SIP Developer Suite
for developing SIP applications

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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 and Add-Ons that enable 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, TISPAN and PacketCable 2.0), 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
- Softphones
- Voice/Video Messaging IAD
- Session Border Controllers

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.

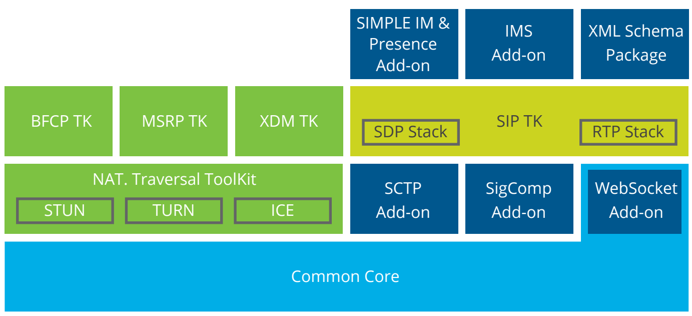
SIP Basics

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, TISPAN and PacketCable 2.0).

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 redirecting clients to 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 and Events Server for handling presence subscription requests from watchers and notifying them about changes in presence status



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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 Stack, a library for Session Description Protocol message processing. The SDP Stack was written in compliance with RFC 2327 and 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 IMS SIP Add-On includes IMS SIP extensions for developing IMS-compliant SIP applications. The IMS Add-On enables full compliancy with IETF, 3GPP IMS, TISPAN and PacketCable 2.0 standards and specifications, and offers a rich feature set that includes:

- Support of all IMS P-headers (RFC 3325, RFC 3455, RFC 3313)
- Support of AKA-MD5 and IKE
- Security Agreement (RFC 3329)
- IPsec with ESP transport mode, tunnel mode and manual keying
- Support of mobile registration using Service-route and Path headers

SIP Developer Suite Components

SigComp Add-On

The SigComp Add-on compresses SIP signaling. It includes support of LZSS and DEFLATE algorithms that 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 2,000 bytes or more. For these protocols to be used in wireless and cellular handsets, in accordance with 3GPP and TISPAN requirements for IMS, this large message size is problematic and message compression is required.

ICE, STUN & TURN NAT Traversal Toolkits

Softil's NAT Traversal Toolkits deliver a complete NAT traversal solution for developers, comprising the latest NAT-related solutions. The Softil NAT Traversal Toolkits comply with STUN, TURN and the latest ICE NAT Traversal mechanism.

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

Instant Messaging and Presence Add-On—SIMPLE

The SIMPLE-Client 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 for which SIP is used share much in common with IMP, and because SIP is such a mature and widely deployed protocol, the adaptation of SIP for IMP is a natural choice for IMP. Softil's SIMPLE-Client Add-on implements the logic/information layer for the Presence User Agent (PUA).

XDM Toolkit

The XDM, XML Document Management Toolkit provides a standard method that makes user-specific, service-related information accessible to service enablers that require it - for example, Push-to-talk over Cellular (PoC), Mission Critical Communication over LTE and Instant Messaging. The XDM Stack is designed in compliance with OMA XML Document Management V2.0 specifications for developing XDM Clients.

MSRP Toolkit

The MSRP Toolkit is a software development tool designed for building MSRP-enabled devices. The Message Session Relay Protocol enables point-to-point messaging and file transfer, and handles messages as media. It is part of the OMA SIMPLE standard.
The MSRP Stack enables development of OMA and RCS complaint Instant Messaging applications.

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

The Stream Control Transmission Protocol (SCTP) 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.

WebSocket Transport

WebSocket is a popular IP transport protocol in Web systems. Using SIP over WebSocket allows communicating with WebRTC clients that are running in the browser. WebSocket is implemented in the WebRTC Transport Extensions Add-on Module for Common Core.

*Softil also offers a standalone Advanced RTP/RTCP Toolkit (RFC 3550/3551 compliant) that provides IPv6 and other advanced functionality, such as secured RTP (SRTP as defined in RFC 3711).AP and PIDF.

For more information, contact Softil at info@softil.com

Rev. A | 07/17

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