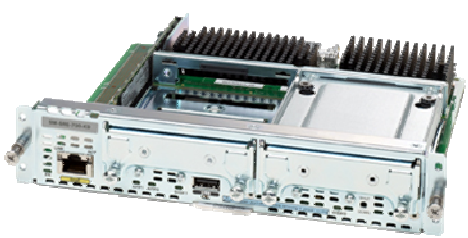
Cisco Unified SIP Proxy

Product Overview

The Cisco® Unified SIP Proxy (USP) is a high-performance, highly available Session Initiation Protocol (SIP) server for centralized routing and SIP signaling normalization. By forwarding requests between call-control domains, the Cisco Unified SIP Proxy provides the means for routing sessions within enterprise and service provider networks. The application is delivered in Network Module and Service Module form factors on Cisco 2900, 3800, 3900, and 3900E Series Integrated Services Routers.

Cisco Unified SIP Proxy aggregates SIP elements and enables application of highly developed routing rules. These rules enable greater control, management, and flexibility of SIP networks. Cisco Unified SIP Proxy simplifies large Cisco Unified Communications Manager, Cisco Unified Communications Manager Express (CME), Cisco Unified Border Element, and Cisco Unified Customer Voice Portal (CVP) deployments as well as other Cisco and multivendor deployments. Cisco Unified SIP Proxy provides great scalability, and network design can provide high availability (Figure 1).

Figure 1. Cisco Unified SIP Proxy Service Module



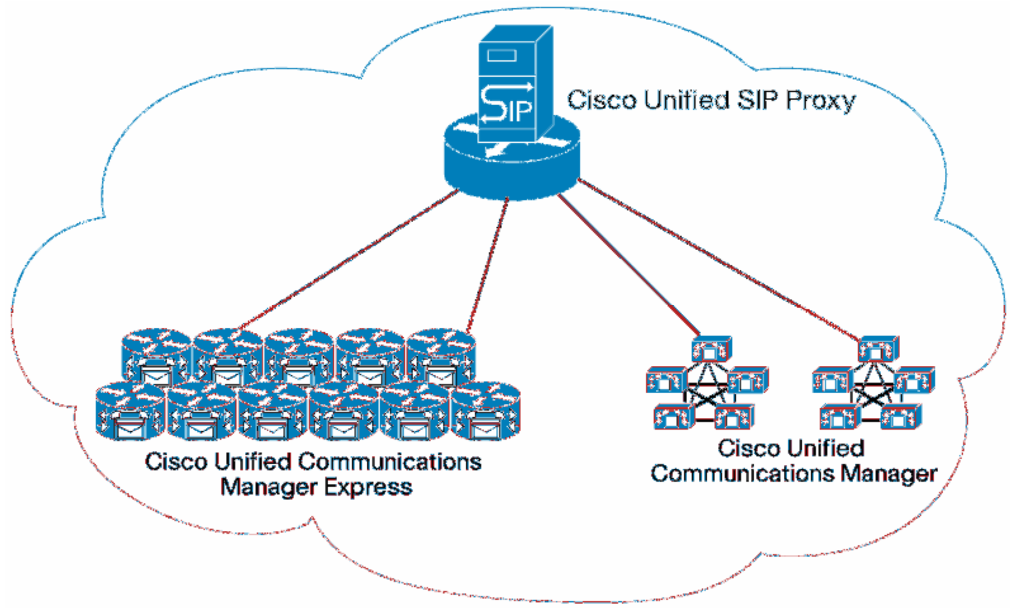
Applications

Cisco Unified SIP Proxy routes among SIP elements and helps enable a broad range of Unified Communications services.

Cisco Unified Communications Manager and Cisco Unified Communications Manager Express SIP Aggregation

Management of SIP dial peers across midsize and large Cisco Unified Communications Manager and Cisco Unified Communications Manager Express networks presents a challenge. As opposed to a full mesh, dial peers can be pointed to Cisco Unified SIP Proxy, which provides a central route point. This process also simplifies the addition and removal of new call-processing agents. If a call-processing agent is unavailable, alternate routing and recovery can be provided. You can apply dial normalization and load balancing as needed (Figure 2).

Figure 2. Cisco Unified Communications Manager and Cisco Unified Communications Manager Express SIP Aggregation

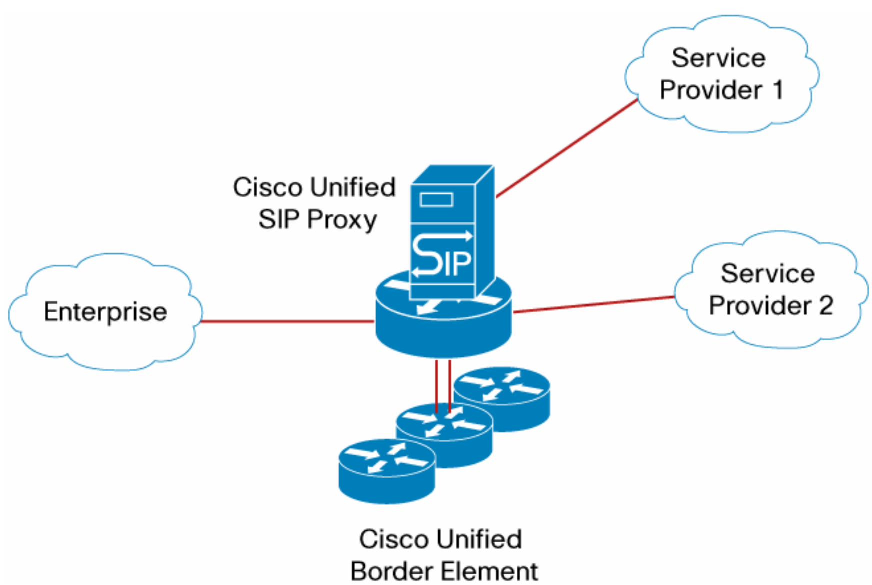


Cisco Unified Border Element Scalability and Load Balancing

Cisco Unified SIP Proxy provides a central route point for management of multiple Cisco Unified Border Elements. You can establish logical separations and use a single Cisco Unified SIP Proxy for either or both ingress and egress traffic. You can apply load balancing and rule-based routing, and provide interfacing to the SIP trunk signal normalization where needed (Figure 3).

If a Cisco Unified Border Element is unavailable, Cisco Unified SIP Proxy can intelligently reroute to an alternate Cisco Unified Border Element. When the Cisco Unified Border Element returns to service, Cisco Unified SIP Proxy resumes sending traffic to the Cisco Unified Border Element. This design enables need-based growth of the service provider interconnect and also avoids risk associated with a single point of failure for the border element.

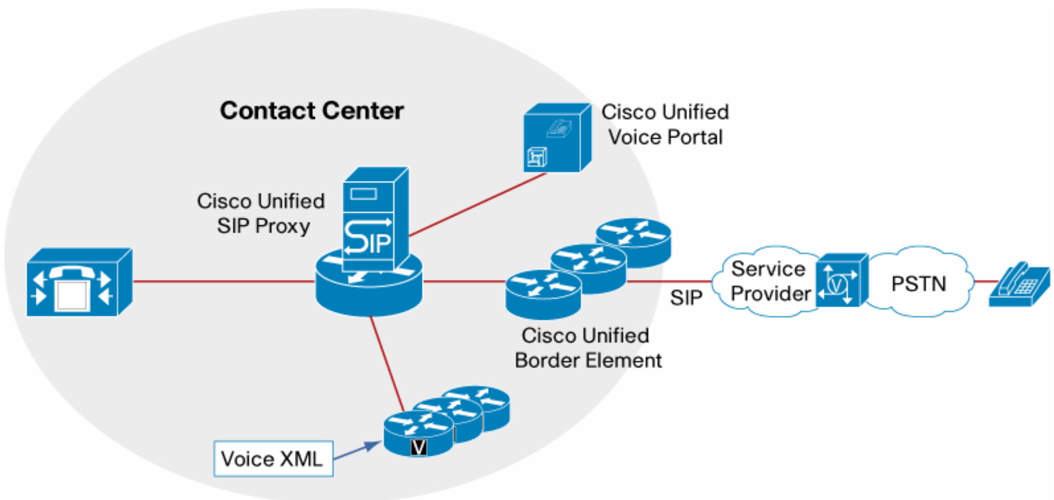
Figure 3. Cisco Unified Border Element Scalability and Load Balancing



SIP Trunk for Contact Center

Whether for inbound or outbound traffic, Cisco Unified SIP Proxy enables routing and management across contact center time-division multiplexing (TDM) and IP trunks. Routing is provided across gateways connecting outside the network as well as across multiple Cisco Unified Customer Voice Portals. If a Cisco Unified Customer Voice Portal or gateway is unavailable, Cisco Unified SIP Proxy can intelligently reroute to an alternate Cisco Unified Customer Voice Portal or gateway. When the Cisco Unified Customer Voice Portal or gateway returns to service, Cisco Unified SIP Proxy resumes sending traffic to the Cisco Unified Customer Voice Portal or gateway. You can apply load balancing and rule-based routing, and provide interfacing to the SIP trunk signal normalization where needed (Figure 4).

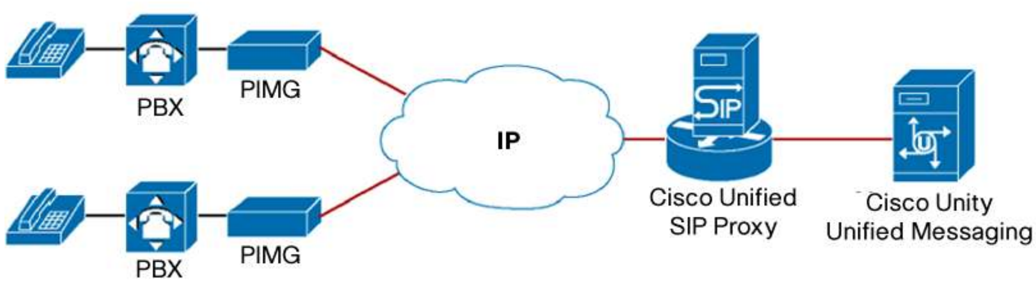
Figure 4. SIP Trunk for Contact Center



Cisco Unity PBX IP Media Gateway Integration

PBX IP Media gateways (PIMGs) are used to connect TDM-based private branch exchanges (PBXs) into Cisco Unity® voice messaging systems. Placement of Cisco Unified SIP Proxy in front of the Cisco Unity application enables PIMGs to share Cisco Unity ports, in turn enabling scalability of hybrid TDM PBX and IP messaging deployments (Figure 5).

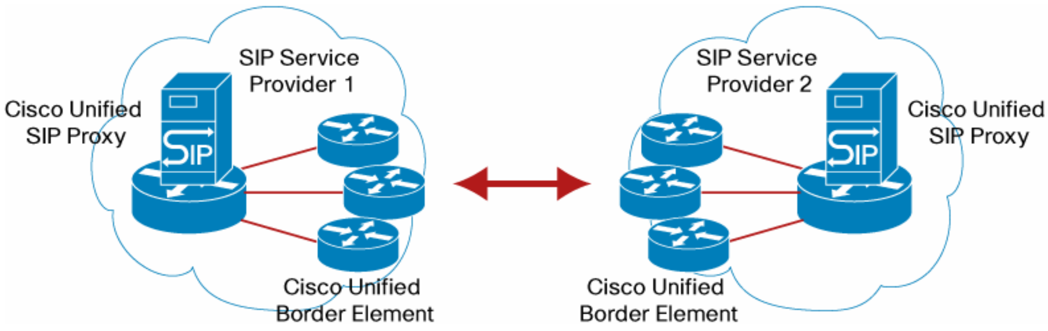
Figure 5. Cisco Unity PBX IP Media Gateway Integration



Service Provider SIP Interconnect Services

For interconnection among service providers, Cisco Unified SIP Proxy enables normalization of dial strings and SIP signaling variants. Cisco Unified SIP Proxy also provides routing and load balancing among SIP elements, including Cisco Unified Border Elements (Figure 6).

Figure 6. Service Provider SIP Interconnect



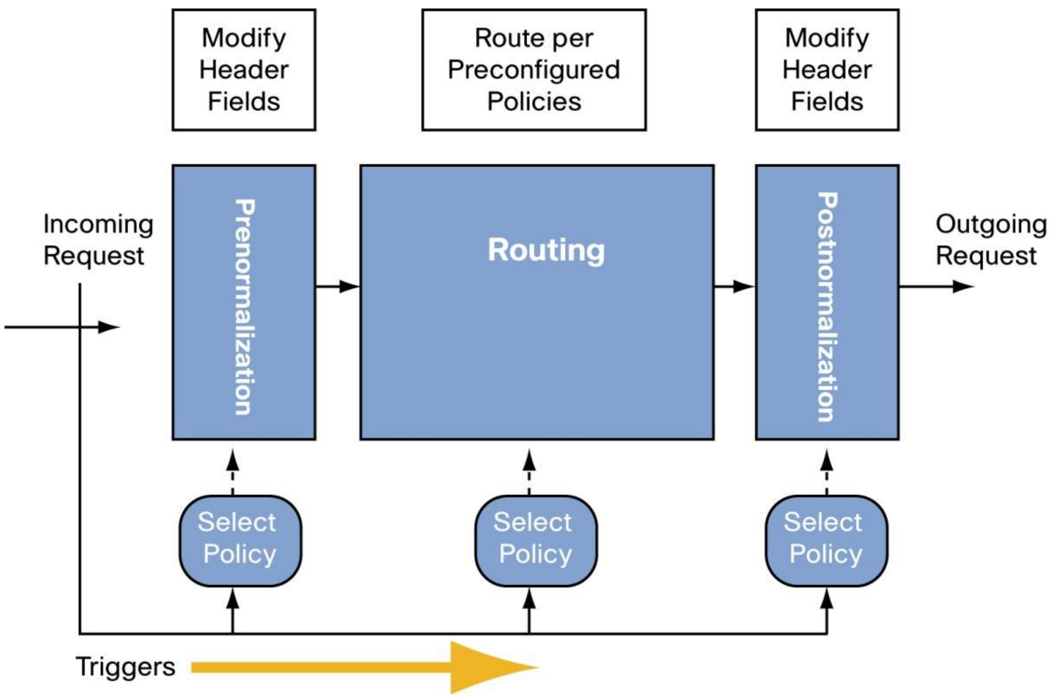
Product Architecture

Cisco Unified SIP Proxy Call Processing

The Cisco Unified SIP Proxy is a call and dialog stateless SIP proxy. Media flows around the proxy and signaling goes through the proxy. The proxy can also modify SIP headers (normalization). Routing and normalization are determined based on administrator-configured policies. Policies are selected based on triggers, administrator-configured conditions that are matched based on information in the SIP message.

As SIP messages come into the proxy, a determination is made as to whether any pre-normalization policies need to be applied. Following pre-normalization, new triggers are used to determine application of routing policies. A further series of triggers provides for further header modifications; for example, post-normalization policies after the routing decision has been made. In cases where policy is not asserted, the proxy provides for pass-through of the SIP message (Figure 7).

Figure 7. Cisco Unified SIP Proxy Call-Processing Model



You can apply distinct rules to groups of requests to create independent “virtualized proxies” within a single Cisco Unified SIP Proxy. The rules are highly flexible and scalable to form routing or normalization policies.

You can deploy Cisco Unified SIP Proxy in redundant network designs. Redundancy is provided using multiple Cisco Unified SIP Proxies in active-active and/or active-standby modes. In the active-active mode, both Cisco Unified SIP Proxies will be active and if one Cisco Unified SIP Proxy fails, the second Cisco Unified SIP Proxy will assume the load of the failed one. In the active-active mode, you can place two Cisco Unified SIP Proxies in a single router or enable redundancy across multiple routers, even taking advantage of different geographic locations thereby avoiding the risk of a single location failure. In the active-standby mode, one Cisco Unified SIP Proxy is in standby mode and if the active Cisco Unified SIP Proxy fails, the standby Cisco Unified SIP Proxy takes over the load. Active-standby mode is achieved using the HSRP (Hot Standby Router Protocol) function on the ISR.

Hierarchical designs provide for high scalability. Cisco Unified SIP Proxies might be placed regionally. You can form multiple Cisco Unified SIP Proxies as a cluster with a higher-order Cisco Unified SIP Proxy.

Features

- Proxy for SIP Unified Communications signaling
- Signaling support: Voice, video, fax, physical terminal line (tty), modem, caller ID, caller name, updates, transfer, forward, hold, conference, status, message waiting indicator (MWI), dual tone multi-frequency (DTMF) relay, and SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) (presence)
- Address resolution (Domain Name System [DNS]: Type A and SRV and Type NAPTR)
    - Domain name resolution based on RFC 3263, Locating SIP Servers
- TCP (Transmission Control Protocol), User Datagram Protocol (UDP), and Transport Layer Security (TLS)
- Lite Mode: Lite Mode provides the ability to run Cisco Unified SIP Proxy at a higher call rate than the standard licensed call rate by disabling record-route functionality. For more details on Lite Mode, refer to the Performance section
- Available as a network module on Cisco 2900, 3800, 3900, and 3900E Integrated Services Routers
    - Minimal performance effect on the router allows for concurrent router applications

Routing

- Routing based on policy
- Configurable multistep routing policies with route table lookup
    - Configurable match rules (for example, longest prefix, exact match, and fixed-length match)
    - Configurable keys selected from the SIP request: Remote address, local address, request uniform resource identifier (URI), P-Asserted-Identity (caller ID), diversion, remote-party ID, To, and From; within these headers Cisco Unified SIP Proxy can select the user, host, port, domain, phone number, URI, carrier codes, and location routing numbers
    - Configurable key modifiers (for example, case insensitivity, ignore plus, ignore display characters, etc.)
    - Numerous routing decisions: Forward to a single route, forward to a route group, reject, and chain to another route policy
- Table-based routing for mapping of requests to destinations
    - Support for large number of routes in a table (10,000+)
    - Routes populated through command-line interface or upload of a route file
- Example routing scenarios
    - URI-based routing (number and name)
    - Call block between specified sources and destinations, including policy-based transit routing (policy may require certain calls to either avoid or to take certain routes)

  - Class of restriction
  - Translation of on-net to off-net dial plans (including public switched telephone network [PSTN] and IP-IP); simplifies network management, eliminating the need to configure translations in each call agent
- Percentage and weight-based routing
  - Load balancing among downstream elements based on preset weight
  - Priority values assignable for routing of selected calls; also enables configuration for least-cost routing
- Time policy routing
  - Time(s) in a day, day(s) in a week, day(s) in a month, and month(s) in a year
- Ability to form downstream elements into a single logical group for load balancing and failover
- SIP element health management and monitoring
  - Rerouting around unavailable SIP elements
  - Ping for service availability and restoration of routing when unavailable SIP element is restored
- Rerouting based on redirect responses (Routing policy and post-normalization applies to the new destination specified in the contact header of the redirect response, and provides for sequential forking.)
- Transport protocol conversion: TCP, UDP, TLS (For example, an incoming call received over UDP can be forwarded to a destination over TLS.)
- Configurable record-routing (on/off)
- Global unique caller ID pass-through

Normalization

- Normalization of SIP headers based on configurable policy
  - Ability to add, remove, or update headers and header parameters
  - Ability to update URI components such as user, domain, and host and ability to add, remove, and update URI parameters
  - Digit manipulation
  - Address manipulation
  - TEL URI <=> SIP URI conversion
  - Domain conversions
  - Regular-expression processing
- Construction of multistep normalization policies
- Pre- and postnormalization
  - Pre-normalization prior to proxy application of routing rules (for example, applied to message coming into the proxy)
  - Post-normalization after proxy application of routing rules (for example, applied to message going out from the proxy)

Rules-Based Selection of Routing and Normalization Policies

- Rich set of configurable rules
  - SIP message type (for example, request and response)
  - SIP method: INVITE, UPDATE, REFER, PRACK, BYE, SUBSCRIBE, NOTIFY, unsolicited NOTIFY, MESSAGE, PUBLISH, REGISTER, INFO, OPTIONS, and any custom or future SIP extensions
  - Request-URI: User, host, phone number, etc.

   - Local and remote IP, port, and protocol of the received SIP message
   - Network name of the incoming and outgoing request (A network is a set of SIP listening points.)
   - Transport protocol
   - Regular expression match on any SIP header
   - Time policy check
   - SIP response code
   - Mid-dialog message check
- Call Admission Control
   - Call counting based

Security and Privacy

- TLS (bidirectional)
- Through-header stripping (for topology hiding)
- User privacy (RFC 3325 P-Asserted ID: Removes P-Asserted ID when receiving a message from an element not configured as trusted, and removes P-Asserted ID and Privacy header when forwarding a message to an element not configured as trusted)

Network Design

- Multiple IP addresses (up to eight) to provide for flexible configuration and network topology design; you can group IP addresses to form networks and apply rules on these networks
- Multiple SIP listening points; each listen point can have configurable port
- "Virtualized proxies" with multiple independent routing and normalization processing in a single server
- Redundancy through clustered network design for high availability
   - Clusters addressed as Fully Qualified Domain Names (FQDNs). Domain Name System (DNS) resolution via Service (SRV) record
   - Virtual IP addressing using Hot Standby Router Protocol (HSRP)
- Very high scalability with clustering of multiple Cisco Unified SIP Proxies
   - Hierarchical and peer requests among clustered Cisco Unified SIP Proxies
   - Up to two Cisco Unified SIP Proxy network modules within the same router and / or multiple Cisco Unified SIP Proxy network modules across different routers

Management

- Flexible management through Graphical User Interface and Command-Line Interface
- Monitor system status using SNMP MIBs
- Load of preexisting configurations onto the Cisco Unified SIP Proxy module
- Copy of configurations off the Cisco Unified SIP Proxy module
- Graceful shutdown and restore, allowing for completion of transactions in process
- RADIUS accounting for SIP events
- SIP message logging for call monitoring
- Trace logging for troubleshooting
- FTP access to Cisco Unified SIP Proxy for easy download of trace logs, SIP message logs, configuration files, and route files and upload of configuration files and route files

- SIP message metrics logging (peg counting); for example, count of incoming and outgoing messages over a period of time and logging to a file
- Detailed call statistics with call attempt, success and failure rates per element
- Database store for debugs and logs
    - Selectively log messages using regular expressions
    - Search through stored log messages
    - Log up to one million log messages

Supported Standards as a SIP Proxy

- IETF RFC 2246: The TLS Protocol Version 1.0
- IETF RFC 2327 SDP: Session Description Protocol
- IETF RFC 2617 HTTP Authentication: Basic and Digest Access Authentication
- IETF RFC 2782: A DNS RR for specifying the location of services (DNS SRV)
- IETF RFC 2806: URLs for Telephone Calls
- IETF RFC 2976: The SIP INFO Method
- IETF RFC 3204: MIME media types for ISUP and QSIG Objects
- IETF RFC 3261 SIP: Session Initiation Protocol
- IETF RFC 3262: Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
- IETF RFC 3263: Session Initiation Protocol (SIP): Locating SIP Servers
- IETF RFC 3264: An Offer/Answer Model with the Session Description Protocol (SDP)
- IETF RFC 3265: Session Initiation Protocol (SIP)-Specific Event Notification
- IETF RFC 3311: The Session Initiation Protocol (SIP) UPDATE Method
- IETF RFC 3325: Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
- IETF RFC 3326: The Reason Header Field for the Session Initiation Protocol (SIP)
- IETF RFC 3515: The Session Initiation Protocol (SIP) Refer Method
- IETF RFC 3665: Session Initiation Protocol (SIP) Basic Call Flow Examples
- IETF RFC 3666: Session Initiation Protocol (SIP) Public Switched Telephone Network (PSTN) Call Flows
- IETF RFC 3725: Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)
- IETF RFC 3842: A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
- IETF RFC 3856: A Presence Event Package for the Session Initiation Protocol (SIP)
- IETF RFC 3891: The Session Initiation Protocol (SIP) "Replaces" Header
- IETF RFC 3892: The Session Initiation Protocol (SIP) Referred-By Mechanism
- IETF RFC 4480 RPID: Rich Presence Extensions to the Presence Information Data Format (PIDF)
- SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE)

Ordering

To order, visit the Cisco Ordering Tool.

Cisco Unified SIP Proxy is available in Network Module and Service Module form factors.

Table 1. Cisco Unified SIP Proxy modules

Part Number	Description	Minimum Version of Cisco Unified SIP Proxy Software Required
NME-CUSP-522	Cisco Unified SIP Proxy Network Module	1.1.2
SM-SRE-700-K9	Cisco Service Ready Engine 700 Service Module	8.5.1
SM-SRE-900-K9	Cisco Service Ready Engine 900 Service Module	8.5.1

Ordering Cisco Unified SIP Proxy with Network Module Enhanced (NME)

Cisco Unified SIP Proxy on the NME is supported on specific ISR platforms.

Table 2. Supported ISR Platforms for Cisco Unified SIP Proxy on the NME-CUSP-522

	ISR Platform				
	2800	3800	2911/2921	2951/3900	3900E
NME-CUSP-522	No	Yes	No	Yes**	Yes**
Minimum IOS Version on ISR		12.4(22)T		15.0(1)M	15.1(1)T

*** Requires SM-NM-ADPTR*

Cisco Unified SIP Proxy employs a counted feature license based on the maximum number of new incoming SIP requests per second. Requests that belong to an existing dialog, including SIP responses, are not counted. Cisco Unified SIP Proxy feature licenses are available at multiple levels. Refer to Tables 3, 4, and 5 for the Cisco Unified SIP Proxy feature licenses available for NME-CUSP-522.

Table 3. Cisco Unified SIP Proxy Feature Licenses for NME-CUSP-522

Part Number	Description
FL-CUSP-10	CUSP Feature License for 10 SIP requests/second
FL-CUSP-30	CUSP Feature License for 30 SIP requests/second
FL-CUSP-100	CUSP Feature License for 100 SIP requests/second

Note: The licensed number of requests/second refers to new incoming SIP requests. Requests that belong to an existing dialog, including SIP responses are not counted.

Table 4. Cisco Unified SIP Proxy Upgrade Licenses for NME-CUSP-522

Part Number	Description
FL-CUSP-10U30=	CUSP Upgrade License for 10 to 30 SIP requests/second
FL-CUSP-10U100=	CUSP Upgrade License for 10 to 100 SIP requests/second
FL-CUSP-30U100=	CUSP Upgrade License for 30 to 100 SIP requests/second

Cisco Unified SIP Proxy upgrade licenses can be received over e-mail. In order to receive upgrade licenses over e-mail, e-delivery type license has to be chosen. Refer to Table 5 for Cisco Unified SIP Proxy e-delivery upgrade licenses available for NME-CUSP-522.

Table 5. Cisco Unified SIP Proxy Upgrade E-delivery Licenses

Part Number	Description
L-FL-CUSP-10U30=	CUSP Upgrade License for 10 to 30 SIP requests/second (e-delivery)
L-FL-CUSP-10U100=	CUSP Upgrade License for 10 to 100 SIP requests/second (e-delivery)
L-FL-CUSP-30U100=	CUSP Upgrade License for 30 to 100 SIP requests/second (e-delivery)

Ordering Cisco Unified SIP Proxy with Service Ready Engine (SRE)

Cisco Unified SIP Proxy on the SRE is supported on specific ISR platforms.

Table 6. Cisco Unified SIP Proxy module ISR Platform Support

	ISR Platform				
	2800	3800	2911/2921	2951/3900	3900E
SM-SRE-700-K9	No	No	Yes	Yes	Yes
SM-SRE-900-K9	No	No	Yes	Yes	Yes
Minimum IOS Version on ISR			15.0(1)M		15.1(1)T

Cisco Unified SIP Proxy employs a counted feature license based on the maximum number of new incoming SIP requests per second. Requests that belong to an existing dialog, including SIP responses, are not counted. Refer to Table 7 for Cisco Unified SIP Proxy SIP requests per second supported on the SRE modules.

Table 7. Cisco Unified SIP Proxy Licenses supported on SRE modules

SIP requests per second	SM-SRE-700-K9	SM-SRE-900-K9
2	Yes	Yes
10	Yes	Yes
30	Yes	Yes
100	Yes	Yes
200		Yes

Refer to Tables 8, 9, and 10 for the Cisco Unified SIP Proxy feature licenses available for the Service Ready Engine modules.

Table 8. Cisco Unified SIP Proxy Feature Licenses—Used for pre-install on SRE shipments from the factory

Part Number	Description
FL-CUSP-2	CUSP Feature License for 2 SIP requests/second
FL-CUSP-10	CUSP Feature License for 10 SIP requests/second
FL-CUSP-30	CUSP Feature License for 30 SIP requests/second
FL-CUSP-100	CUSP Feature License for 100 SIP requests/second
FL-CUSP-200	CUSP Feature License for 200 SIP requests/second

Note:

The licensed number of requests/second refers to new incoming SIP requests. Requests that belong to an existing dialog, including SIP responses are not counted.

FL-CUSP-200 is supported on SM-SRE-900-K9 only.

Cisco Unified SIP Proxy license can be added to an SM-SRE-700-K9 or SM-SRE-900-K9 Service Ready Engine after shipment from the factory. In this case a spare feature license is used. These spare licenses can be received either via standard delivery or over e-mail.

- Refer to Table 9 for Cisco Unified SIP Proxy spare feature licenses available for the SRE modules that are to be received via standard delivery.
- Refer to Table 10 for Cisco Unified SIP Proxy feature licenses available for the SRE modules that are to be received via e-mail (e-delivery).

Table 9. Cisco Unified SIP Proxy Spare Licenses for SRE modules – Standard delivery

Part Number	Description
FL-CUSP-2	CUSP Feature License for 2 SIP requests/second Spare
FL-CUSP-10	CUSP Feature License for 10 SIP requests/second Spare
FL-CUSP-30	CUSP Feature License for 30 SIP requests/second Spare
FL-CUSP-100	CUSP Feature License for 100 SIP requests/second Spare
FL-CUSP-200	CUSP Feature License for 200 SIP requests/second Spare

Note: FL-CUSP-200= is supported on SM-SRE-900-K9 only.

Table 10. Cisco Unified SIP Proxy Spare E-delivery Licenses for SRE modules

Part Number	Description
L-FL-CUSP-2	CUSP Feature License for 2 SIP requests/second Spare (e-delivery)
L-FL-CUSP-10	CUSP Feature License for 10 SIP requests/second (e-delivery)
L-FL-CUSP-30	CUSP Feature License for 30 SIP requests/second (e-delivery)
L-FL-CUSP-100	CUSP Feature License for 100 SIP requests/second (e-delivery)
L-FL-CUSP-200	CUSP Feature License for 200 SIP requests/second (e-delivery)

Note: L-FL-CUSP-200= is supported on SM-SRE-900-K9 only.

Cisco Unified SIP Proxy supports upgrade from one license level to a higher one according to the capacity of the module. Refer to Table 11 for Cisco Unified SIP Proxy upgrade licenses. In the case of a multi-level upgrade, for example, to upgrade from 2 SIP requests per second to 100 SIP requests per second, you would use FL-CUSP-2U10= and then FL-CUSP-10U100= licenses.

Table 11. Cisco Unified SIP Proxy Upgrade Licenses

Part Number	Description
FL-CUSP-2U10=	CUSP Upgrade License for 2 to 10 SIP requests/second
FL-CUSP-10U30=	CUSP Upgrade License for 10 to 30 SIP requests/second
FL-CUSP-10U100=	CUSP Upgrade License for 10 to 100 SIP requests/second
FL-CUSP-30U100=	CUSP Upgrade License for 30 to 100 SIP requests/second
FL-CUSP-100U200=	CUSP Upgrade License for 100 to 200 SIP requests/second

Note: FL-CUSP-100U200= is supported on SM-SRE-900-K9 only.

Cisco Unified SIP Proxy upgrade licenses can be received over e-mail. In order to receive upgrade licenses over e-mail, e-delivery type license has to be chosen. Refer to Table 12 for Cisco Unified SIP Proxy e-delivery upgrade licenses.

Table 12. Cisco Unified SIP Proxy Upgrade E-delivery Licenses

Part Number	Description
L-FL-CUSP-2U10=	CUSP Upgrade License for 2 to 10 SIP requests/second (e-delivery)
L-FL-CUSP-10U30=	CUSP Upgrade License for 10 to 30 SIP requests/second (e-delivery)
L-FL-CUSP-10U100=	CUSP Upgrade License for 10 to 100 SIP requests/second (e-delivery)

Part Number	Description
L-FL-CUSP-30U100=	CUSP Upgrade License for 30 to 100 SIP requests/second (e-delivery)
L-FL-CUSP-100U200=	CUSP Upgrade License for 100 to 200 SIP requests/second (e-delivery)

Note: L-FL-CUSP-200= is supported on SM-SRE-900-K9 only.

Restrictions

The following limitation only applies to Cisco Unified SIP Proxy 1.1.5 or earlier versions of the application. Cisco Unified SIP Proxy 1.1.5 or earlier may not co-reside in the same router when Cisco Unified Communications Manager Express or Cisco Unified Survivable Remote Site Telephony is configured for Skinny Client Control Protocol (SCCP) controlled phones. Nor may Cisco Unified SIP Proxy 1.1.5 or earlier co-reside in the same router with TDM gateways or configuration of H.323 dial peers (including Cisco Unified Border Element). Cisco Unified Communications Manager Express, Cisco Unified Survivable Remote Site Telephony, and Cisco Unified Border Element configured for SIP may co-reside in the same router. Other voice and router functions are also available for use in the same router with Cisco Unified SIP Proxy.

With Cisco Unified SIP Proxy 8.5 or later versions of the application, Cisco Unified SIP Proxy may co-reside in the same router with any other application supported on the Integrated Services Router.

Performance

Performance is limited by both the number of incoming SIP requests specified in the feature license and module processing capability. With Cisco Unified SIP Proxy version 8.5 onwards, Cisco Unified SIP Proxy can be operated in Standard and Lite modes. Standard mode will provide the standard performance described by the feature license installed. Lite mode will enable Cisco Unified SIP Proxy to run at a higher SIP requests/second rate when the Record Route feature is disabled.

Table 13 describes the Cisco Unified SIP Proxy performance on the NME-CUSP-522 module.

Table 13. Cisco Unified SIP Proxy performance on NME-CUSP-522 module

Cisco Unified SIP Proxy Feature License	NME-CUSP-522	
	Standard Mode (SIP requests/second)	Lite Mode (SIP requests/second)
FL-CUSP-10	10	10
FL-CUSP-30	30	30
FL-CUSP-100	100	450

Table 14 describes the Cisco Unified SIP Proxy performance on the SM-SRE-700-K9 module.

Table 14. Cisco Unified SIP Proxy performance on SM-SRE-700-K9 module

Cisco Unified SIP Proxy Feature License	SM-SRE-700-K9	
	Standard Mode (SIP requests/second)	Lite Mode (SIP requests/second)
FL-CUSP-2	2	5
FL-CUSP-10	10	25
FL-CUSP-30	30	75
FL-CUSP-100	100	450

Table 15 describes the Cisco Unified SIP Proxy performance on the SM-SRE-900-K9 module.

Table 15. Cisco Unified SIP Proxy performance on SM-SRE-900-K9 module

Cisco Unified SIP Proxy Feature License	SM-SRE-900-K9	
	Standard Mode (SIP requests/second)	Lite Mode (SIP requests/second)
FL-CUSP-2	2	5
FL-CUSP-10	10	25
FL-CUSP-30	30	75
FL-CUSP-100	100	450
FL-CUSP-200	200	750

Performance is the same for both TCP and UDP based deployments.

Performance will vary depending on call flows. Maximum module performance will be lower when DNS lookup, SIP logging, or RADIUS logging services are enabled.

Hardware Specifications

Table 16. Hardware Specifications for NME-CUSP-522-K9

Hardware part number	Cisco NME-CUSP-522-K9
Form factor	Enhanced network module (NME)
CPU	1.4-GHz Intel Pentium M
Memory (RAM)	2 GB
Storage	160-GB hard disk
Internal network interfaces	10/100/1000 Gigabit Ethernet connectivity to router backplane
Physical characteristics	Dimensions (H x W x D): 1.55 x 7.10 x 7.2 in. (3.9 x18.0 x 18.3 cm) Weight: 1.5 lb (0.7 kg) maximum
Operating environment	Operating temperature: 41 to 104℉ (5 to 40℃) Nonoperating and storage temperature: –40 to 158℉ (–40 to 70℃) Operating humidity: 5 to 85% (noncondensing) Operating altitude: –197 to 6000 ft (–60 to 1800m)
Safety	UL 60950-1, Safety of Information Technology Equipment-Safety-Part 1: General Requirements (USA); plastic materials that are exposed to the end user shall meet the requirements of fire enclosure (UL94V-1) as defined in UL 60950
EMC	Emission: • 47 CFR Part 15 Class A • CISPR22 Class A • EN300386 Class A • EN55022 Class A • EN61000-3-2 • EN61000-3-3 • SD/EMI (India) • KN22 (Korea) • VCCI Class I • AS/NZS CISPR 22 • Class A Immunity: • CISPR24 • EN300386 • EN50082-1 • EN55024 • SD/EMI (India) • KN22 (Korea) • EN61000-6-1

Table 17. Cisco SRE Module Product Specifications

Feature	Cisco SRE-700 SM	Cisco SRE-900 SM
Product SKU	SM-SRE-700-K9	SM-SRE-900-K9
Form factor	SM	SM
CPU	Intel® Core™2 1.86 GHz	Intel® Core™2 Duo 1.86 GHz
DRAM	2 GB	4 GB
Compact Flash memory	2-GB internal USB flash-memory module	2-GB internal USB flash-memory module
Hard disk	1 x 500 GB	2 x 500 GB (1 TB in non-RAID mode)
Hot-swappable HDD	None	Yes
Redundant Array of Independent Disks (RAID) support	None	RAID 1
Internal network interfaces	Gigabit Ethernet connectivity to router backplane	Gigabit Ethernet connectivity to router backplane
External network interfaces	1 USB connector 1 RJ-45 Gigabit Ethernet connector	1 USB connector 1 RJ-45 Gigabit Ethernet connector
Router platforms	2911, 2921, 2951, 3925, 3945	2911, 2921, 2951, 3925, 3945
Cisco IOS® Software (on Router)	IOS release 15.0(1)M	IOS release 15.0(1)M
Embedded hardware-based cryptography acceleration	No	Yes
Power Specifications		
Power consumption (maximum)	50W	50W
Physical Specifications		
Dimensions (H x W x D)	1.58 x 7.44 x 7.5 in. (4 x 18.9 x 19.1 cm)	1.58 x 7.44 x 7.5 in. (4 x 18.9 x 19.1 cm)
Shipping dimensions (H x W x D with packaging)	9.5 x 7.5 x 2.5 in. (24.1 x 19.1 x 6.4 cm)	9.5 x 7.5 x 2.5 in. (24.1 x 19.1 x 6.4 cm)
Maximum weight	2.5 lb (1.1 kg)	2.5 lb (1.1 kg)
Environmental Specifications		
Operating Conditions		
Operating temperature	32 to 104℉ (0 to 40℃) normal 23 to 131℉ (-5 to +55℃) short term	32 to 104℉ (0 to 40℃) normal 23 to 131℉ (-5 to +55℃) short term
Humidity	10 to 85% operating	10 to 85% operating
Altitude (operating)	104℉ (40℃) at sea level 104℉ (40℃) at 6,000 ft (1,800m) 86℉ (30℃) at 13,000 ft (4,000m) 27.2℃ (81℉) at 15,000 ft (4,600m) Note: De-rate 34.5℉ (1.4℃) per 1,000 ft above 6,00 0 ft (per 300 m above 2,600 m)	
Transportation/Storage Conditions		
Temperature	-4 to 149℉ (-20 to +65℃)	-4 to 149℉ (-20 to +65° C)
Relative humidity	5 to 95%	5 to 95%
Altitude	15,000 ft (4,600 m)	15,000 ft (4,600 m)
Regulatory Compliance		
Safety	• UL 60950-1, First Edition, Standard for safety for information technology equipment (US) • CAN/CSA-C22.2 No. 60950-1-03, Safety of information technology equipment including electrical business equipment (Canada) • IEC 60950-1:2001, Safety of information technology equipment / Second Edition -2005) (World-Wide)- 2nd Ed. 2005 (is optional and will roll in by Dec. 1, 2010) • EN 60950 -1:2001, Safety of information technology equipment (CENELEC; includes EU and EFTA) • GB4943-2001, Safety of information technology equipment (PRC) • AS/NZS 60950-1, Safety of information technology equipment including electrical business equipment (Australia) • NOM-019, Safety of data processing equipment (Mexico)	

EMC	Emission: • 47 CFR Part 15 Class A • CISPR22 Class A • EN300386 Class A • EN55022 Class A • EN61000-3-2 • EN61000-3-3 • SD/EMI (India) • KN22 (Korea) • VCCI Class I • AS/NZS CISPR 22 Class A Immunity: • CISPR24 • EN300386 • EN50082-1 • EN55024 • SD/EMI (India) • KN22 (Korea) • EN61000-6-1

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