NetIQ® AppManager® for VoIP Quality

Management Guide

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About this Book and the Library

The NetIQ AppManager product (AppManager) is a comprehensive solution for managing, diagnosing, and analyzing performance, availability, and health for a broad spectrum of operating environments, applications, services, and server hardware.

AppManager provides system administrators with a central, easy-to-use console to view critical server and application resources across the enterprise. With AppManager, administrative staff can monitor computer and application resources, check for potential problems, initiate responsive actions, automate routine tasks, and gather performance data for real-time and historical reporting and analysis.

Intended Audience

This guide provides information for individuals responsible for installing an AppManager module and monitoring specific applications with AppManager.

Other Information in the Library

The library provides the following information resources:

Installation Guide for AppManager

Provides complete information about AppManager pre-installation requirements and step-by-step installation procedures for all AppManager components.

User Guide for AppManager Control Center

Provides complete information about managing groups of computers, including running jobs, responding to events, creating reports, and working with Control Center. A separate guide is available for the AppManager Operator Console.

Administrator Guide for AppManager

Provides information about maintaining an AppManager management site, managing security, using scripts to handle AppManager tasks, and leveraging advanced configuration options.

Upgrade and Migration Guide for AppManager

Provides complete information about how to upgrade from a previous version of AppManager.

Management guides

Provide information about installing and monitoring specific applications with AppManager.

Help

Provides context-sensitive information and step-by-step guidance for common tasks, as well as definitions for each field on each window.

The AppManager library is available in Adobe Acrobat (PDF) format from the NetIQ Web site: www.netiq.com/support/am/extended/documentation/default.asp?version=AMDocumentation.

Conventions

The library uses consistent conventions to help you identify items throughout the documentation. The following table summarizes these conventions.

Convention	Use
Bold	◆ Window and menu items ◆ Technical terms, when introduced
Italics	◆ Book and CD-ROM titles ◆ Variable names and values ◆ Emphasized words
`Fixed Font`	◆ File and folder names ◆ Commands and code examples ◆ Text you must type ◆ Text (output) displayed in the command-line interface
Brackets, such as *[value]*	◆ Optional parameters of a command
Braces, such as *{value}*	◆ Required parameters of a command
Logical OR, such as *value1*|*value2*	◆ Exclusive parameters. Choose one parameter.

About NetIQ Corporation

NetIQ, an Attachmate business, is a global leader in systems and security management. With more than 12,000 customers in over 60 countries, NetIQ solutions maximize technology investments and enable IT process improvements to achieve measurable cost savings. The company's portfolio includes award-winning management products for IT Process Automation, Systems Management, Security Management, Configuration Audit and Control, Enterprise Administration, and Unified Communications Management. For more information, please visit www.netiq.com.

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Contacting the Online User Community

Qmunity, the NetIQ online community, is a collaborative network connecting you to your peers and NetIQ experts. By providing more immediate information, useful links to helpful resources, and access to NetIQ experts, Qmunity helps ensure you are mastering the knowledge you need to realize the full potential of IT investments upon which you rely. For more information, please visit http://community.netiq.com.

1 Introducing AppManager for VoIP Quality

For most organizations, managing voice over IP (VoIP) systems and networks is essential for maintaining the health and performance of mission-critical resources. AppManager for VoIP Quality helps you optimize call performance and ensure the availability of your VoIP systems and networks by providing automated problem detection and correction.

AppManager for VoIP Quality provides three separate but related components:

- *AppManager for Call Performance*, which allows you to monitor VoIP performance between NetIQ Performance Endpoints
- *AppManager for Cisco SAA*, which allows you to monitor VoIP performance between Cisco routers using the Cisco Service Assurance Agent
- *AppManager for Call Setup*, which allows you to monitor the response time and availability of H.323 gateways and gatekeepers, and the response time and availability of SIP (Session Initiation Protocol) devices

This chapter provides an overview of AppManager for VoIP Quality components and examples of the Call Performance proxy architecture.

1.1 Features and Benefits

The following are a few of the features and benefits of monitoring VoIP call quality with AppManager:

- Monitors end-to-end call quality using simulated VoIP traffic between key points in the network
- Measures objective call quality using the Mean Opinion Score (MOS)
- Identifies the issues impairing call quality, such as delay, jitter, and lost data
- Identifies call quality and network problems before end users are affected
- Uses NetIQ Performance Endpoints to generate real-time transport protocol (RTP) streams that mimic complex VoIP traffic
- Uses the Cisco SAA agent to collect network performance data from simulated VoIP traffic between routers
- Supports the leading codecs, including G.711a, G.711u, G.723-ACELP, G.723-MPMLQ, G.726, G.729, and G.729A
- Ensures “digital dial tone” on your IP network
- Monitors availability and response time of H.323 and SIP devices by simulating client activities
- Identifies issues that impair service availability: registration, call setup, or response time
- Pinpoints registration and call setup problems before end users are affected
- Supports H.323 version 2

1.2 Licensing AppManager

AppManager for VoIP Quality consumes one AppManager license for each discovered agent. The number of discovered agents is the sum of CallPerf agents, CallPerfProxy agents, H.323 agents, SIP agents, and SAAProxy agents.

If you have multiple instances of the same VoIPQuality CallPerfProxy object in the TreeView that refer to the same endpoint, you will be charged for more than one AppManager license. To be charged the correct number of licenses, do not discover a given endpoint more than once.

1.3 Reviewing AppManager for Call Performance

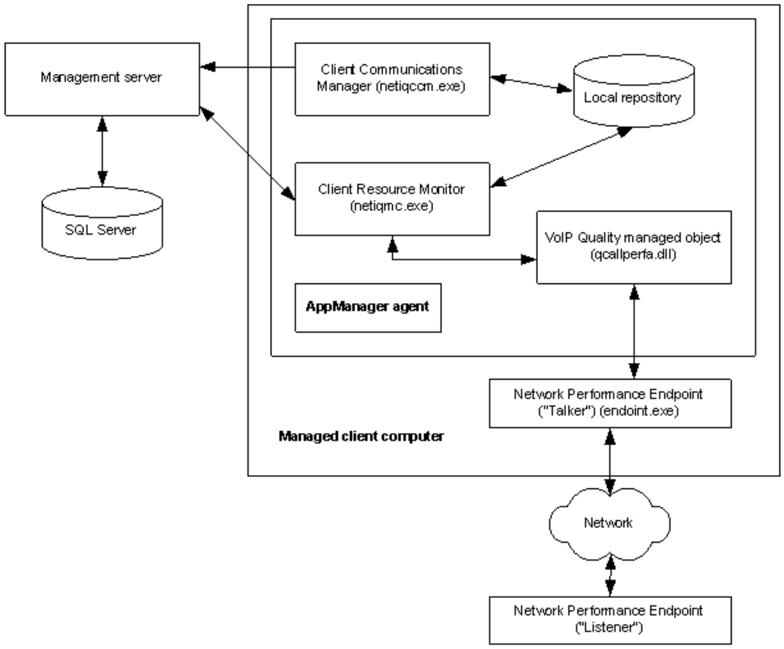
AppManager for Call Performance is one component of the AppManager for VoIP Quality module. With AppManager for Call Performance, you monitor VoIP quality between Performance Endpoints (endpoints) using either the standard architecture or the proxy architecture. In the standard architecture, the endpoints and the AppManager agent are on the same computer. The proxy architecture allows the agent and endpoints to be on separate computers.

1.3.1 Monitoring Between Endpoints

AppManager for VoIP Quality monitors call quality and VoIP network performance by actively driving synthetic VoIP traffic between endpoints placed at key points in the network. Endpoints are lightweight software agents installed on client and server computers to collect information about network transactions and send it back for analysis and reporting. AppManager for VoIP Quality installs the latest endpoint. Endpoints are also available free from NetIQ from the Current Performance Endpoints Product Upgrades page.

VoIPQuality_CallPerf Knowledge Scripts can emulate complex VoIP traffic, and monitor critical network metrics, such as delay, jitter, and lost data.

This architecture provides for the monitoring of VoIP traffic over Windows platforms. Within this architecture, the AppManager agent and the endpoint are on the same computer, as shown in the following diagram:



The agent consists of two Windows services: the *managed client* service (`NetIQmc`) and the *client communications manager* service (`NetIQccm`). The `NetIQccm` service handles communication back to the management server for events and data insertion into the SQL database. The `NetIQmc` service handles the scheduling and housekeeping of Knowledge Scripts, and uses the *managed objects* to collect performance and event data. Knowledge Scripts provide the rules for what to do with the information gathered by the managed objects.

The managed objects collect specific types of data, such as counters and event logs.

The VoIP Quality managed object uses the endpoints to collect call quality and network performance data. To that end, the VoIP Quality managed object sends the Knowledge Script parameters to the endpoint configured as the *talker*. The *talker* then performs setup and job flows with the endpoint configured as the *listener*, which in turn collects the information requested by the VoIP Quality managed object and sends it back to the talker. The talker then passes the collected information to the VoIP Quality managed object.

1.3.2 Understanding the Proxy Architecture

The architecture of the Call Performance proxy lets you monitor VoIP traffic using the following platforms:

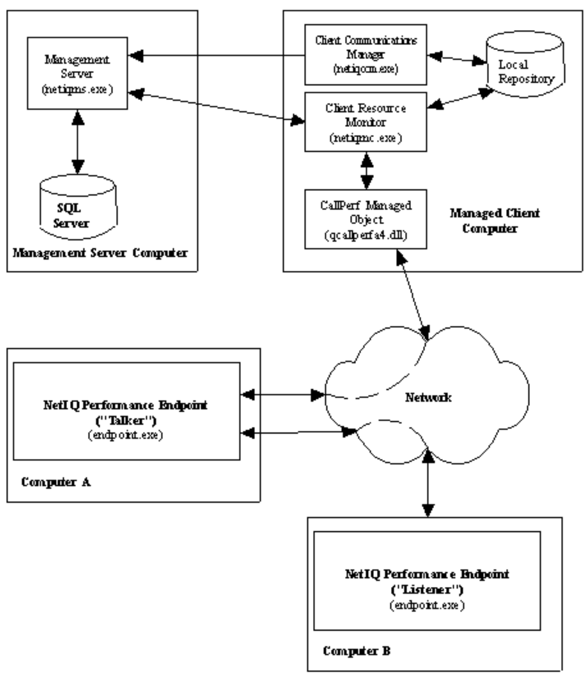
- Sun Solaris for SPARC
- Sun Solaris for x86
- Linux for Cobalt RaQ3
- Linux x86
- Linux x86 for RPM
- Spirient Terametrics (Linux)
- Microsoft Windows

With the AppManager proxy architecture, the endpoint and the VoIP Quality managed object do not need to be installed on the same Windows computer.

NOTE: Install the AppManager for VoIP Quality module on agent computers spread across the network to simulate user VoIP traffic. Do not install the module on critical application servers, such as Cisco Unified Communication Manager servers.

Within the proxy architecture, the managed object, `qCallPerfa4.dll`, is installed on the managed server computer. When you run a Knowledge Script job, the managed object runs on the managed client (acting as proxy) and sends messages to and from the endpoint on the computer configured as the talker. The proxy communicates with the endpoint using port 10115.

The following diagram illustrates the Call Performance proxy architecture:



1.4 Reviewing AppManager for Cisco SAA

The Cisco Service Assurance Agent (SAA) is a network performance measurement agent in Cisco IOS software. SAA, or the SAA Response Time Reporter (RTR), allows network performance monitoring between a Cisco router and a remote device, such as another Cisco router or an IP host. The agent measures performance by sending one or more simulated packets to a target IP device or Cisco router. SAA enables you to test and collect delay, jitter, and packet loss statistics on the data network. It also provides the mechanism to monitor performance for different classes of traffic over the same connection.

Delay and jitter can be measured by deploying Cisco routers 17xx or later with Cisco IOS software version 12.0(5)T or later, and then configuring the Cisco IOS SAA features. AppManager for Cisco SAA uses SNMP to drive the SAA tests from source Cisco routers (clients or talkers) to target routers (responders or listeners). The target router must be configured as a responder before you can run successful jitter tests. For more information, see Section 2.7, "Considerations for Cisco SAA Agent Installation," on page 28.

AppManager is designed to help you gain easy access to Cisco SAA data, and to help you analyze and manage that data. The AppManager for Cisco SAA solution minimizes the cost of maintaining an SAA system, aids in capacity planning, and can prevent downtime. VoIPQuality_CiscoSAA Knowledge Scripts can emulate complex VoIP traffic between routers, and monitor critical network metrics such as latency, jitter, and lost data.

1.5 Reviewing AppManager for H.323 Call Setup

AppManager for H.323 Call Setup monitors the response time and availability of H.323 gateways and gatekeepers. Knowledge Scripts provide the ability to emulate H.323 client processes, including call registration and call setup using the H.323 protocol, ensuring that key H.323 devices are available and performing well on the network.

1.5.1 Understanding the H.323 Protocol

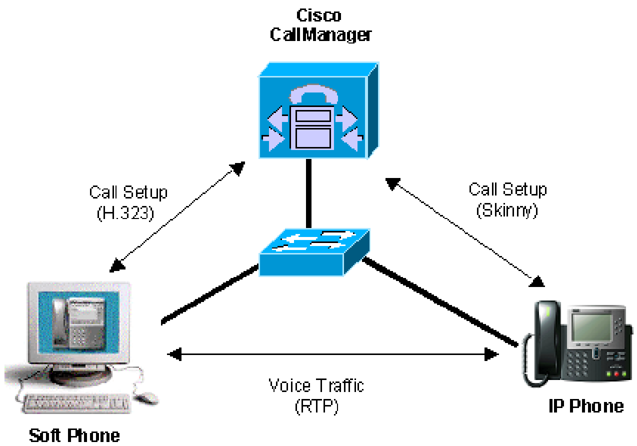
H.323 is a packet-based multimedia communications system that uses two protocols, H.225 and H.245, to enable video conferencing over LANs and other packet-switched networks, as well as video over the Internet.

An H.323 trace follows the H.225 and H.245 messages that are used to set up, process, and tear down calls. The H.323 architecture involves devices called terminals, gateways, gatekeepers, and Multipoint Controller Units (MCUs). Audio, video, and data streams are exchanged between endpoints, which can be terminals, gateways, or MCUs.

Device	Description
Gatekeeper	The device that controls permissions and authorizes network access when calls are attempted. It also monitors bandwidth and restricts phone calls based on current data. In addition, the gatekeeper provides some aliasing and name resolution. Terminals, gateways, and MCUs register with the gatekeeper if one is available. However, the gatekeeper is an optional component in an H.323 network.
Gateway	A bridge between protocols. It converts PSTN (Public Switched Telephone Network) protocols and switched voice to packet network protocols and voice techniques. It acts as an interpreter to translate data between a terminal and other network types, such as the regular phone network or an ISDN (Integrated Services Digital Network). For example, Cisco CallManager can act as an H.323 gateway.
MCU	Enables multipoint conferencing between multiple terminals and gateways. AppManager does not support monitoring or communicating with MCUs.
Port and firewall	H.323 uses fixed ports for RAS Signaling and Call Signaling. Control Signaling and the actual call itself use dynamic ports, although if Fast Connect or H.245 tunneling is used, the dynamic port is not used. Gatekeeper discovery and Gatekeeper RAS use different, but fixed, ports as well.
Terminal	A terminal, such as an IP phone or Microsoft NetMeeting, offers real-time communication with other H.323 endpoints.

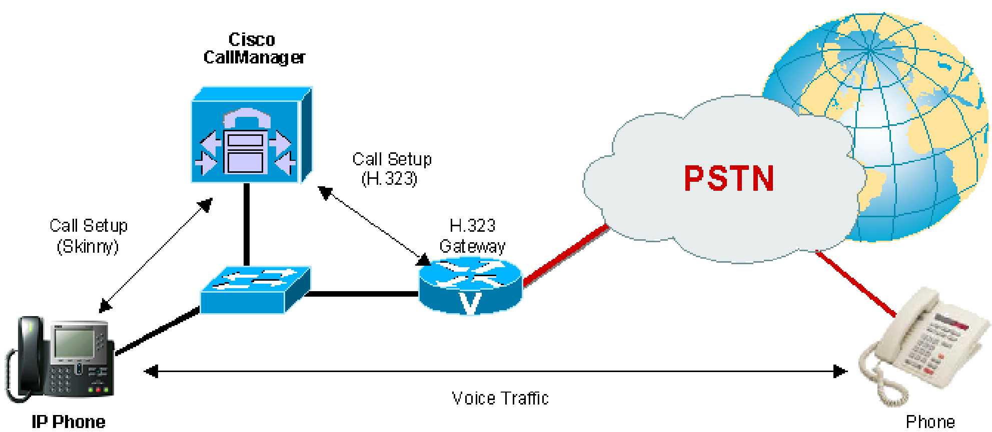
1.5.2 Network Flow Example: Soft Phone to IP Phone Using CallManager H.323 Call Setup

In a Cisco IP telephony environment, whenever a soft phone makes a call to a regular IP phone, the call must go through a gateway to translate the phone call setup from H.323 to Skinny (SCCP) protocol. Usually, the Cisco CallManager is the gateway that handles this type of protocol translation. First the soft phone communicates the need to make a call with the CallManager. Then the CallManager determines whether the receiving IP phone is available and chooses the path of the call. Once the call is setup and connected, the voice traffic travels directly between the two devices.



1.5.3 Network Flow Example: IP Phone to the PSTN Using an H.323 Gateway

In a gateway flow, one terminal (the talker) communicates the need to make a call with a gateway, which in turn determines whether the receiving terminal (the listener) is available and then, choosing the path of the call, sends the call on its way.



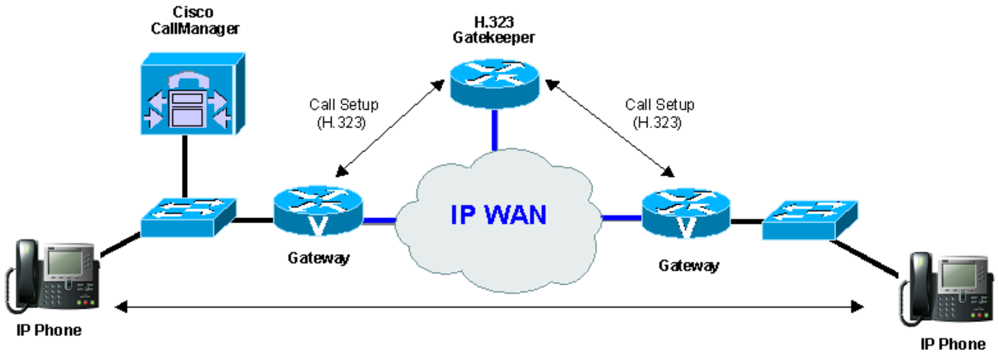
1.5.4 Network Flow Example: IP Phones Using an H.323 Gatekeeper

Each endpoint must register with the gatekeeper when one is available:

- The endpoint sends out a UDP broadcast on a specified port and waits for a response
- After receiving the response, the endpoint attempts to register with the gatekeeper

NOTE: The registration process is a separate flow — the terminal can also specify the gatekeeper with which to register.

When initiating a call, the terminal (talker) contacts the gateway for permission to make the call. When the gatekeeper replies, the terminal then asks to set up the call. The gatekeeper determines whether the second terminal (listener) is available and then grants permission to the talker. The talker then establishes the call to the listener.

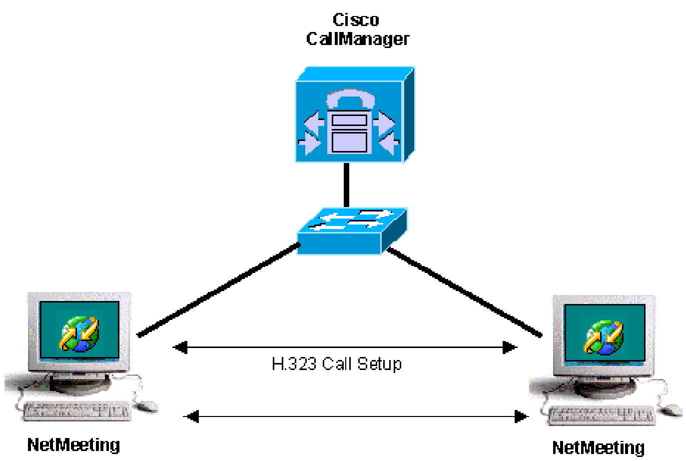


You set up the test for this flow with the H.323_Listen and H.323_CallSetup_Gatekeeper Knowledge Scripts.

Because you can continue to place multiple gatekeepers between terminals and endpoints, H.323 architecture has unlimited growth potential.

1.5.5 Network Flow Example: NetMeeting to NetMeeting with No Gatekeeper or Gateway

Terminals (talkers and listeners) can communicate directly without a gatekeeper or a gateway, if neither is available.



You set up the test for this flow with the H.323_Listen and H.323_CallSetup_Direct Knowledge Scripts.

1.6 Reviewing AppManager for SIP Call Setup

AppManager for SIP Call Setup monitors the response time and availability of the Session Initiation Protocol (SIP). Knowledge Scripts provide the ability to emulate SIP client processes, including call registration and call setup using the SIP protocol, ensuring that key SIP devices are available and performing well on the network.

SIP is a simple, HTTP-like protocol for performing call setup. Developed by the IETF MMUSIC Working Group as an alternative to H.323, SIP equips platforms to signal the setup of voice and multimedia calls over IP networks. Three components comprise the SIP architecture:

- User agent client, which makes calls.
- User agent server, which answers or rejects calls.
- SIP server, which locates called parties. The SIP server can be a proxy server or a redirect server.

SIP is based on a set of several commands, which operate in much the same way as the `HTTP GET` and `POST` commands. The SIP command set includes the following:

- **INVITE** - Use this command to begin a call.
- **BYE** - Use this command to end a call.
- **REGISTER** - Use this command to register a SIP endpoint with a server.

As with HTTP, a SIP request includes a command line, a set of headers (key/value pairs), and a body. For example:

```
COMMAND target_address SIP/2.0
Via:
From:
To:
.
.
.
Contact-Length:
<SDP data>
```

SIP can operate in a server-based environment or in a direct-calling environment. In a *server-based* environment, SIP messages are transmitted from the talker to the listener using a server or a set of servers. Signaling requests are sent to the talker's local server, which then routes the request to the listener (either directly or using additional SIP servers). In contrast, in a *direct-calling* environment all signaling occurs directly between the talker and the listener.

In a server-based environment, a client must register with its *registration server*, also known as a registrar. The registrar maintains lists of active SIP clients on the network. To place a call, a SIP client sends an INVITE request to its call server, which can act as a *proxy* or as a *redirector* (and is frequently the same computer as the registrar). In proxy mode, the server forwards the `INVITE` to the listener's SIP server. Listeners send requests to the proxy server, which either takes care of the request or sends it to other servers. Those servers have no idea that the proxy server is not the source of the message.

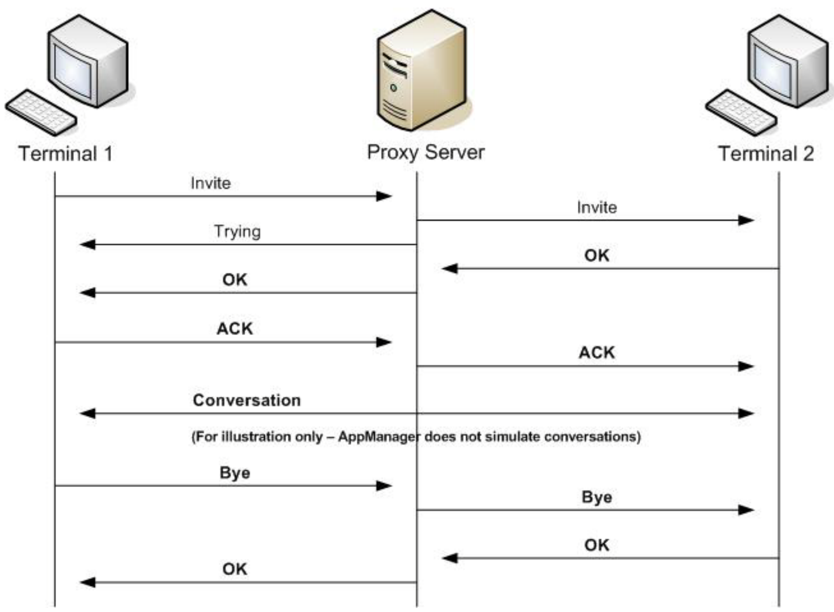
In redirect mode, the server sends to the talker both a redirection message and the address of the server where the listener can be reached. In redirect mode, it is the talker's responsibility to send another `INVITE` request to the listener's SIP server. A redirect server does not generate requests; instead, it provides the information the talker needs to generate the request itself. Thus, SIP elements can act as relays or as redirectors. The relay method is more common. It is the only method supported by AppManager.

Using the relay method, a SIP call involves the following flows between the talker and its DIP server:

```
C->S: INVITE (ring the other phone)
C<-S: 100 TRYING (indicates that the other phone is ringing)
C<-S: 200 OK (indicates that the call has been accepted)
C->S: BYE (end the call)
C<-S: 200 OK (acknowledge the hang up)
```

NOTE: AppManager uses the UDP protocol for SIP signaling.

The following is an example of a SIP call between Terminal 1 and Terminal 2 using a proxy server:



2 Installing AppManager for VoIP Quality

This chapter provides installation instructions and describes system requirements for AppManager for VoIP Quality.

This chapter assumes you have AppManager installed. For more information about installing AppManager or about AppManager system requirements, see the *Installation Guide for AppManager*, which is available on the AppManager Documentation page.

2.1 System Requirements

For the latest information about supported software versions and the availability of module updates, visit the AppManager Supported Products page. Unless noted otherwise, this module supports all updates, hotfixes, and service packs for the releases listed below.

AppManager for VoIP Quality has the following system requirements:

Software/Hardware	Version
NetIQ AppManager installed on the AppManager repository (QDB) computer, on the proxy agent computer, and on all console computers	7.0 or later Support for Windows Server 2008 on AppManager 7.x requires AppManager Windows Agent hotfix 71704 or later. For more information, see the AppManager Suite Hotfixes page.
Microsoft operating system installed on the proxy agent computers	One of the following: ◆ Windows 7 (32- or 64-bit) ◆ Windows Server 2008 R2 **Note** Support for Quality of Service (QoS) settings on Windows 7 and Windows 2008 R2 endpoints requires the NetIQ Performance Endpoints version 5.1.15368.0 or later. You can download the Performance Endpoints for your operating system at the Current Performance Endpoints Product Upgrades page. ◆ Windows Server 2008 (32- or 64-bit) ◆ Windows Server 2003 R2 (32- or 64-bit) ◆ Windows Vista (32- or 64-bit) **Note** Windows Vista does not support DSCP-based QoS. ◆ Windows XP (32-bit)

Software/Hardware	Version
AppManager for Microsoft Windows module installed on repository, agent, and console computers	Support for Windows Server 2008 R2 on AppManager 7.x requires the AppManager for Windows module, version 7.6.170.0 or later. For more information, see the AppManager Module Upgrades & Trials page.
NetIQ Performance Endpoints	Version 5.1.15368.0 or later Performance Endpoints software is included with AppManager for VoIP Quality and is installed automatically. For more information, see Section 2.2, “Pre-installation Considerations,” on page 20.
Cisco IOS	12.0 or later Monitoring for delay requires version 12.2(2)T or later
Microsoft Internet Explorer installed on the agent computers	7.0 or later
Hardware	◆ Port 10115 configured in your firewall to permit bidirectional TCP and UDP packets: **Source** — **Destination** <local machine>:10115 — <any machine>:<any port> <any machine>: 10115 — <local machine>:<any port> ◆ Port numbers of your choice configured in your firewall to permit bidirectional TCP and UDP packets. You can use these port numbers in the *Advanced Configuration* parameters in the VoIPQuality Knowledge Scripts.

If you encounter problems using this module with a later version of your application, contact NetIQ Technical Support.

2.2 Pre-installation Considerations

Consider the following before installing the AppManager for VoIP Quality module and when deciding where to install the proxy agent:

- If you are running the proxy agent on the managed server, limit the number of remote computers to improve performance.
- If you have one agent computer acting as the proxy for all remote computers, then you have a single point of failure for your proxy environment. When the proxy agent computer is not running for any reason, no jobs will run on the remote computers.
- If the proxy agent computer cannot contact the endpoint when it is ready to run a test, then the test will fail. The endpoint does not keep any scheduling or result information, which is all kept at the agent.
- Each remote computer should be associated with only one proxy agent computer.

Ensure that you install the Performance Endpoint on the remote computer and that you install the AppManager for VoIP Quality module on the proxy agent computer. Module installation installs the latest endpoint. However, automatic installation of the Performance Endpoint might fail on computers where you have enabled User Access Control (UAC). To workaround this problem, disable UAC during installation, or install the Performance Endpoint manually. You can download the free endpoint software from the Current Performance Endpoints Product Upgrades page.

For more information about monitoring between endpoints, see Section 1.3, "Reviewing AppManager for Call Performance," on page 10.

2.3 Installing the Module

Run the module installer only once on any computer. The module installer automatically identifies and updates all relevant AppManager components on a computer.

Access the `AM70-VoIPQuality-7.x.x.0.msi` module installer from the `AM70_VoIPQuality_7.x.x.0` self-extracting installation package on the AppManager Module Upgrades & Trials page.

You can install the Knowledge Scripts into local or remote AppManager repositories (QDBs). Install these components only once per QDB.

The module installer now installs Knowledge Scripts for each module directly into the QDB instead of to the `\AppManager\qdb\kp` folder as in previous releases of AppManager.

You can install the module manually, or you can use Control Center to deploy the module on a remote computer where an agent is installed. For more information, see Section 2.4, "Deploying the Module with Control Center," on page 22. However, if you do use Control Center to deploy the module, Control Center only installs the *agent* components of the module. The module installer installs the QDB and console components as well as the agent components on the agent computer.

To install the module manually:

1. Double-click the module installer `.msi` file.
2. Accept the license agreement.
3. Review the results of the pre-installation check. You can expect one of the following three scenarios:
    - *No AppManager agent is present*. In this scenario, the pre-installation check fails, and the installer does not install agent components.
    - *An AppManager agent is present, but some other prerequisite fails*. In this scenario, the default is to not install agent components because of one or more missing prerequisites. However, you can override the default by selecting **Install agent component locally**. A missing application server for this particular module often causes this scenario. For example, installing the AppManager for Microsoft SharePoint module requires the presence of a Microsoft SharePoint server on the selected computer.
    - *All prerequisites are met*. In this scenario, the installer will install the agent components.
4. To install the Knowledge Scripts into the QDB and to install the Analysis Center reports into the Analysis Center Configuration Database:

    4a Select **Install Knowledge Scripts** to install the repository components, including the Knowledge Scripts, object types, and SQL stored procedures.

    4b Select Install report package to install the Analysis Center reports.

    4c Specify the SQL Server name of the server hosting the QDB, as well as the case-sensitive QDB name.

    4d Specify the SQL Server name of the server hosting the Analysis Center Configuration Database.

5 ***If you use Control Center 7.x,*** run the module installer for each QDB attached to Control Center.

6 ***If you use Control Center 8.x,*** run the module installer only for the primary QDB, and Control Center will automatically replicate this module to secondary QDBs.

7 Run the module installer on all console computers to install the Help and console extensions.

8 ***If you will use a proxy agent computer to monitor call performance***, review deployment considerations before installing the module on the proxy agent computer. For more information, see Section 2.2, "Pre-installation Considerations," on page 20.

9 Run the module installer on all proxy agent computers to install the agent components.

10 ***If you will be monitoring the response time for H.323 call setup***, perform required configuration tasks. For more information, see Section 2.8, "Configuring the H.323 Managed Object Computer in Communications Manager," on page 29.

11 ***If you have not discovered VoIP Quality resources***, run one or more of the applicable Discovery Knowledge Scripts. For more information, see Section 2.5, "Discovering VoIP Quality Resources," on page 23.

12 To get the updates provided in this release, upgrade any running Knowledge Script jobs. For more information, see Section 2.6, "Upgrading Knowledge Script Jobs," on page 26.

After the installation has completed, the `VoIPQuality_Install.log` file, located in the `\NetIQ\Temp\NetIQ_Debug\`<*ServerName*> folder, lists any problems that occurred.

2.4 Deploying the Module with Control Center

You can use Control Center to deploy the module on a remote computer where an agent is installed. This topic briefly describes the steps involved in deploying a module and provides instructions for checking in the module installation package. For more information, see the *Control Center User Guide for AppManager*, which is available on the AppManager Documentation page.

2.4.1 Deployment Overview

This section describes the tasks required to deploy the module on an agent computer.

To deploy the module on an agent computer:

1. Verify the default deployment credentials.
2. Check in an installation package.
3. Configure an email address to receive notification of a deployment.
4. Create a deployment rule or modify an out-of-the-box deployment rule.
5. Approve the deployment task.
6. View the results.

2.4.2 Checking In the Installation Package

You must check in the installation package, `AM70-VoIPQuality-7.x.x.0.xml`, before you can deploy the module on an agent computer.

To check in a module installation package:

1. Log on to Control Center and navigate to the Administration pane.
2. In the Deployment folder, select **Packages**.
3. On the Tasks pane, click **Check in Packages**.
4. Navigate to the folder where you saved `AM70-VoIPQuality-7.x.x.0.xml` and select the file.
5. Click **Open**. The Deployment Package Check in Status dialog box displays the status of the package check in.

2.5 Discovering VoIP Quality Resources

Use the Discovery Knowledge Scripts to discover VoIP Quality Call Performance managed objects, Cisco SAA managed objects, and configuration and resource information for SIP (Session Initiation Protocol) and H.323.

2.5.1 Discovery_VoIPQuality_CallPerf

Use this Knowledge Script to discover the VoIP Quality Call Performance managed object on Windows servers on which the AppManager agent and a Performance Endpoint are installed. By default, this script runs once.

Set the Values tab parameters as needed:

Parameter	How To Set It
Raise event if discovery succeeds?	This script always raises an event when the job fails for any reason. In addition, you can set this parameter to **y** to raise an event when the job succeeds. The default is n.
Event severity when discovery succeeds	Set the event severity level, from 1 to 40, to reflect the importance of an event in which discovery succeeds. The default is 25.
Event severity when discovery fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which discovery fails. The default is 5.
Event severity when discovery partially succeeds	Set the event severity level, from 1 to 40, to reflect the importance of an event in which discovery returns some data but also generates warning messages. The default is 10.

2.5.2 Discovery_VoIPQuality_CallPerfProxy

Use this Knowledge Script to discover a Call Performance computer that does not have the AppManager agent installed but does have a NetIQ Performance Endpoint installed. Discovery is successful *only* if the managed object is able to communicate with the endpoint on the target computer.

Use this script when you want a single AppManager agent computer to act as a proxy for one or more Performance Endpoints installed on remote computers. When discovering Call Performance proxy resources, keep in mind the following:

- Run this script on only one computer at a time. Each remote computer should be associated with only one managed client computer that acts as its proxy.
- The endpoint on the remote computer must be running in order to be discovered. Discovery will fail if the managed client cannot contact the endpoint.

Only one computer should act as a proxy for a given remote computer. Therefore, run this script on only one computer at a time. By default, this script runs once.

Set the Values tab parameters as needed:

Parameter	How To Set It
List of remote computers to monitor	Provide a list of the remote computers for which you want to discover Call Performance resources. Specify at least one remote computer. Use a comma to separate the names in the list: `raldbellijm02,raldattixlm`
Full path to file with list of computers to monitor	Instead of listing each remote computer separately, you can specify the full path to a file on the agent computer that contains a computer name on each line of the file.
Raise event if discovery succeeds?	This script always raises an event when the job fails for any reason. In addition, you can set this parameter to **y** to raise an event when the job succeeds. The default is n.
Event severity when discovery succeeds	Set the event severity level, from 1 to 40, to reflect the importance of an event in which discovery succeeds. The default is 25.
Event severity when discovery fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which discovery fails. The default is 5.

2.5.3 Discovery_VoIPQuality_CiscoSAA

Use this Knowledge Script to discover a computer on which the Cisco SAA managed object, `qCiscoSAAa4.dll`, is installed. Discovery is successful *only* if the managed object is able to communicate with the Cisco router on the target computers.

Run this script on the Windows server that will act as proxy for all routers on which Cisco SAA software is installed. Only one computer should act as a proxy for any given set of routers. Therefore, run this script on only one computer at a time. By default, this script runs once.

Set the Values tab parameters as needed:

Parameter	How To Set It
List of remote computers to monitor	Provide a list of the remote computers for which you want to discover Cisco SAA resources. You must specify at least one remote computer. Use a comma to separate multiple names in the list: `raldserver02,raldserver03`
Full path to file with list of computers to monitor	Instead of listing each remote computer separately, you can specify the full path to a file on the agent computer that contains a computer name on each line of the file.
Raise event if discovery succeeds?	This script always raises an event when the job fails for any reason. In addition, you can set this parameter to **y** to raise an event when the job succeeds. The default is n.
Event severity when discovery succeeds	Set the event severity level, from 1 to 40, to reflect the importance of an event in which discovery succeeds. The default is 25.
Event severity when discovery fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which discovery fails. The default is 5.

2.5.4 Discovery_VoIPQuality_CallSetup_H.323

Use this Knowledge Script to discover the H.323 resources and configuration on a Windows server. By default, this script runs once.

Set the Values tab parameters as necessary:

Parameter	How To Set It
Raise event if discovery succeeds?	This script always raises an event when the discovery fails for any reason. In addition, you can set this parameter to **y** to raise an event when the job succeeds. The default is n.
Event severity when discovery succeeds	Set the event severity level, from 1 to 40, to reflect the importance of an event in which discovery succeeds. The default is 25.
Event severity when discovery fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which discovery fails. The default is 5.

2.5.5 Discovery_VoIPQuality_CallSetup_SIP

Use this Knowledge Script to discover SIP (Session Initiation Protocol) resources and configuration on a Windows server. By default, this script runs once.

Set the Values tab parameters as necessary:

Parameter	How To Set It
Raise event if discovery succeeds?	This script always raises an event when the discovery fails for any reason. In addition, you can set this parameter to **y** to raise an event when the job succeeds. The default is n.
Event severity when discovery succeeds	Set the event severity level, from 1 to 40, to reflect the importance of an event in which discovery succeeds. The default is 25.
Event severity when discovery fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which discovery fails. The default is 5.

2.6 Upgrading Knowledge Script Jobs

This release of AppManager for VoIP Quality may contain updated Knowledge Scripts. You can push the changes for updated scripts to running Knowledge Script jobs in one of the following ways:

- Use the AMAdmin_UpgradeJobs Knowledge Script.
- Use the Properties Propagation feature.

2.6.1 Running AMAdmin_UpgradeJobs

The AMAdmin_UpgradeJobs Knowledge Script can push changes to running Knowledge Script jobs. Your AppManager repository (`QDB`) must be at version 7.0 or later. In addition, the repository computer must have hotfix 72040 installed, or the most recent AppManager Repository hotfix. To download the hotfix, see the AppManager Suite Hotfixes Web page.

Upgrading jobs to use the most recent script version allows the jobs to take advantage of the latest script logic while maintaining existing parameter values for the job.

For more information, see the Help for the AMAdmin_UpgradeJobs Knowledge Script.

2.6.2 Propagating Knowledge Script Changes

You can propagate script changes to jobs that are running and to Knowledge Script Groups, including recommended Knowledge Script Groups and renamed Knowledge Scripts.

Before propagating script changes, verify that the script parameters are set to your specifications. Customized script parameters may have reverted to default parameters during the installation of the module. New parameters may need to be set appropriately for your environment or application.

You can choose to propagate only properties (specified in the Schedule and Values tabs), only the script (which is the logic of the Knowledge Script), or both. Unless you know specifically that changes affect only the script logic, you should propagate both properties and the script.

For more information about propagating Knowledge Script changes, see the "Running Monitoring Jobs" chapter of the *Operator Console User Guide for AppManager*.

Propagating Changes to Ad Hoc Jobs

You can propagate the properties and the logic (script) of a Knowledge Script to ad hoc jobs started by that Knowledge Script. Corresponding jobs are stopped and restarted with the Knowledge Script changes.

To propagate changes to ad hoc Knowledge Script jobs:

1. In the Knowledge Script view, select the Knowledge Script for which you want to propagate changes.
2. Click **Properties Propagation** > **Ad Hoc Jobs**.
3. Select the components of the Knowledge Script that you want to propagate to associated ad hoc jobs:

Select	To propagate
Script	The logic of the Knowledge Script.
Properties	Values from the Knowledge Script Schedule and Values tabs, such as schedule, monitoring values, actions, and advanced options.

Propagating Changes to Knowledge Script Groups

You can propagate the properties and logic (script) of a Knowledge Script to corresponding Knowledge Script Group members.

After you propagate script changes to Knowledge Script Group members, you can propagate the updated Knowledge Script Group members to associated running jobs. For more information, see "Propagating Changes to Ad Hoc Jobs" on page 27.

To propagate Knowledge Script changes to Knowledge Script Groups:

1. In the Knowledge Script view, select the Knowledge Script Group for which you want to propagate changes.
2. On the KS menu, select **Properties propagation** > **Ad Hoc Jobs**.
3. ***If you want to exclude a Knowledge Script member from properties propagation***, deselect that member from the list in the Properties Propagation dialog box.
4. Select the components of the Knowledge Script that you want to propagate to associated Knowledge Script Groups:

Select	To propagate
Script	The logic of the Knowledge Script.
Properties	Values from the Knowledge Script Schedule and Values tabs, including the schedule, actions, and Advanced properties.

5. Click **OK**. Any monitoring jobs started by a Knowledge Script Group member are restarted with the job properties of the Knowledge Script Group member.

2.7 Considerations for Cisco SAA Agent Installation

Before monitoring call performance between Cisco SAA-enabled routers, you must configure SNMP permissions in Security Manager, configure your SAA router as a responder, ensure sufficient memory, and synchronize clocks.

2.7.1 Configuring SNMP Permissions in Security Manager

Before you can discover Cisco SAA resources, configure AppManager Security Manager with the SNMP community string information for the routers on which you will be running VoIP tests.

If your community string information is the same for all Cisco SAA-enabled routers, then complete the following procedure once. If your community string information is different for different routers, then complete the following procedure once for each different community string.

After you configure all your SNMP community strings, run Discovery_VoIPQuality_CiscoSAA on the computer that will be used as the proxy for all routers on which the Cisco SAA software is installed. Ensure you have installed the Cisco SAA managed object, `qCiscoSAAa4.dll`, on the proxy computer.

Complete the following fields in the Custom tab of Security Manager for the proxy agent computer.

Field	Description
Label	`SAA`
Sub-label	Indicates whether the community string information will be used for a single router or for all routers. ◆ *For a single router*, provide the name of the router. ◆ *For all routers using the proxy agent computer*, provide the name of the proxy agent computer. ◆ *For all routers*, type `default`.
Value 1	Read/write community string value, such as private or public.

2.7.2 Configuring SAA Router as Responder

The Cisco Service Assurance Agent (SAA) allows network performance monitoring between a Cisco router and a remote device, such as another Cisco router or an IP host. SAA, or the SAA Response Time Reporter (RTR), supports running jitter and delay tests on many Cisco routers. Two routers, each running SAA, are required: one is a packet sender, the other a responder. To test jitter, the responder must be enabled.

To find out whether a router has SAA enabled, issue the following at a command prompt:

```
show rtr responder
```

To enable SAA, issue the following commands:

```
configure terminal
rtr responder
```

2.7.3 Ensuring Sufficient Router Memory for SAA Tests

Occasionally, a router may not have enough memory available to run an SAA test. The amount of memory available to SAA RTR operations is a configurable value; about 25% of the router's total memory is the recommended amount. To set or change the amount of memory available to SAA RTR operations, issue the following commands:

`show memory summary` (to show the largest process memory block available)

`show rtr application` (to show what the memory water mark is currently set to)

From router config, you can set the memory watermark by executing the following command:

```
rtr low-memory <value>
```

2.7.4 Synchronizing Clocks

In order for a VoIPQuality_CiscoSAA Knowledge Script to retrieve delay data from Cisco SAA-enabled routers, the clocks on all the routers must be synchronized with one other.

2.8 Configuring the H.323 Managed Object Computer in Communications Manager

Before running gateway tests, configure Cisco Unified Communications Manager to recognize the name of the listener and talker computers on which you installed the H.323 managed object, `qH323a4.dll`.

To configure Cisco Unified Communications Manager:

1. Navigate to your Cisco CallManager Administration Web page.
2. Provide the appropriate information in the User Name and Password text boxes.
3. On the Device menu, click Phone.
4. On the Find and List Phones page, click Add a New Phone.
5. In the Phone Type field, select H.323 client, and then click Next.
6. In the Device Name field, type the name of the device that you are adding. For example, type the hostname of the Performance Endpoint computer on which the H.323 managed object is installed.
7. In the Description field, type a description of the device. For example, type the hostname followed by the phone number extension: `ralduser01-1505`. Consider entering a description using terminology that will simplify subsequent searches.
8. In the Device pool field, select Default, and then click Insert.
9. When prompted to enter a directory number, which is the same as the phone number extension, click OK.
10. Enter the Directory Number (such as 1505), and then click Insert. You are prompted to return to the Phone Configuration page.
11. Click Reset Phone.
12. Click Reset. The alias information is stored in CallManager.

13 Discover resources using the Discovery_VoIPQuality_CallSetup_H.323 Knowledge Script.

14 Use the H.323_UpdateAlias Knowledge Script to change the alias of *each* talker and listener resource to the Directory Number you configured on the Communications Manager.

3 Reporting with Analysis Center

NetIQ Analysis Center is designed to import raw data from multiple AppManager repositories, transform that data into useful information about the computing infrastructure that supports your business, and publish that information in the form of reports.

Beginning with version 2.6, Analysis Center ships with operational and service level reports designed specifically for VoIP quality data. With these reports, you can capture and distribute vital information, such as MOS, R-value, jitter, data loss, and delay.

You can find the reports within the **Reports > AppManager > VoIP Quality** folder in the Analysis Center Navigation pane. These reports have been configured to filter for VoIP and call quality data, so you can use them pretty much right out of the box.

3.1 Operational Reports

The operational side of your organization may be one of the most vital in terms of VoIP functionality. Operational reports provide the details behind the service-level management reports and help you isolate servers that are experiencing problems.

The following table describes the operational reports available from Analysis Center for VoIP and call quality data. For more information, see the Configuration Card details for each report.

Report Name	Description
VoIP Quality Availability	Displays the availability of the various VoIP Quality or call setup tests that are being run by AppManager. Use the Metric context to select the types of tests you are interested in; expand the Metric folder to select the specific instances (data streams) for which you want to see the availability. This report presents the availability of your VoIP Quality Knowledge Script test results as a percentage. A test is considered unavailable if the Knowledge Script cannot run for any reason, such as network connectivity.
VoIP Quality Good-Acceptable-Poor Metric by Stream	Displays the VoIP Quality metrics between each pair of endpoints (in other words, a listener/talker pair). Use the Metric context to expand the individual metric you want to display and then select each individual stream instance. By default, thresholds are set for the Delay metric. Use the **Properties** section of the **Parameters** tab to adjust these thresholds for the metric you are including in the report.

Report Name	Description
VoIP Quality Performance Data Metrics by Date and Time	Compares multiple metrics by date and time. For example, you can compare the percent of lost data vs. the percent of loss due to the jitter buffer. Or compare multiple instances of the same metric, such as the average jitter between each of your VoIP Quality endpoints. Use the Metric context to select the metrics to include in the report. To select specific instances of a metric, expand the metric description and then select the instances that you want to include in the report. Use the Time context to set the time range and interval (for example, **Last 28 Days** by **Day**). The interval you select determines the time aggregation (for example, if you select **Day**, there is one value for each date; if you select **Hour**, there are 24 values for each date). Use the other context controls as data filters. For example, use the Group context to select which computers or groups to include in the report.
VoIP Quality Performance Data Metrics by Hour	Examines individual or multiple metrics by hour of the day. For example, use this report to look at the average or maximum delay being measured between the VoIP Quality endpoints by the hour of the day. This type of report can give you insight into whether call quality is being affected during specific times of the day. Use the Metric context to select one or more metrics to include in the report. To select specific instances of a metric, expand the metric description and then select the appropriate instances that you want to include in the report. Use the other context controls as data filters.

3.2 Service Level Reports

The reporting capability of Analysis Center enables you to demonstrate the value of IT and how well IT is aligned with business objectives. To these ends, run the service level management reports to reflect server availability and call quality.

The following table describes the service level reports available from Analysis Center for VoIP and call quality data. For more information, see the Configuration Card details for each report.

Report Name	Description
VoIP Quality Delay Good-Acceptable-Poor	Displays a pie chart of good, acceptable, and poor delay values. By default, this report uses the **Delay of VoIP Call Performance** data stream generated by the AppManager VoIPQuality_CallPerf_(codec) Knowledge Script. Good delay values are those below 150 ms.; poor delay values are those above 400 ms. To change these values, use the **Parameters** section of the **Properties** tab. Use the other contexts to filter data. For example, to show results for a particular codec type, use the Knowledge Script context to select the Knowledge Script running tests for that codec.
VoIP Quality Jitter Good-Acceptable-Poor	Displays a pie chart of good, acceptable, and poor jitter values. By default, this report uses the **Jitter of VoIP Call Performance** data stream generated by the AppManager VoIPQuality_CallPerf_(codec) Knowledge Script. Good jitter values are those below 40 ms.; poor jitter values are those above 60 ms. To change these standards, use the **Parameters** section of the **Properties** tab. Use the other contexts to filter data. For example, to show results for a particular codec type, use the Knowledge Script context to select the Knowledge Script running tests for that codec.

Report Name	Description
VoIP Quality Lost Data Good-Acceptable-Poor	Displays a pie chart of good, acceptable, and poor lost data values. By default, this report uses the **Percent Lost Data of VoIP Call Performance** data stream generated by the AppManager VoIPQuality_CallPerf_(codec) Knowledge Script. The Good Standard is data loss below 0.50%; the Poor Standard is data loss above 1.00%. To change these standards, use the **Parameters** section of the **Properties** tab. Use the other contexts to filter data. For example, to show results for a particular codec type, use the Knowledge Script context to select the Knowledge Script running tests for that codec.
VoIP Quality MOS Good-Acceptable-Poor	Displays a pie chart of good, acceptable, and poor MOS (mean opinion score) values. By default, this report uses the **MOS score of VoIP Call Performance** data stream generated by the AppManager VoIPQuality_CallPerf_(codec) Knowledge Script. Good MOS scores are those above 4.03; poor MOS scores are those below 3.6. To change this report to show R-value instead, use the Metric context to select **R-Value of VoIP Call Performance**, and then change the **Good** and **Poor Standards** defined in the **Parameters** section of the **Properties** tab. If you are using R-value, set the **Good Standard** to a value above 80 and the **Poor Standard** to a value below 70. Use the other contexts to filter data. For example, to show results for a particular codec type, use the Knowledge Script context to select the Knowledge Script used for that codec.
VoIP Quality Service Levels Overview	This dashboard report provides an overview of underlying reports that reflect key VoIP quality metrics. The member reports show the good-acceptable-poor levels for MOS, jitter, delay, and lost data. Click on the title of any member report to see the full view of that report. When deploying this report, be sure to deploy each member report first.

4 VoIPQuality Knowledge Scripts

From the Knowledge Script view of Control Center, you can access more information about any NetIQ-supported Knowledge Script by selecting it and clicking **Help**. Or in the Operator Console, click any Knowledge Script in the Knowledge Script pane and press **F1**.

Knowledge Script	What It Does
CallPerf_G711a	Runs a VoIP test between Performance Endpoints for the G.711a codec.
CallPerf_G711u	Runs a VoIP test between Performance Endpoints for the G.711u codec.
CallPerf_G723.1-ACELP	Runs a VoIP test between Performance Endpoints for the G.723.1-ACELP codec.
CallPerf_G723.1-MPMLQ	Runs a VoIP test between Performance Endpoints for the G.723.1-MPMLQ codec.
CallPerf_G726	Runs a VoIP test between Performance Endpoints for the G.726 codec.
CallPerf_G729	Runs a VoIP test between Performance Endpoints for the G.729 codec.
CallPerf_G729A	Runs a VoIP test between Performance Endpoints for the G.729 Annex A codec.
CiscoSAA_G711a	Runs a VoIP test between Cisco SAA-enabled routers for the G.711a codec.
CiscoSAA_G711u	Runs a VoIP test between Cisco SAA-enabled routers for the G.711u codec.
CiscoSAA_G723.1-ACELP	Runs a VoIP test between Cisco SAA-enabled routers for the G.723.1-ACELP codec.
CiscoSAA_G723.1-MPMLQ	Runs a VoIP test between Cisco SAA-enabled routers for the G.723.1-MPMLQ codec.
CiscoSAA_G726	Runs a VoIP test between Cisco SAA-enabled routers for the G.726 codec.
CiscoSAA_G729	Runs a VoIP test between Cisco SAA-enabled routers for the G.729 codec.
CiscoSAA_G729A	Runs a VoIP test between Cisco SAA-enabled routers for the G.729 Annex A codec.
Report_Configuration	Summarizes VoIP Quality configuration information and summary information about which talkers are talking to which listeners for selected computers.
Report_GroupSummary	Summarizes group-to-group comparisons for VoIP Quality data streams.
Report_MOSAvailMatrix	Summarizes the average MOS and availability for valid talker-listener pairs.
Report_MOSSummary	Summarizes group-to-group comparisons for MOS breakdown.

Knowledge Script	What It Does
Report_RvalueSummary	Summarizes group-to-group comparisons for R-value breakdown.
Report_TimeDetail	Summarizes time series charts for selected groups.
Report_VoIPQualitySummary	Summarizes VoIP quality statistics: MOS, delay, jitter, and packet loss.

4.1 CallPerf_G711a

Use this Knowledge Script to run a VoIP test between Performance Endpoints using the G.711a codec, which uses the A-law for companding, a popular standard in Europe. This script raises an event if a metric exceeds or falls below a threshold and generates data streams for network delay, MOS, R-value, delay, jitter, jitter buffer loss, and lost data.

4.1.1 Understanding Packet Loss Concealment

Packet loss concealment (PLC) is enabled by default in the G.711u and G.711a codecs. PLC describes a number of techniques for minimizing or masking the effects of data loss during a VoIP conversation. When PLC is enabled, the adverse affects of data loss are not as severe. AppManager calculates the call quality, factoring in the behavior of PLC, which is enabled by default in the CallPerf_G711a and CallPerf_G711u Knowledge Scripts.

In the following table, *packetization delay* refers to the delay these codecs introduce as they convert a signal from analog to digital; this delay is included in the MOS estimate, as is the *jitter buffer delay*, the delay introduced by the effects of buffering to reduce interarrival delay variations.

Codec	Default Data Rate	Default Datagram Size	Packetization Delay	Default Jitter Buffer Delay	Theoretical Maximum MOS
G.711u G.711a	64 kbps	20 ms	1.0 ms	2 datagrams (40 ms)	4.40
G.726	32kbps	20 ms	1.25 ms	2 datagrams (40 ms)	4.22
G.729 G.729A	8 kbps	20 ms	35.0 ms	2 datagrams (40 ms)	4.07
G.723.1-MPMLQ	6.3 kbps	30 ms	67.5 ms	3 datagrams (60 ms)	3.87
G.723.1-ACELP	5.3 kbps	30 ms	67.5 ms	3 datagrams (60 ms)	3.69

4.1.2 Resource Objects

Call Perf object

Call Perf proxy object

When you run this script on an agent computer that acts as proxy for multiple remote computers, AppManager creates only one job that drives the tests for all talkers on that computer. These tests run simultaneously. Running multiple tests at one time can take an undesirable toll on your bandwidth resources. Use the Objects tab on the Knowledge Script Properties dialog box to include or exclude remote resources from the tests.

4.1.3 Default Schedule

By default, this script runs every 15 minutes.

4.1.4 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
General Settings	
Select listener(s)	Select the listener computers from the Select Desired Computers dialog box.
Collect data?	Select **Yes** to collect data about MOS, R-value, delay, jitter, jitter buffer loss, and lost data for charts and graphs. The default is Yes.
Collect network delay data?	Select **Yes** to collect data about network delay for charts and graphs. The default is unselected.
Configuration Settings	
Test duration	Specify the duration of a test event in seconds, between one and 300. The default is 60 seconds.
Service Quality	Select a DiffServ (Differentiated Services) codepoint for classifying the bits in the IP header: ◆ **None**. Default setting. No special treatment is given to packets. ◆ **EF0-101000**. Deprecated Expedited Flow codepoint in use by most phones. Equivalent to the TOS "CRITIC/ECP" setting reserved for voice. ◆ **EF-101110**. Expedited Forwarding per-hop behavior (PHB) codepoint, represents the highest-priority service. ◆ **AF-011000**. Deprecated Assured Flow per-hop behavior (PHB) codepoint, represents a medium-quality service. Equivalent to the TOS "flash" setting. ◆ **AF11-001010** (Assured Forwarding, Class 1, low drop precedence) ◆ **AF12 - 001100** (Assured Forwarding, Class 1, medium drop precedence) ◆ **AF13-001110** (Assured Forwarding, Class 1, high drop precedence) ◆ **AF2 -010010** (Assured Forwarding, Class 2, low drop precedence) ◆ **AF22-010100** (Assured Forwarding, Class 2, medium drop precedence) ◆ **AF23-010110** (Assured Forwarding, Class 2, high drop precedence) ◆ **AF31 011010** (Assured Forwarding, Class 3, low drop precedence) ◆ **AF32-011100** (Assured Forwarding, Class 3, medium drop precedence) ◆ **AF33-011110** (Assured Forwarding, Class 3, high drop precedence) ◆ **AF4 -100010** (Assured Forwarding, Class 4, low drop precedence) ◆ **AF42-100100** (Assured Forwarding, Class 4, medium drop precedence) ◆ **AF43-100110** (Assured Forwarding, Class 4, high drop precedence) ◆ **802.1p-011** (For medium-priority traffic, often used for call setup packets) ◆ **802.1p-101** (For high-priority traffic, recommended for VoIP data packets)

Parameter	How To Set It
Use Service Quality in data stream legend?	Select **Yes** to allow service quality to be used in the dynamic legend for data streams. If you select Yes, the job generates unique data stream legends based on Quality of Service (QoS) settings as well as Talker endpoint (E1) and Listener endpoint (E2) settings. Analysis Center does not collapse unique data stream legends. However, Analysis Center does collapse data stream legends that are not unique, like the legends that get created if you select No for this parameter.
Voice activity rate	Specify a voice activity rate percentage. For example, enter 50 to indicate that data is being sent during 50% of a call's duration. The default is 50%.
Delay between voice datagrams	Specify a delay, in milliseconds, between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Use silence suppression?	Select **Yes** to enable silence suppression, which means that no data is sent during periods of “call silence” (i.e. when no one is talking). The default is unselected.
Enable packet loss concealment?	Select **Yes** to enable packet loss concealment (PLC). PLC is a technique for minimizing or masking the effects of data loss. When PLC is enabled, AppManager assumes that the quality of a conversation would be improved, but the improvement is factored into a MOS calculation **only** if data is lost. The default is Yes.
Advanced Configuration Settings	
Absolute jitter buffer size	Specify the size of the absolute jitter buffer in milliseconds. The jitter buffer size is a critical component of the MOS calculation. For example, a jitter buffer of 43 ms could hold two 20-ms datagram packets and allow for three extra milliseconds of variability. The default is 40 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Event Notification	
Raise event if test fails?	Select **Yes** to raise an event if the VoIP test does not complete successfully. The default is Yes.
Event severity when test fails	Set the event severity level, from 1 to 40, to indicate the importance of an event in which the VoIP test does not complete successfully. The default is 5.
Raise event if MOS/R-value falls below threshold	Select **Yes** to raise an event if the MOS score or R-value falls below the threshold that you set. The default is Yes.

Parameter	How To Set It
VoIP quality metric for event	Select the VoIP quality metric you want to use for the VoIP test. The default is MOS. ◆ **MOS**. The Mean Opinion Score (MOS) is an overall score representing the quality of a call. The MOS is a number between 1 and 5. A MOS of 5 is excellent; a MOS of 1 is unacceptably bad. The MOS is calculated based on measured items plus jitter buffer size. The jitter buffer size is constant based on the codec/script. Only one jitter buffer size is specified and used throughout the test. ◆ **R-value**. A single score that is derived from delays and equipment impairment factors. An R-value can be mapped to an estimated MOS. R-values range from 100 (excellent) to 0 (poor).
Threshold - Minimum MOS	Specify the minimum Mean Opinion Score (MOS) that must be reached to prevent an event from being raised. The default is 3.60.
Threshold - Minimum R-value	Specify the minimum acceptable R-value that must be reached to prevent an event from being raised. The default is 70.
Event severity when VoIP quality falls below threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which MOS or R-value falls below the threshold that you set. The default is 5.
Raise event if delay exceeds threshold?	Select **Yes** to raise an event if delay exceeds the threshold that you set. The default is Yes.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets. The default is 400 ms.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold that you set. The default is 15.
Raise event if jitter exceeds threshold?	Select **Yes** to raise an event if jitter exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets. The default is 60 ms.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold that you set. The default is 15.
Raise event if jitter buffer loss exceeds threshold?	Select **Yes** to raise an event if jitter buffer loss exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter buffer loss	Specify the maximum percentage of jitter buffer loss that can occur before an event is raised. The default is 1.0%. Jitter buffer loss is the amount of data that is lost when jitter exceeds that which the jitter buffer can hold. Jitter buffer loss affects call clarity, which affects the overall MOS score.
Event severity when jitter buffer loss exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter buffer loss exceeds the threshold that you set. The default is 15.
Raise event if lost data exceeds threshold?	Select **Yes** to raise an event if lost data exceeds the threshold that you set. The default is Yes.

Parameter	How To Set It
Threshold - Maximum lost data	Specify the maximum percentage of lost data that can occur before an event is raised. The default is 1.0%. To calculate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold that you set. The default is 15.

4.2 CallPerf_G711u

Use this Knowledge Script to run a VoIP test between Performance Endpoints using the G.711u codec, which uses the u-law for companding, the most frequently used method in the USA. This script raises an event if a metric exceeds or falls below a threshold and generates data streams for network delay, MOS, R-value, delay, jitter, jitter buffer loss, and lost data.

4.2.1 Understanding Packet Loss Concealment

Packet loss concealment (PLC) is enabled by default in the G.711u and G.711a codecs. PLC describes a number of techniques for minimizing or masking the effects of data loss during a VoIP conversation. When PLC is enabled, the adverse affects of data loss are not as severe. AppManager calculates the call quality, factoring in the behavior of PLC, which is enabled by default in the CallPerf_G711a and CallPerf_G711u Knowledge Scripts.

In the following table, *packetization delay* refers to the delay these codecs introduce as they convert a signal from analog to digital; this delay is included in the MOS estimate, as is the *jitter buffer delay*, the delay introduced by the effects of buffering to reduce interarrival delay variations.

Codec	Default Data Rate	Default Datagram Size	Packetization Delay	Default Jitter Buffer Delay	Theoretical Maximum MOS
G.711u G.711a	64 kbps	20 ms	1.0 ms	2 datagrams (40 ms)	4.40
G.726	32kbps	20 ms	1.25 ms	2 datagrams (40 ms)	4.22
G.729 G.729A	8 kbps	20 ms	35.0 ms	2 datagrams (40 ms)	4.07
G.723.1-MPMLQ	6.3 kbps	30 ms	67.5 ms	3 datagrams (60 ms)	3.87
G.723.1-ACELP	5.3 kbps	30 ms	67.5 ms	3 datagrams (60 ms)	3.69

4.2.2 Resource Objects

Call Perf object

Call Perf proxy object

When you run this script on an agent computer that acts as proxy for multiple remote computers, AppManager creates only one job that drives the tests for all talkers on that computer. These tests run simultaneously. Running multiple tests at one time can take an undesirable toll on your bandwidth resources. Use the Objects tab on the Knowledge Script Properties dialog box to include or exclude remote resources from the tests.

4.2.3 Default Schedule

By default, this script runs every 15 minutes.

4.2.4 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
General Settings	
Select listener(s)	Select the listener computers from the Select Desired Computers dialog box.
Collect data?	Select **Yes** to collect data about MOS, R-value, delay, jitter, jitter buffer loss, and lost data for charts and graphs. The default is Yes.
Collect network delay data?	Select **Yes** to collect data about network delay for charts and graphs. The default is unselected.
Configuration Settings	
Test duration	Specify the duration of a test event in seconds, between one and 300. The default is 60 seconds.

Parameter	How To Set It
Service Quality	Select a DiffServ (Differentiated Services) codepoint for classifying the bits in the IP header: ◆ **None**. Default setting. No special treatment is given to packets. ◆ **EF0-101000**. Deprecated Expedited Flow codepoint in use by most phones. Equivalent to the TOS "CRITIC/ECP" setting reserved for voice. ◆ **EF-101110**. Expedited Forwarding per-hop behavior (PHB) codepoint, represents the highest-priority service. ◆ **AF-011000**. Deprecated Assured Flow per-hop behavior (PHB) codepoint, represents a medium-quality service. Equivalent to the TOS "flash" setting. ◆ **AF11-001010** (Assured Forwarding, Class 1, low drop precedence) ◆ **AF12 - 001100** (Assured Forwarding, Class 1, medium drop precedence) ◆ **AF13-001110** (Assured Forwarding, Class 1, high drop precedence) ◆ **AF2 -010010** (Assured Forwarding, Class 2, low drop precedence) ◆ **AF22-010100** (Assured Forwarding, Class 2, medium drop precedence) ◆ **AF23-010110** (Assured Forwarding, Class 2, high drop precedence) ◆ **AF31 011010** (Assured Forwarding, Class 3, low drop precedence) ◆ **AF32-011100** (Assured Forwarding, Class 3, medium drop precedence) ◆ **AF33-011110** (Assured Forwarding, Class 3, high drop precedence) ◆ **AF4 -100010** (Assured Forwarding, Class 4, low drop precedence) ◆ **AF42-100100** (Assured Forwarding, Class 4, medium drop precedence) ◆ **AF43-100110** (Assured Forwarding, Class 4, high drop precedence) ◆ **802.1p-011** (For medium-priority traffic, often used for call setup packets) ◆ **802.1p-101** (For high-priority traffic, recommended for VoIP data packets)
Use Service Quality in data stream legend?	Select **Yes** to allow service quality to be used in the dynamic legend for data streams. If you select Yes, the job generates unique data stream legends based on Quality of Service (QoS) settings as well as Talker endpoint (E1) and Listener endpoint (E2) settings. Analysis Center does not collapse unique data stream legends. However, Analysis Center does collapse data stream legends that are not unique, like the legends that get created if you select No for this parameter.
Voice activity rate	Specify a voice activity rate percentage. For example, enter 50 to indicate that data is being sent during 50% of a call's duration. The default is 50%.
Delay between voice datagrams	Specify a delay, in milliseconds, between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Use silence suppression?	Select **Yes** to enable silence suppression, which means that no data is sent during periods of “call silence” (i.e. when no one is talking). The default is unselected.

Parameter	How To Set It
Enable packet loss concealment?	Select **Yes** to enable packet loss concealment (PLC). PLC is a technique for minimizing or masking the effects of data loss. When PLC is enabled, AppManager assumes that the quality of a conversation would be improved, but the improvement is factored into a MOS calculation **only** if data is lost. The default is Yes.
Advanced Configuration Settings	
Absolute jitter buffer size	Specify the size of the absolute jitter buffer in milliseconds. The jitter buffer size is a critical component of the MOS calculation. For example, a jitter buffer of 43 ms could hold two 20-ms datagram packets and allow for three extra milliseconds of variability. The default is 40 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Event Notification	
Raise event if test fails?	Select **Yes** to raise an event if the VoIP test does not complete successfully. The default is Yes.
Event severity when test fails	Set the event severity level, from 1 to 40, to indicate the importance of an event in which the VoIP test does not complete successfully. The default is 5.
Raise event if MOS/R-value falls below threshold	Select **Yes** to raise an event if the MOS score or R-value falls below the threshold that you set. The default is Yes.
VoIP quality metric for event	Select the VoIP quality metric you want to use for the VoIP test. The default is MOS. ◆ **MOS**. The Mean Opinion Score (MOS) is an overall score representing the quality of a call. The MOS is a number between 1 and 5. A MOS of 5 is excellent; a MOS of 1 is unacceptably bad. The MOS is calculated based on measured items plus jitter buffer size. The jitter buffer size is constant based on the codec/script. Only one jitter buffer size is specified and used throughout the test. ◆ **R-value**. A single score that is derived from delays and equipment impairment factors. An R-value can be mapped to an estimated MOS. R-values range from 100 (excellent) to 0 (poor).
Threshold - Minimum MOS	Specify the minimum Mean Opinion Score (MOS) that must be reached to prevent an event from being raised. The default is 3.60.
Threshold - Minimum R-value	Specify the minimum acceptable R-value that must be reached to prevent an event from being raised. The default is 70.
Event severity when VoIP quality falls below threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which MOS or R-value falls below the threshold that you set. The default is 5.

Parameter	How To Set It
Raise event if delay exceeds threshold?	Select **Yes** to raise an event if delay exceeds the threshold that you set. The default is Yes.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets. The default is 400 ms.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold that you set. The default is 15.
Raise event if jitter exceeds threshold?	Select **Yes** to raise an event if jitter exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets. The default is 60 ms.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold that you set. The default is 15.
Raise event if jitter buffer loss exceeds threshold?	Select **Yes** to raise an event if jitter buffer loss exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter buffer loss	Specify the maximum percentage of jitter buffer loss that can occur before an event is raised. The default is 1.0%. Jitter buffer loss is the amount of data that is lost when jitter exceeds that which the jitter buffer can hold. Jitter buffer loss affects call clarity, which affects the overall MOS score.
Event severity when jitter buffer loss exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter buffer loss exceeds the threshold that you set. The default is 15.
Raise event if lost data exceeds threshold?	Select **Yes** to raise an event if lost data exceeds the threshold that you set. The default is Yes.
Threshold - Maximum lost data	Specify the maximum percentage of lost data that can occur before an event is raised. The default is 1.0%. To calculate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold that you set. The default is 15.

4.3 CallPerf_G723.1-ACELP

Use this Knowledge Script to run a VoIP test between Performance Endpoints using the G.723.1-ACELP codec, which uses the conjugate structure algebraic code excited linear predictive compression (ACELP) algorithm. This script raises an event if a metric exceeds or falls below a threshold and generates data streams for network delay, MOS, R-value, delay, jitter, jitter buffer loss, and lost data.

4.3.1 Resource Objects

Call Perf object

Call Perf proxy object

When you run this script on an agent computer that acts as proxy for multiple remote computers, AppManager creates only one job that drives the tests for all talkers on that computer. These tests run simultaneously. Running multiple tests at one time can take an undesirable toll on your bandwidth resources. Use the Objects tab on the Knowledge Script Properties dialog box to include or exclude remote resources from the tests.

4.3.2 Default Schedule

By default, this script runs every 15 minutes.

4.3.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
General Settings	
Select listener(s)	Select the listener computers from the Select Desired Computers dialog box.
Collect data?	Select **Yes** to collect data about MOS, R-value, delay, jitter, jitter buffer loss, and lost data for charts and graphs. The default is Yes.
Collect network delay data?	Select **Yes** to collect data about network delay for charts and graphs. The default is unselected.
Configuration Settings	
Test duration	Specify the duration of a test event in seconds, between one and 300. The default is 60 seconds.

Parameter	How To Set It
Service Quality	Select a DiffServ (Differentiated Services) codepoint for classifying the bits in the IP header: ◆ **None**. Default setting. No special treatment is given to packets. ◆ **EF0-101000**. Deprecated Expedited Flow codepoint in use by most phones. Equivalent to the TOS "CRITIC/ECP" setting reserved for voice. ◆ **EF-101110**. Expedited Forwarding per-hop behavior (PHB) codepoint, represents the highest-priority service. ◆ **AF-011000**. Deprecated Assured Flow per-hop behavior (PHB) codepoint, represents a medium-quality service. Equivalent to the TOS "flash" setting. ◆ **AF11-001010** (Assured Forwarding, Class 1, low drop precedence) ◆ **AF12 - 001100** (Assured Forwarding, Class 1, medium drop precedence) ◆ **AF13-001110** (Assured Forwarding, Class 1, high drop precedence) ◆ **AF2 -010010** (Assured Forwarding, Class 2, low drop precedence) ◆ **AF22-010100** (Assured Forwarding, Class 2, medium drop precedence) ◆ **AF23-010110** (Assured Forwarding, Class 2, high drop precedence) ◆ **AF31 011010** (Assured Forwarding, Class 3, low drop precedence) ◆ **AF32-011100** (Assured Forwarding, Class 3, medium drop precedence) ◆ **AF33-011110** (Assured Forwarding, Class 3, high drop precedence) ◆ **AF4 -100010** (Assured Forwarding, Class 4, low drop precedence) ◆ **AF42-100100** (Assured Forwarding, Class 4, medium drop precedence) ◆ **AF43-100110** (Assured Forwarding, Class 4, high drop precedence) ◆ **802.1p-011** (For medium-priority traffic, often used for call setup packets) ◆ **802.1p-101** (For high-priority traffic, recommended for VoIP data packets)
Use Service Quality in data stream legend?	Select **Yes** to allow service quality to be used in the dynamic legend for data streams. If you select Yes, the job generates unique data stream legends based on Quality of Service (QoS) settings as well as Talker endpoint (E1) and Listener endpoint (E2) settings. Analysis Center does not collapse unique data stream legends. However, Analysis Center does collapse data stream legends that are not unique, like the legends that get created if you select No for this parameter.
Voice activity rate	Specify a voice activity rate percentage. For example, enter 50 to indicate that data is being sent during 50% of a call's duration. The default is 50%.
Delay between voice datagrams	Specify a delay, in milliseconds, between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Use silence suppression?	Select **Yes** to enable silence suppression, which means that no data is sent during periods of “call silence” (i.e. when no one is talking). The default is unselected.
Advanced Configuration Settings	

Parameter	How To Set It
Absolute jitter buffer size	Specify the size of the absolute jitter buffer in milliseconds. The jitter buffer size is a critical component of the MOS calculation. For example, a jitter buffer of 43 ms could hold two 20-ms datagram packets and allow for three extra milliseconds of variability. The default is 40 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Event Notification	
Raise event if test fails?	Select **Yes** to raise an event if the VoIP test does not complete successfully. The default is Yes.
Event severity when test fails	Set the event severity level, from 1 to 40, to indicate the importance of an event in which the VoIP test does not complete successfully. The default is 5.
Raise event if MOS/R-value falls below threshold	Select **Yes** to raise an event if the MOS score or R-value falls below the threshold that you set. The default is Yes.
VoIP quality metric for event	Select the VoIP quality metric you want to use for the VoIP test. The default is MOS. ◆ **MOS**. The Mean Opinion Score (MOS) is an overall score representing the quality of a call. The MOS is a number between 1 and 5. A MOS of 5 is excellent; a MOS of 1 is unacceptably bad. The MOS is calculated based on measured items plus jitter buffer size. The jitter buffer size is constant based on the codec/script. Only one jitter buffer size is specified and used throughout the test. ◆ **R-value**. A single score that is derived from delays and equipment impairment factors. An R-value can be mapped to an estimated MOS. R-values range from 100 (excellent) to 0 (poor).
Threshold - Minimum MOS	Specify the minimum Mean Opinion Score (MOS) that must be reached to prevent an event from being raised. The default is 3.60.
Threshold - Minimum R-value	Specify the minimum acceptable R-value that must be reached to prevent an event from being raised. The default is 70.
Event severity when VoIP quality falls below threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which MOS or R-value falls below the threshold that you set. The default is 5.
Raise event if delay exceeds threshold?	Select **Yes** to raise an event if delay exceeds the threshold that you set. The default is Yes.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets. The default is 400 ms.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold that you set. The default is 15.

Parameter	How To Set It
Raise event if jitter exceeds threshold?	Select **Yes** to raise an event if jitter exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets. The default is 60 ms.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold that you set. The default is 15.
Raise event if jitter buffer loss exceeds threshold?	Select **Yes** to raise an event if jitter buffer loss exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter buffer loss	Specify the maximum percentage of jitter buffer loss that can occur before an event is raised. The default is 1.0%. Jitter buffer loss is the amount of data that is lost when jitter exceeds that which the jitter buffer can hold. Jitter buffer loss affects call clarity, which affects the overall MOS score.
Event severity when jitter buffer loss exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter buffer loss exceeds the threshold that you set. The default is 15.
Raise event if lost data exceeds threshold?	Select **Yes** to raise an event if lost data exceeds the threshold that you set. The default is Yes.
Threshold - Maximum lost data	Specify the maximum percentage of lost data that can occur before an event is raised. The default is 1.0%. To calculate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold that you set. The default is 15.

4.4 CallPerf_G723.1-MPMLQ

Use this Knowledge Script to run a VoIP test between Performance Endpoints using the G.723.1-MPMLQ codec, which uses the multipulse maximum likelihood quantization (MPMLQ) compression algorithm. This script raises an event if a metric exceeds or falls below a threshold and generates data streams for network delay, MOS, R-value, delay, jitter, jitter buffer loss, and lost data.

4.4.1 Resource Objects

Call Perf object

Call Perf proxy object

When you run this script on an agent computer that acts as proxy for multiple remote computers, AppManager creates only one job that drives the tests for all talkers on that computer. These tests run simultaneously. Running multiple tests at one time can take an undesirable toll on your bandwidth resources. Use the Objects tab on the Knowledge Script Properties dialog box to include or exclude remote resources from the tests.

4.4.2 Default Schedule

By default, this script runs every 15 minutes.

4.4.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
General Settings	
Select listener(s)	Select the listener computers from the Select Desired Computers dialog box.
Collect data?	Select **Yes** to collect data about MOS, R-value, delay, jitter, jitter buffer loss, and lost data for charts and graphs. The default is Yes.
Collect network delay data?	Select **Yes** to collect data about network delay for charts and graphs. The default is unselected.
Configuration Settings	
Test duration	Specify the duration of a test event in seconds, between one and 300. The default is 60 seconds.
Service Quality	Select a DiffServ (Differentiated Services) codepoint for classifying the bits in the IP header: ◆ **None**. Default setting. No special treatment is given to packets. ◆ **EF0-101000**. Deprecated Expedited Flow codepoint in use by most phones. Equivalent to the TOS "CRITIC/ECP" setting reserved for voice. ◆ **EF-101110**. Expedited Forwarding per-hop behavior (PHB) codepoint, represents the highest-priority service. ◆ **AF-011000**. Deprecated Assured Flow per-hop behavior (PHB) codepoint, represents a medium-quality service. Equivalent to the TOS "flash" setting. ◆ **AF11-001010** (Assured Forwarding, Class 1, low drop precedence) ◆ **AF12 - 001100** (Assured Forwarding, Class 1, medium drop precedence) ◆ **AF13-001110** (Assured Forwarding, Class 1, high drop precedence) ◆ **AF2 -010010** (Assured Forwarding, Class 2, low drop precedence) ◆ **AF22-010100** (Assured Forwarding, Class 2, medium drop precedence) ◆ **AF23-010110** (Assured Forwarding, Class 2, high drop precedence) ◆ **AF31 011010** (Assured Forwarding, Class 3, low drop precedence) ◆ **AF32-011100** (Assured Forwarding, Class 3, medium drop precedence) ◆ **AF33-011110** (Assured Forwarding, Class 3, high drop precedence) ◆ **AF4 -100010** (Assured Forwarding, Class 4, low drop precedence) ◆ **AF42-100100** (Assured Forwarding, Class 4, medium drop precedence) ◆ **AF43-100110** (Assured Forwarding, Class 4, high drop precedence) ◆ **802.1p-011** (For medium-priority traffic, often used for call setup packets) ◆ **802.1p-101** (For high-priority traffic, recommended for VoIP data packets)

Parameter	How To Set It
Use Service Quality in data stream legend?	Select **Yes** to allow service quality to be used in the dynamic legend for data streams. If you select Yes, the job generates unique data stream legends based on Quality of Service (QoS) settings as well as Talker endpoint (E1) and Listener endpoint (E2) settings. Analysis Center does not collapse unique data stream legends. However, Analysis Center does collapse data stream legends that are not unique, like the legends that get created if you select No for this parameter.
Voice activity rate	Specify a voice activity rate percentage. For example, enter 50 to indicate that data is being sent during 50% of a call's duration. The default is 50%.
Delay between voice datagrams	Specify a delay, in milliseconds, between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Use silence suppression?	Select **Yes** to enable silence suppression, which means that no data is sent during periods of “call silence” (i.e. when no one is talking). The default is unselected.
Advanced Configuration Settings	
Absolute jitter buffer size	Specify the size of the absolute jitter buffer in milliseconds. The jitter buffer size is a critical component of the MOS calculation. For example, a jitter buffer of 43 ms could hold two 20-ms datagram packets and allow for three extra milliseconds of variability. The default is 40 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Event Notification	
Raise event if test fails?	Select **Yes** to raise an event if the VoIP test does not complete successfully. The default is Yes.
Event severity when test fails	Set the event severity level, from 1 to 40, to indicate the importance of an event in which the VoIP test does not complete successfully. The default is 5.
Raise event if MOS/R-value falls below threshold	Select **Yes** to raise an event if the MOS score or R-value falls below the threshold that you set. The default is Yes.

Parameter	How To Set It
VoIP quality metric for event	Select the VoIP quality metric you want to use for the VoIP test. The default is MOS. ◆ **MOS**. The Mean Opinion Score (MOS) is an overall score representing the quality of a call. The MOS is a number between 1 and 5. A MOS of 5 is excellent; a MOS of 1 is unacceptably bad. The MOS is calculated based on measured items plus jitter buffer size. The jitter buffer size is constant based on the codec/script. Only one jitter buffer size is specified and used throughout the test. ◆ **R-value**. A single score that is derived from delays and equipment impairment factors. An R-value can be mapped to an estimated MOS. R-values range from 100 (excellent) to 0 (poor).
Threshold - Minimum MOS	Specify the minimum Mean Opinion Score (MOS) that must be reached to prevent an event from being raised. The default is 3.60.
Threshold - Minimum R-value	Specify the minimum acceptable R-value that must be reached to prevent an event from being raised. The default is 70.
Event severity when VoIP quality falls below threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which MOS or R-value falls below the threshold that you set. The default is 5.
Raise event if delay exceeds threshold?	Select **Yes** to raise an event if delay exceeds the threshold that you set. The default is Yes.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets. The default is 400 ms.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold that you set. The default is 15.
Raise event if jitter exceeds threshold?	Select **Yes** to raise an event if jitter exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets. The default is 60 ms.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold that you set. The default is 15.
Raise event if jitter buffer loss exceeds threshold?	Select **Yes** to raise an event if jitter buffer loss exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter buffer loss	Specify the maximum percentage of jitter buffer loss that can occur before an event is raised. The default is 1.0%. Jitter buffer loss is the amount of data that is lost when jitter exceeds that which the jitter buffer can hold. Jitter buffer loss affects call clarity, which affects the overall MOS score.
Event severity when jitter buffer loss exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter buffer loss exceeds the threshold that you set. The default is 15.
Raise event if lost data exceeds threshold?	Select **Yes** to raise an event if lost data exceeds the threshold that you set. The default is Yes.

Parameter	How To Set It
Threshold - Maximum lost data	Specify the maximum percentage of lost data that can occur before an event is raised. The default is 1.0%. To calculate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold that you set. The default is 15.

4.5 CallPerf_G726

Use this Knowledge Script to run a VoIP test between Performance Endpoints using the G.726 codec, a waveform coder that uses Adaptive Differential Pulse Code Modulation (ADPCM). ADPCM is a variation of pulse code modulation (PCM), which sends only the difference between two adjacent samples, producing a lower bit rate. This script raises an event if a metric exceeds or falls below a threshold and generates data streams for network delay, MOS, R-value, delay, jitter, jitter buffer loss, and lost data.

4.5.1 Resource Objects

Call Perf object

Call Perf proxy object

When you run this script on an agent computer that acts as proxy for multiple remote computers, AppManager creates only one job that drives the tests for all talkers on that computer. These tests run simultaneously. Running multiple tests at one time can take an undesirable toll on your bandwidth resources. Use the Objects tab on the Knowledge Script Properties dialog box to include or exclude remote resources from the tests.

4.5.2 Default Schedule

By default, this script runs every 15 minutes.

4.5.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
General Settings	
Select listener(s)	Select the listener computers from the Select Desired Computers dialog box.
Collect data?	Select **Yes** to collect data about MOS, R-value, delay, jitter, jitter buffer loss, and lost data for charts and graphs. The default is Yes.
Collect network delay data?	Select **Yes** to collect data about network delay for charts and graphs. The default is unselected.
Configuration Settings	
Test duration	Specify the duration of a test event in seconds, between one and 300. The default is 60 seconds.

Parameter	How To Set It
Service Quality	Select a DiffServ (Differentiated Services) codepoint for classifying the bits in the IP header: ◆ **None**. Default setting. No special treatment is given to packets. ◆ **EF0-101000**. Deprecated Expedited Flow codepoint in use by most phones. Equivalent to the TOS "CRITIC/ECP" setting reserved for voice. ◆ **EF-101110**. Expedited Forwarding per-hop behavior (PHB) codepoint, represents the highest-priority service. ◆ **AF-011000**. Deprecated Assured Flow per-hop behavior (PHB) codepoint, represents a medium-quality service. Equivalent to the TOS "flash" setting. ◆ **AF11-001010** (Assured Forwarding, Class 1, low drop precedence) ◆ **AF12 - 001100** (Assured Forwarding, Class 1, medium drop precedence) ◆ **AF13-001110** (Assured Forwarding, Class 1, high drop precedence) ◆ **AF2 -010010** (Assured Forwarding, Class 2, low drop precedence) ◆ **AF22-010100** (Assured Forwarding, Class 2, medium drop precedence) ◆ **AF23-010110** (Assured Forwarding, Class 2, high drop precedence) ◆ **AF31 011010** (Assured Forwarding, Class 3, low drop precedence) ◆ **AF32-011100** (Assured Forwarding, Class 3, medium drop precedence) ◆ **AF33-011110** (Assured Forwarding, Class 3, high drop precedence) ◆ **AF4 -100010** (Assured Forwarding, Class 4, low drop precedence) ◆ **AF42-100100** (Assured Forwarding, Class 4, medium drop precedence) ◆ **AF43-100110** (Assured Forwarding, Class 4, high drop precedence) ◆ **802.1p-011** (For medium-priority traffic, often used for call setup packets) ◆ **802.1p-101** (For high-priority traffic, recommended for VoIP data packets)
Use Service Quality in data stream legend?	Select **Yes** to allow service quality to be used in the dynamic legend for data streams. If you select Yes, the job generates unique data stream legends based on Quality of Service (QoS) settings as well as Talker endpoint (E1) and Listener endpoint (E2) settings. Analysis Center does not collapse unique data stream legends. However, Analysis Center does collapse data stream legends that are not unique, like the legends that get created if you select No for this parameter.
Voice activity rate	Specify a voice activity rate percentage. For example, enter 50 to indicate that data is being sent during 50% of a call's duration. The default is 50%.
Delay between voice datagrams	Specify a delay, in milliseconds, between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Use silence suppression?	Select **Yes** to enable silence suppression, which means that no data is sent during periods of “call silence” (i.e. when no one is talking). The default is unselected.
Advanced Configuration Settings	

Parameter	How To Set It
Absolute jitter buffer size	Specify the size of the absolute jitter buffer in milliseconds. The jitter buffer size is a critical component of the MOS calculation. For example, a jitter buffer of 43 ms could hold two 20-ms datagram packets and allow for three extra milliseconds of variability. The default is 40 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Event Notification	
Raise event if test fails?	Select **Yes** to raise an event if the VoIP test does not complete successfully. The default is Yes.
Event severity when test fails	Set the event severity level, from 1 to 40, to indicate the importance of an event in which the VoIP test does not complete successfully. The default is 5.
Raise event if MOS/R-value falls below threshold	Select **Yes** to raise an event if the MOS score or R-value falls below the threshold that you set. The default is Yes.
VoIP quality metric for event	Select the VoIP quality metric you want to use for the VoIP test. The default is MOS. ◆ **MOS**. The Mean Opinion Score (MOS) is an overall score representing the quality of a call. The MOS is a number between 1 and 5. A MOS of 5 is excellent; a MOS of 1 is unacceptably bad. The MOS is calculated based on measured items plus jitter buffer size. The jitter buffer size is constant based on the codec/script. Only one jitter buffer size is specified and used throughout the test. ◆ **R-value**. A single score that is derived from delays and equipment impairment factors. An R-value can be mapped to an estimated MOS. R-values range from 100 (excellent) to 0 (poor).
Threshold - Minimum MOS	Specify the minimum Mean Opinion Score (MOS) that must be reached to prevent an event from being raised. The default is 3.60.
Threshold - Minimum R-value	Specify the minimum acceptable R-value that must be reached to prevent an event from being raised. The default is 70.
Event severity when VoIP quality falls below threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which MOS or R-value falls below the threshold that you set. The default is 5.
Raise event if delay exceeds threshold?	Select **Yes** to raise an event if delay exceeds the threshold that you set. The default is Yes.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets. The default is 400 ms.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold that you set. The default is 15.

Parameter	How To Set It
Raise event if jitter exceeds threshold?	Select **Yes** to raise an event if jitter exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets. The default is 60 ms.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold that you set. The default is 15.
Raise event if jitter buffer loss exceeds threshold?	Select **Yes** to raise an event if jitter buffer loss exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter buffer loss	Specify the maximum percentage of jitter buffer loss that can occur before an event is raised. The default is 1.0%. Jitter buffer loss is the amount of data that is lost when jitter exceeds that which the jitter buffer can hold. Jitter buffer loss affects call clarity, which affects the overall MOS score.
Event severity when jitter buffer loss exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter buffer loss exceeds the threshold that you set. The default is 15.
Raise event if lost data exceeds threshold?	Select **Yes** to raise an event if lost data exceeds the threshold that you set. The default is Yes.
Threshold - Maximum lost data	Specify the maximum percentage of lost data that can occur before an event is raised. The default is 1.0%. To calculate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold that you set. The default is 15.

4.6 CallPerf_G729

Use this Knowledge Script to run a VoIP test between Performance Endpoints using the G.729 codec, which offers compression with high quality. This script raises an event if a metric exceeds or falls below a threshold and generates data streams for network delay, MOS, R-value, delay, jitter, jitter buffer loss, and lost data.

4.6.1 Resource Objects

Call Perf object

Call Perf proxy object

When you run this script on an agent computer that acts as proxy for multiple remote computers, AppManager creates only one job that drives the tests for all talkers on that computer. These tests run simultaneously. Running multiple tests at one time can take an undesirable toll on your bandwidth resources. Use the Objects tab on the Knowledge Script Properties dialog box to include or exclude remote resources from the tests.

4.6.2 Default Schedule

By default, this script runs every 15 minutes.

4.6.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
General Settings	
Select listener(s)	Select the listener computers from the Select Desired Computers dialog box.
Collect data?	Select **Yes** to collect data about MOS, R-value, delay, jitter, jitter buffer loss, and lost data for charts and graphs. The default is Yes.
Collect network delay data?	Select **Yes** to collect data about network delay for charts and graphs. The default is unselected.
Configuration Settings	
Test duration	Specify the duration of a test event in seconds, between one and 300. The default is 60 seconds.
Service Quality	Select a DiffServ (Differentiated Services) codepoint for classifying the bits in the IP header: ◆ **None**. Default setting. No special treatment is given to packets. ◆ **EF0-101000**. Deprecated Expedited Flow codepoint in use by most phones. Equivalent to the TOS "CRITIC/ECP" setting reserved for voice. ◆ **EF-101110**. Expedited Forwarding per-hop behavior (PHB) codepoint, represents the highest-priority service. ◆ **AF-011000**. Deprecated Assured Flow per-hop behavior (PHB) codepoint, represents a medium-quality service. Equivalent to the TOS "flash" setting. ◆ **AF11-001010** (Assured Forwarding, Class 1, low drop precedence) ◆ **AF12 - 001100** (Assured Forwarding, Class 1, medium drop precedence) ◆ **AF13-001110** (Assured Forwarding, Class 1, high drop precedence) ◆ **AF2 -010010** (Assured Forwarding, Class 2, low drop precedence) ◆ **AF22-010100** (Assured Forwarding, Class 2, medium drop precedence) ◆ **AF23-010110** (Assured Forwarding, Class 2, high drop precedence) ◆ **AF31 011010** (Assured Forwarding, Class 3, low drop precedence) ◆ **AF32-011100** (Assured Forwarding, Class 3, medium drop precedence) ◆ **AF33-011110** (Assured Forwarding, Class 3, high drop precedence) ◆ **AF4 -100010** (Assured Forwarding, Class 4, low drop precedence) ◆ **AF42-100100** (Assured Forwarding, Class 4, medium drop precedence) ◆ **AF43-100110** (Assured Forwarding, Class 4, high drop precedence) ◆ **802.1p-011** (For medium-priority traffic, often used for call setup packets) ◆ **802.1p-101** (For high-priority traffic, recommended for VoIP data packets)

Parameter	How To Set It
Use Service Quality in data stream legend?	Select **Yes** to allow service quality to be used in the dynamic legend for data streams. If you select Yes, the job generates unique data stream legends based on Quality of Service (QoS) settings as well as Talker endpoint (E1) and Listener endpoint (E2) settings. Analysis Center does not collapse unique data stream legends. However, Analysis Center does collapse data stream legends that are not unique, like the legends that get created if you select No for this parameter.
Voice activity rate	Specify a voice activity rate percentage. For example, enter 50 to indicate that data is being sent during 50% of a call's duration. The default is 50%.
Delay between voice datagrams	Specify a delay, in milliseconds, between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Use silence suppression?	Select **Yes** to enable silence suppression, which means that no data is sent during periods of “call silence” (i.e. when no one is talking). The default is unselected.
Advanced Configuration Settings	
Absolute jitter buffer size	Specify the size of the absolute jitter buffer in milliseconds. The jitter buffer size is a critical component of the MOS calculation. For example, a jitter buffer of 43 ms could hold two 20-ms datagram packets and allow for three extra milliseconds of variability. The default is 40 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Event Notification	
Raise event if test fails?	Select **Yes** to raise an event if the VoIP test does not complete successfully. The default is Yes.
Event severity when test fails	Set the event severity level, from 1 to 40, to indicate the importance of an event in which the VoIP test does not complete successfully. The default is 5.
Raise event if MOS/R-value falls below threshold	Select **Yes** to raise an event if the MOS score or R-value falls below the threshold that you set. The default is Yes.

Parameter	How To Set It
VoIP quality metric for event	Select the VoIP quality metric you want to use for the VoIP test. The default is MOS. ◆ **MOS**. The Mean Opinion Score (MOS) is an overall score representing the quality of a call. The MOS is a number between 1 and 5. A MOS of 5 is excellent; a MOS of 1 is unacceptably bad. The MOS is calculated based on measured items plus jitter buffer size. The jitter buffer size is constant based on the codec/script. Only one jitter buffer size is specified and used throughout the test. ◆ **R-value**. A single score that is derived from delays and equipment impairment factors. An R-value can be mapped to an estimated MOS. R-values range from 100 (excellent) to 0 (poor).
Threshold - Minimum MOS	Specify the minimum Mean Opinion Score (MOS) that must be reached to prevent an event from being raised. The default is 3.60.
Threshold - Minimum R-value	Specify the minimum acceptable R-value that must be reached to prevent an event from being raised. The default is 70.
Event severity when VoIP quality falls below threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which MOS or R-value falls below the threshold that you set. The default is 5.
Raise event if delay exceeds threshold?	Select **Yes** to raise an event if delay exceeds the threshold that you set. The default is Yes.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets. The default is 400 ms.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold that you set. The default is 15.
Raise event if jitter exceeds threshold?	Select **Yes** to raise an event if jitter exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets. The default is 60 ms.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold that you set. The default is 15.
Raise event if jitter buffer loss exceeds threshold?	Select **Yes** to raise an event if jitter buffer loss exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter buffer loss	Specify the maximum percentage of jitter buffer loss that can occur before an event is raised. The default is 1.0%. Jitter buffer loss is the amount of data that is lost when jitter exceeds that which the jitter buffer can hold. Jitter buffer loss affects call clarity, which affects the overall MOS score.
Event severity when jitter buffer loss exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter buffer loss exceeds the threshold that you set. The default is 15.
Raise event if lost data exceeds threshold?	Select **Yes** to raise an event if lost data exceeds the threshold that you set. The default is Yes.

Parameter	How To Set It
Threshold - Maximum lost data	Specify the maximum percentage of lost data that can occur before an event is raised. The default is 1.0%. To calculate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold that you set. The default is 15.

4.7 CallPerf_G729A

Use this Knowledge Script to run a VoIP test between Performance Endpoints using the G.729 Annex A codec, which offers compression with high quality. This script raises an event if a metric exceeds or falls below a threshold and generates data streams for network delay, MOS, R-value, delay, jitter, jitter buffer loss, and lost data.

4.7.1 Resource Objects

Call Perf object

Call Perf proxy object

When you run this script on an agent computer that acts as proxy for multiple remote computers, AppManager creates only one job that drives the tests for all talkers on that computer. These tests run simultaneously. Running multiple tests at one time can take an undesirable toll on your bandwidth resources. Use the Objects tab on the Knowledge Script Properties dialog box to include or exclude remote resources from the tests.

4.7.2 Default Schedule

By default, this script runs every 15 minutes.

4.7.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
General Settings	
Select listener(s)	Select the listener computers from the Select Desired Computers dialog box.
Collect data?	Select **Yes** to collect data about MOS, R-value, delay, jitter, jitter buffer loss, and lost data for charts and graphs. The default is Yes.
Collect network delay data?	Select **Yes** to collect data about network delay for charts and graphs. The default is unselected.
Configuration Settings	
Test duration	Specify the duration of a test event in seconds, between one and 300. The default is 60 seconds.

Parameter	How To Set It
Service Quality	Select a DiffServ (Differentiated Services) codepoint for classifying the bits in the IP header: ◆ **None**. Default setting. No special treatment is given to packets. ◆ **EF0-101000**. Deprecated Expedited Flow codepoint in use by most phones. Equivalent to the TOS "CRITIC/ECP" setting reserved for voice. ◆ **EF-101110**. Expedited Forwarding per-hop behavior (PHB) codepoint, represents the highest-priority service. ◆ **AF-011000**. Deprecated Assured Flow per-hop behavior (PHB) codepoint, represents a medium-quality service. Equivalent to the TOS "flash" setting. ◆ **AF11-001010** (Assured Forwarding, Class 1, low drop precedence) ◆ **AF12 - 001100** (Assured Forwarding, Class 1, medium drop precedence) ◆ **AF13-001110** (Assured Forwarding, Class 1, high drop precedence) ◆ **AF2 -010010** (Assured Forwarding, Class 2, low drop precedence) ◆ **AF22-010100** (Assured Forwarding, Class 2, medium drop precedence) ◆ **AF23-010110** (Assured Forwarding, Class 2, high drop precedence) ◆ **AF31 011010** (Assured Forwarding, Class 3, low drop precedence) ◆ **AF32-011100** (Assured Forwarding, Class 3, medium drop precedence) ◆ **AF33-011110** (Assured Forwarding, Class 3, high drop precedence) ◆ **AF4 -100010** (Assured Forwarding, Class 4, low drop precedence) ◆ **AF42-100100** (Assured Forwarding, Class 4, medium drop precedence) ◆ **AF43-100110** (Assured Forwarding, Class 4, high drop precedence) ◆ **802.1p-011** (For medium-priority traffic, often used for call setup packets) ◆ **802.1p-101** (For high-priority traffic, recommended for VoIP data packets)
Use Service Quality in data stream legend?	Select **Yes** to allow service quality to be used in the dynamic legend for data streams. If you select Yes, the job generates unique data stream legends based on Quality of Service (QoS) settings as well as Talker endpoint (E1) and Listener endpoint (E2) settings. Analysis Center does not collapse unique data stream legends. However, Analysis Center does collapse data stream legends that are not unique, like the legends that get created if you select No for this parameter.
Voice activity rate	Specify a voice activity rate percentage. For example, enter 50 to indicate that data is being sent during 50% of a call's duration. The default is 50%.
Delay between voice datagrams	Specify a delay between voice datagrams. The delay determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Use silence suppression?	Select **Yes** to enable silence suppression, which means that no data is sent during periods of “call silence” (i.e. when no one is talking). The default is unselected.
Advanced Configuration Settings	

Parameter	How To Set It
Absolute jitter buffer size	Specify the size of the absolute jitter buffer in milliseconds. The jitter buffer size is a critical component of the MOS calculation. For example, a jitter buffer of 43 ms could hold two 20-ms datagram packets and allow for three extra milliseconds of variability. The default is 40 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Event Notification	
Raise event if test fails?	Select **Yes** to raise an event if the VoIP test does not complete successfully. The default is Yes.
Event severity when test fails	Set the event severity level, from 1 to 40, to indicate the importance of an event in which the VoIP test does not complete successfully. The default is 5.
Raise event if MOS/R-value falls below threshold	Select **Yes** to raise an event if the MOS score or R-value falls below the threshold that you set. The default is Yes.
VoIP quality metric for event	Select the VoIP quality metric you want to use for the VoIP test. The default is MOS. ◆ **MOS**. The Mean Opinion Score (MOS) is an overall score representing the quality of a call. The MOS is a number between 1 and 5. A MOS of 5 is excellent; a MOS of 1 is unacceptably bad. The MOS is calculated based on measured items plus jitter buffer size. The jitter buffer size is constant based on the codec/script. Only one jitter buffer size is specified and used throughout the test. ◆ **R-value**. A single score that is derived from delays and equipment impairment factors. An R-value can be mapped to an estimated MOS. R-values range from 100 (excellent) to 0 (poor).
Threshold - Minimum MOS	Specify the minimum Mean Opinion Score (MOS) that must be reached to prevent an event from being raised The default is 3.60.
Threshold - Minimum R-value	Specify the minimum acceptable R-value that must be reached to prevent an event from being raised. The default is 70.
Event severity when VoIP quality falls below threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which MOS or R-value falls below the threshold that you set. The default is 5.
Raise event if delay exceeds threshold?	Select **Yes** to raise an event if delay exceeds the threshold that you set. The default is Yes.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets. The default is 400 ms.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold that you set. The default is 15.

Parameter	How To Set It
Raise event if jitter exceeds threshold?	Select **Yes** to raise an event if jitter exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets. The default is 60 ms.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold that you set. The default is 15.
Raise event if jitter buffer loss exceeds threshold?	Select **Yes** to raise an event if jitter buffer loss exceeds the threshold that you set. The default is Yes.
Threshold - Maximum jitter buffer loss	Specify the maximum percentage of jitter buffer loss that can occur before an event is raised. The default is 1.0%. Jitter buffer loss is the amount of data that is lost when jitter exceeds that which the jitter buffer can hold. Jitter buffer loss affects call clarity, which affects the overall MOS score.
Event severity when jitter buffer loss exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter buffer loss exceeds the threshold that you set. The default is 15.
Raise event if lost data exceeds threshold?	Select **Yes** to raise an event if lost data exceeds the threshold that you set. The default is Yes.
Threshold - Maximum lost data	Specify the maximum percentage of lost data that can occur before an event is raised. The default is 1.0%. To calculate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold that you set. The default is 15.

4.8 CiscoSAA_G711a

Use this Knowledge Script to run a VoIP test between Cisco SAA-enabled routers using the G.711a codec, which uses the A-law for companding, a popular standard in Europe. This script raises an event if a threshold is exceeded. In addition, this script generates data streams for delay, jitter, and lost data.

4.8.1 Resource Object

Cisco SAA object

4.8.2 Default Schedule

By default, this script runs every 15 minutes.

4.8.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Collect data?	Select **y** to collect data about delay, jitter, and lost data for charts and graphs. The default is y.
Select target router(s)	Select the target routers (listeners) that you want to include in the VoIP test.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The default is 400 ms. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. The default is 60 ms. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets.
Threshold - Maximum lost data	Specify the maximum percentage of data that can be lost before an event is raised. The default is 1.0%. To generate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when test fails	Set a severity level, between 1 and 40, to indicate the importance of an event in which the VoIP test fails. The default is 5. Select 0 if you do not want to raise an event.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Test duration	Specify the duration of a test event in seconds. The default is 60 seconds.
Service Quality	Select a quality of service (QoS) option for the IP Type of Service (TOS) precedence bits in the IP header: ◆ **None - 000**. This is the default setting. No special treatment is given to packets. ◆ **DiffServAF - 011**. The DiffServ Assured Flow setting represents a medium-quality service. This setting is equivalent to the TOS "flash" setting. ◆ **DiffServEF - 101**. The DiffServ Expedited Flow setting represents the highest-priority service. This setting is equivalent to the TOS "CRITIC/ECP" setting reserved for voice.

Parameter	How To Set It
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Delay between voice datagrams	Specify a delay in milliseconds between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.

4.9 CiscoSAA_G711u

Use this Knowledge Script to run a VoIP test between Cisco SAA-enabled routers using the G.711u codec, which uses the u-law for companding, the most frequently used method in the USA. This script raises an event if a threshold is exceeded. In addition, this script generates data streams for delay, jitter, and lost data.

4.9.1 Resource Object

Cisco SAA object

4.9.2 Default Schedule

By default, this script runs every 15 minutes.

4.9.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Collect data?	Select **y** to collect data about delay, jitter, and lost data for charts and graphs. The default is y.
Select target router(s)	Select the target routers (listeners) that you want to include in the VoIP test.

Parameter	How To Set It
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The default is 400 ms. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. The default is 60 ms. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets.
Threshold - Maximum lost data	Specify the maximum percentage of data that can be lost before an event is raised. The default is 1.0%. To generate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when test fails	Set a severity level, between 1 and 40, to indicate the importance of an event in which the VoIP test fails. The default is 5. Select 0 if you do not want to raise an event.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Test duration	Specify the duration of a test event in seconds. The default is 60 seconds.
Service Quality	Select a quality of service (QoS) option for the IP Type of Service (TOS) precedence bits in the IP header: ◆ **None - 000**. This is the default setting. No special treatment is given to packets. ◆ **DiffServAF - 011**. The DiffServ Assured Flow setting represents a medium-quality service. This setting is equivalent to the TOS "flash" setting. ◆ **DiffServEF - 101**. The DiffServ Expedited Flow setting represents the highest-priority service. This setting is equivalent to the TOS "CRITIC/ECP" setting reserved for voice.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.

Parameter	How To Set It
Delay between voice datagrams	Specify a delay in milliseconds between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.

4.10 CiscoSAA_G723.1-ACELP

Use this Knowledge Script to run a VoIP test between Cisco SAA-enabled routers using the G.723.1-ACELP codec, which uses the conjugate structure algebraic code excited linear predictive compression (ACELP) algorithm. This script raises an event if a threshold is exceeded. In addition, this script generates data streams for delay, jitter, and lost data.

4.10.1 Resource Object

Cisco SAA object

4.10.2 Default Schedule

By default, this script runs every 15 minutes.

4.10.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Collect data?	Select **y** to collect data about delay, jitter, and lost data for charts and graphs. The default is y.
Select target router(s)	Select the target routers (listeners) that you want to include in the VoIP test.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The default is 400 ms. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets.

Parameter	How To Set It
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. The default is 60 ms. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets.
Threshold - Maximum lost data	Specify the maximum percentage of data that can be lost before an event is raised. The default is 1.0%. To generate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when test fails	Set a severity level, between 1 and 40, to indicate the importance of an event in which the VoIP test fails. The default is 5. Select 0 if you do not want to raise an event.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Test duration	Specify the duration of a test event in seconds. The default is 60 seconds.
Service Quality	Select a quality of service (QoS) option for the IP Type of Service (TOS) precedence bits in the IP header: ◆ **None - 000**. This is the default setting. No special treatment is given to packets. ◆ **DiffServAF - 011**. The DiffServ Assured Flow setting represents a medium-quality service. This setting is equivalent to the TOS "flash" setting. ◆ **DiffServEF - 101**. The DiffServ Expedited Flow setting represents the highest-priority service. This setting is equivalent to the TOS "CRITIC/ECP" setting reserved for voice.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Delay between voice datagrams	Specify a delay in milliseconds between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.

Parameter	How To Set It
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.

4.11 CiscoSAA_G723.1-MPMLQ

Use this Knowledge Script to run a VoIP test between Cisco SAA-enabled routers using the G.723.1-MPMLQ codec, which uses the multipulse maximum likelihood quantization (MPMLQ) compression algorithm. This script raises an event if a threshold is exceeded. In addition, this script generates data streams for delay, jitter, and lost data.

4.11.1 Resource Object

Cisco SAA object

4.11.2 Default Schedule

By default, this script runs every 15 minutes.

4.11.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Collect data?	Select **y** to collect data about delay, jitter, and lost data for charts and graphs. The default is y.
Select target router(s)	Select the target routers (listeners) that you want to include in the VoIP test.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The default is 400 ms. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. The default is 60 ms. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets.
Threshold - Maximum lost data	Specify the maximum percentage of data that can be lost before an event is raised. The default is 1.0%. To generate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.

Parameter	How To Set It
Event severity when test fails	Set a severity level, between 1 and 40, to indicate the importance of an event in which the VoIP test fails. The default is 5. Select 0 if you do not want to raise an event.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Test duration	Specify the duration of a test event in seconds. The default is 60 seconds.
Service Quality	Select a quality of service (QoS) option for the IP Type of Service (TOS) precedence bits in the IP header: ◆ **None - 000**. This is the default setting. No special treatment is given to packets. ◆ **DiffServAF - 011**. The DiffServ Assured Flow setting represents a medium-quality service. This setting is equivalent to the TOS "flash" setting. ◆ **DiffServEF - 101**. The DiffServ Expedited Flow setting represents the highest-priority service. This setting is equivalent to the TOS "CRITIC/ECP" setting reserved for voice.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Delay between voice datagrams	Specify a delay in milliseconds between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.

4.12 CiscoSAA_G726

Use this Knowledge Script to run a VoIP test between Cisco SAA-enabled routers using the G.726 codec, a waveform coder that uses Adaptive Differential Pulse Code Modulation (ADPCM). ADPCM is a variation of pulse code modulation (PCM), which sends only the difference between two adjacent samples, producing a lower bit rate. This script raises an event if a threshold is exceeded. In addition, this script generates data streams for delay, jitter, and lost data.

4.12.1 Resource Object

Cisco SAA object

4.12.2 Default Schedule

By default, this script runs every 15 minutes.

4.12.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Collect data?	Select **y** to collect data about delay, jitter, and lost data for charts and graphs. The default is y.
Select target router(s)	Select the target routers (listeners) that you want to include in the VoIP test.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The default is 400 ms. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. The default is 60 ms. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets.
Threshold - Maximum lost data	Specify the maximum percentage of data that can be lost before an event is raised. The default is 1.0%. To generate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when test fails	Set a severity level, between 1 and 40, to indicate the importance of an event in which the VoIP test fails. The default is 5. Select 0 if you do not want to raise an event.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.

Parameter	How To Set It
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Test duration	Specify the duration of a test event in seconds. The default is 60 seconds.
Service Quality	Select a quality of service (QoS) option for the IP Type of Service (TOS) precedence bits in the IP header: ◆ **None - 000**. This is the default setting. No special treatment is given to packets. ◆ **DiffServAF - 011**. The DiffServ Assured Flow setting represents a medium-quality service. This setting is equivalent to the TOS "flash" setting. ◆ **DiffServEF - 101**. The DiffServ Expedited Flow setting represents the highest-priority service. This setting is equivalent to the TOS "CRITIC/ECP" setting reserved for voice.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Delay between voice datagrams	Specify a delay in milliseconds between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.

4.13 CiscoSAA_G729

Use this Knowledge Script to run a VoIP test between Cisco SAA-enabled routers using the G.729 codec, which offers compression with high quality. This script raises an event if a threshold is exceeded. In addition, this script generates data streams for delay, jitter, and lost data.

4.13.1 Resource Object

Cisco SAA object

4.13.2 Default Schedule

By default, this script runs every 15 minutes.

4.13.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Collect data?	Select **y** to collect data about delay, jitter, and lost data for charts and graphs. The default is y.
Select target router(s)	Select the target routers (listeners) that you want to include in the VoIP test.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The default is 400 ms. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets.
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. The default is 60 ms. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets.
Threshold - Maximum lost data	Specify the maximum percentage of data that can be lost before an event is raised. The default is 1.0%. To generate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when test fails	Set a severity level, between 1 and 40, to indicate the importance of an event in which the VoIP test fails. The default is 5. Select 0 if you do not want to raise an event.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Test duration	Specify the duration of a test event in seconds. The default is 60 seconds.
Service Quality	Select a quality of service (QoS) option for the IP Type of Service (TOS) precedence bits in the IP header: ◆ **None - 000**. This is the default setting. No special treatment is given to packets. ◆ **DiffServAF - 011**. The DiffServ Assured Flow setting represents a medium-quality service. This setting is equivalent to the TOS "flash" setting. ◆ **DiffServEF - 101**. The DiffServ Expedited Flow setting represents the highest-priority service. This setting is equivalent to the TOS "CRITIC/ECP" setting reserved for voice.

Parameter	How To Set It
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Delay between voice datagrams	Specify a delay in milliseconds between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.

4.14 CiscoSAA_G729A

Use this Knowledge Script to run a VoIP test between Cisco SAA-enabled routers using the G.729 Annex A codec, which offers compression with high quality. This script raises an event if a threshold is exceeded. In addition, this script generates data streams for delay, jitter, and lost data.

4.14.1 Resource Object

Cisco SAA object

4.14.2 Default Schedule

By default, this script runs every 15 minutes.

4.14.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Collect data?	Select **y** to collect data about delay, jitter, and lost data for charts and graphs. The default is y.
Select target router(s)	Select the target routers (listeners) that you want to include in the VoIP test.
Threshold - Maximum delay	Specify the maximum amount of delay that can occur before an event is raised. The default is 400 ms. The delay is calculated on each packet. The delay of the test is the average of the delay for all packets.

Parameter	How To Set It
Threshold - Maximum jitter	Specify the maximum amount of jitter that can occur before an event is raised. The default is 60 ms. Jitter indicates the variation in packet arrival time. The jitter for the test is the average of the jitter between all packets.
Threshold - Maximum lost data	Specify the maximum percentage of data that can be lost before an event is raised. The default is 1.0%. To generate this percentage, the script divides the number of datagrams lost by the number of datagrams sent.
Event severity when test fails	Set a severity level, between 1 and 40, to indicate the importance of an event in which the VoIP test fails. The default is 5. Select 0 if you do not want to raise an event.
Event severity when delay exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which delay exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when jitter exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which jitter exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Event severity when lost data exceeds threshold	Set a severity level, between 1 and 40, to indicate the importance of an event in which lost data exceeds the threshold. The default is 15. Select 0 if you do not want to raise an event.
Test duration	Specify the duration of a test event in seconds. The default is 60 seconds.
Service Quality	Select a quality of service (QoS) option for the IP Type of Service (TOS) precedence bits in the IP header: ◆ **None - 000**. This is the default setting. No special treatment is given to packets. ◆ **DiffServAF - 011**. The DiffServ Assured Flow setting represents a medium-quality service. This setting is equivalent to the TOS "flash" setting. ◆ **DiffServEF - 101**. The DiffServ Expedited Flow setting represents the highest-priority service. This setting is equivalent to the TOS "CRITIC/ECP" setting reserved for voice.
Source port number	Specify the port number of the test source computer. Accept the default of AUTO to automatically locate the port number.
Destination port number	Specify the port number of the test destination computer. Accept the default of AUTO to automatically locate the port number.
Delay between voice datagrams	Specify a delay in milliseconds between voice datagrams. The delay between datagrams determines the size of the datagram packets. A delay of 20 ms means that accumulated data is sent every 20 milliseconds. A smaller delay value results in smaller, more frequent datagrams that increase processing overhead; a larger delay value results in fewer, larger datagrams that increase delay. The default is 20 ms.

Parameter	How To Set It
Additional fixed delay	Specify any additional fixed delay in milliseconds. Use this parameter to add a delay value from a known, constant source. For instance, if your test equipment adds 10 ms of delay to each datagram, enter 10 here. The default is 0 ms.

4.15 Report_Configuration

Use this Knowledge Script to summarize the VoIP Quality configuration for selected computers.

4.15.1 Resource Object

Report agent

4.15.2 Default Schedule

By default, this script runs once.

4.15.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Data Source	
Computer selection by	Select the computers that you want to include in the report. You can select computers by category: View, Server Group, or Computer. The default is View.
Select computers	Select the view in which you want to see the results. The default is Master.
Report Settings	
Include parameter help card?	Select **y** to include a table in the report that lists parameter settings for the script. The default is y.
Select output folder	Select the name and location of the folder in which the report will be saved. The default is VoIPQuality_Configuration.
Add job ID to output folder name?	Select **y** to add the job ID to the name of the output folder. The default is n. The job ID is helpful to make the correlation between a specific instance of a Report script and the corresponding report.
Select properties	Select and enter miscellaneous report properties in the Report Properties dialog box.
Add time stamp to title	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. Adding a time stamp is useful for running consecutive iterations of the same report without overwriting previous output.

Parameter	How To Set It
Event Notification	
Raise event if report succeeds?	This script automatically raises an event if the report is not generated successfully. Select **y** to raise an event when the report is generated successfully. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is successful. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

4.16 Report_GroupSummary

Use this Knowledge Script to summarize average VoIP Quality data stream values (MOS, R-value, availability, delay, jitter, jitter buffer loss, lost data) for a specified period.

4.16.1 Resource Object

Report agent

4.16.2 Default Schedule

By default, this script runs once.

4.16.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Data Source	
Values grouped by	Select the category by which you want to group data for display in the report. The default is Talker.
Select Knowledge Script(s)	Select the Knowledge Scripts that generated the data that you want to include in the report.
Computer selection by	Select the category by which you want to select the computers that you want to include in the report: View, Server Group, or Computer. The default is View.
Select computers	Select the computers that you want to include in the report.
Select time range	Select a **Specific** or **Sliding** date/time range from which the report should pull data. The default is Sliding.
Chart Thresholds	

Parameter	How To Set It
MOS threshold	Specify the MOS threshold to display on the MOS charts in the report. The default is 0.000.
R-value threshold	Specify the R-value threshold to display on the R-value charts in the report. The default is 0.000.
Availability threshold	Specify the Availability threshold to display on the Availability charts in the report. The default is 0%.
Delay threshold	Specify the Delay threshold to display on the Delay charts in the report. The default is 0 ms.
Jitter threshold	Specify the Jitter threshold to display on the Jitter charts in the report. The default is 0 ms.
Percent Jitter Buffer Loss threshold	Specify the Jitter Buffer Loss threshold to display on the Jitter Buffer Loss charts in the report. The default is 0.000%.
Percent Lost Data threshold	Specify the Lost Data threshold to display on the Percent Lost Data charts in the report. The default is 0.000%.
Report Settings	
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Include charts?	Select **y** to include a chart in the report. The default is y.
Include table?	Select **y** to include a table of information in the report. The default is y.
Select chart style	Select and enter chart properties in the Chart Settings dialog box. The default style is Bar.
Select output folder	Select the name and location of the folder in which the report will be saved. The default name is VoIPQualityGroupSum.
Add job ID to output folder name?	Select **y** to add the job ID to the name of the output folder. The default is n. The job ID is helpful for making the correlation between a specific instance of a Report script and the corresponding report.
Select properties	Select and enter report properties in the Report Properties dialog box. The default name is VoIP Quality Group Summary.
Add time stamp to title	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. Adding a time stamp is useful for running consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	This script automatically raises an event if the report is not generated successfully. Select **y** to raise an event when the report is generated successfully. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is successful. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.

Parameter	How To Set It
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

4.17 Report_MOSAvailMatrix

Use this Knowledge Script to summarize average MOS and availability between a talker and a listener within a selected period.

4.17.1 Resource Object

Report agent

4.17.2 Default Schedule

By default, this script runs once.

4.17.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Data Source	
Select Knowledge Script(s)	Select the Knowledge Scripts that generated the data that you want to include in the report.
Computer selection by	Select the category by which you want to select the computers that you want to include in the report: View, Server Group, or Computer. The default is View.
Select computers	Select the computers that you want to include in the report.
Select time range	Select a **Specific** or **Sliding** date/time range from which the report should pull data. The default is Sliding.
Report Settings	
Decimal accuracy for % values	Specify the number of decimal places that you want to see in the percentage values generated by this report. The default is 3.
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Select output folder	Select the name and location of the folder in which the report will be output. The default name is VoIPQuality_MOSAvailMatrix
Add job ID to output folder name?	Select **y** to add the job ID to the name of the output folder. The default is n. The job is helpful for making the correlation between a specific instance of a Report script and the corresponding report.
Select properties	Select and enter report properties in the Report Properties dialog box. The default report name is VoIP Quality MOS - Availability Matrix.

Parameter	How To Set It
Add time stamp to title	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. Adding a time stamp is useful for running consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	This script automatically raises an event if the report is not generated successfully. Select **y** to raise an event when the report is generated successfully. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is generated successfully. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

4.18 Report_MOSSummary

Use this Knowledge Script to summarize average MOS quality within a group within a selected time range.

4.18.1 Resource Object

Report agent

4.18.2 Default Schedule

By default, this script runs once.

4.18.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Data Source	
Calls grouped by	Select the category by which you want to group data for display in the report. The default is Talker.
Select Knowledge Script(s)	Select the Knowledge Scripts that generated the data that you want to include in the report.
Computer selection by	Select the category by which you want to select the computers that you want to include in the report: View, Server Group, or Computer. The default is View.
Select computers	Select the computers that you want to include in the report.

Parameter	How To Set It
Select time range	Select a **Specific** or **Sliding** date/time range from which the report should pull data. The default is Sliding.
Chart Thresholds	
Good-Acceptable threshold	Specify a MOS value below which the call is acceptable and equal to or above which the call is good. This value appears on the chart as a thick horizontal line. The default is 4.030.
Acceptable-Poor threshold	Specify a MOS value below which the call is poor and above which the call is acceptable. This value appears on the chart as a thick horizontal line. The default is 3.600.
Chart Settings	
Chart size	Select the size of the rendered chart. Choose from Large, Medium, and Small. The default is Medium.
Horizontal chart?	Select **y** to create a horizontal bar chart or accept the default to create a vertical bar chart.
Chart color scheme	Select a color scheme template. The default is NetIQ.
Report Settings	
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Include charts?	Select **y** to include a chart in the report. The default is y.
Include table?	Select **y** to include a table of information in the report. The default is y.
Select output folder	Select the name and location of the folder in which the report will be output. The default name is VoIPQualityMOSSum.
Add job ID to output folder name?	Select **y** to add the job ID to the name of the output folder. The default is n. The job ID is helpful for making the correlation between a specific instance of a Report script and the corresponding report.
Select properties	Select and enter report properties in the Report Properties dialog box. The default name is VoIP Quality MOS Summary.
Add time stamp to title	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. Adding a time stamp is useful for running consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	This script automatically raises an event if the report is not generated successfully. Select **y** to raise an event when the report is generated successfully. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is generated successful. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.

Parameter	How To Set It
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

4.19 Report_RvalueSummary

Use this Knowledge Script to summarize average R-value quality within a group within a selected time range.

4.19.1 Resource Object

Report agent

4.19.2 Default Schedule

By default, this script runs once.

4.19.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Data Source	
Calls grouped by	Select the category by which you want to group data for display in the report. The default is Talker.
Select Knowledge Script(s)	Select the Knowledge Scripts that generated the data that you want to include in the report.
Computer selection by	Select the category by which you want to select the computers that you want to include in the report: View, Server Group, or Computer. The default is View.
Select computers	Select the computers that you want to include in the report.
Select time range	Select a **Specific** or **Sliding** date/time range from which the report should pull data. The default is Sliding.
Chart Thresholds	
Good-Acceptable threshold	Specify an R-value below which the call is acceptable and equal to or above which the call is good. This value appears on the chart as a thick horizontal line. The default is 80.0.
Acceptable-Poor threshold	Specify an R-value below which the call is poor and above which the call is acceptable. This value appears on the chart as a thick horizontal line. The default is 70.0.
Chart Settings	
Chart size	Select the size of the rendered chart. Choose from Large, Medium, and Small. The default is Medium.

Parameter	How To Set It
Horizontal chart?	Select **y** to create a horizontal bar chart or accept the default to create a vertical bar chart.
Chart color scheme	Select a color scheme template. The default is NetIQ.
Report Settings	
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Include charts?	Select **y** to include a chart in the report. The default is y.
Include table?	Select **y** to include a table of information in the report. The default is y.
Select output folder	Select the name and location of the folder in which the report will be output. The default name is VoIPQualityRvalueSum.
Add job ID to output folder name?	Select **y** to add the job ID to the name of the output folder. The default is n. The job ID is helpful for making the correlation between a specific instance of a Report script and the corresponding report.
Select properties	Select and enter report properties in the Report Properties dialog box. The default name is VoIP Quality R-value Summary.
Add time stamp to title	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. Adding a time stamp is useful for running consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	This script automatically raises an event if the report is not generated successfully. Select **y** to raise an event when the report is generated successfully. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is generated successfully. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

4.20 Report_TimeDetail

Use this Knowledge Script to summarize the average values by minute of the VoIP Quality data streams (MOS, R-value, availability, delay, jitter, jitter buffer loss, lost data) within a selected period.

4.20.1 Resource Object

Report agent

4.20.2 Default Schedule

By default, this script runs once.

4.20.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Data Source	
Select Knowledge Scripts	Select the Knowledge Scripts that generated the data that you want to include in the report.
Computer selection by	Select the category by which you want to select the computers that you want to include in the report: View, Server Group, or Computer. The default is View.
Select computers	Select the computers that you want to include in the report.
Select time range	Select a **Specific** or **Sliding** date/time range from which the report should pull data. The default is Sliding.
Aggregate by n minutes	Specify the interval in minutes in which time data will be grouped. The default is 30 minutes.
Chart Thresholds	
MOS threshold	Specify the MOS threshold to display on the MOS charts in the report. The default is 0.000.
R-value threshold	Specify the R-value threshold to display on the R-value charts in the report. The default is 0.000.
Availability threshold	Specify the Availability threshold to display on the Availability charts in the report. The default is 0%.
Delay threshold	Specify the Delay threshold to display on the Delay charts in the report. The default is 0 ms.
Jitter threshold	Specify the Jitter threshold to display on the Jitter charts in the report. The default is 0 ms.
Percent Jitter Buffer Loss threshold	Specify the Jitter Buffer Loss threshold to display on the Jitter Buffer Loss charts in the report. The default is 0.000%.
Percent Lost Data threshold	Specify the Lost Data threshold to display on the Percent Lost Data charts in the report. The default is 0.000%.
Report Settings	
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Include charts?	Select **y** to include a chart in the report. The default is y.
Include table?	Select **y** to include a table of information in the report. The default is y.
Select chart style	Select and enter chart properties in the Chart Settings dialog box. The default style is Line.

Parameter	How To Set It
Select output folder	Select the name and location of the folder in which the report will be output. The default name is VoIPQualityTimeDetail.
Add job ID to output folder name?	Select **y** to add the job ID to the name of the output folder. The default is n. The job ID is helpful for making the correlation between a specific instance of a Report script and the corresponding report.
Select properties	Select and enter report properties in the Report Properties dialog box. The default name is VoIP Quality Time Detail.
Add time stamp to title	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. Adding a time stamp is useful for running consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	This script automatically raises an event if the report is not generated successfully. Select **y** to raise an event when the report is generated successfully. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is generated successfully. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

4.21 Report_VoIPQualitySummary

Use this Knowledge Script to summarize VoIP quality statistics (MOS, R-value, delay, jitter, jitter buffer loss, and lost data) within a group within a selected period.

4.21.1 Resource Object

Report agent

4.21.2 Default Schedule

By default, this script runs once.

4.21.3 Setting Parameter Values

Set the following parameters as needed:

Parameter	How To Set It
Data Source	

Parameter	How To Set It
Calls grouped by	Select the category by which you want to group data for display in the report. The default is Talker.
Select Knowledge Script(s)	Select the Knowledge Scripts that generated the data that you want to include in the report.
Computer selection by	Select the category by which you want to select the computers that you want to include in the report: View, Server Group, or Computer. The default is View.
Select computers	Select the computers that you want to include in the report.
Select time range	Select a **Specific** or **Sliding** date/time range from which the report should pull data. The default is Sliding.
Chart Thresholds	
Good-Acceptable MOS threshold	Specify the MOS threshold for the good-to-acceptable range to display in the report. The default is 4.030.
Acceptable-Poor MOS threshold	Specify the MOS threshold for the acceptable-to-poor range to display in the report. The default is 3.600.
Good-Acceptable R-value threshold	Specify the R-value threshold for the good-to-acceptable range to display in the report. The default is 80.0.
Acceptable-Poor R-value threshold	Specify the R-value threshold for the acceptable-to-poor range to display in the report. The default is 70.0.
Good-Acceptable delay threshold	Specify the delay threshold for the good-to-acceptable range to display in the report. The default is 150 ms.
Acceptable-Poor delay threshold	Specify the delay threshold for the acceptable-to-poor range to display in the report. The default is 400 ms.
Good-Acceptable jitter threshold	Specify the jitter threshold for the good-to-acceptable range to display in the report. The default is 40 ms.
Acceptable-Poor jitter threshold	Specify the jitter threshold for the acceptable-poor range to display in the report. The default is 60 ms.
Good-Acceptable packet loss threshold	Specify the packet loss threshold for the good-acceptable range to display in the report. The default is 0.5%.
Acceptable-Poor packet loss threshold	Specify the packet loss threshold for the acceptable-poor range to display in the report. The default is 1.0%.
Good-Acceptable jitter buffer loss threshold	Specify the jitter buffer loss threshold for the good-acceptable range to display in the report. The default is 0.5%.
Acceptable-Poor jitter buffer loss threshold	Specify the jitter buffer loss threshold for the acceptable-poor range to display in the report. The default is 1.0%.
Chart Settings	
Chart size	Select the size of the rendered chart. Choose from Large, Medium, and Small. The default is Medium.
Horizontal chart?	Select **y** to create a horizontal bar chart or accept the default to create a vertical bar chart.
Chart color scheme	Select a color scheme template. The default is NetIQ.

Parameter	How To Set It
Report Settings	
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Include charts?	Select **y** to include a chart in the report. The default is y.
Include table?	Select **y** to include a table of information in the report. The default is y.
Select output folder	Select the name and location of the folder in which the report will be output. The default name is VoIPQualityCallSum.
Add job ID to output folder name?	Select **y** to add the job ID to the name of the output folder. The default is n. The job ID is helpful for making the correlation between a specific instance of a Report script and the corresponding report.
Select properties	Select and enter report properties in the Report Properties dialog box. The default name is VoIP Quality Summary.
Add time stamp to title	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. Adding a time stamp is useful for running consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	This script automatically raises an event if the report is not generated successfully. Select **y** to raise an event when the report is generated successfully. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is generated successfully. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

4.22 Reviewing Call Performance Metrics

The VoIPQuality_CallPerf and VoIPQuality_CiscoSAA Knowledge Scripts generate data streams, based on some or all of the following call performance metrics, for use in graphs and reports.

Metric	Description
MOS	The Mean Opinion Score (MOS) is an overall score representing the quality of a call. The MOS is a number between 1 and 5. A MOS of 5 is excellent; a MOS of 1 is unacceptably bad. The MOS is calculated based on measured items plus jitter buffer size. The jitter buffer size is constant based on the codec. Jitter buffers minimize the call disruptions from delay and jitter. However, jitter buffers themselves contribute to the overall delay experienced by the call. Additionally, in situations with high network delay, jitter buffers can cause data to be lost. Thus the jitter buffer must be factored into the MOS calculation. NetIQ uses a modified version of the ITU (International Telecommunications Union) G.107 standard E-model equation to calculate the MOS. The E-model, developed by the European Telecommunications Standards Institute (ETSI), has become ITU standard G.107. This algorithm is used to evaluate the quality of a transmission by factoring in the “mouth to ear” characteristics of a speech path. **NOTE**: AppManager for Cisco SAA does not generate MOS metrics.
R-value	Defined by ITU (International Telecommunication Union) recommendation G.107, the E-model is a complex calculation, the output of which is a single score called an R-value that is derived from delays and equipment impairment factors. An R-value can be mapped to an estimated MOS. R-values range from 100 (excellent) to 0 (poor). An estimated MOS can be directly calculated from an R-value. **NOTE**: AppManager for Cisco SAA does not generate R-value metrics.
Delay	Measured in milliseconds, delay is perhaps the most common hindrance to VoIP call quality. The delay is calculated on each packet. This value includes all delay factors between the endpoints. The end-to-end delay, or latency, as measured between the endpoints is a key factor in determining voice over IP call quality. The AppManager delay measurement is taken for datagrams traveling between the endpoints in a single direction and includes the following factors: ◆ **Network delay in one direction**—Datagram's RTP timestamp subtracted from time it was received by Endpoint 2. Endpoints must synchronize their high-precision timers to calculate one-way delay. This delay factor actually includes the propagation delay (time spent on the actual network) and the transport delay (time spent getting through intermediate network devices). ◆ **Packetization delay**—Fixed value; dependant on selected codec. ◆ **Jitter buffer delay**—Fixed value; dependant on type and size of configured jitter buffer. ◆ **Additional fixed delay**—Fixed value; user-configured. Most callers notice delay when it exceeds 250ms. ITU-T standard G.114 specifies 150ms as the maximum one-way delay that is tolerable for high-quality VoIP.

Metric	Description
Jitter	As simulated calls run during a VoIP quality test, the endpoints calculate jitter, a factor known to adversely affect call quality. Jitter is also called delay variation, and it indicates the variance of the arrival rate of datagrams sent during a simulated VoIP call. When a datagram is sent, the sender (one of the endpoints) gives it a timestamp. When a datagram is received, the receiver adds another timestamp. These two timestamps are used to calculate the datagram's transit time. If the transit times for datagrams within the same call are different, the call contains jitter. In a video application, jitter manifests itself as a flickering image, while in a telephone call, its effect may be similar to the effect of packet loss: some words may be missing or garbled. The amount of jitter in a call depends on the degree of difference between the datagrams' transit times. If the transit time for all datagrams is the same (no matter how long it took for the datagrams to arrive), the call contains no jitter. If the transit times differ slightly, the call contains some jitter. Jitter values in excess of 50ms probably indicate poor call quality. They provide a short-term measurement of network congestion and can also show the effects of queuing within the network.
Jitter Buffer Loss	Jitter buffer loss is the amount of data that is lost when jitter exceeds that which the jitter buffer can hold. Jitter buffer loss affects call clarity, which affects the overall MOS score. Jitter buffers smooth out variations in calls by holding some datagrams to feed them to the application sequentially. Datagrams that are not contained by the jitter buffer due to excessive delay variation would be lost to the application and are thus called jitter buffer lost datagrams. This statistic includes datagrams with delay too great for the jitter buffer you set ("overruns") as well as those that arrive too quickly while the jitter buffer is still full and must be discarded ("underruns"). Any jitter detected in the call is compared to the size of the jitter buffer in the call script. Jitter buffer lost datagrams are then expressed as a percentage of all datagrams sent. AppManager uses the jitter buffer loss statistic when calculating a MOS for simulated VoIP traffic sent between the target devices.
Percent Lost Data	When a datagram is lost during a VoIP transmission, you can lose an entire syllable or word in a conversation. Obviously, data loss can severely impair call quality. AppManager therefore includes data loss as a call quality impairment factor in calculating the MOS of each simulated VoIP call. To measure data loss, the sending endpoint reports to the receiving endpoint how many bytes it sent, and the receiver compares that value to the amount received to determine lost data. Because *packet loss concealment* (PLC) is enabled by default in the G.711 codecs, call quality is less adversely affected if any data is lost during the VoIP test. PLC makes the codec itself more expensive to manufacture, but does not otherwise add delay or have other bad side effects.

4.23 Diagnosing VoIP Quality Problems

AppManager includes Knowledge Scripts that monitor and detect problems with VoIP quality and call quality. These scripts raise informational events as a result of the detected problems. By using NetIQ Vivinet Diagnostics and AppManager together, you have the means to diagnose more precisely any problems with VoIP quality between phones, endpoints, or other target devices such as routers and gateways.

Using an existing methodology (launching an Action script based on an event), the VoIPQuality_CallPerf Knowledge Scripts can run the Action_DiagnoseVoIPQuality Knowledge Script, which in turn launches Vivinet Diagnostics to diagnose the problem when MOS, R-value, delay, jitter, jitter buffer loss, and percentage of lost data exceed their thresholds.

The Action script runs by default only if Vivinet Diagnostics version 1.1 or later is installed on the computer on which it runs.

To enable a VoIPQuality_CallPerf Knowledge Script to launch Vivinet Diagnostics:

1. In the script's Properties dialog box, click the **Actions** tab. Action_DiagnoseVoIPQuality is selected by default.
2. Click **Properties**.
3. Enter values for all parameters. For more information about the parameter values, click **Help** on the Properties for Action_DiagnoseVoIPQuality dialog box.
4. Continue entering values on the other tabs of the Properties dialog box, or click **OK** to run the job.

4.24 Reviewing Quality of Service

In order for VoIP users to receive an acceptable level of voice quality, VoIP traffic must be given priority over other kinds of network traffic, such as data. The main goal of *Quality of Service* (QoS) is to ensure that VoIP traffic receives the preferential treatment it deserves, thereby reducing or eliminating the delay of voice packets that travel across a network.

In order to work with multiple types of network traffic, QoS first *classifies* the traffic, and then *handles* it. Once traffic is sorted into classes, QoS can determine how the traffic should be treated.

One QoS standard for sorting traffic is *DiffServ* (Differentiated Services). DiffServ sorts packets into groups that have similar QoS requirements and then gives those groups the required treatment at every hop in the network, also known as the *per-hop behavior* (PHB). DiffServ uses the second byte (eight bits) in the IP header to define and sort a packet. Of the eight bits, the last two are reserved for future use. DiffServ uses only the first six bits (the *Differentiated Services* [DS] field) in classifying a packet into one of three *codepoints*: Best Effort, Expedited Forwarding, and Assured Forwarding.

DiffServ-enabled routers can subdivide networks into DiffServ (DS) domains, within which all IP traffic competes for a finite share of bandwidth determined by a committed information rate, or CIR. To ensure that traffic that exceeds the CIR is still delivered without compromising the performance of high-priority traffic, packets within a DS domain are placed into PHB groups, including Expedited Forwarding and Assured Forwarding. Of these two groups, Expedited Forwarding receives slightly lower drop precedence and slightly higher bandwidth allocation than Assured Forwarding.

These groups allow for very exact policy-based QoS: they can be further subdivided to determine which packets are least likely to be dropped and most likely to be forwarded quickly despite congestion. Assured Forwarding includes four classes, AF1-AF4. Within each class, three subclasses may be defined, with increasing drop precedence. For example, AF1 may be the highest class of traffic, but within that class, AF13 will be dropped before AF11 or AF12.

Two DiffServ settings once used as part of the standard implementation may soon be deprecated, or rendered obsolete. Those settings, Expedited Flow and Assured Flow, used only the first three bits of the DiffServ codepoint, the type of service (TOS) bits. More recent DiffServ implementations use all six bits, for a total of 64 possible settings.

NOTE: TOS (Type of Service) is the previous name used to identify the second byte of the IP header. Used in early TCP/IP specs, TOS is described in RFC 791. In more recent TCP/IP specs, the same byte is referred to as the DS field, which is described in RFC2474.

IEEE 802.1p is an OSI Layer 2 standard for prioritizing and queuing network traffic at the data link/ MAC sub-layer. It can also be defined as best-effort QoS at Layer 2. 802.1p traffic is classified and sent to the destination with no bandwidth reservation.

The three-bit Prioritization field in the 802.1p tag establishes eight levels of priority, similar to the IP Precedence bits. A level-eight priority is the highest, and is thus reserved for router-update traffic. However, AppManager supports only the two priority levels that are appropriate for VoIP traffic:

- 011--(3) For medium-priority traffic. Often used for call setup packets.
- 101--(5) For high-priority traffic. Recommended for VoIP data packets.

Network adapters and switches route traffic based on the priority level. The hardware itself-usually a NIC card or an IP phone-does the tagging. Many recently developed IP phones are marking voice packets with a priority of five (101).

NOTE: Some older switches support just two or three priority queues, so their implementation of 802.1p does not support all of the eight priority levels in the IEEE 802.1p specification. Such switches place 802.1p values of 0 through 3 in a low-priority queue, and priority levels 4 through 7 in a high-priority queue, using only two different priority levels.

When monitoring call performance with AppManager, you can run a VoIP test using a DiffServ codepoint and one of the VoIP Quality Knowledge Scripts.

5 CallSetup Knowledge Scripts

AppManager for CallSetup/H.323 monitors the response time and availability of H.323 gateways and gatekeepers. Knowledge Scripts provide the ability to emulate H.323 client processes, including call registration and call setup using the H.323 protocol, ensuring that key H.323 devices are available and performing well on the network.

AppManager for CallSetup/SIP monitors the response time and availability of SIP. Knowledge Scripts provide the ability to emulate SIP client processes, including call registration and call setup using the SIP protocol, ensuring that key SIP devices are available and performing well on the network.

From the Knowledge Script view of Control Center, you can access more information about any NetIQ-supported Knowledge Script by selecting it and clicking **Help**. Or in the Operator Console, click any Knowledge Script in the Knowledge Script pane and press **F1**.

Knowledge Script	What It Does
H.323_CallSetup_Direct	Sets up a VoIP call directly between NetIQ endpoints over H.323 without the use of a gatekeeper or a gateway.
H.323_CallSetup_Gatekeeper	Sets up a VoIP call between NetIQ endpoints over H.323 using a gatekeeper.
H.323_CallSetup_Gateway	Sets up a VoIP call between NetIQ endpoints over H.323 using a gateway.
H.323_Listen	Causes the NetIQ endpoint to begin listening for H.323 communications on a given port.
H.323_Registration	Registers a NetIQ endpoint with a gatekeeper over H.323.
H.323_UpdateAlias	Updates the alias name of the H.323 resource in the repository.
Report_H.323Configuration	Displays the Call Setup H.323 configuration for the selected computers.
Report_H.323ResponseAvailMatrix	Displays the average response time and availability between talkers and listeners.
Report_H.323ResponseTimeDetail	Displays the average response time by minute.
Report_SIPConfiguration	Displays the Call Setup SIP configuration for the selected computers.
Report_SIPResponseAvailMatrix	Displays the average response time and availability between talkers and listeners.
Report_SIPResponseTimeDetail	Displays the average response time by minute.
SIP_CallSetup_Direct	Performs all of the steps associated with setting up a VoIP call using SIP directly between two endpoints.
SIP_CallSetup_Server	Performs all of the steps associated with setting up a VoIP call using a SIP server.

Knowledge Script	What It Does
SIP_Listen	Causes the NetIQ endpoint to begin listening for SIP communications on a given port.
SIP_Registration	Performs basic registration with the server and checks for response time, a valid response, and a successful return code.
SIP_UpdateAlias	Updates the alias name of the SIP resource in the repository.

5.1 H.323_CallSetup_Direct

Use this Knowledge Script to set up a VoIP call between NetIQ endpoints (one listener and one talker) over H.323 without the use of a gatekeeper or a gateway.

A direct call setup test tears down the call. The tear down is not included in the response time measurement. This script raises an event if response time exceeds the threshold that you set.

You can set up an H.323 call across a LAN to ensure that an H.323 device, such as NetMeeting, can make direct calls. This “loop back” monitoring job helps you verify proper installation and configuration of H.323 agent devices. In addition, you can determine the response time, which indicates LAN performance, for H.323 call setup between client devices.

5.1.1 Prerequisite

Run the H.323_Listen script without specifying gatekeeper information on the Values tab. You cannot initiate a CallSetup Knowledge Script job unless you first run the Listen job.

5.1.2 Resource Object

H.323 object

5.1.3 Default Schedule

By default, this script runs every 15 minutes.

5.1.4 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How to Set It
Collect data?	Select **y** to collect data about response time for reports and graphs. The default is y.
Select listener(s)	Select the names of the computers that you want to act as listeners.
Threshold - Maximum response time	Specify the maximum amount of response time that can occur before an event is raised. The default is 500 milliseconds.
Event severity when response time exceeds the threshold	Set the event severity level, from 1 to 40, to reflect the importance of an event in which response time exceeds the threshold. The default is 15.

Parameter	How to Set It
Event severity when call setup fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which the call setup test fails. The default is 5.
Listening port number	Enter the port number of the listening endpoint. This port number must match that of the listening port you specified in the H.323_Listen script. The default port number is 1720.
Use H.245 tunneling?	Select **y** to enable H.245 message encapsulation. The default is n.

5.2 H.323_CallSetup_Gatekeeper

Use this Knowledge Script to set up a VoIP call between NetIQ endpoints (one listener and one talker) over H.323 using a gatekeeper. A gatekeeper call is one in which the gatekeeper confirms the availability of the listener before allowing the talker to complete a call.

A gatekeeper call setup test tears down the call. The tear down is not included in the response time measurement. This script raises an event if response time exceeds the threshold that you set.

You can set up an H.323 call through a router gatekeeper to determine the response time, which indicates performance, for H.323 call setup across a WAN or to the PSTN (Public Switched Telephone Network).

TIP: Run the H.323_Registration script on the agent computer to monitor the gatekeeper's availability by registering and deregistering the agent computer with the gatekeeper.

5.2.1 Prerequisite

Run the H.323_Listen script and specify *all* of the gatekeeper parameters on the Values tab. The Listen script prepares the listener endpoint for listening. You cannot initiate a CallSetupKnowledge Script job unless you first run the Listen job.

5.2.2 Resource Object

H. 323 object

5.2.3 Default Schedule

By default, this script runs every 15 minutes.

5.2.4 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Collect data?	Select **y** to collect data about response time for reports and graphs. The default is y.
Select listener(s)	Select the names of the computers that you want to act as listeners.

Parameter	How To Set It
Threshold - Maximum response time	Specify the maximum amount of response time that can occur before an event is raised. The default is 500 milliseconds.
Event severity when response time exceeds the threshold	Set the event severity level, from 1 to 40, to reflect the importance of an event in which response time exceeds the threshold. The default is 15.
Event severity when call setup fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which the call setup test fails. The default is 5.
Name or IP address of gatekeeper	Enter the name or IP address of the gatekeeper. If you enter an incorrect IP address or leave the field blank, then this job will fail.
Port to communicate with gatekeeper	Enter the port on the gatekeeper with which the signaling (talker) and listening ports will communicate in order to communicate with the gatekeeper. If you entered a name or IP address in *Name or IP address of gatekeeper*, then you *must* provide the port number. If you enter an incorrect port number or leave the field blank, then this job will fail. The default port number is 1719.
Password for use of gatekeeper	Enter the password, if any, associated with the gatekeeper. If you entered a name or IP address in *Name or IP address of gatekeeper*, then you *must* provide the applicable password. If you enter an incorrect password or leave the field blank, then this job will fail.
Use H.245 tunneling?	Select **y** to enable H.245 message encapsulation. The default is n.

5.3 H.323_CallSetup_Gateway

Use this Knowledge Script to set up a VoIP call between NetIQ endpoints (one listener and one talker) over H.323 using a gateway. A gateway call is one in which the talker funnels a call through a gateway, which determines the route of the call and availability of the listener. The gateway sends the call on to the listener.

A gateway call setup test tears down the call. The tear down is not included in the response time measurement.This script raises an event when response time exceeds the threshold that you set.

Some gateways may not be able to handle call setup tests for multiple calls or multiple listeners. For example, your test may fail if you attempt to set up a test with two listeners (from talker A to listeners B and C). In this case, run two separate call setup tests for the two calls: one from talker A to listener B, and another from talker A to listener C.

You can set up an H.323 call through a router gateway to determine the response time, which indicates performance, for H.323 call setup across a WAN or to the PSTN (Public Switched Telephone Network).

You can set up an H.323 call through Cisco Unified Communications Manager to ensure that Cisco soft phones can make calls. This “loop back” monitoring job helps you determine the response time, which indicates performance, for H.323 call setup through Communications Manager.

NOTE

- Cisco Unified Communications Manager provides H.323 gateway services for soft phones because the phones use H.323, not SCCP, for call setup. You must manually configure all soft phones on a Communications Manager.

- If the name of the agent phone in AppManager is not the same as the “Device Name” in Communications Manager, run H.323_UpdateAlias. In the *New alias name* parameter, enter the same description that is used for the Communications Manager phone configuration “Device Name.”
- Because Communications Manager is not a gatekeeper for H.323 devices, do not run H.323_CallSetup_Gatekeeper or H.323_Registration against a Communications Manager server.

5.3.1 Prerequisites

- Configure the gateway to associate the hostname of the endpoint computer with an H.323 alias, an arbitrary value that you determine.
- Run the H.323_UpdateAlias script to pull the configured alias information into AppManager.
- Run the H.323_Listen script without specifying any gatekeeper information on the Values tab. The Listen script prepares the listener endpoint for listening. You cannot initiate a CallSetup Knowledge Script job unless you first run a Listen job.

5.3.2 Resource Object

H. 323 object

5.3.3 Default Schedule

By default, this script runs every 15 minutes.

5.3.4 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Collect data?	Select **y** to collect data about response time for reports and graphs. The default is y.
Select listener(s)	Select the names of the computers that you want to act as listeners. **NOTE**: Some gateways may not be able to handle call setup tests for multiple listeners. For example, your test may fail if you select two computers for this parameter. In this case, simply run this script twice — once for the first computer and then again for the second.
Threshold - Maximum response time	Specify the maximum amount of response time that can occur before an event is raised. The default is 500 milliseconds.
Event severity when response time exceeds the threshold	Set the event severity level, from 1 to 40, to reflect the importance of an event in which response time exceeds the threshold. The default is 15.
Event severity when call setup fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which the call setup test fails. The default is 5.
Name or IP address of gateway	Enter the name or IP address of the gatekeeper. If you enter an incorrect IP address or leave the field blank, then this job will fail.

Parameter	How To Set It
Port to communicate with gateway	Enter the port on the gatekeeper with which the signaling (talker) and listening ports will communicate in order to communicate with the gatekeeper. If you enter an incorrect port number or leave the field blank, then this job will fail. The default port number is 1720.
Use H.245 tunneling?	Select **y** to enable H.245 message encapsulation. The default is n.

5.4 H.323_Listen

Use this Knowledge Script to cause the NetIQ endpoint to begin listening for H.323 communications on a given port. This script restarts the listening process on any agent computer whose endpoint goes down for any reason. It does not check against running multiple listening jobs on the same computer for the same alias and port — that is the job of the managed object.

The listening test will perform registration if the gatekeeper is specified in the script parameters. It is not necessary to perform both the registration test and the listening test. However, you must run the listen test before running any of the call setup tests.

5.4.1 Resource Object

H.323 object

5.4.2 Default Schedule

By default, this script runs on an asynchronous schedule.

5.4.3 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Event severity when listening job fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which the listening job fails. The default is 5.
Listening port number	Enter the number of the listening port. The default is 1720.
Name of gatekeeper	Enter the IP address of the gatekeeper.
Port to communicate with gatekeeper	Enter the port on the gatekeeper with which the signaling port, or talker, will communicate in order to register with the gatekeeper. The default is 1719.
Password for use of gatekeeper	Enter the password, if any, associated with the gatekeeper.

5.5 H.323_Registration

Use this Knowledge Script to register a NetIQ endpoint computer (also called the signaling port or the talker) with a gatekeeper over H.323. Registration verifies the availability of the gatekeeper and allows a gatekeeper to map dialed numbers to an IP addressing structure.

You do not need to run this script before running the H.323_Listen script.The Listen script automatically performs registration if a gatekeeper is present.

The registration test returns one measurement: total response time for registering with a gatekeeper.

TIP: To monitor gatekeeper availability, run this script on the agent computer to register and deregister the agent computer with the gatekeeper.

5.5.1 Resource Object

H323Agent

5.5.2 Default Schedule

By default, this script runs every 15 minutes.

5.5.3 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Collect data?	Select **y** to collect data for reports and graphs. The default is y.
Event severity when registration fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which registration fails. The default is 5.
Name or IP address of gatekeeper	Enter the name or IP address of the gatekeeper.
Port to communicate with gatekeeper	Enter the port on the gatekeeper with which the signaling port, or talker, will communicate in order to register with the gatekeeper. The default is 1719.
Password for use of gatekeeper	Enter the password, if any, associated with the gatekeeper.

5.6 H.323_UpdateAlias

Use this Knowledge Script to update the alias name of the H.323 resource in the repository. The alias name is the arbitrary name that you assigned the H.323 resource (the computer with the endpoint installed) when you configured it. By assigning an alias to the endpoint computer and then running H.323_UpdateAlias, you can pull all of the configured information into AppManager.

The new alias name is used when you stop and restart existing jobs or when you create new jobs. If you rerun the Discovery_VoIPQuality_CallSetup_H.323 Knowledge Script, the alias name is reset to the default name.

AppManager uses <*devicename*>`.netiq` as the default alias for H.323 devices. If a router gateway requires an access list, consider changing the alias to an IP address or to a fully qualified hostname so that the router can locate the AppManager managed object.

5.6.1 Resource Object

H.323 object

Run this script on only one resource at a time.

5.6.2 Default Schedule

By default, this script runs once.

5.6.3 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
New alias name	Enter a new alias name for the H.323 resource.
Event severity when update fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which the update fails. The default is 5.

5.7 Report_H.323Configuration

Use this Knowledge Script to summarize Call Setup H.323 configuration information for the selected computers.

5.7.1 Resource Object

Report agent

5.7.2 Default Schedule

By default, this script runs once.

5.7.3 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Data Source	
Select computer(s)	Select the computers to include in the report. You can select computers by category: View, Server Group, or Computer. The default is View.
Report Settings	

Parameter	How To Set It
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Select output folder	Select the name and location of the folder in which the report will be output. The default folder name is H323Config.
Add job ID to output folder name?	Select **y** to append the job ID to the name of the output folder. The default is n. A job ID helps you correlate a specific instance of a Report script with the corresponding report.
Select properties	Select report properties in the Report Properties dialog box. The default report name is Call Setup H.323 Configuration Report.
Add time stamp to title?	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. A time stamp lets you run consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	Select **y** to raise an event when the report is successfully generated. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is generated successfully. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

5.8 Report_H.323ResponseAvailMatrix

Use this Knowledge Script to summarize the average H.323 response time and availability between a talker and a listener within a time frame that you select.

5.8.1 Resource Object

Report agent

5.8.2 Default Schedule

By default, this script runs once.

5.8.3 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Data Source	
Select Knowledge Script(s)	Select the Knowledge Scripts to include in the report.
Select computer(s)	Select which computers to include in the report. You can select computers by category: View, Server Group, or Computer. The default is View.
Select time range	Select a **Specific** or **Sliding** date/time range from which the report should pull data. The default is Sliding
Report Settings	
Decimal accuracy for % values	Specify the number of decimal places that you want to see in the values generated by this report. The default is 3.
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Select output folder	Select the name and location of the folder in which the report will be output. The default folder name is H323RespAvailMatrix.
Add job ID to output folder name?	Select **y** to append the job ID to the name of the output folder. The default is n. A job ID lets you correlate a specific instance of a Report script with the corresponding report.
Select properties	Set the report properties as desired. The default report name is Call Setup H.323 Response Time Availability.
Add time stamp to title?	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. A time stamp lets you run consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	Select **y** to raise an event when the report is successfully generated. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is generated successfully. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

5.9 Report_H.323ResponseTimeDetail

Use this Knowledge Script to summarize the average H.323 response time by minute within a time range that you select.

5.9.1 Resource Object

Report agent

5.9.2 Default Schedule

By default, this script runs once.

5.9.3 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Data Source	
Select Knowledge Script(s)	Select the Knowledge Scripts to include in the report.
Select computer(s)	Select which computers to include in the report. You can select computers by category: View, Server Group, or Computer. The default is View.
Select time range	Select a **Specific** or **Sliding** date/time range from which the report should pull data. The default is Sliding.
Aggregate by n minute(s)	Enter the number of minutes in which time data will be grouped. The default is 15 minutes.
Report Settings	
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Include charts?	Select **y** to include a chart in the report. The default is y.
Include table?	Select **y** to include a table of information in the report. The default is y.
Select output folder	Select the name and location of the folder in which the report will be output. The default folder name is H323RespTimeDetail.
Add job ID to output folder name?	Select **y** to append the job ID to the name of the output folder. The default is n. A job ID helps you correlate a specific instance of a Report script with the corresponding report.
Select properties	Set the report properties as desired. The default report name is Call Setup H.323 Response Time Detail.
Add time stamp to title?	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. A time stamp lets you run consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	Select **y** to raise an event when the report is successfully generated. The default is y.

Parameter	How To Set It
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is generated successfully. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

5.10 Report_SIPConfiguration

Use this Knowledge Script to summarize SIP call setup configuration information for the selected computers.

5.10.1 Resource Object

Report agent

5.10.2 Default Schedule

By default, this script runs once.

5.10.3 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Data Source	
Select computer(s)	Select the computers to include in the report. You can select computers by category: View, Server Group, or Computer. The default is View.
Report Settings	
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Select output folder	Select the name and location of the folder in which the report will be output. The default folder name is SIPConfig.
Add job ID to output folder name?	Select **y** to append the job ID to the name of the output folder. The default is n. A job ID lets you correlate a specific instance of a Report script with the corresponding report.
Select properties	Set the properties parameters as desired. The default report name is Call Setup SIP Configuration Report.

Parameter	How To Set It
Add time stamp to title?	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. A time stamp lets you run consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	Select **y** to raise an event when the report is successfully generated. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is generated successfully. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

5.11 Report_SIPResponseAvailMatrix

Use this Knowledge Script to summarize the average SIP response time and availability between a talker and a listener within a time frame that you select.

5.11.1 Resource Object

Report agent

5.11.2 Default Schedule

By default, this script runs once.

5.11.3 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Data Source	
Select Knowledge Script(s)	Select the Knowledge Scripts to include in the report.
Select computer(s)	Select which computers to include in the report. You can select computers by category: View, Server Group, or Computer. The default is View.
Select time range	Select a **Specific** or **Sliding** date/time range from which the report should pull data. The default is Sliding
Report Settings	

Parameter	How To Set It
Decimal accuracy for % values	Enter the number of decimal places that you want to see in the values generated by this report. The default is 3.
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Select output folder	Select the name and location of the folder in which the report will be output. The default folder name is SIPRespAvailMatrix.
Add job ID to output folder name?	Select **y** to append the job ID to the name of the output folder. The default is n. A job ID helps you correlate a specific instance of a Report script with the corresponding report.
Select properties	Set the report properties as desired. The default report name is Call Setup SIP Response Time Availability.
Add time stamp to title?	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. A time stamp lets you run consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	Select **y** to raise an event when the report is successfully generated. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is generated successfully. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

5.12 Report_SIPResponseTimeDetail

Use this Knowledge Script to summarize the average SIP response time by minute within a time range that you select.

5.12.1 Resource Object

Report agent

5.12.2 Default Schedule

By default, this script runs once.

5.12.3 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Data Source	
Select Knowledge Script(s)	Select the Knowledge Scripts to include in the report.
Select computer(s)	Select which computers to include in the report. You can select computers by category: View, Server Group, or Computer. The default is View.
Select time range	Select a **Specific** or **Sliding** date/time range from which the report should pull data. The default is Sliding.
Aggregate by n minute(s)	Enter the interval in minutes in which time data will be grouped. The default is 15 minutes.
Report Settings	
Include parameter card?	Select **y** to include a table in the report that lists parameter settings for the report script. The default is y.
Include charts?	Select **y** to include a chart in the report. The default is y.
Include table?	Select **y** to include a table of information in the report. The default is y.
Select chart style	Define the graphic properties for the charts in your report. The default style is Line.
Select output folder	Select the name and location of the folder in which the report will be output. The default folder name is SIPRespTimeDetail.
Add job ID to output folder name?	Select **y** to append the job ID to the name of the output folder. The default is n. A job ID lets you correlate a specific instance of a Report script with the corresponding report.
Select properties	Set the report properties as desired. The default report name is Call Setup SIP Response Time Detail.
Add time stamp to title?	Select **y** to append a time stamp to the title of the report, making each title unique. The default is n. The time stamp consists of the date and time the report was generated. A time stamp lets you run consecutive iterations of the same report without overwriting previous output.
Event Notification	
Raise event if report succeeds?	Select **y** to raise an event when the report is successfully generated. The default is y.
Event severity when report succeeds	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report is generated successfully. The default is 35.
Event severity when report has no data	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report contains no data. The default is 25.
Event severity when report fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which the report fails. The default is 5.

5.13 SIP_CallSetup_Direct

Use this Knowledge Script to perform all of the steps associated with setting up a VoIP call using SIP directly between two endpoints: 1) send a SIP `INVITE` message to the listener endpoint, 2) encapsulate media information, including codec and RTP ports in the body of the `INVITE` message using SDP, and 3) process the server's response to determine whether the call was successfully initiated.

This script raises an event when response time exceeds the threshold that you set.

You can set up SIP call across a LAN to ensure that a SIP device can make direct calls. This "loop back" monitoring job list you verify proper installation and configuration of SIP agent devices. In addition, you can determine the response time, which indicates LAN performance, for SIP call setup between client devices.

5.13.1 Prerequisites

- Run SIP_Listen to verify that the endpoint is available and capable of receiving incoming calls.
- This script assumes that the SIP registrar has the same IP address as the proxy agent computer.

5.13.2 Resource Object

SIP object

5.13.3 Default Schedule

By default, this script runs every 15 minutes.

5.13.4 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Collect data?	Select **y** to collect data for reports and graphs. The default is y.
Select listener(s)	Select the names of the computers that you want to act as listeners.
Threshold - Maximum response time	Specify the maximum amount of response time that can occur before an event is raised. The default is 800 milliseconds.
Event severity when response time exceeds the threshold	Set the severity level, from 1 to 41, to indicate the importance of an event in which response time exceeds the threshold. The default is 15.
Event severity when SIP call is unsuccessful	Set the severity level, from 1 to 41, to indicate the importance of an event in which the SIP call does not complete successfully. The default is 5.
Talker port number	Enter the port number of the talker computer. The default is 5060.
Listener port number	Enter the port number of the listener computer. The default is 5060.

5.14 SIP_CallSetup_Server

Use this Knowledge Script to perform all of the steps associated with setting up a VoIP call that is routed from one endoint to another through a SIP server.

- First, the script sends a SIP `INVITE` message to the listener endpoint via a SIP server.
- *If the SIP server is a proxy server*, then the script performs all signaling functions through the proxy.
- *If the SIP server is a redirect server*, then the script parses the new URL (in the Contact field) and contacts that server to actually set up the call.
- Next, the script encapsulates media information, including codec and RTP ports in the body of the `INVITE` message using SDP.
- Finally, the script processes the server's response to determine whether the call was successfully initiated.

This script raises an event when response time exceeds the threshold that you set.

5.14.1 Prerequisites

- Run SIP_Listen, which verifies that the endpoint is available and capable of receiving incoming calls.
- This script assumes that the SIP registrar has the same IP address as the proxy agent computer.

5.14.2 Resource Object

SIPAgent

5.14.3 Default Schedule

By default, this script runs every 15 minutes.

5.14.4 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Collect data?	Select **y** to collect data about response time for reports and graphs. The default is y.
Select listener(s)	Select the names of the computers that you want to act as listeners.
Threshold - Maximum response time	Specify the maximum amount of response time that can occur before an event is raised. The default is 800 milliseconds.
Event severity when response time exceeds the threshold	Set the severity level, from 1 to 41, to indicate the importance of an event in which response time exceeds the threshold. The default is 15.
Event severity when SIP call is unsuccessful	Set the severity level, from 1 to 41, to indicate the importance of an event in which the SIP call does not complete successfully. The default is 5.

Parameter	How To Set It
Talker port number	Enter the port number of the talker computer. The default is 5060.
Listener port number	Enter the port number of the listener computer. The default is 5060.
Name or IP address of proxy	Enter the name or IP address of the proxy computer.
Proxy port number	Enter the port number of the proxy computer. The default is 5060.
Register talker with proxy?	Enter **y** to register the talker computer with the proxy computer. The default is n.

5.15 SIP_Listen

Use this Knowledge Script to start a new thread, open a specified port, listen for incoming SIP messages, and then automatically return a response that indicates success. If necessary, the test registers the SIP endpoint with a SIP server so that the SIP endpoint can receive incoming calls in a server-based environment. SIP_Listen continues to run asynchronously until the job is stopped. It polls the endpoint every 60 seconds and refreshes the registration, if necessary, and it issues a `stop listen` command when the job is stopped.

5.15.1 Prerequisite

This script assumes that the SIP registrar has the same IP address as the proxy agent computer.

5.15.2 Resource Object

SIPAgent

5.15.3 Default Schedule

By default, this script runs on an asynchronous schedule. Regardless of the schedule that you select, once you start the script, its job status appears as Running.

5.15.4 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Event severity when registration fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which registration is unsuccessful. The default is 5.
Listener port number	Enter the port number of the listener computer. The default is 5060.
Name or IP address of proxy	Enter the name or IP address of the proxy computer.
Proxy port number	Enter the port number of the proxy computer. The default is 5060.

5.16 SIP_Registration

Use this Knowledge Script to perform basic registration with the server and checks for response time, a valid response, and a successful return code. Upon completion, this script immediately deregisters with the server by sending a `REGISTER` request with a TTL of **0**.

You do not need to run this script before running SIP_Listen. The Listen script automatically performs registration.

This script raises an event when response time exceeds the threshold that you set.

5.16.1 Prerequisite

This script assumes that the SIP registrar has the same IP address as the proxy agent computer.

5.16.2 Resource Object

SIP object

5.16.3 Default Schedule

By default, this script runs every 15 minutes.

5.16.4 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
Collect data?	Select **y** to collect data about response time for reports and graphs. The default is y.
Threshold - Maximum response time	Specify the maximum amount of response time that can occur before an event is raised. The default is 800 milliseconds.
Event severity when response time exceeds the threshold	Set the severity level, from 1 to 40, to indicate the importance of an event in which response time exceeds the threshold. The default is 15.
Event severity when registration fails	Set the severity level, from 1 to 40, to indicate the importance of an event in which registration fails. The default is 5.
Registrant port number	Enter the port number of the computer that you want to register. The default is 5060.
Name or IP address of proxy	Enter the name or IP address of the proxy computer.
Proxy port number	Enter the port number of the proxy computer. The default is 5060.

5.17 SIP_UpdateAlias

Use this Knowledge Script to update the alias or DNS name of the SIP resource in the repository. You can drop this script on only one resource at a time. Once you update the alias/DNS name, the new name is used when you stop and restart existing jobs or when you create new jobs. If you rerun the Discovery_VoIPQuality_CallSetup_SIP script, the alias/DNS name is reset to the default name.

NOTE: The alias/DNS name is the arbitrary name you assigned the SIP resource (the computer with the endpoint installed) when you configured it. To pull all configured information into AppManager, assign an alias/DNS name to the endpoint computer and then run SIP_UpdateAlias.

5.17.1 Prerequisite

This script assumes that the SIP registrar has the same IP address as the proxy agent computer.

5.17.2 Resource Object

SIP object

5.17.3 Default Schedule

By default, this script runs once.

5.17.4 Setting Parameter Values

Set the following parameters as necessary:

Parameter	How To Set It
New alias name	Provide a new alias name for the SIP resource. You can enter an alias name in addition to a DNS name or instead of a DNS name.
New DNS name	Provide a new DNS name for the SIP resource. You can enter a DNS name in addition to an alias name or instead of an alias name.
Event severity when update fails	Set the event severity level, from 1 to 40, to reflect the importance of an event in which the alias update fails. The default is 5.