# Cisco Unified Service Monitor 2.3 Deployment Best Practices

## Deployment Guide

# Contents

## Introduction

This white paper provides insight into voice quality and the different methods to measure voice quality.

Voice over IP (VoIP) has passed its infancy stage and is a mature technology that has been widely adopted by customers hoping to take advantage of the cost savings offered by VoIP in addition to a range of advanced features that improve efficiency. Voice quality is the qualitative and quantitative measure of sound and conversation quality on an IP phone call. Voice quality measurement describes and evaluates the clarity of voice conversation.

The shift from the traditional time-division multiplexing (TDM) world to a packet-based IP telephony solution poses challenges for voice quality. Unlike data, which is bursty in nature and tolerant to delay and packet loss, voice and video are extremely sensitive to jitter, packet loss, and delay. In a converged network with voice, video, and data residing on the same network, there is a huge demand for the network infrastructure to be reliable and scalable and to offer different levels of service for advanced technologies such as voice, video, wireless, and data.

### Voice Impairment Parameters

The real-time nature of voice drives strict service-level agreements (SLAs) to be implemented in the network. The primary voice impairment parameters are jitter, packet loss, and delay.

#### Packet Loss

In data networks, even if a few packets are lost during transmission, TCP ensures the retransmission and assembly of the packets, and the user will not notice any difference. But in the transmission of voice packets across the IP backbone, the missing packets cause distortion in voice quality on the receiving end, and retransmission of missing voice packets is useless. It is tolerable to have occasional packet loss, but consecutive loss of voice packets can affect the overall quality of the transmitted voice.

#### Jitter

Delay variation, or jitter, occurs when voice packets arrive at the destination at different time intervals. This can happen because of the connectionless nature of IP. Depending on the congestion and load on the network, the arrival rate of these packets at the destination may vary. The devices on the receiving end should be capable of buffering these packets and playing them back to the user at a consistent interframe interval. These types of devices are called dejitter buffers. A dejitter buffer usually adds a forced delay (default 60 milliseconds [ms]) to every VoIP packet received. Typically, this delay is in the 20 to 60 ms range. This delay is commonly called the play out delay.

#### Delay

Delay is the finite amount of time it takes a packet to reach the receiving endpoint after being transmitted from the sending endpoint. In the case of voice, this is the amount of time it takes for a sound to travel from the speaker’s mouth to the listener’s ear. Delay (or latency) does not affect voice fidelity. Extended network delay is perceived as echo in the conversation. Even though network delay is not a direct cause of echo, it does amplify the perception of any echo present in the media path. Extremely long delays can lead to "collisions" in the conversation, when both parties seem to be speaking simultaneously.

The network infrastructure must meet the following requirements:

- Packet loss must not be more than 1 percent.
- Average one-way jitter must not be more than 30 ms.
- One-way delay must be under 150 ms.

To help ensure good voice quality, it is imperative to keep jitter and packet loss under control by paying close attention to voice impairment factors.

### Requirements of a Voice Quality Measurement Product

By now, you have a basic understanding of the voice impairment parameters and their importance in voice quality measurement. Some of the key requirements for a voice quality measurement product follow. It must:

- Be able to calculate voice quality for actual calls. Although it is possible to generate synthetic voice traffic and calculate voice quality for these calls, the voice quality generated from synthetic calls does not represent end-user experience. The capability to simulate voice traffic must be made use of to verify voice quality when a real-time alert is received.
- Provide details of voice impairment parameters such as jitter, packet loss, and so on. It is important to understand the cause of voice quality degradation; for example, knowing whether jitter or packet loss is causing the problem will help in fine-tuning the network, if required.
- Report voice quality in real time. For example, in a call that lasts for 5 minutes, if the user encounters voice quality problems in the second minute, it is important for the administrator to get an alert at the second minute rather than at the end of the conversation.
- Provide details about the endpoints involved in a conversation, the type of codec used, and the IP addresses and phone numbers of the endpoints. Such detailed information is important in troubleshooting a voice quality problem.
- Be scalable and easily deployable.

### Components of Cisco Unified Communications

The Cisco® Unified Communications Family of products can be divided into two parts:

- Cisco Unified Communications infrastructure
- Application infrastructure

The Cisco Unified Communications infrastructure consists primarily of call-processing devices such as Cisco Unified Communications Manager, Cisco Unified Communications Manager Express, Cisco Unity® software, and Cisco Unity Express. The Cisco Unified Communications infrastructure layer is the brain of the Cisco Unified Communications Family of products, and it performs the critical functions of call setup, call routing, and call tear-down.

The application infrastructure consists of software applications such as Cisco Emergency Responder, MeetingPlace®, MeetingPlace Express, IP Contact Center, IP Contact Center Express, and Personal Assistant. These applications provide added functionality to the Cisco Unified Communications Family of products.

## Cisco Unified Service Monitor

Given the dynamic nature of IP networks and the strong dependency of the Cisco Unified Communications solution on network infrastructure, it is imperative for network administrators to have voice quality information on real calls (not simulated calls) at their fingertips to help enable them to resolve voice quality problems. Cisco Unified Service Monitor meets user requirements in reporting voice quality issues.

There are two ways to measure call quality using Service Monitor:

- **Real-time call quality measurement and reporting progresses:** Use the Cisco 1040 or Network Appliance Module (NAM) with version 4.x or later hardware.
- **Call quality measurement and reporting at the end of the call:** Use the Cisco Unified Communications Manager cluster (4.2.x and 5.x/6.x/7.x provide Cisco Voice Transmission Quality - based mean opinion scores [MOSs]; for earlier call manager versions, Service Monitor reports jitter and packet loss for the call).

Service Monitor analyzes the data that it received from Cisco 1040/NAM or Cisco Unified Communications Manager and sends Simple Network Management Protocol (SNMP) traps when a MOS falls below a threshold. Service Monitor provides a set of default global thresholds, one per supported codec. Service Monitor also allows users to change the default global thresholds and to override them by creating Cisco 1040 threshold groups and cluster threshold groups.

### Data Collection and Analysis

Service Monitor receives and analyzes MOSs from the following sources when they are installed and configured properly in your voice network:

- **Cisco 1040s:** Cisco 1040s compute MOSs for each Real-Time Transport Protocol (RTP) stream and send syslog messages to Service Monitor every 60 seconds.
- **Cisco Network Analysis Module 4.0 or later:** NAM computes MOSs for each RTP stream, and Service Monitor pulls the data from each NAM. This integration requires Service Monitor to have user credentials with collect view permissions on the NAM.
- **Cisco Voice Transmission Quality:** Cisco Unified Communications Manager collects data from Cisco MGCP Voice Gateways and Cisco IP phones; MOSs are calculated on the Media Gateway Control Protocol (MGCP) gateways and Cisco IP phones using the Cisco Voice Transmission Quality algorithm. At the termination of a call, Cisco Unified Communications Manager stores the data in call management records (CMRs).

### Call Classification and Call Detail Record - Based Reports

The following are new features in the Cisco Unified Service Monitor 2.3 release:

- **Call Classification:** Cisco Unified Service Monitor 2.3 now classifies various call types based on Cisco Unified Communications Manager configuration and call detail records (CDRs). The Cisco Unified Service Monitor Call Classification feature includes the ability to classify calls into the categories listed in Table 1.

**Table 1.** Categories for the Cisco Unified Service Monitor Call Classification Feature

| System Categories | User-Defined Categories |
|---|---|
| **Unity Voicemail**<br>**Conference Bridge**<br>**VG/Trunk Outgoing**<br>• MGCP Gateway Outgoing<br>• H.323 Gateway Outgoing<br>• H.323 Trunk Outgoing<br>• SIP Trunk Outgoing<br>**VG/Trunk Incoming**<br>• MGCP Gateway Incoming<br>• H.323 Gateway Incoming<br>• H.323 Trunk Incoming<br>• SIP Trunk Incoming<br>**Tandem**<br>**Intercluster Trunk (ICT)**<br>• ICT GateKeeper Controlled<br>• ICT Non-GateKeeper Controlled<br>**OnNet Trunk**<br>• OnNet H.323 Trunk<br>• OnNet SIP Trunk<br>**Internal** | **System Defaults:**<br>Local<br>Long Distance<br>International<br>Emergency<br>Service<br>Toll Free<br><br>**User-Created:**<br>Any custom-created category, such as:<br>• Conference calls to Web Ex<br>• Calls to Legacy Voice mail<br>• Long Distance to East Coast |

Cisco Unified Service Monitor supports two category types: System categories and user categories. Service Monitor can classify calls under system categories without any user input. All calls are classified based on Cisco Unified Communications Manager device configuration (for example device type, OnNet/OffNet configuration).

Apart from classifying calls under these categories, Service Monitor also classifies calls as OffNet or OnNet. For a call to be classified as OffNet, one of the endpoints has to be a gateway or trunk and the:

- Gateway or trunk is configured as OffNet in Cisco Unified Communications Manager
- Gateway or trunk is configured to use the system default (the system default is OffNet, configurable from Cisco Unified Communications Manager > Service Parameters)
- Gateway is analog

- **Call Detail Record - Based Reports:** Cisco Unified Service Monitor 2.3 performs call classification based on Cisco Unified Communications Manager call detail records. it has the ability to provide on-demand call reports based on various call types and to filter calls by call category, device type, and the success or failure of calls where call termination cause codes are determined and grouped.

As from previous Service Monitor releases, data from these new call classification and call detail record features will be retrieved and stored by Cisco Unified Service Statistics Manager for long-term reporting and trend analysis that displays call types, for example call volume reports and call analysis reports.

**Note:** The call classification user interface and analytics had been moved from Service Statistics Manager's logic to Service Monitor's.

For a more detailed guide on these new Service Monitor features, refer to the Service Monitor Call Classification White Paper at http://www.cisco.com/en/US/products/ps6536/prod_white_papers_list.html.

### Service Monitor Server Requirements

The Cisco Unified Service Monitor 2.3 software runs on an Intel-based machine, with a server running Windows 2003 Server (Standard/Enterprise) with Service Pack 1 or 2. The software license must be procured by the customers.

For minimum system requirements, please refer to http://www.cisco.com/en/US/prod/collateral/netmgtsw/ps6491/ps6705/ps6536/data_sheet_c78-602379.html.

### Cisco 1040 or NAM-Based Call Quality: R-Factor

The ITU E model, as defined in G.107 (03/2003), predicts the subjective quality that is experienced by an average listener by combining the impairment caused by transmission parameters (such as loss and delay) into a single rating: the transmission rating factor R (the R-factor). This rating, expressed on a scale of 0 (worst) to 100 (best), can be used to predict subjective user reactions, such as the MOS. The MOS can be obtained from the R-factor with a converting formula. Thus the R value is an estimate of the quality that can be expected if the network is realized the way it is planned.

#### Cisco 1040/NAM Deployment Locations

The Cisco 1040 is the hardware component that will be deployed on the Switched Port Analyzer (SPAN) port of an access switch, as close to the IP phones and other problem areas (gateways, and so on) as possible. NAM could be a hardware module or an appliance. The module will be deployed on an integrated services router or Cisco 65xx/76xx. The number of concurrent RTP streams supported by Cisco 1040 and NAM are different. This plays a critical role in defining the deployment location of Cisco 1040 or NAM components. Table 2 outlines the number of concurrent RTP streams monitored by various components.

**Table 2.** Number of Concurrent RTP Streams Measured

| Cisco 1040 Sensor | NME-NAM | NAM-2 | NAM 2204 | NAM 2220 |
|---|---|---|---|---|
| 100 RTP streams/minute | 100 RTP streams/minute | 400 RTP streams/minute | 1500 RTP streams/minute | 4000 RTP streams/minute |

### Cisco 1040 Hardware

The Cisco 1040 sensor has two Fast Ethernet interfaces:

- Management port
- SPAN port

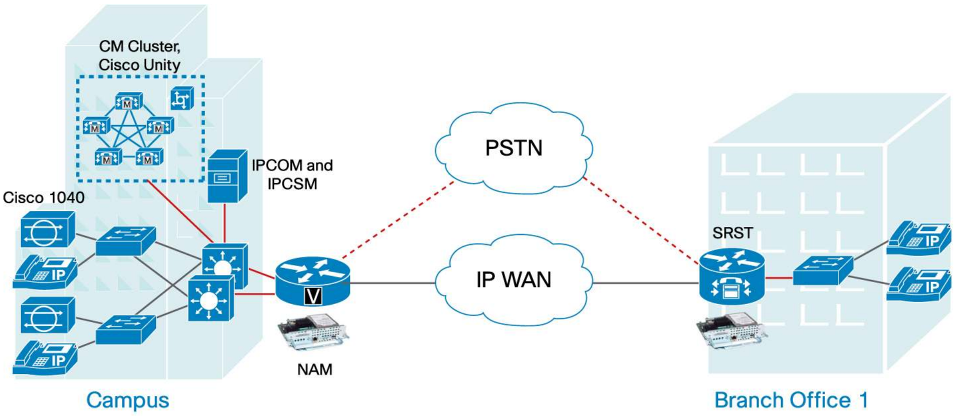
The Cisco 1040 uses IEEE 802.3af standard Power over Ethernet (PoE) from the switch to which it connects. When the Cisco 1040 boots up, it uses Dynamic Host Configuration Protocol (DHCP) option 150 to retrieve its configuration and image files on a Trivial File Transfer Protocol (TFTP) server. Similar to the way that an IP phone registers with Cisco Unified Communications Manager, a Cisco 1040 registers (using Skinny Client Control Protocol [SCCP]) to the Service Monitor application. On the TFTP server, the Cisco 1040 first looks for its configuration file, named QOV[Cisco 1040 MAC address].cnf. If that file does not exist, the Cisco 1040 looks for a file named QOVDefault.cnf. These CNF files provide the image (SvcMonAB2_102.img) filename for the Cisco 1040 to download in addition to the Service Monitor IP addresses. The Cisco 1040 then downloads this image and registers to the Service Monitor, just like a phone registers to Cisco Unified Communications Manager, using SCCP.

Then the Cisco 1040 utilizes the SPAN port on a switch to monitor the actual voice calls. It collects voice impairment parameters such as jitter and packet loss from the RTP stream and computes MOS values. The Cisco 1040 works in passive mode to collect voice impairment statistics.

The Cisco 1040 reports voice quality details and MOS values every 60 seconds, providing near real-time voice quality measurement. Each Cisco 1040 can monitor 100 RTP streams with optimal SPAN port configuration. NAM can monitor 100 to 5000 RTP streams based on the type of NAM or appliance.

As shown in Figure 1, multiple Cisco 1040s/NAMs can be deployed in the network and configured to register to the Service Monitor software component. Each instance of a Service Monitor software component can report call quality for 45,000 IP phones and can support up to a maximum of 5000 RTP streams/minute.

**Figure 1.** Multiple Cisco 1040s Register to Cisco Unified Service Monitor



**Note:** In Figure 1, the Cisco Unified Service Monitor and Cisco Unified Operations Manager software instances coreside on a single machine. This type of deployment is supported for medium-size networks (up to 10,000 phones). For networks with more than 10,000 phones, the Cisco Unified Service Monitor and Cisco Unified Operations Manager software should be deployed to run on separate machines.

Strategic Versus Tactical

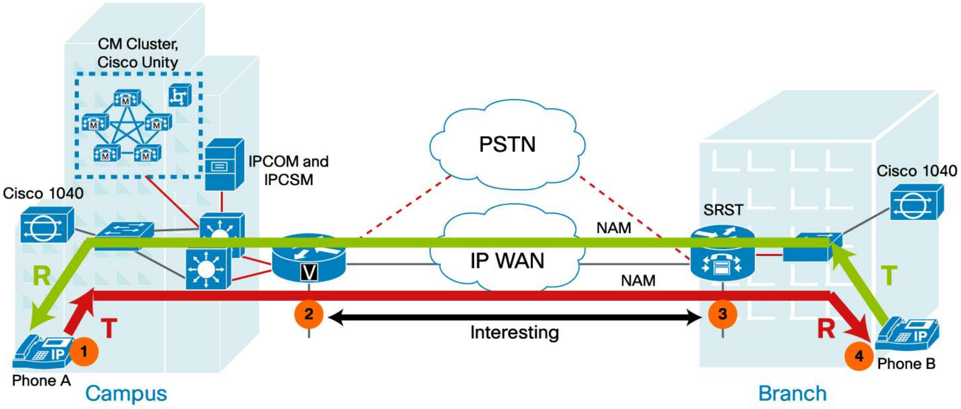
Service Monitor satisfies most quality monitoring requirements for enterprise IP telephony. Deployment strategies include:

- **Strategic monitoring:** The Cisco 1040 and NAM are installed to continuously monitor RTP streams to the IP phones at some or all locations in the managed environment. Depending on monitoring goals, significant coverage of all or most sites could be included, or by using sampling techniques, representative sites would be selected for monitoring and would determine the location of the Cisco 1040s. Service Monitor can scale to support up to 5000 RTP streams per minute total from multiple Cisco 1040s/NAMs and can provide real-time alerting on call quality.
- **Tactical monitoring:** Cisco 1040s/NAM can be inexpensively shipped overnight and installed at the site (such as a branch office) that has voice quality concerns or problems. Once it is installed and configured, it can immediately begin monitoring and assessing the quality of IP-based calls without complex setup or installation. The Cisco 1040 is FCC Class B compliant and can easily be installed in any office environment.

Figure 2 shows centralized Cisco Unified Communications Manager deployment with one remote branch office connected through a WAN circuit. To monitor voice quality for calls between headquarters and branch across the WAN circuit, two Cisco 1040s can be deployed close to the phone and two NAMs can be deployed at the edge of the network as shown. The key recommendation is to deploy the Cisco 1040s as close to the IP phone as possible and the NAMs in the network exchange points such as WAN in/out or core or distribution layers. In most cases, the Cisco 1040s will sit on the access layer switch in the campus.

Example of Deployment Scenario

**Figure 2.** Centralized Cisco Unified Communications Manager Deployment



- For each phone, there are transmit and receive RTP streams.
- For the RTP stream originating from Phone A (TX RTP stream), the segment between 1 and 2 in Figure 2 experiences the least impairment, and the probability of voice quality degrading in this segment is slim to none. The RTP stream between segments 2 and 3 traverses several network devices in the WAN and is prone to network conditions.
- The previous statement is also true for RTP streams originating from Phone B.
- For the Cisco 1040/NAM on the left in Figure 2, MOS will be calculated based on the RTP stream between Phone A/B covering headquarters LAN segments 1 and 2.
- For the NAM/NAM on the middle in Figure 2, MOS will be calculated based on the RTP stream coming between Phone A/B in the WAN covering segments 2 and 3.

- For the NAM/Cisco 1040 on the right in Figure 2, MOS will be calculated based on the RTP stream between Phone A/B in the branch LAN covering segment 3, and 4 is of importance; the switch on the left must be configured to span the incoming RTP stream, to span the destination port seen by the left Cisco 1040.

Service Monitor collects and correlates the collected MOS data from different segments, which will clearly identify the segment with poor call quality during the voice call. With optimal SPAN port configuration, each Cisco 1040 can monitor up to 100 RTP streams.

### Deciding on the Number of Cisco 1040s and NAMs for Call Quality Monitoring

The number of Cisco 1040s/NAMs required depends on the busy hour call completion (BHCC) handled by the switch. For a 10,000-phone network, for example, the cluster could handle 1000 to 4000 calls simultaneously. As the size of the network increases, it becomes more appropriate to take samples of the calls generated from the cluster and measure voice quality for a subset of these calls. On the other hand, if the network consists of only about 1000 phones, it is easier to monitor voice quality on all of the calls; however, the same sampling technique can also be applied to a 1000-phone network.

The Unified Communications deployment follows one of the following call processing models:

- Single site with centralized call processing
- Multiple-site WAN with centralized call processing
- Multiple-site WAN with distributed call processing

For a single site with centralized call processing, most often a Catalyst® 6500 is used in the access layer/wiring closet to connect the IP phones. It is common to expect about 200 IP phones (4 blades with 48 ports = approximately 200) on a single Catalyst 6500. If 4 out of 10 phones are active at any point, a NAM line card can be placed on the Catalyst 6500 to monitor the active calls. It is also possible to deploy multiple Cisco 1040s on the same switch to address situations in which the switch is handling high call volume.

It is not necessary to measure voice quality for every call. The general practice is to measure voice quality for a subset of the calls on a continuous basis and use a tactical approach for troubleshooting voice quality problems.

Based on this analysis, for a 1000-phone deployment, if you were to sample 30 percent of the active calls and measure voice quality on a continuous basis, you would need three Cisco 1040s (30 percent of 1000 is 300, and each Cisco 1040 can monitor 100 RTP streams; hence, you would need three Cisco 1040s). Usually, the Cisco 1040s are deployed in pairs (one each at the origination and termination endpoints).

This sampling could be quite aggressive for most common deployments. Based on the simultaneous calls that are active on a switch, the number of Cisco 1040s required for voice quality measurement varies. As the network size increases, the sampling policy can be reduced.

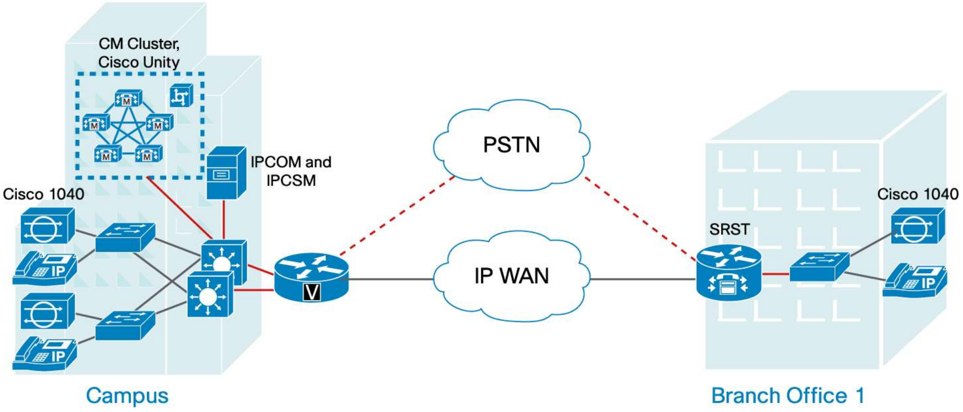
The previous scenario had a dense population of phones on a single switch; another scenario could have phones scattered among multiple smaller switches. Typically, the number of phones in a branch is less, and you will see smaller-density switches used in the branch. For this scenario, it is excessive to have one Cisco 1040 per switch, especially if the number of RTP streams on these switches is small (20 - 40 RTP streams). To address this issue, you can use Remote SPAN (RSPAN) and combine RTP streams on multiple switches. Another alternative is to use an active hub and connect the SPAN destination port from multiple switches to the same active hub. With this alternative, there is potential for Layer 2 loops, and you must evaluate the best option before embarking on any approach.

In summary, Service Monitor can monitor actual voice calls in real time and provide details of the parameters that cause voice quality degradation. The combination of Service Monitor and Operations Manager provides a powerful product for monitoring and troubleshooting voice quality problems.

### Placement of Cisco 1040s

The Cisco 1040 is FCC Class B certified and can be deployed in a wiring closet or on a desk. The Cisco 1040 uses the SPAN port on the switch to monitor the RTP stream. As shown in Figure 3, the Cisco 1040s are deployed as close to the IP phone as possible so that the voice quality measurement will be close to what the user experiences. The Cisco 1040 reports voice quality measurements every 60 seconds. For each conversation, there are four RTP streams: two each from originating and terminating phones. In the default SPAN port configuration, Service Monitor receives four MOS values every 5 seconds for each conversation. As discussed in the previous section, of the four RTP streams, two provide meaningful statistics; hence, the MOS value calculated for the interesting RTP stream is of importance and should be considered for further analysis.

**Figure 3.** Cisco 1040 Placements



The number of Cisco 1040s per switch depends on the following:

- Type of switch
- Type of customer
- Number of simultaneous calls

The type of switch determines the number of SPAN destination ports that can be configured on the switch. Modular switches such as the 65xx support two SPAN destination ports with different source ports. Most of the fixed configuration switches support a single destination port. On the modular configuration switches, you can have two sensors deployed on the same switch, and on the fixed configuration switch, you can have a single Cisco 1040 deployed.

A typical enterprise customer has various organizations, such as engineering, human resources, marketing, sales, and support. The traffic patterns are different in these organizations. You can expect more calls for support, sales, and marketing groups, and the switches that house these users will have a higher busy hour call attempt (BHCA) value. In a call center environment, you can expect high call volume and, therefore, high BHCA, which will be the deciding factor for the number of Cisco 1040s required.

The number of simultaneous calls is tied to the previous argument about the BHCA value. It is important, therefore, to understand the BHCA value and average call hold (ACH) time to determine the number of Cisco 1040s required. Keep in mind the limit on the number of RTP streams supported by each sensor (100) and the number of simultaneous calls generated by the phones connected to the switch where the Cisco 1040 is deployed.

You could have a situation in which the number of RTP streams exceeds 100, in which case you can add Cisco 1040s or configure the SPAN source port in such a way that you monitor only selected phones on the switch. If the SPAN destination port sees more than 100 RTP streams, the Cisco 1040 goes into sampling mode, and the MOS value reported is diluted.

### Bandwidth Used by Cisco 1040

Cisco 1040 bandwidth requirements are very little. Each syslog message takes up around 60 bytes per stream. So a total of 6000 bytes will be used up reporting call quality for the 100 streams each minute.

### Cisco 1040 in Sampling Mode

A Cisco 1040 can monitor 100 RTP streams. If it is deployed on a switch that has more than 100 RTP streams, it performs sampling, in which case some of the RTP streams are not considered for MOS value generation. You must avoid this situation at all times. In sampling mode, the MOS value reported is diluted because some of the RTP streams are not considered. The Cisco 1040 monitors the RTP stream and collects the information necessary to compute the MOS value. This information is stored in a buffer, from where the computation process picks data to compute the MOS value. If the packets arrive at a faster rate than the rate at which the buffer is emptied, part of the RTP stream is dropped before the Cisco 1040 starts the collection process. Keep in mind that CPU resources are constantly utilized; hence, it is not just the buffer that becomes a bottleneck when the Cisco 1040 is overwhelmed with more RTP streams: the CPU also falls short in serving the different processes. The MOS value reported by the Cisco 1040 gets worse as the number of simultaneous RTP streams increases beyond 100. It is important to plan ahead and optimize the SPAN port configuration in these scenarios.

### Cisco 1040s in the Branch

In a branch office, the density of IP phones is less when compared with the density seen in a campus. Typically, the branch office contains fixed-configuration switches, and the number of simultaneous calls is lower.

In a fairly large branch, it is common to see multiple fixed-configuration switches stacked to provide more density and avoid the need to run gigabit uplink to aggregate switches and routers. The Cisco 1040 fits into this model the same as with any other switch. It utilizes the SPAN port to monitor the RTP stream.

In a scenario in which the switches are not stacked but have gigabit uplink to an aggregate switch, if the number of RTP streams is below 100, then one Cisco 1040 per switch is excessive. In this situation, RSPAN is useful. The configuration done on the switch with SPAN, RSPAN, or Enhanced SPAN (ESPAN) is transparent to the Cisco 1040; the Cisco 1040 functions normally as long as it sees the RTP stream.

For cases in which RSPAN is not a desirable configuration or not an approved configuration, a simple active hub can be used to connect the individual SPAN port from the various switches, and the Cisco 1040 can be deployed on the hub. It is very important to keep spanning tree loops in mind when such a configuration is attempted. The use of a hub must be selected as a last resort.

### SPAN Port Limitation

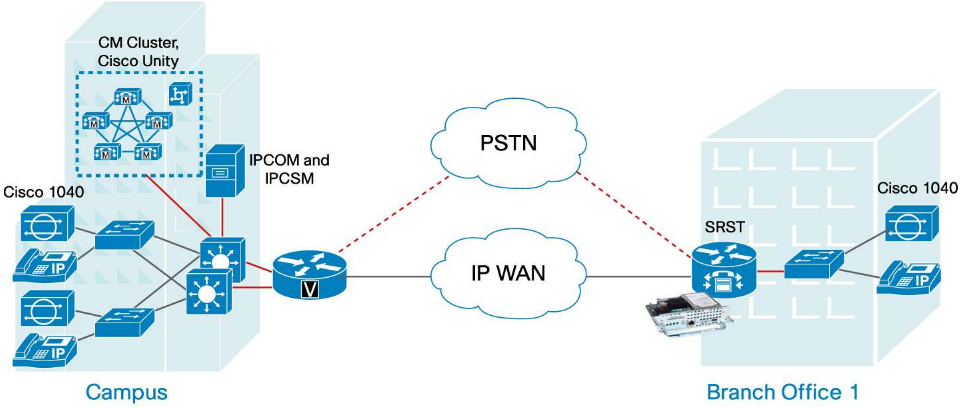
SPAN ports are widely used to connect packet sniffers for troubleshooting. In the contact center world, the SPAN port is used to record the voice conversation. In the service monitor world, the SPAN port is used to monitor voice quality. If the SPAN port is required for a packet sniffer, contact center, and Service Monitor at the same time, the SPAN port does not allow configuration of the same source port tied to multiple SPAN destination ports in fixed configuration switches. This is a limitation of SPAN port configuration on fixed configuration switches. The only alternative is to use an active splitter that offers one-to-many streams. The simplest splitter must be an active hub that offers a one-to-many stream. In this model, the packet sniffer, contact center application, and Cisco 1040 connect to the hub, and the hub connects to the SPAN destination port on the switch.

It is also recommended that all fax devices be aggregated in one voice gateway and that the 1040s not span the switch to which the gateway is connected. Currently the 1040 is known not to correctly handle certain cases of calls going to fax machines.

Placement of Inline NAM and NAM Appliance

The Cisco NAM can be deployed in a wiring closet/core or edge of the network (WAN entry/exit). The NAM uses the Cisco 1040 code to monitor the RTP streams, and it provides higher scalability to monitor up to 4000 streams per second. As shown in Figure 4, the NAMs are deployed as close to the WAN entry and exit points as possible so that the voice quality measurement will provide the segment if the voice quality is dropping the WAN. The Cisco 1040 reports voice quality measurements every 3 seconds and MOS every 60 seconds. As discussed in the previous section, of the four RTP streams, two provide meaningful statistics; hence, the MOS value calculated for the interesting RTP stream is of importance and should be considered for further analysis.

**Figure 4.** Cisco NAM Placements



The number of Cisco 1040s per switch depends on the following:

- Type of router/switch
- Type of customer
- Number of simultaneous calls

The type of switch/router determines the NAM or line card support. Modular switches such as the 6500/7000 series support the NAM card. Most of the fixed configuration switches won’t support NAMs or NAM line cards. Integrated services routers such as the 28xx/38xx series support NAM. Different configurations of NAM support different numbers of concurrent RTP streams. Refer to Table 2 for concurrent RTP stream support.

### Cisco Voice Transmission Quality - Based Call Quality: K-Factor

K-factor (Klirrfaktor) is a mean opinion score (MOS) estimator of the endpoint type defined in ITU standard P.564. This standard relates to the testing and performance requirements of such a device. K-factor predates the standard. A P.564-compliant version will follow. K-factor is trained using thousands of speech samples and impairment scenarios, along with target P.862.1 MOS values for each scenario. The trained K-factor device in the IP phone or gateway can then recognize the current impairment and produce a running MOS value prediction.

R-factor (see the section “Cisco 1040 or NAM-Based Call Quality: R-Factor”) is based on three dimensions: loss, delay, and echo. K-factor and other P.564 MOS estimators measure only packet loss, which is a network effect. They are packet loss metrics projected onto a psychological scale. In general, primary statistics (packet loss, jitter, and concealment ratio) show visible degradation well before MOSs start to degrade. Hence, MOSs are a secondary indication of network problems, because MOS is essentially a packet loss metric. Packet loss counts, jitter, concealment ratio, and concealment second counters are primary statistics, based on direct observation. MOSs are a secondary statistic. Hence, you should use MOSs as a flag, but then use primary statistics to investigate or qualify the alarm. Use primary metrics in SLAs rather than MOSs.

Cisco Voice Transmission Quality - based call quality reports can be obtained using Service Monitor in conjunction with Cisco Unified Communications Manager and the latest Cisco Unified IP Phones and Cisco Gateways.

When to Use Cisco Voice Transmission Quality - Based Call Quality Reporting

Some key points to keep in mind when choosing to go with Cisco Voice Transmission Quality - based call quality reporting are:

- Cisco Voice Transmission Quality is supported from Cisco Unified Communications Manager 4.2 or later versions.
- Cisco 7906, 7911, 7931, 7921, 7962-G, 7962-G/GE, 7942-G, 7942-G/GE, 7972-G/GE, 7940, 7960, 7941, 7961, 7970, and 7971 IP Phones support Cisco Voice Transmission Quality in SCCP and Session Initiation Protocol (SIP) mode. (You must have new firmware; the firmware can be downloaded from Cisco Unified Communications Manager 4.2 or 5.x or 6.x or 7.x.) For an updated list of supported devices, please refer to the release notes for Cisco Unified Service Monitor 2.3 on Cisco.com at http://cisco.com/en/US/docs/net_mgmt/cisco_unified_service_monitor/2.3/release/ReleaseNotesforCiscoUnifiedServiceMonitor2_2.html.
- Sampling rate is every 8 seconds.
- Score is sent at the end of the call by using CMRs.

Unlike Cisco 1040-based call quality reporting, Cisco Voice Transmission Quality - based call quality is reported at the end of the call. You can use Cisco Voice Transmission Quality - based reporting if you prefer not to have call quality reporting as the call progresses. Also, the Cisco Voice Transmission Quality feature is inherent to Cisco Unified Communications Manager 4.2 and later. Therefore, if you do not want to invest in a Cisco 1040, Cisco Voice Transmission Quality - based call quality still provides MOSs to estimate user experience.

## Scalability

Service Monitor scalability is dependent on the estimated call rates generated in the network. The call volume supported by a fully loaded Service Monitor 2.3 system with various scenarios is given in Table 3.

**Table 3.** Call Volume Supported by a Fully Loaded Service Monitor 2.3 System

| Scenario | Cisco Voice Transmission Quality CDRs/Minute | Cisco 1040/NAM Segments/Minute |
|---|---|---|
| **Cisco Voice Transmission Quality and Cisco 1040/NAM** | 666 | 1500 |
| **Cisco Voice Transmission Quality Only** | 1600 | - |
| **Cisco 1040/NAM Only** | - | 5000 |

## Preparing the Server for Service Monitor

This section describes how to prepare your server for Service Monitor installation.

### Operating System and Server

Service Monitor is supported on Windows 2003 Server Standard/Enterprise Edition with Service Pack 1/Service Pack 2 and Windows 2003 Server R2 Enterprise Edition and Service Pack 2. No other operating systems are supported. It is recommended that software other than the operating system and antivirus software not be installed on this computer system.

Server suggestions:

- For a small network (fewer than 5000 phones), Serial ATA (SATA) disks are required.
- For a medium network (5000 to 15,000 phones), SCSI disks are required (suggestion: MCS 7845-I1 or MCS 7845-H1 comes with SCSI).

- For networks with more than 15,000 phones, Serial Attached SCSI (SAS) disks are required (suggestion: MCS 7845-I2 or MCS 7845-H2 comes with SAS).

### Hostname

It is recommended that you configure the hostname for the Service Monitor server before you start installing Service Monitor. Specify the hostname when you are installing the operating system or subsequently; select My Computer, right-click, and select Properties > Computer Name.

Once Service Monitor is installed, changing the hostname is a very laborious process involving file manipulation and the execution of scripts. The User Guide for Cisco Unified Service Monitor documents all the steps involved in changing the hostname of the Service Monitor server.

### Verify Locale Settings

Service Monitor supports only the U.S. English and Japanese locales. Using other locales means that you are running on an unsupported configuration. Further, Service Monitor may display erratic behavior, such as JRunProxyServer services not starting automatically. However, non-U.S. English keyboard layouts should work.

### Verify ODBC Driver Manager

Some components of Service Monitor require the presence of the correct version of Open Database Connectivity (ODBC) on the Service Monitor server.

To verify the ODBC Driver Manager version, do the following:

Step 1. On the Service Monitor server, select Start > Settings > Control Panel > Administrative Tools > Data Sources (ODBC).

Step 2. Click the About tab.

Step 3. Make sure that all ODBC core components have the same version number (3.5xx or later). ODBC is not available from Microsoft as a standalone installation but is packaged along with Microsoft Data Access Component (MDAC).

**Note:** If the necessary OBDC is not listed, install MDAC 2.5 or later by referring to the Microsoft website.

### Browser Version and Flash Plug-in

The recommended browser is Microsoft Internet Explorer 6.0/7.0.

When using Service Monitor, disable any software on your desktop that you use to prevent popup windows from opening. Service Monitor must be able to open multiple windows to display information.

### Network Time Protocol

The clocks on Service Monitor, call manager servers, NAMs, and Cisco 1040s must be synchronized for Service Monitor reports to include complete and up-to-date information and accurately reflect activity during a given time period. CDR/CMR stream correlation won’t work if the clock is not in sync on all of the Cisco Unified Service Monitor components (Cisco Unified Communications Manager/NAM/Cisco 1040). These notes offer a starting point and do not provide complete instructions for configuring Network Time Protocol (NTP).

To get started:

1. Talk with your call manager administrators to determine the time server with which Service Monitor should synchronize. You might find Cisco IP Telephony Clock Synchronization: Best Practices, a white paper on Cisco.com, useful; you can read it at http://cisco.com/en/US/products/sw/voicesw/ps556/prod_white_papers_list.html.

2. Use your system documentation to configure NTP on the Windows Server 2003 system where Service Monitor will be installed. Configure NTP with the time server being used by the call manager in your network. You might find How to configure an authoritative time server in Windows Server 2003 at http://support.microsoft.com/kb/816042 useful

### Connectivity

Make sure that the Service Monitor server can reach the TFTP server and the phones or devices in the IP address range where the Cisco 1040 would be deployed.

### Terminal Server Services

Remote Desktop Service and Virtual Network Computing (VNC) Services are recommended to remotely manage the Service Monitor server. VNC Services and Remote Desktop can be used to remotely install the Operations Manager and Service Monitor software.

### Antivirus and Platform Agents

You should enable virus protection on the Service Monitor server, using antivirus software. Active scanning of drives and memory should be performed during off-peak hours. Please exclude from scanning the “CSCOpx” folder. You may experience delays, and performance may be degraded, when the virus protection software is scanning all files. Service Monitor has undergone interoperability testing with the following:

- Third-party virus protection software:
  - Symantec Antivirus Corporate Edition Version 9.0
  - McAfee VirusScan Enterprise 8.0
- Platform agents:
  - (Optional) Cisco Security Agent 5.2.0

### Check Routing and Firewalls

Make sure that any firewalls between the Service Monitor server and the call manager, TFTP server, and Cisco 1040s are configured to allow management traffic through. See the “Port Availability” section below for information on which ports should be opened.

Also make sure that there is connectivity between devices and the Service Monitor server. Even if a route exists to a network behind a device, it does not mean that one exists to (and from) the device itself.

### Port Availability

Table 4 lists the ports used by Service Monitor. These ports should not be scanned.

**Table 4.** Service Monitor Port Usage

| Protocol | Port Number | Service Name |
|---|---|---|
| **TCP** | 22 | Secure Shell (SSH) Protocol |
| **User Datagram Protocol (UDP)** | 53 | Domain Name System (DNS) |
| **UDP** | 67 and 68 | DHCP |
| **UDP** | 5666 | Syslog: Service Monitor receives syslog messages from Cisco 1040. |
| **TCP** | 2000 | SCCP: Service Monitor uses SCCP to communicate with Cisco 1040s. |
| **TCP** | 43459 | Database |
| **TCP** | 5665 - 5680 | Interprocess communication between the user interface and back-end processes |

### TFTP Server

Cisco Unified Communications Manager 5.x or 4.2 can be used as the TFTP server.

If you use Cisco Unified Communications Manager as a TFTP server, Service Monitor cannot copy configuration files to Cisco Unified Communications Manager due to security settings on the latter. You will need to manually upload the configuration file. After uploading the configuration file, reset the TFTP server on Cisco Unified Communications Manager. For more information, see Cisco Unified Communications Manager documentation.

**Note:** Due to known security limitations of TFTP, it is recommended to disable the TFTP service in Service Monitor if Cisco 1040 sensors are not in use. The CiscoWorks Common Services TFTP Service could be disabled in Windows Control Panel > Services.

### Cisco Unified Communications Manager Configuration

Service Monitor can collect and analyze data from Cisco Unified Communications Manager. Follow the configuration steps described in the user guide in the “Cisco Unified Communications Manager Configuration” section or go to http://www.cisco.com/en/US/docs/net_mgmt/cisco_unified_service_monitor/2.3/user/guide/confccm.html#wpxref75795.

These are some important points that you should be aware of when you are configuring Cisco Unified Communications Manager:

- If you don’t see call records in the Cisco Voice Transmission Quality reports, make sure CDR and CMR are enabled on all call manager nodes of a cluster.
- Make sure that the cluster ID is unique in the system; System Monitor will generate an error when a duplicate cluster is entered.
- Make sure that on the call manager Enterprise Parameter Configuration page the CDR File Time Interval is set to 1 (minute). This determines how frequently the call manager generates CDRs. (Typically this is not changed.)
- Check the owner of the sm_record_create_table in the CDR database after adding the call manager to the Service Monitor server. Make sure the table owner is dbo. Caution: The call manager cannot write CDRs to the database if the sm_record_create_table owner is not dbo.
- Cisco Unified Service Monitor IP should be added to the Unified Communications Manager server 5.x and later as a billing server.

Depending on the Unified Communications Manager version that you use, you need to perform some subset of the tasks listed in the above section. Where tasks themselves differ slightly from one Unified Communications Manager version to another, version-specific steps are noted in the procedures.

Table 5 lists the configuration tasks you must complete for each version of Unified Communications Manager that you want Service Monitor to obtain Cisco Voice Transmission Quality data from.

**Table 5.** Configuration Tasks for Unified Communications Manager

| Task | Perform Task for These Unified Communications Manager Versions | |
|---|---|---|
| | 5.x and later | 4.x |
| Setting Unified Communications Manager Service Parameters | Yes | Yes |
| Setting Unified Communications Manager Enterprise Parameters | Yes | Yes |
| Adding Service Monitor to Unified Communications Manager 5.x and Later as a Billing Server | Yes | - |
| Activating the AXL Web Service on Unified Communications Manager 5.x and Later | Yes | - |

| Task | Perform Task for These Unified Communications Manager Versions | |
|---|---|---|
| **Configuring Database Authentication on Unified Communications Manager 4.x Systems** | - | Yes |
| **Configuring Voice Gateways** | Yes | Yes |

## Service Monitor Installation

This section describes the tasks that should be performed during Service Monitor installation.

### Preinstallation Checks

Perform the following checks before installing Service Monitor:

- Dual homing (dual network interface cards [NICs]), using two different IP addresses, is not supported on Service Monitor. If, during installation, you receive a warning message to edit a file named gatekeeper.cfg, then your server is dual homed, and you must disable one of the NIC interfaces before adding any devices to Service Monitor. Using two NICs with a single IP address (a failover configuration, in case one of the NICs fails) is supported.
- Make sure that you change the default call manager cluster ID setting (located at call manager Administration > Enterprise parameters). The default setting is Standalone Cluster. Unless you change this entry, all of the clusters will have the same cluster ID. This causes problems in Service Monitor. Changing the cluster ID requires a restart of RIS Collector service, Windows SNMP service, and the CCMadmin service. Perform these restarts on the publisher and then on the subscribers.

**Note:** If Service Monitor is already using Cisco Unified Communications Manager and you are changing the cluster name, then you have to delete and readd the Cisco Unified Communications Managers in Service Monitor for it to reflect the new cluster name.

If Operations Manager is already managing Cisco Unified Communications Manager and you are changing the cluster name, then the cluster names in the service-level view will not reflect the new cluster name. You have to delete and readd the Cisco Unified Communications Manager in Operations Manager for it reflect the new cluster name.

- Make sure that the Service Monitor server's hostname is resolvable using DNS. If DNS is not being used, edit the Windows hosts file and enter the Service Monitor hostname and IP address. The hosts file is located at C:\Windows\system32\drivers\etc.

### Installation Procedures

If you do not have a license key, then during the installation, select the evaluation version. The evaluation version can manage up to 1000 phones for up to 90 days. When the Service Monitor license has been acquired, simply upload the license into the Service Monitor server by clicking http://hostname:1741/cwhp/maas.licenseInfo.do. Operations Manager and Service Monitor require separate licenses.

### Licensing and Registering the Software

Licensing grants you permission to manage a certain number of phones. You can enter licenses for Service Monitor during installation or add them later. There is a separate license for Cisco Unified Service Monitor and Cisco Unified Operations Manager.

### Uninstallation

The uninstallation process may cause a warning message similar to the following to appear:

The uninstallation is waiting for a process to stop, do you wish to continue to wait?

If you see this message, click Yes and continue to wait.

It is a good practice to delete the C:\Program Files\CSCOpx folder and then reboot the server after the Service Monitor application has been uninstalled from any server. Remember to save any Cisco 1040-related call metrics, performance, or node-to-node archived files that you might want to keep from the C:\Program Files\CSCOpx\data folder.

## Failover and Redundancy

Service Monitor supports failover for only Cisco 1040 functionality. A primary and secondary Service Monitor server can be configured to provide redundancy and failover support to Cisco 1040. In case of a primary Service Monitor server going down, the probe would automatically switch over to the secondary Service Monitor server. However there is no synchronization between the primary Service Monitor and secondary Service Monitor servers.

### Cisco 1040 Failover Mechanism

The 1040 establishes a connection with Service Monitor and periodically sends an SCCP Keep Alive message. Service Monitor acknowledges the Keep Alive message to maintain the connection.

The following scenario describes when a 1040 will fail over to the secondary server:

1. The Cisco 1040 stops receiving Keep Alive acknowledgement messages from the primary Service Monitor server.
2. After sending three Keep Alive messages without any acknowledgement, the Cisco 1040 sends a Keep Alive message to the secondary Service Monitor server.
3. The secondary Service Monitor server sends a Keep Alive acknowledgement message.
4. The Cisco 1040 sends a StationRegister message with the station user ID set to the Cisco 1040's ID.
5. Secondary Service Monitor goes to the TFTP server to get the latest configuration file for this Cisco 1040.
6. Secondary Service Monitor sends a StationRegister acknowledgement message.
7. Now Cisco 1040 will start sending syslog messages to the secondary Service Monitor server while still sending Keep Alive messages to the primary Service Monitor server to see whether it’s back up again.

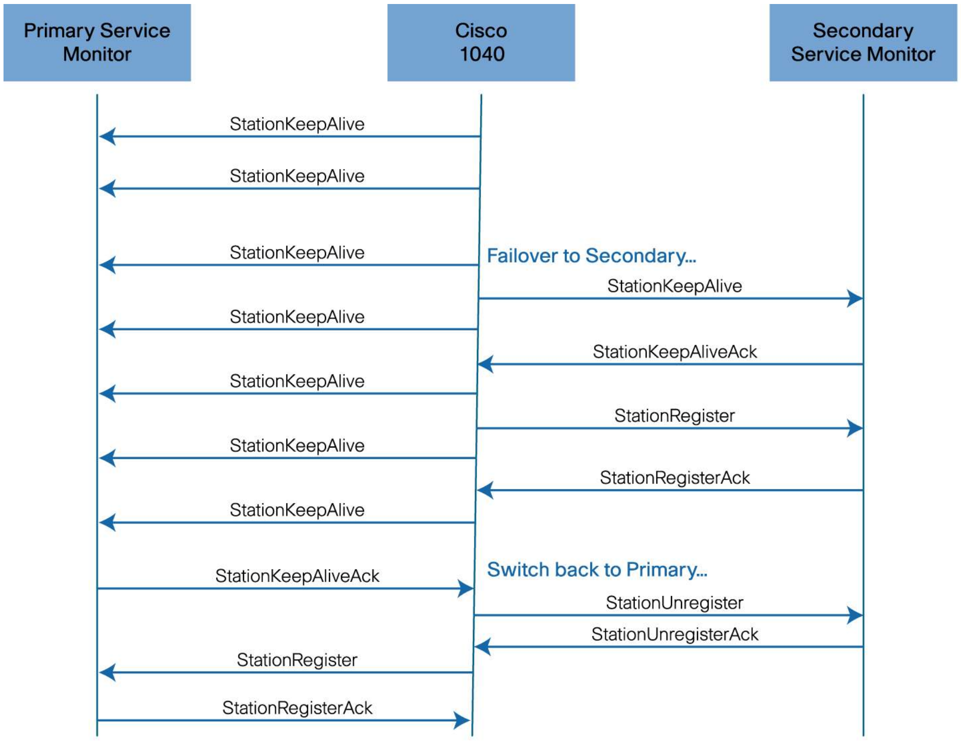
**Note:** Users cannot set the time of a failover Cisco 1040. The only way to make any configuration changes to a failover Cisco 1040 is to first make this Service Monitor server its primary Service Monitor server.

The following scenario describes how the Cisco 1040 will revert back to the primary server:

1. Cisco 1040 begins to receive a Keep Alive acknowledgement from its primary Service Monitor server once it comes back up again.
2. Cisco 1040 sends a StationUnregister message to the secondary Service Monitor server.
3. Secondary Service Monitor server sends a StationUnregister acknowledgement message to the Cisco 1040.
4. Cisco 1040 sends a StationRegister message with the station user ID set to the Cisco 1040's ID to the primary Service Monitor server.
5. Primary Service Monitor server sends back a StationRegister acknowledgement message.
6. Cisco 1040 starts to send syslog messages to this Service Monitor server now.

See Figure 5.

**Figure 5.** Cisco Failover Scenario



### Cisco 1040 Failover Deployment

Preparing for Failover

The first step is to have two Service Monitor servers available for configuration. One server acts as the primary server and the second as a secondary server. Please refer to the Service Monitor installation guide for the hardware specification of these servers.

It is recommended that these servers connect to the network through redundant paths. This helps ensure that a failure in one part of the network that affects the primary server does not also affect the connectivity of the secondary server.

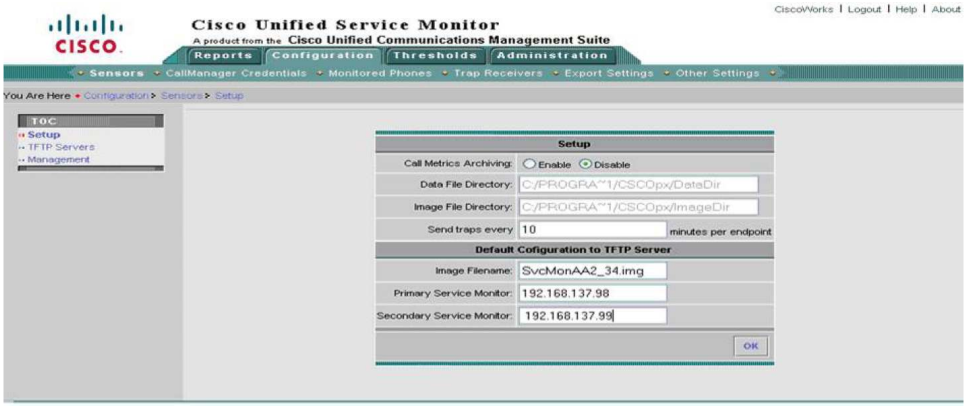
Setting Up Failover

Failover can be set up globally or for specific Cisco 1040s. If a Service Monitor server acts as the primary or secondary Service Monitor server for any Cisco 1040s across all locations, failover can be set up globally. A Service Monitor server can act as the primary Service Monitor server for one domain or location and at the same time as the secondary Service Monitor server for another domain or location. In this case, failover needs to be set up for specific Cisco 1040s.

Setting Up Failover in Default Configuration

Go to the primary Service Monitor server, select the Configuration tab, the Cisco 1040s option, and Setup from the TOC. The Setup dialog box is displayed (Figure 6). Enter the IP address or DNS name of the primary Service Monitor server and the IP address or DNS name of the secondary Service Monitor server to the default configuration for failover operations of any Cisco 1040s.

**Figure 6.** The Setup Dialog Box


CiscoWorks | Logout | Help | About
Cisco Unified Service Monitor
A product from the Cisco Unified Communications Management Suite
Reports
Configuration
Thresholds
Administration
Sensors
CallManager Credentials
Monitored Phones
Trap Receivers
Export Settings
Other Settings
You Are Here • Configuration > Sensors > Setup
TOC
Setup
TFTP Servers
Management
OK

Setup

Call Metrics Archiving: ○ Enable ● Disable

Data File Directory: C:/PROGRA~1/CSCOpx/DataDir

Image File Directory: C:/PROGRA~1/CSCOpx/ImageDir

Send traps every 10 minutes per endpoint

Default Cofiguration to TFTP Server

Image Filename: SvcMonAA2_34.img

Primary Service Monitor: 192.168.137.98

Secondary Service Monitor: 192.168.137.99

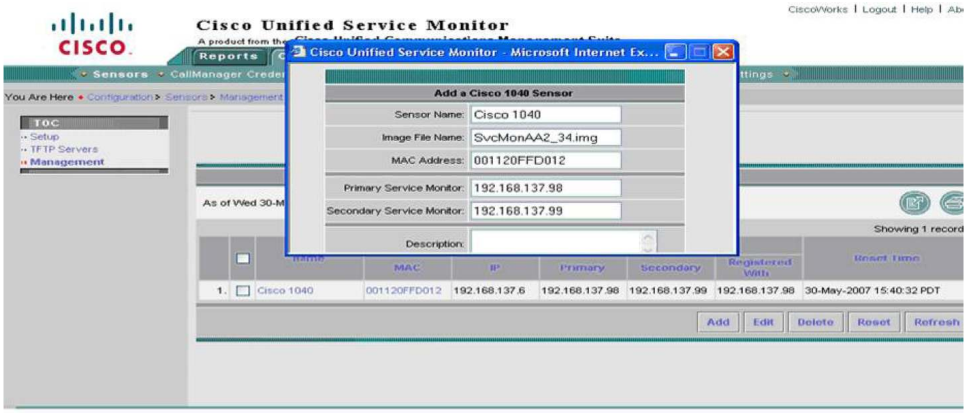
Setting Up Failover for a Specific Cisco 1040 Sensor

Go to the primary Service Monitor server, select the Configuration tab, the Cisco 1040s option, and Management from the TOC.

The Cisco 1040 Details dialog box opens showing a list of any previously defined or registered Cisco 1040s. Select Add to create a specific configuration for a Cisco 1040.

The Add a Cisco 1040 dialog box opens (Figure 7). Enter the IP address or DNS name of the primary Service Monitor server and the IP address or DNS name of the secondary Service Monitor server to the configuration for failover operations of a specific Cisco 1040.

**Figure 7.** Dialog Box for Adding a Cisco 1040


CiscoWorks | Logout | Help | Ab
Cisco Unified Service Monitor
Reports
Sensors
CallManager Creden
ttings
You Are Here • Configuration > Sensors > Management
TOC
Setup
TFTP Servers
Management
Cisco Unified Service Monitor - Microsoft Internet Ex...
As of Wed 30-M
Showing 1 record
Add
Edit
Delete
Reset
Refresh

Add a Cisco 1040 Sensor

Sensor Name: Cisco 1040

Image File Name: SvcMonAA2_34.img

MAC Address: 001120FFD012

Primary Service Monitor: 192.168.137.98

Secondary Service Monitor: 192.168.137.99

Description:

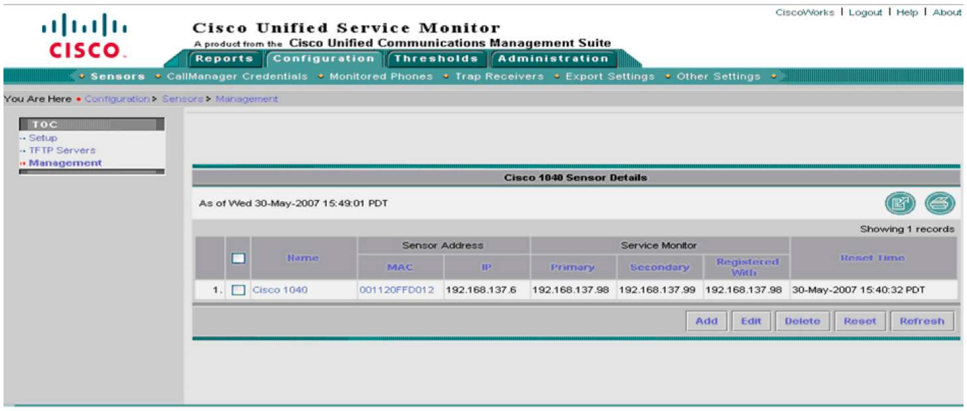
| | □ | Name | MAC | IP | Primary | Secondary | Registered With | Reset Time |
|---|---|---|---|---|---|---|---|---|
| 1. | □ | Cisco 1040 | 001120FFD012 | 192.168.137.6 | 192.168.137.98 | 192.168.137.99 | 192.168.137.98 | 30-May-2007 15:40:32 PDT |

Viewing Failover Status

Enter http://<IP address> in your browser where IP address is the address of your Cisco 1040. The Current Service Monitor field will show the Service Monitor to which the Cisco 1040 is sending data; this could be a primary or secondary Service Monitor server.

Go to the primary Service Monitor server or secondary Service Monitor server if failover took place, select the Configuration tab, the Cisco 1040 option, and Management from the TOC. The Cisco 1040 details page will show the primary Service Monitor server, secondary Service Monitor server, and the Service Monitor server to which the Cisco 1040 is registering (Figure 8).

**Figure 8.** The Service Monitor Cisco 1040 Details Page



Cisco 1040 Sensor Details

As of Wed 30-May-2007 15:49:01 PDT

Showing 1 records

|  | □ | Name | Sensor Address |  | Service Monitor |  |  | Reset Time |
|---|---|---|---|---|---|---|---|---|
|  |  |  | MAC | IP | Primary | Secondary | Registered With |  |
| 1. | □ | Cisco 1040 | 001120FFD012 | 192.168.137.6 | 192.168.137.98 | 192.168.137.99 | 192.168.137.98 | 30-May-2007 15:40:32 PDT |

### Cisco Voice Transmission Quality Redundancy

There is no failover capability for the Cisco Voice Transmission Quality capability. However the call manager can be added to multiple Service Monitor servers. Up to three Service Monitor servers can be configured as billing application servers in Cisco Unified Communications Manager 5.x or later to receive Cisco Voice Transmission Quality data. So when one Service Monitor server is down, the other Service Monitor server will still be able to obtain the Cisco Voice Transmission Quality data from Cisco Unified Communications Manager.

**Note:** Cisco Unified Communications Manager publisher server is responsible for transferring Cisco Voice Transmission Quality data to the Service Monitor server. If the publisher server is unavailable, there is no mechanism for Cisco Unified Service Manager to obtain the Cisco Voice Transmission Quality data in the cluster.

## Backup and Restore

The Service Monitor 2.x database normally is very big. Backup can take hours, so the backup process is not included when the Service Monitor software is reinstalled or upgraded.

On servers on which Operations Manager and Service Monitor are coresident, the database is not very big so it is OK to use the Common Services backup and restore process. The backup UI might time out, but the backup will be complete after some time (this might take a few hours for a big database). Restore can be done only after backup has completed successfully.

In Service Monitor standalone servers, it is recommended to use the Common Services backup and restore process only when the database is less than 6 GB. For large Service Monitor databases, it is recommend to do backup manually (saving database password and copy database files) as documented in the user guide.

## Configuring Low-Volume Schedule and Database Purging

Service Monitor needs 8 hours of low-volume time during a day. During a low-volume schedule, Service Monitor handles roughly 20 percent of the number records that are processed during a peak period and performs database maintenance. For Cisco 1040/NAM data, during regular call volume the maximum segment rate allowed is 5000 per minute. Anything over this rate will be discarded. During low call volume, the maximum segment rate allowed is 25 percent of the maximum regular call volume. Throttling of Cisco 1040 data is based on the total amount of data received in any 5-minute interval; it is not per minute. This allows accommodation of temporary spikes in traffic while blocking continuous high-rate traffic that is over the supported limit. There is no throttling of Cisco Voice Transmission Quality data. Service Monitor standalone server has been tested to support a maximum rate of 1500 Cisco Voice Transmission Quality calls per minute.

The default low-volume schedule is 10 p.m. through 6 a.m. To change the schedule, on the Service Monitor server, change the values of these properties in the NMSROOT\qovr\qovrconfig.properties file:

```
lowcallvolume-Mon=0-6,22-24
lowcallvolume-Tue=0-6,22-24
lowcallvolume-Wed=0-6,22-24
lowcallvolume-Thu=0-6,22-24
lowcallvolume-Fri=0-6,22-24
lowcallvolume-Sat=0-20,22-24
lowcallvolume-Sun=0-20,22-24
```

You can configure more than one low-volume period as long as the total time adds up to 8 hours and it covers midnight to 1 a.m. Here are some examples:

```
lowcallvolume-Mon=1-7,21-23
lowcallvolume-Tue=0-6,21-22,23-24
```

To put changes into effect after you edit qovrconfig.properties, you must stop and start the QOVR process. While logged on to the server where Service Monitor is installed, from the command line, enter these commands:

```
pdterm QOVR
pdexec QOVR
```

Service Monitor needs 4 hours data purge time. Data purging must occur during the low-volume schedule and must not run from midnight to 2 a.m. The default schedule is 2 a.m. to 6 a.m. To change the schedule on the Service Monitor server, change the values of these properties in the NMSROOT\qovr\qovrconfig.properties file:

```
datapurge-Mon=2-6;
datapurge-Tue=2-6;
datapurge-Wed=2-6;
datapurge-Thu=2-6;
datapurge-Fri=2-6;
datapurge-Sat=2-6;
datapurge-Sun=2-6;
```

Data purge need not run continuously for 4 hours. You can configure more than one data purge period as long as:

- The total time adds up to 4 hours
- Data purging occurs during low-volume schedule
- No data purging occurs from midnight through 2 a.m.

Here are some examples:

```
datapurge-Mon=2-5;22-23;
datapurge-Tue=2-3;4-6;23-24
```

**Note:** Do not edit the properties files using Wordpad as it introduces a carriage return. Use Notepad to edit the file instead.

The data retention period determines the number of days that data is retained in the Service Monitor database before being purged. The default value depends on the deployment scenario:

- Service Monitor alone on a server: 7 days
- Any coresident server: 3 days

### Troubleshooting

This section provides a few troubleshooting tips.

**Q. I don’t see any service quality alerts in the Operations Manager dashboard. What could be the problem?**

**A.** Before debugging this problem, do the following:

- Go to the Service Monitor application and verify that the probes are registered and visible in the Service Monitor GUI.
- Go to the Service Monitor setup page and verify that Operations Manager is entered as a trap recipient, even if Service Monitor is on the same machine.
- In Operations Manager, go to Administration > Service Quality and add Service Monitor, even if it is on the same machine.
- If all of the previous are correct, then only calls that fall below the Service Monitor threshold will be shown in the Alerts and Activities page.

When all of the previous have been done, debug the problem by doing the following:

- Check syslog.log under NMSROOT\CSCOpx\log\qovr\syslog.log in Cisco Unified Operations Manager 1.x Cisco Unified Service Monitor 2.x. Check whether you see recent syslogs. Check the D=<value>. This is the MOS value multiplied by a factor of 10. Check to see whether this value is below the threshold multiplied by 10.
- If this is true, check NMSROOT\log\qovr\trapgen.log. This file should contain the traps that are being generated by the system. If traps are available in this file, then it means that the Service Monitor portion is functional and ready. If not, then check for exceptions in probemanager.log and datahandler.log.

**Q. A few rows in the Cisco 1040 diagnostic reports do not show directory numbers. What could be the issue?**

**A.** Make sure that the call is over. CDRs and CMRs are generated at the end of the call only, whereas Cisco 1040s send data even while the call is in progress.

Make sure that the corresponding call in the Cisco Voice Transmission Quality diagnostic report shows the directory numbers for both the caller and callee. Some gateway ports do not have directory numbers; Cisco 1040 sensor rows for those calls will not have directory number information.

**Q. Why is the directory number missing in sensor reports?**

**A.** Please check the following:

- The Cisco 1040s monitor RTP traffic; they do not report directory numbers.
- Directory numbers can be seen in the diagnostic Cisco 1040 reports only if the call whose streams are being reported by Cisco 1040s to Service Monitor is also reported by the call manager to Service Monitor, meaning that the call manager server must be added to the Service Monitor server.
- Service Monitor and call managers have to be time synched in order to display the directory numbers.

**Q. Can you give me some general Cisco 1040 troubleshooting tips?**

**A.** Please check the following:

- If the Cisco 1040 is not receiving the IP address, check the DHCP configuration on the DHCP server.
- Start with http://<ip-addr>/Communication and see what the Cisco 1040 communication debug page says. If there are any startup issues on the Cisco 1040 (such as the TFTP server not being reachable or an inability to download the image file), this page should tell you.
- If you are suspecting a Cisco 1040/Service Monitor issue, then use the sniffer to check the communication from the Cisco 1040 to the rest of the world.

You should use the sniffer on the management port of the Cisco 1040 to begin with.

- If Cisco Unified Communications Manager 5.x is used for the TFTP server, please restart the TFTP service after the configuration files and image files are manually copied over for changes to take effect.
- The spanning port doesn't play a role in booting the Cisco 1040. But do make sure SPAN is configured on the switch port instead of the routed port.
- ● Service Monitor installs the TFTP server by default; if the customer installs another TFTP server, make sure the TFTP service in Common Services is shut down to avoid any potential issue.
- ● Do not edit configuration files using Wordpad (where the TFTP server is not editable), as it introduces a carriage return.

Detailed troubleshooting tips are documented at http://cisco.com/en/US/products/ps6536/prod_troubleshooting_guides_list.html.

**Q.** **We do not have a PoE switch to power our Cisco 1040 sensor. Cisco discontinued selling the SM-1040-PWR power adaptor. What are our options to power the sensors?**

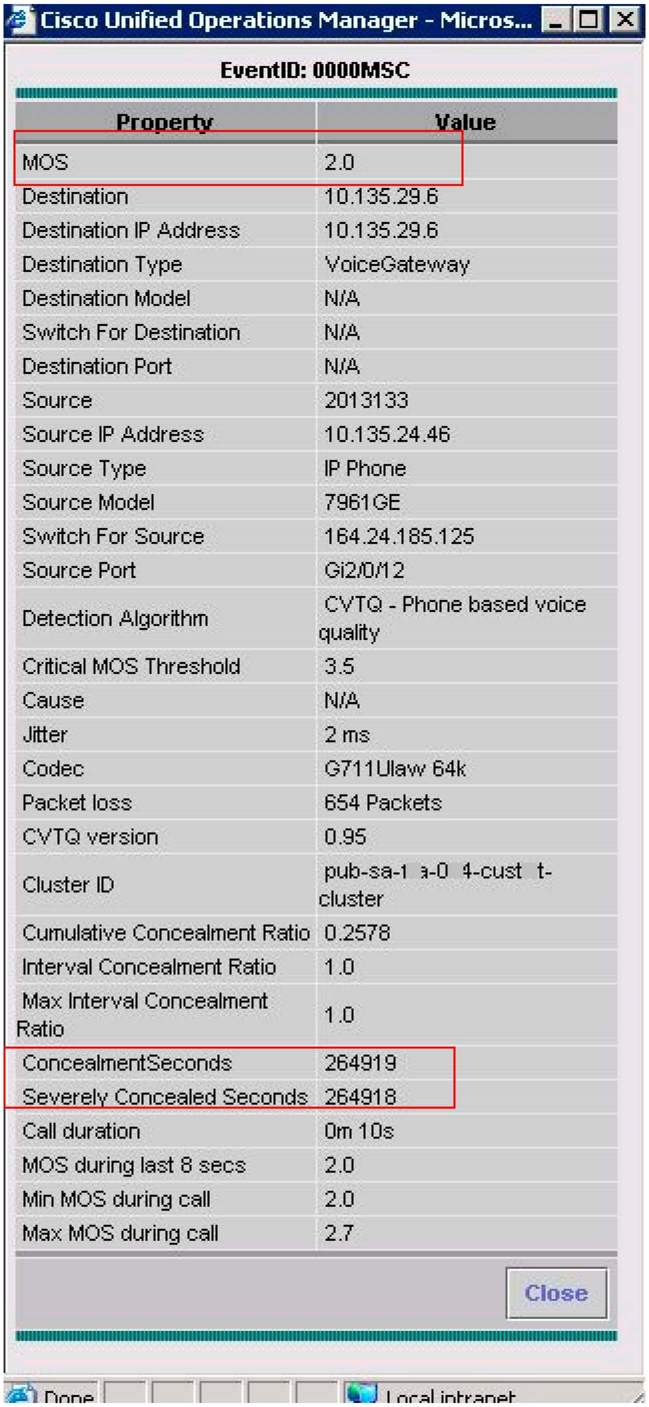
**A.** Procure an industry-standard DC 5-volt, 2.6-amp power adapter.

**Q.** **Why are Cisco Voice Transmission Quality MOS values constantly very low, at the 2.0 range, and why are the concealment seconds values too high?**

**A.** Please add the following configuration under dial-peers in the voice gateway:

- *rtp payload-type comfort-noise 13*
- Or if there is no need to conserve bandwidth, turn off Voice Activity Detection (VAD) with the no vad command
- Examples below

```
Dial-peer voice 101 voip
Destination-pattern 11…
Ip qos dscp cs3 signaling
Dtmf-relay h245-alphanumeric
No vad
Codec g711ulaw
Preference 1
Session target ipv4:10.10.1.1
```



EventID: 0000MSC

| Property | Value |
|---|---|
| MOS | 2.0 |
| Destination | 10.135.29.6 |
| Destination IP Address | 10.135.29.6 |
| Destination Type | VoiceGateway |
| Destination Model | N/A |
| Switch For Destination | N/A |
| Destination Port | N/A |
| Source | 2013133 |
| Source IP Address | 10.135.24.46 |
| Source Type | IP Phone |
| Source Model | 7961GE |
| Switch For Source | 164.24.185.125 |
| Source Port | Gi2/0/12 |
| Detection Algorithm | CVTQ - Phone based voice quality |
| Critical MOS Threshold | 3.5 |
| Cause | N/A |
| Jitter | 2 ms |
| Codec | G711Ulaw 64k |
| Packet loss | 654 Packets |
| CVTQ version | 0.95 |
| Cluster ID | pub-sa-1[redacted]a-0[redacted]4-cust[redacted]t-cluster |
| Cumulative Concealment Ratio | 0.2578 |
| Interval Concealment Ratio | 1.0 |
| Max Interval Concealment Ratio | 1.0 |
| ConcealmentSeconds | 264919 |
| Severely Concealed Seconds | 264918 |
| Call duration | 0m 10s |
| MOS during last 8 secs | 2.0 |
| Min MOS during call | 2.0 |
| Max MOS during call | 2.7 |

**Q. Why is the MOS value unavailable from reports?**

**A.** The following scenarios could lead to the MOS value being unavailable:

- The phone device does not support Cisco Voice Transmission Quality. Refer to your Cisco IP Phone documentation for Cisco Voice Transmission Quality support; Cisco Unified Personal Communicator, voicemail, and H.323 calls do not support Cisco Voice Transmission Quality.
- The call duration is less than 8 seconds.
- By default, voice quality statistics reporting is turned off in Media Gateway Control Protocol (MGCP) gateways. Make sure to enable Cisco Voice Transmission Quality in gateways by adding this configuration in global mode: mgcp voice-quality-stats all.

- Minimum of Cisco IOS® Software Release 12.4.(4)T and 5510 DSP hardware: use the Cisco IOS Software show version and show voice dsp detailed commands.
- If using a Session Initiation Protocol phone and Unified Communications Manager version 7.1(2) or earlier, enable the Call Stats checkbox in the phone's SIP profile.

**Q. How do I force Service Monitor to store and report zero duration calls, which are often failed calls?**

**A.** Make sure that the CDR Log Calls With Zero Duration Flag in Cisco Unified Communications Manager > Service parameters is off for all nodes.

**Q. What UDP and TCP ports are in use by Service Monitor and must be opened in the firewall?**

**A.** The following ports must be allocated for Service Monitor and exempted in the firewall inspection:

- **UDP:** 53 (DNS), 67 and 68 (DHCP), and 5666 (syslog)
- **TCP:** 22 (SFTP), 2000 (SCCP), 43459 (database communication), 5665 - 5680 (Service Monitor internal communications

**Q. Can we apply the latest Microsoft Windows OS hot fixes in the Service Monitor server?**

**A.** We recommend following the security guidance given at the National Security Agency (NSA) website for Microsoft products at http://www.nsa.gov/ia/guidance/security_configuration_guides/operating_systems.shtml#microsoft.

**Q. Can Service Monitor run in VMware?**

**A.** Yes Service Monitor is certified to run in VMware ESX 3.5 and ESXi 4.0. A minimum of 4 vCPU, 4 GB memory, 80 GB vDisk, and 1 vNIC with static MAC address is required.

## Useful URLs

### Cisco.com URLs for Customers and Partners

#### Product Information

Product Page: http://www.cisco.com/en/US/products/ps6536/prod_literature.html

Installation and Upgrade Guides: http://www.cisco.com/en/US/products/ps6536/prod_installation_guides_list.html

Release Notes: http://www.cisco.com/en/US/products/ps6536/prod_release_note09186a00807ee746.html#wp90281

Data Sheet: http://www.cisco.com/en/US/products/ps6536/products_data_sheets_list.html

Cisco Unified Service Monitor 2.3 Data Sheet:
http://www.cisco.com/en/US/products/ps6536/products_data_sheets_list.html

User Guide for Cisco Unified Service Monitor:
http://www.cisco.com/en/US/docs/net_mgmt/cisco_unified_service_monitor/2.3/user/guide/UserGuideforCiscoUnifiedServiceMonitor2_3.html

#### Training

Instructor-led Cisco Unified Operations Manager and Cisco Unified Service Monitor 2-day training. Customer and partners can send an email to aeskt_registration@cisco.com.

**Evaluation Downloads**

For Partners and Customers

Step 4. Go to the Marketplace site at http://www.cisco.com/go/marketplace. Note that you must log in with a Cisco employee or authorized Cisco Partner login and password.

Step 5. Select the Collateral & Subscription Store link.

Step 6. Read the notice to Cisco employees and click Continue.

Step 7. From the navigation menu at the top-left corner of the page (above the Subscriptions link), select the Marketing Collateral link. From the Marketing Collateral navigation menu, select Network Management Evaluation Kits, and then select the desired evaluation kit.

Step 8. Use Add to cart and Checkout to place the order for the desired kit, using your Access Visa or personal credit card.

For further questions on Cisco Unified Operations Manager or Cisco Unified Service Monitor, or for any other Cisco Unified Management - related questions, send an email to ask-ipc-management@cisco.com.

**Patch Download:** http://www.cisco.com/cgi-bin/tablebuild.pl/servmon

**Miercom Review of Cisco Unified Operations Manager and Cisco Unified Service Monitor:**
http://www.cisco.com/en/US/products/ps6535/prod_white_papers_list.html

Americas Headquarters
Cisco Systems, Inc.
San Jose, CA

Asia Pacific Headquarters
Cisco Systems (USA) Pte. Ltd.
Singapore

Europe Headquarters
Cisco Systems International BV
Amsterdam, The Netherlands

Cisco has more than 200 offices worldwide. Addresses, phone numbers, and fax numbers are listed on the Cisco Website at **www.cisco.com/go/offices.**

Printed in USA C07-554560-01 06/10