and 3GPP standards, is IMS-compliant (3GPP and TISPA), delivers high performance, and provides multiple API layers for full user control and flexibility.

Products developed with the SIP Developer Suite include:

- X-CSCFs
- UA/UE IMS terminals
- MRFC
- IM-MGW
- Application Servers
- BGCF
- Softswitches
- Gateways
- 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/Video Messaging IAD
- Session Border Controllers

The SIP Developer Suite Architecture

The SIP Developer Suite is comprised of building blocks that simplify and dramatically reduce development time of SIP applications. This modular set of extendable, highly versatile development and testing tools allows "mix and match" of components, so that developers can create the exact environment needed for specific applications, while retaining a small footprint and boosting performance. High Level APIs hide IMS and SIP complexity to accelerate development time.



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 is also the core signaling protocol in IMS networks (3GPP and TISPA). 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



The SIP Developer Suite Components

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 ensuring reliability. The SIP Toolkit complies with the latest IETF and 3GPP standards and is comprised of:

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

SDP (Session Description Protocol) Stack, 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.

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

IMS Add-On

The RADVISION IMS SIP Add-On includes IMS SIP extensions to develop IMS compliant SIP applications. The IMS Add-On allows full compliance with IETF and 3GPP and TISPAN IMS standards and offers a rich feature set including:

- Support of IMS P-headers (RFC3325 RFC3455 RFC3313)
- AKA-MD5 and IKE Support
- Security Agreement (RFC 3329)
- IPsec with ESP transport mode, tunnel mode and manual keying
- Support for mobile registration using Service-route and Path headers

SigComp Add-On

The Signaling Compression (SigComp) add-on module compresses SIP signaling. It includes support of LZSS and DEFLATE algorithms which are implemented with Dynamic and Static compression. SIP messages are text-based and therefore not optimized in terms of size. For example, a typical SIP message ranges from a few hundred bytes to over two thousand bytes or more. In order for these protocols to be used as planned in wireless and cellular handsets, and in accordance with 3GPP and TISPAN requirements for IMS, this large message size is problematic and requires message compression.

Together, the SIP Toolkit, IMS Add-On and SigComp Add-On components make up the RADVISION IMS SIP Toolkit, a powerful and highly versatile set of tools to facilitate development of IMS SIP applications while reducing development time and costs.

STUN NAT Traversal Toolkit

RADVISION's IETF standards-based STUN (Simple Traversal Underneath NAT) NAT Traversal Toolkit is a complete NAT traversal solution for developers, comprising the latest NAT-related solutions available for SIP, RTSP, MEGACO/H.248, DIAMETER and others. The RADVISION NAT Traversal Toolkit complies with IETF Standard RFC 3489 and the latest draft-ietf-behave-rfc3489bis-05. This enables support for both existing STUN servers as well as those being developed, and serves as a foundation for ICE (Interactive Connectivity Establishment), which is based on the updated STUN standard.

Presence & Instant Messaging Add-On - SIMPLE

The SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) add-on focuses on applying the Session Initiation Protocol (SIP, RFC 3261) to the suite of services collectively known as Instant Messaging and Presence (IMP). Since the services SIP is used for share much in common with IMP, and it is such a mature and widely deployed protocol, the adaptation of SIP for IMP is a natural choice for IMP. RADVISION's SIMPLE add-on implements logic/information layer for PUA (Presence User Agent).

XML Schema Package

The XML Schema package includes a wide range of service specific XML schemas, such as support for XCAP and PIDF.

SCTP Add-On

SCTP (Stream Control Transmission Protocol) is an improved transport layer used for SIP signaling. SCTP provides advanced features, such as multi-homing, multi-stream control, improved reliability and security. SCTP is based on socket interface programming, a convenient alternative for network programmers.

ProLab™ SIP Test Solution

The RADVISION IMS ready ProLab SIP Test Solution is a powerful and highly versatile set of tools to facilitate testing and pre-deployment of SIP applications while reducing time and costs. The ProLab IMS SIP Test Solution is fully compliant with IETF and 3GPP standards. The entire ProLab Test Management Suite performs advanced signaling tests, such as evaluating the number of calls handled, media tests, such as checking for packet loss, delay, and jitter, and voice and video quality. Additionally, the ProLab Suite enables simulating different network topologies.

*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).

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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