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Avaya Aura™ SIP Enablement Services

STANDARDS AND REGULATIONS

- SIP Forum's SIPconnect Interface Specification
- Internet Engineering Task Force (IETF)
- Internet Engineering Task Force (IETF)

SIP Forum's SIPconnect Interface Specification

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Communication Manager 3.0 and later and SIP Enablement Services 3.0 are certified as SIPconnect Compliant by the SIP Forum.

SIPconnect is an industry standards-based approach to direct IP peering between SIP-enabled IP PBXs and VoIP service provider networks. The SIP Forum began working on SIPconnect subsequent to the initial release of the SIPconnect Technical Recommendation in February of 2005 based on a proposal submitted by members of the SIPconnect initiative. Avaya is a founding member of the SIPconnect initiative.

Recently introduced by the SIP Forum, the SIPconnect Compliant program was created to help validate and ensure multimedia communication interoperability among IP communications equipment manufacturers, software providers and service providers, as well as to promote further adoption of Session Initiation Protocol (SIP) as the standard for IP-based communications.

Visit the SIPconnect technical working group at www.sipforum.org for further details or to download the technical recommendation.
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Internet Engineering Task Force (IETF)

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SIP Enablement Services supports the following SIP-related IETF RFCs and Internet Drafts.

- 2246 The TLS Protocol
- 2327 SDP:Session Description Protocol
- 2396 Uniform Resource Identifiers (URI): Generic Syntax
- 2617 HTTP Authentication: Basic and Digest Access Authentication
- 2737 Entity MIB (Version 2)
- 2782 DNS RR for specifying the location of services (DNS SRV)
- 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- 2863 The Interfaces Group MIB
- 2976 The SIP INFO Method
- 3066 Tags for the Identification of Languages
- 3261 SIP: Session Initiation Protocol
- 3262 Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
- 3263 Session Initiation Protocol (SIP): Locating SIP Servers
- 3264 An Offer/Answer Model with the Session Description Protocol (SDP)
- 3265 Session Initiation Protocol (SIP) - Specific Event Notification
- 3275 (Extensible Markup Language) XML-Signature Syntax and Processing
- 3311 The Session Initiation Protocol (SIP) UPDATE Method
- 3323 A Privacy Mechanism for the Session Initiation Protocol (SIP)
- 3324 Short Term requirements for Network Asserted Identity
- 3325 Private Extensions to Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
- 3326 The Reason Header Field for the Session Initiation Protocol (SIP)
- 3327 Session Initiation Protocol (SIP) Extension Header Field for Registering Non-Adjacent Contacts
- 3362 Real-time Facsimile (T.38) - image/t38 MIME sub-type Registration
- 3388 Grouping of Media Lines in the Session Description Protocol (SDP)
- 3411 An Architecture for Describing Simple Network Management Protocol (SNMP) Management Frameworks
- 3412 Message Processing and Dispatching for the Simple Network Management Protocol (SNMP)
- 3413 Simple Network Management Protocol (SNMP) Applications\
- 3414 User-based Security Model (USM) for version 3 of the Simple Network Management Protocol (SNMPv3)
- 3415 View-based Access Control Model (VACM) for the Simple Network Management Protocol (SNMP)
- 3418 Management Information Base (MIB) for the Simple Network Management Protocol (SNMP)
- 3420 Internet Media Type message/sipfrag
- 3428 Session Initiation Protocol (SIP) Extension for Instant Messaging
- 3433 Entity Sensor Management Information Base
- 3515 The Session Initiation Protocol (SIP) Refer Method
- 3551 RTP Profile for Audio and Video Conferences with Minimal Control
- 3578 Mapping of Integrated Services Digital Network (ISDN) User Part (ISUP) Overlap Signaling to the Session Initiation Protocol (SIP)
- 3581 An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing
- 3605 Real Time Control Protocol (RTCP) attribute in Session Description Protocol (SDP)
- 3608 Session Initiation Protocol (SIP) Extension Header Field for Service Route Discovery During Registration
- 3665 Session Initiation Protocol (SIP) Basic Call Flow Examples
- 3666 Session Initiation Protocol (SIP) Public Switched Telephone Network (PSTN) Call Flows
- 3680 A Session Initiation Protocol (SIP) Event Package for Registrations
- 3711 The Secure Real-time Transport Protocol (SRTP)
- 3725 Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)
- 3840 Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)
- 3841 Caller Preferences for the Session Initiation Protocol (SIP)
- 3842 A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
- 3856 A Presence Event Package for the Session Initiation Protocol (SIP)
- 3857 A Watcher Information Event Template-Package for the Session Initiation Protocol (SIP)
- 3858 An Extensible Markup Language (XML) Based Format for Watcher

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- 3860 Common Profile for Instant Messaging (CPIM)
- 3863 Presence Information Data Format (PIDF)
- 3891 The Session Initiation Protocol (SIP) "Replaces" Header
- 3892 The Session Initiation Protocol (SIP) Referred-By Mechanism
- 3903 Session Initiation Protocol (SIP) Extension for Event State Publication
- 3911 The Session Initiation Protocol (SIP) "Join" Header
- 3960 Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)
- 3966 The tel URI for Telephone Numbers
- 4028 Session Timers in the Session Initiation Protocol (SIP)
- 4235 An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)
- 4240 Basic Network Media Services with SIP
- 4244 An Extension to the Session Initiation Protocol (SIP) for Request History Information
- 4320 Actions Addressing Identified Issues with the Session Initiation Protocol's (SIP) Non-INVITE Transaction
- 4424 Real-Time Transport Protocol (RTP) Payload Format for the Variable-Rate Multimode Wideband (VMR-WB) Extension Audio Codec
- 4458 Session Initiation Protocol (SIP) URIs for Applications such as Voicemail and Interactive Voice Response (IVR)
- 4474 Enhancements for Authenticated Identity Management in the Session Initiation Protocol (SIP)
- 4497 Interworking between the Session Initiation Protocol (SIP) and QSIG
- 4566 SDP: Session Description Protocol
- 4568 Session Description Protocol (SDP) Security Descriptions for Media Streams
- 4575 A Session Initiation Protocol (SIP) Event Package for Conference State
- 4579 Session Initiation Protocol (SIP) Call Control - Conferencing for User Agents

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Internet Engineering Task Force (IETF) ^ Back to Top

Communication Manager supports the following SIP-related IETF RFCs and Internet Drafts:

- RFC 1889 RTP: Real-Time Transport Protocol
- RFC 2246 The TLS Protocol
- RFC 2327 SDP: Session Description Protocol
- RFC 2396 URI generic syntax
- RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 3261 SIP: Session Initiation Protocol
- RFC 3262 Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
- RFC 3263 Session Initiation Protocol (SIP): Locating SIP Servers
- RFC 3264 An Offer/Answer Model with the Session Description Protocol (SDP)
- RFC 3265 SIP-Specific Event Notification
- RFC 3311 UPDATE method
- RFC 3324 Short Term Requirements for Network Asserted Identity
- RFC 3325 Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
- RFC 3515 REFER method
- RFC 3551 RTP Profile for Audio and Video Conferences with Minimal Control
- RFC 3578 Mapping of Integrated Services Digital Network (ISDN) User Part (ISUP) Overlap Signaling to the Session Initiation Protocol (SIP)
- RFC 3711 Secure Real-time Transport Protocol (SRTP)
- RFC 3840 Indicating User Agent Capabilities in the Session Initiation Protocol (SIP) (partial support)
- RFC 3841 Caller Preferences for the Session Initiation Protocol (SIP)
- RFC 3842 A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
- RFC 3891 The Session Initiation Protocol (SIP) "Replaces" Header
- RFC 3892 The SIP Referred-By Mechanism
- RFC 3911 The SIP "Join" Header
- RFC 3960 Early Media
- RFC 4028 Session Timers in SIP
- RFC 4235 An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)
- RFC 4244 Request History Information
- RFC 4497 Interworking between the Session Initiation Protocol (SIP) and QSIG
- RFC 4568 Security Descriptions for Media Streams
- RFC 4579 Session Initiation Protocol (SIP) Call Control - Conferencing for User Agents
- draft-elwell-sipping-redirection-reason-00
- draft-ietf-sipping-cc-transfer-02
- draft-ietf-sipping-conference-package-01
- draft-ietf-sipping-realtimefax-01
- draft-johnston-sipping-cc-uui-03

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