Miercom

Lab Testing Summary Report

March 2006
Report 060302

Product Category:

IP Telephony and Converged Applications

Products Tested:

Cisco Unified Communications System

Featuring:

Cisco Unified CallManager 5.0

Cisco Unified CallManager Express 3.4

Survivable Remote Site Telephony 3.4

Cisco Unified Personal Communicator 1.0

Cisco Unified CallManager 5.0

Cisco Unified Presence Server

Miercom NETWORKS AS ADVERTISED™ Independent Lab Tested

Key Findings and Conclusions:

- **Driven by the new Cisco Unified CallManager 5.0, the Cisco Unified Communications System natively integrates SIP alongside Cisco "SCCP"**
- **Testing confirms seamless, full-featured, user-transparent SIP-SCCP telephony interoperability**
- **"Presence" support has been added throughout, for both SIP *and* SCCP phone users, as well as new presence-enabled applications**
- **New SIP-based applications include Cisco Unified Personal Communicator, providing PC and Mac users with desktop video, Web collaboration, and access to voicemail, MeetingPlace, and presence-enabled directory**

Cisco Systems engaged Miercom to independently review the Cisco Unified Communications System–the consolidation of Cisco's IP-telephony wares, featuring the new SIP-supporting CallManager 5.0. Key among the verified new capabilities: user-transparent, full-featured interoperability between SIP and SCCP environments; straightforward upgrade to CallManager 5.0 on Linux; and verifying that comparable IP-

SIP and/or SCCP: Cisco Users Now Can Choose

Test results from Miercom review of Cisco Unified Communications System
SIP = New Model 7961 phone, new v8.0 SIP firmware
SCCP = Standard Model 7960 phone, current version SCCP firmware
All testing performed on new Unified CallManager 5.0

Connection Scenarios	Metric	Result
SIP to SCCP **SCCP to SIP** **SIP to SIP** **SCCP to SCCP**	**One-way latency,** phone to phone	**Under 80 ms, excellent call quality** in all scenarios
	Fail-over: CallManagers in same cluster	**Same effective fail-over in all scenarios** – for new calls and established calls
	Fail-over: Survivable Remote Site Telephony (SRST)	**Same effective fail-over in all scenarios** – for new calls and established calls
	Security: RTP link encryption, Key exchange, Authentication, Secure call control	**Same security** in all scenarios
	Features	**170+ features** in all scenarios

Cisco Unified Communications System

Key Components Reviewed

SIP and SCCP Call Control

Call Controller	Version	Platform	IP-Station Environment
Cisco Unified CallManager	5.0	Linux	For 100 to 30,000 stations; single server, or up to eight in cluster
CallManager Express	3.4	Cisco IOS router	Up to 240 stations; Unity Express module adds voicemail, auto-attendant
Survivable Remote Site Telephony (SRST)	3.4	Cisco IOS router	For branch/remote-office survivability, to 720 stations
Cisco IOS Gateway	12.4T	Cisco IOS router	Expands gateway capability from H.323 & MGCP currently, to include all the latest SIP capabilities

Cisco Unified IP Phones

IP Phone Model	# Lines	Protocol Support	Key features
7971G-GE	8	SIP or SCCP	Color display, speaker, Gigabit Ethernet
7970G	8	SIP or SCCP	Color display, speaker
7961G-GE	6	SIP or SCCP	Display, speaker, Gigabit Ethernet
7961G	6	SIP or SCCP	Display, speaker
7960G	6	SIP or SCCP	Display, speaker
7941G-GE	2	SIP or SCCP	Display, speaker, Gigabit Ethernet
7941G	2	SIP or SCCP	Display, speaker
7940G	2	SIP or SCCP	Display, speaker
7912G	1	SIP or SCCP	Display, speaker
7905G	1	SIP or SCCP	Display, speaker
7920	6	SCCP	IEEE 802.11b Wifi phone
7985G	1	SCCP	Color Video, speaker

SIP-based, SIP-enabled Applications

Application	Description	Key capabilities
Unified Personal Communicator v1.0	SIP-based client software, Mac and PC versions	Desktop video; optional softphone, Web collaboration, conferencing; presence-enabled directory, contact lists, click-to-call; recent call log; access to MeetingPlace, voice mail; (IM and call routing coming in next release)
MeetingPlace Express v1.0	Linux based conferencing	For 20 to 120 concurrent users; concurrent SIP and H.323 support; audio conferencing and screen sharing
Cisco Unified Presence Server	Presence aggregation and integration	New with SIP; monitors and distributes presence status; also supports "IP Phone Messenger" application
Mobility Manager server	Manages mobility applications	Supports "Mobile Connect" application – simultaneous call delivery to up to four phone devices
Unity Connection	Voice mail and unified messaging	For up to 1,500 subscribers; SIP-enabled; voice mail message store; integrates with Exchange, Outlook
IP Contact Center Express v4.5	SIP supporting contact center	Up to 300 agents; SIP agents now supported; supervisor call-control functions are via JTAPI to CallManager 5.0

Managing the SIP Environment

Application	Description	Key capabilities
Cisco Unified Operations Manager v1.1	Win 2003-based management	Auto-discovery and hierarchical mapping; identifies and details all IP phones, including 3rd-party SIP; runs diagnostic tests of individual phones
Cisco Unified Services Monitor v1.1	Multi-protocol call monitoring	Works with probes; monitors up to 80 concurrent calls (RTP streams); supports SIP and SCCP concurrently

telephony performance is delivered to users in the SIP environment, and in a mixed SIP-and-SCCP environment, as users enjoyed in the previous SCCP-only environment.

SIP and SCCP Call Control

Our testing to compare SIP with SCCP and mixed SIP-SCCP environments focused on three critical areas: voice and connection quality; reliability and survivability; and security. As the table on page 1 summarizes, we confirmed that Cisco delivers the same fast call set-up, connection and voice quality, and all the security, failover and survivability features for the SIP environment.

Security-wise, even Secure-RTP (voice-stream) encryption works seamlessly – between new Cisco phones running the new SIP firmware (v8.0) and (a) other new Cisco phones (running SIP or SCCP), or (b) new or old Cisco phones running SCCP. So whether running all SIP phones, or a mix of SIP and SCCP phones, voice conversations can still be effectively protected. And we confirmed that there is no perceptible degradation in voice quality or latency due to the (128-bit-key, AES-based, SRTP) encrypted media (or between SIP and SCCP phones), as between SCCP phones before.

We also validated key aspects of survivability by setting up and exercising various fail-over scenarios. With calls between SIP phones and between a mix of SIP and SCCP phones, we failed a redundant CallManager server in a cluster, and observed the effect on established calls, and delay in placing new calls.

The only phones affected are those registered to the failed server, and SIP and SCCP phones are impacted to the same extent. Calls that are up at the time of a server failure remain up, whether between SIP phones or SIP and SCCP phones. The same reliability applies in this case whether it is a CallManager 5.0 on a Cisco MCS server, or a CallManager Express 3.4, which runs on an IOS router platform. We also exercised the case where a branch-office, with a mix of SIP and SCCP phones, loses remote call control. We employed the remote-branch local-survivability form of CallManager, SRST, v3.4, which also runs on an IOS router. In all these environments, there were no differences in the reliability or survivability that SIP-phone users get, compared to what SCCP phone users have long enjoyed.

The newest, SIP-supporting version of CallManager, 5.0, runs on Linux, unlike previous versions, which run on Windows 2000. Users who buy CallManager 5.0 can have it pre-installed on Cisco MCS servers.

For upgrading, all the files and CD's needed to upgrade an existing Windows-based CallManager to 5.0 and Linux are also included. The Miercom team went through the upgrade procedure. It takes time, but it is straightforward (see below screenshot of the upgrade wizard), and in the end, it does work.

Win2000 to Linux. The process to upgrade from a Windows-based CallManager to a 5.0 version on Linux is detailed, but straightforward. The above screen shows the wizard-based interface that guides the user step-by-step through the procedure.

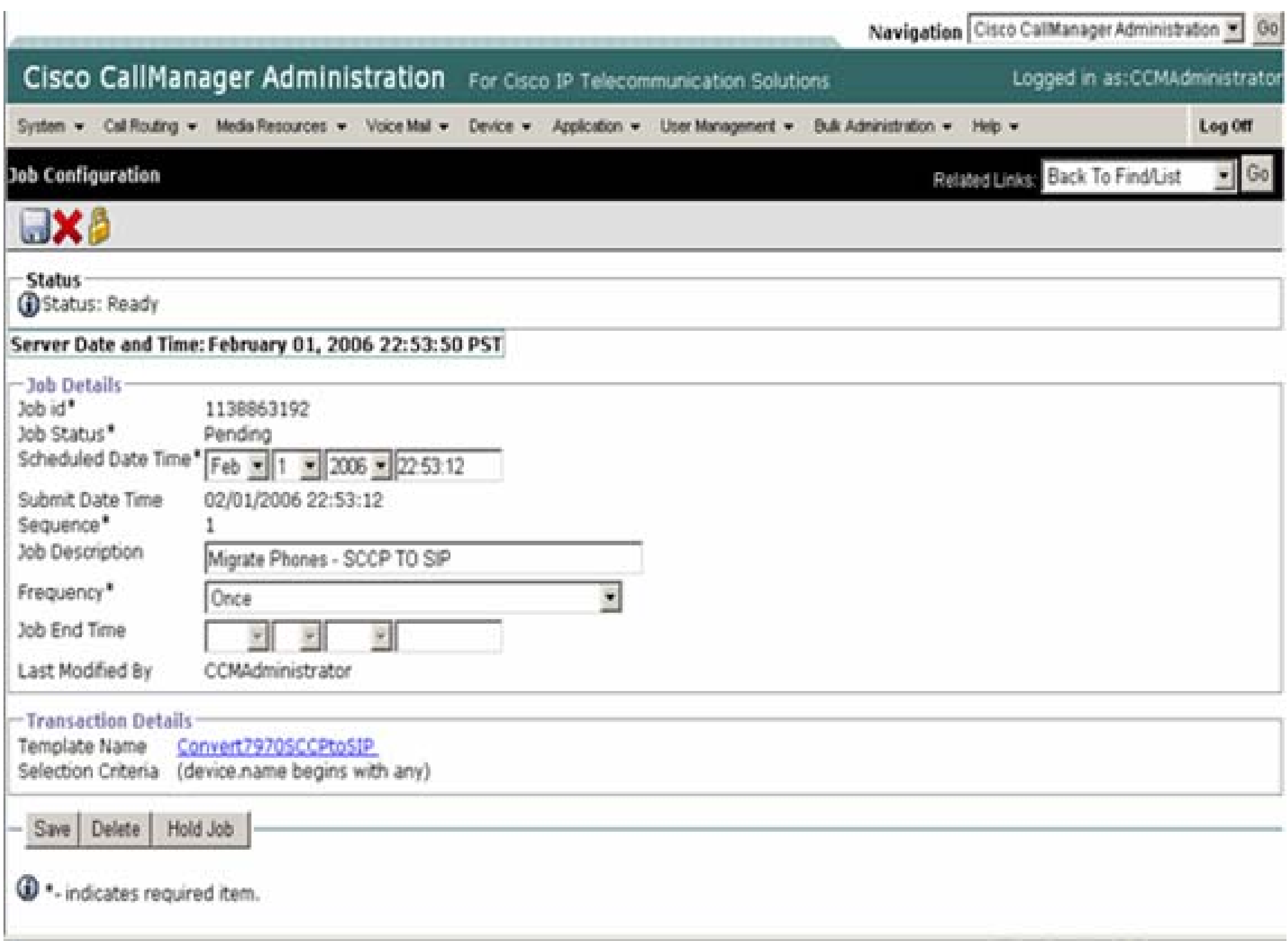
It is important to note that upgrading doesn't mean having to take down your VoIP network for hours. Upgrades can be applied individually to CallManager servers in a cluster, and these can then be brought back on line one at a time.

Linux is widely regarded as generally more secure, and often better performing, than Windows as an IP-telephony call-control platform. Cisco says only that its new Linux version simply offers users the choice.

Practically all of Cisco's current IP phones can run SIP – it is an alternative phone firmware download. The older Cisco phones support a basic SIP feature set. Due to the smooth and seamless interworking of Cisco's SCCP and SIP phones, it may not make much sense to convert existing, older SCCP phones over to SIP, since you'd suffer a net loss of features. Still, the latest Web-based Cisco CallManager Admin interface includes tools for upgrading SCCP phones to SIP. The below screen shot shows one step in this process.

In terms of features, we believe Cisco's latest SIP implementation delivers to users of Cisco's new SIP phones over 90 percent of the features that SCCP-phone users get.

Users get optimal performance and maximum SIP functionality by running SIP on Cisco's latest IP hard phones, which the vendor developed with its forthcoming SIP and CallManager 5.0 environment in mind.



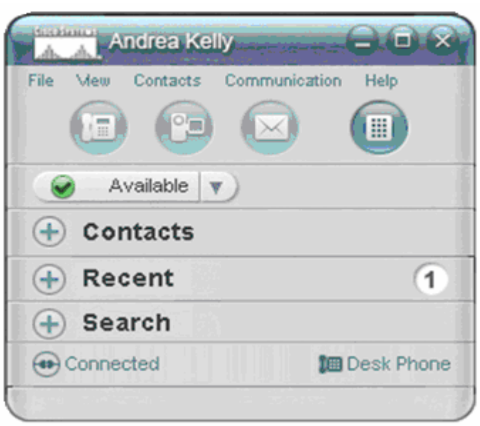
Upgrading phones from SCCP to SIP is addressed with utilities in the latest CallManager Admin interface. The administrator specifies when a phone is upgraded with the appropriate version of SIP firmware. Each phone model has its own SIP firmware version, which this utility determines and applies automatically.

Because SIP involves heavy endpoint processing (compared to most proprietary protocols, where much of the feature processing is done by the call controller), Cisco had to come up with a beefed up line of phones for SIP. In addition, these phones have enhanced graphical displays, support Unicode, and support IEEE inline power.

Cisco's new IP-phone series includes the 7971, 7961, 7941 and 7911 (see summary of the phone models we reviewed in the table on Page 2). These cost a little more than their pre-SIP predecessors, but they offer enhancements including higher-resolution displays, in addition to greater processing power for future features, including SIP. These same new phones run SCCP, too.

SIP-based, SIP-enabled Applications

Cisco also demonstrated the Cisco Resource Reservation Protocol (RSVP) Agent for Call Admission Control, a new feature of Cisco Unified Call Manager 5.0. This agent "guarantees" the call path and available WAN bandwidth via Cisco routers, for voice and video calls. It works with the underlying L2/L3 network of Cisco routers. The use of RSVP is optional.



Unified Client—open and shut. One of the slickest new applications we reviewed was the Unified Client v1.0. The above view shows the small footprint of the client on the desktop. The key functions can be individually expanded, such as "Contacts" and "Recent Call Log," shown in interface to the right.

Cisco is exploiting the Unified Communications environment with new and enhanced applications that tap SIP's inherent capabilities. Popular SIP-enabled capabilities include multimedia support, including video, and "presence" – which lets people see the real-time, on-the-phone status of co-workers. And both of these are featured in Cisco's new Unified Personal Communicator.

The Unified Personal Communicator (see below) is a SIP-based client, with versions that run on PCs or Macs. In general, it lets users reach out and touch a multitude of services. There is a new, optional softphone that loads and runs on the PC or Mac (such as for road warriors). Otherwise, default values for all parameters are provided; the user only needs to customize the installation.

The software works with an associated, external phone (such as in an office or teleworker environment). Desktop video conferencing support and access to Web collaboration (via MeetingPlace) are integral.

Unified Personal Communicator maintains a small footprint on the desktop or laptop, but then expands to provide access to key functions. These include:

- Contacts, providing a quick status view of close associates, and fast click-to-dial calling
- Recent call log
- Presence-enabled directory.

Unified Personal Communicator works with the new Cisco Unified Presence Server, and automatically receives status and availability changes of close associates. The software also provides client access to various server-based resources, including LDAP directories, Unity voice mail, and MeetingPlace collaboration. A forthcoming release of Unified Personal Communicator will include personal call routing and Instant Message.

Miercom reviewers liked the fact that Unified Personal Communicator can be used to access key Cisco server-based applications – MeetingPlace, Unity, and so on – but it is also very useful for individual call handling, requiring just the SIP support of CallManager 5.0.

As the table on Page 2 shows, various other Cisco application packages have been developed or enhanced to exploit the new Cisco Unified Communications System. For example, SIP trunking is now supported for efficient call handling between CallManager 5.0 and MeetingPlace, Unity and Contact Center environments.

A host of other useful applications were reviewed, all working in the new environment:

- **Cisco Unified MeetingPlace Express,** a new Linux-based package, which provides audio conferencing and Web-based publishing. SIP and H.323 are supported.

- **Cisco Unified Presence Server**, which handles the processing and distribution of presence status and change notifications from a wide variety of SIP applications. CallManager can maintain presence information to some extent, but a separate Cisco Unified Presence Server off-loads the CallManager and better handles large networks and scalability. Cisco has also implemented a neat messaging utility, called IP Phone Messenger, which also runs on the presence server. It lets users send and receive messages via the displays of the IP phones.

- **Cisco Unity Connection**, which provides voicemail and unified messaging for up to 1,500 users. The package now supports SIP trunking with PBXs, and also features voice recognition. We reviewed this running on a remote server and exercised the voice recognition for directory access and dialing. This package is also accessible from Unified Personal Communicator, via the IMAP email protocol, or via a Web interface.

- **Cisco Unified Contact Center Express**, which provides a full call-center environment for up to 300 agents on a single server. IPCC Express can now control SIP phones, too. Third-party call control – used to support supervisor functions – is implemented via a JTAPI interface to CallManager.

- **Emergency Responder** v1.3, which tracks user locations for 911 emergency calls, now also supports Cisco and third-party SIP phones.

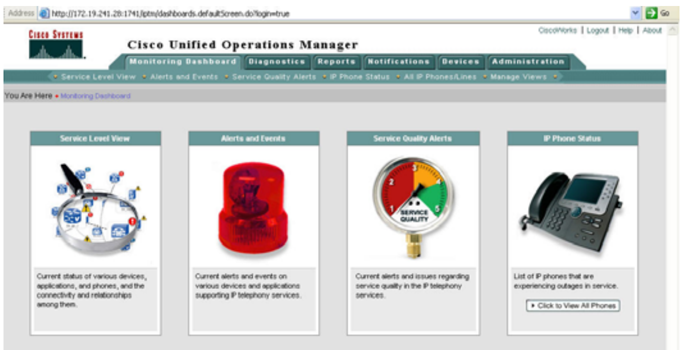
- **Mobility Manager Server,** a Linux-based server package that supports applications including Mobile Connect. With this, up to four phone devices – on or off the CallManager system – can be simultaneously rung for an incoming call. This package also now supports Cisco and third-party SIP phones.

Managing the SIP Environment

The Miercom test team also reviewed the latest versions of two key Cisco management applications, also designed to support the SIP and mixed-protocol environment:

- Cisco Unified Operation Manager v1.1
- Cisco Unified Service Monitor v1.0

The Cisco Unified Operations Manager (see launch window below) auto-discovers the complete network environment and generates hierarchical topology diagrams, in which nodes are continually monitored.

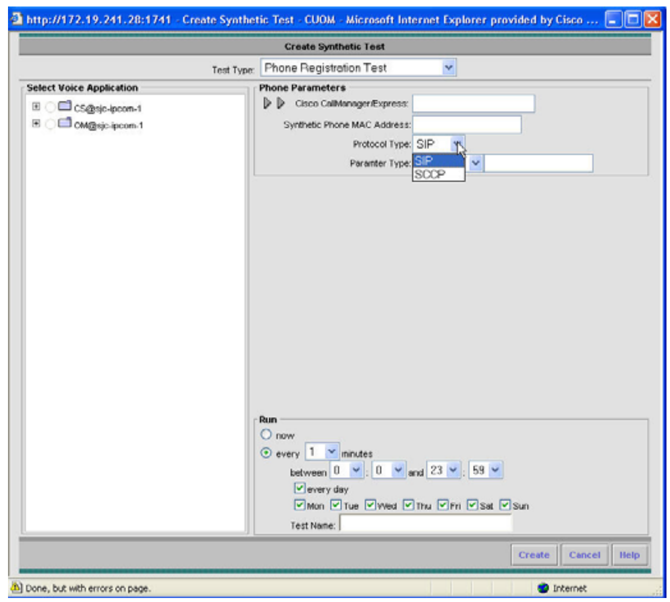


We observed that Operations Manager could also automatically identify and include in its phone listings third-party SIP phones, as well as Cisco SIP and SCCP endpoints.

Third-party SIP phones, too. The detailed breakdown of phone endpoints (shown above) is compiled automatically and shows good detail of Cisco phones, as well as third-party SIP phones. Several protocols are run under the covers to identify devices – CDP, SNMP, Ping.

A valuable aspect of the Operations Manager, we believe, is the hierarchical topology maps. In the top-level network diagram are shown "core voice components," including all servers and gateways. This is an excellent tool for quickly learning the status of key infrastructure nodes. The user can then "drill down" into successive network diagrams to get the latest details on any particular endpoint.

Another noteworthy feature is the ability to launch simulated test traffic for diagnostics and to check if devices, from servers to SIP phones, are functioning properly. The screenshot below shows the set-up of a "synthetic" traffic test.



A second key management application, Service Monitor, can run with Operations Manager or separately. This package runs in conjunction with "probes" (such as the Cisco 1040), which are inserted usually on spanned switch ports, watching WAN and backbone links. The application focuses on watching up to 80 concurrent calls or RTP streams per probe.

Based on its monitoring, the application can issue alerts on any gateway or IP phone. As with Operations Manager, Service Monitor can also track Cisco SCCP or SIP, as well as third-party SIP phones.

NetWORKS as Advertised

Based on Miercom's review of the Cisco Unified Communications System, and the operation, capabilities and features of the key components that were exercised, Miercom is proud to award its "***NetWORKS as Advertised***" certification and attest to these findings:

- The system natively integrates SIP alongside Cisco's proprietary "SCCP"
- Testing confirms user-transparent, full-featured, SIP-SCCP phone interoperability
- "Presence" support has been added throughout, for both SIP *and* SCCP phone users, as well as presence-enabled applications
- Cisco now offers hot new SIP-based applications to exploit the new environment. These include: Cisco Unified Personal Communicator, providing PC and Mac users with desktop video, Web collaboration, and access to voicemail, MeetingPlace, and presence-enabled directory.

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