CA Unified Communications Monitor

Reporting Guide

Version 3.4

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Contents

Chapter 1: Reports for Monitoring

The Monitoring category of reports helps IT staff monitor unified communications system performance and detect problems early. Monitoring reports provide the following information:

- Summaries of system health.
- Detailed information about endpoints in the network.
- Phone, trunk, and voice gateway usage.
- Diagnostic data useful for troubleshooting performance issues.
- Summaries of performance data, with the worst-performing results shown first.
- Links to more detailed data that is narrowly focused on one Location or component.
- Detailed information about active endpoints to help with inventory and license tracking.

If custom groups of Locations or devices are defined in CA Performance Center, more data views showing group-level metrics are included in the applicable reports. You can drill down from first-level groups into data from individual group members.

This section contains the following topics:

Interpreting Data in Reports

For calls on the network, the collector gathers data from the flows that pass through the switch it shares with a call server or cluster. The collector retains only the data for calculating call setup and call quality metrics, which it sends to the management console for evaluation.

Data evaluation occurs when the collected metrics are compared to the performance thresholds assigned to Locations and media devices.

Data Ratings

In most reports, for each assigned threshold, UC Monitor classifies the relevant data for call performance metrics by using one of the following ratings.

Acknowledged

When you acknowledge an incident, all the data the incident covers is marked as acknowledged. Future data for an acknowledge incident is also automatically marked as acknowledged.

No Data

No data is available.

Unrated

One of the following reasons:

- Sufficient data does not exist to establish a rating.
- The number of observations is less than the minimum observations threshold.

Normal

The metric value does not exceed a performance threshold. Performance is good, or normal.

Degraded

The metric value exceeds the degraded threshold.

Excessive

The metric value exceeds the excessively degraded threshold.

Call Attempts and Reporting Intervals

Every attempted call is logged in the database. However, to generate call setup metrics for a call performance report, a call must progress far enough to receive a response-time measurement.

UC Monitor bases response time on the time it takes for the call server to send setup information to the calling endpoint (see definition on page 112). Call server response time is used to calculate the post-dial delay and delay-to-dial-tone metrics.

Call performance data is collected continuously and sent to the management console, but reports are updated every 5 minutes. Call quality metrics are collected when a call completes. The metrics do not appear in reports until they are received at the management console and processed.

A reporting interval is distinct from a monitoring interval: monitoring is continuous, but reporting of call quality and call setup metrics occurs every 5 minutes. A call must end during a reporting interval for the call metrics to appear in reports. Several minutes can elapse before the collector reports all data.

Note: For other types of reporting, such as call server and collector incidents, a 15-minute reporting interval applies. Data from a Call Watch is collected every 15 seconds and reported every minute.

A call that fails or is abandoned must actually reach the ringing phase before UC Monitor logs the associated IP address or directory number.

In a few rare cases, a call attempt has no call setup data. The following example explains how the situation occurs:

- The call server shares a switch with the collector.
- A call is abandoned before a ringing signal is sent.
- Response-time measurements are too negligible to be recorded because the collector is located so closely to the call server.
- A call attempt is logged in such a case, but no call setup metrics are provided.

Call Legs and Bi-Directional Data

When it is first placed, a VoIP call consists of two call legs:

- Origination party: Call setup metrics apply to this leg.
- Destination party: Call setup metrics do not apply to this leg of the call.

But as the call travels across the network, the call legs are considered directional. The origination party is not always the sender of call data. At times during the call, the origination party receives call data from the destination party.

UC Monitor captures these metrics and uses them in calculating and reporting on call performance. For Cisco deployments, it gathers call setup data from call server flows. UC Monitor also bases delay measurements on the timing of flows between the originator and the call server. UC Monitor can derive the delay-to-dial tone and post-dial delay call setup metrics from these timings.

Without eavesdropping on calls, the collector monitors call legs in each direction. The collector also obtains call performance data from call legs that involve endpoints in the PSTN. For PSTN-related data, the collector polls the Cisco voice gateway that routed them, or passively receives call quality reports from Avaya gateways. The collector can gather data from all call legs that are processed by the call servers with which it shares a switch.

UC Monitor reports use the data from each call leg to evaluate the listening quality that users experienced. For example, in Call Watch reports, listener call quality is expressed as MOS from call legs traveling toward each Location.

UC Monitor reports do not show quality statistics for call legs that terminate at a voice mail server or other device where these statistics are not available. For example, Cisco Unity and Unity Express do not report call quality statistics. For calls that are forwarded or transferred, reports show the IP address or directory number of the final destination, not the originally dialed destination.

Call Leg Correlation

UC Monitor can usually correlate call legs belonging to the same end-to-end call so that they are identified in a meaningful way. UC Monitor correlates audio call legs by the IP address and port combination of their two associated endpoints. For typical IP-only calls, the two call legs involve the same endpoints, which are reversed to show the direction of call flow. For PSTN call legs, the voice gateway creates a Global Call Identifier (GCI). The collector or Lync collector correlates the GCI with the ID in the call setup packet for the leg terminating on a gateway port.

Avaya call-processing hardware does not provide information about call direction until the call is complete. This information is available in UC Monitor reports only when the Avaya Communication Manager sends call detail records (CDR) to the collector. An asterisk identifies calls that lack the required CDR data to identify the direction of call data flow.

Call legs that include a video stream are more complicated to correlate. To the management console, they appear to be pairs of separate calls that media devices route and translate in the path. To handle these cases, session-level correlation is available as a filtering option in the Calls Overview.

More information:

Calls Overview (see page 49)
Report Metrics (see page 97)

Why is Direction Important?

The direction of a call leg matters when you are troubleshooting a performance issue. Reported performance is different for each direction of a call, although in most cases the differences are slight. The route that a call takes through your network can be asymmetric. For example, with a longer route, and more delay, in one direction between the conversing parties.

Therefore, consider bi-directional calls and call legs as you interpret the data in reports. And understand which directions are presented in each report.

For example, the Performance by All (see page 16) report contains data from all observed call legs traveling in sending and receiving directions: inbound to the IP network and outbound to the PSTN. When you drill down for more information, the next view usually shows call performance data by Location.

The perspective on the data changes with the Performance by Location (see page 17) report, which presents the call quality data for that Location. No matter which party initiates the call, the call quality perspective is that of the listener. Call quality metrics are from the call legs traveling toward that listener. Call setup metrics are taken only from the calls that originated at that Location.

Monitoring reports generally break out performance for a Location using data from the receiving direction. After you drill down by Location or by media device, the sending Location is also shown, to help you distinguish the performance of calls between pairs of Locations or media devices.

Types of Calls

Calls on the network are classified according to their status at completion or according to their type. Cisco Unified Communications Manager supports many extra phone features, such as hold, forward, and conference. Each feature produces a call of a different type. UC Monitor supports the following types of call.

Abandoned call

A call attempt in which an endpoint initiated a call that was disconnected before the call was answered. Abandoned calls have an origination time and a disconnect time, but no connect time.

Failed call

A call attempt that failed to connect during the setup phase. A call setup failure code is reported.

Call hold and resume

Calls placed on hold and then taken off hold when call conversation data resumes. Putting a call on hold effectively ends a Call Watch for both phones.

Forwarded or redirected call

Calls are forwarded automatically, on busy, or on no answer. Calls are forwarded to voice mail or to another number. An auto-attendant is often used. Cisco Unity and Unity Express provide this capability.

The final destination, not the dialed destination, of the forwarded or redirected call is shown as the call leg in reports.

Conference call

Call that used a conference number and appeared to go into the conference bridge. When the number of callers in the conference is reduced to two, the call server connects them directly to each other to free up the conference resource.

Pickup

The call is answered by a different extension than the one dialed. The extension that picks up the call receives a ringing signal that is based on membership in the pickup group. Similar to forwarded or redirected calls.

The final destination, not the dialed destination, of the forwarded or redirected call is shown as the call leg in reports.

Transfer with consultation

A call is answered and then forwarded to another extension. Multiple call legs are logged:

- Original call, from calling party to called party
- Call from the transferring party to the transfer destination
- Call from the transferred party to the transfer destination

Transfer without consultation

Also known as a blind transfer. The call is not answered before it is forwarded.

What is a Location?

UC Monitor administrators create Locations that correspond to network entities such as branch offices, departments, or buildings. Location definitions determine how data is displayed in reports. UC Monitor reports organize call performance data based on Location or Sending Location. These categories are further identified as <Unassigned>, <None>, or <External>, as explained in the following list:

Receiving Location

Call data that is categorized as Location was *received by* the indicated Location.

Sending Location

Call Data that is categorized as Sending Location was *sent from* the indicated Location.

<Unassigned>

The <Unassigned> Location identifies IP phones that were observed during monitoring but whose IP addresses are not part of the subnets in your Locations. An <Unassigned> Location is not the same as a Location of <None>.

A Location is <Unassigned> for reasons such as:

- Locations have not been defined for the IP address ranges of some phones in the network.
- A device pool is misconfigured in a Cisco Unified Communications Manager.
- The phone is misconfigured, placed in an unexpected VLAN or with an unexpected IP address.

Call traffic from endpoints in new subnets is labeled as <Unassigned> until the administrator creates a Location definition for those subnets. Periodically, the administrator exports a list of current Location definitions and verifies that it reflects recent changes to the network and telephone system.

<None>

A Receiving Location or Sending Location of <None> indicates that the IP address of the endpoint was not determined, for reasons such as the following:

- Call setup failed before the number was determined.
- A directory number in the system is not assigned to an endpoint.
- The origination party abandoned the call before it was answered.

  Note: Abandoned call legs that have metrics are displayed in the Call Leg Details report. In addition, you can see these calls in the Calls Overview when the option to “Show abandoned calls” is enabled in the Settings dialog.

<External>

The <External> Location appears only when a Microsoft environment is monitored. This Location identifies endpoints that are reported to the front-end server as remote. UC Monitor uses this Location definition for endpoint addresses outside your firewall, or behind a NAT device, that make remote access connections through an edge server. Such endpoints are assigned a temporary IP address, usually a private address, that potentially overlaps with internal IP addresses in use by other endpoints.

Units of Observation

UC Monitor reports provide indications of data validity by showing the number of observations for a particular metric. As a rule, the greater the number of observations, the higher the validity of the statistics for a metric. The pool of performance data is based on the number of calls placed (calls originated) or the number of call minutes.

The following table describes the observation units for Call Performance reports:

Unit	Type of Metric	Description
Calls Originated	Call setup	One call = one call leg. The number of calls from endpoints at a given Location. The term *Originated* helps identify the direction of the call leg. Call setup metrics apply to one direction for a VoIP call. Only the applicable call legs are included in the Calls Originated count for call setup.
Call Minutes	Call quality	One call minute = 60 seconds of time that a telephone was off-hook and sending data. Call minutes are computed as follows: *(total seconds of all call data)/60* Normal rounding of seconds into minutes is then applied, but partial minutes are reported.

The observation units correspond to the units used in performance threshold configuration. Administrators can set the minimum number of calls originated or call minutes that must be collected before an incident is created.

Observations are taken continuously and reported to the management console, but observed data is not immediately reflected in Call Performance reports. No data related to a call is reported until the call ends. When it ends, all data from that call is reported during the next five-minute reporting interval. A call that lasts 42 minutes, for example, reports no data until the ninth reporting interval. At which point, the updated Call Performance Overview adds the entire 42 call minutes to the total number of call minute observations.

The number of observations is a helpful means of judging whether relatively heavy usage played a role in performance issues.

More information:

Call Performance Overview (see page 16)
Report Metrics (see page 97)

Call Setup Observations

Call setup observations are computed as one per call because only one call leg of the call contains the call setup data. Only calls originating from a Location are included in call setup measurements for that Location. The length of the call is not used to derive the value of the metric.

Calls originating from a voice gateway or the PSTN only compute post-dial delay.

Call setup availability observations are computed as one per call. The value for this metric is calculated as:

```
(# failed calls)/(total # of calls originated)
```

The number of call setup availability observations is equal to the total number of calls originated.

Call Quality Observations

To derive most call quality metrics, UC Monitor uses a call minute as the unit of observation. Each call quality metric is a weighted average over the number of call minutes for each call that ends during the five-minute reporting interval. Normal rounding of seconds into minutes is also performed. For call quality metrics, each call leg contributes its metrics separately.

The following example clarifies the relationship between observation units and call quality metric calculation. Assume that two calls were made from A to B during a five-minute reporting interval: one call of nearly 2 minutes, and one call of approximately 3 minutes. For the MOS metric, the following data was gathered:

Calls	MOS	Call minutes	Seconds
A → B	3.1	2	116
A → B	3.8	3	182

The UC Monitor calculation for MOS is a weighted MOS for the five-minute reporting interval that uses the actual total duration of calls in seconds. Totals for seconds are rounded to derive the totals for call minutes.

The MOS calculation is:

```
(116*3.1) + (182*3.8) = 1051.2
1051.2/300 seconds = 3.50
```

In this example, the MOS is 3.50 for the reporting interval and is rated as Excessive, using the default thresholds. The number of call minutes reported is 5.

Call Performance Overview

The Call Performance Overview provides a quick evaluation of the overall performance of your entire unified communications system. The overview also is the point of entry when you drill down into specific data about a Location, media device, or server. The overview displays call performance metrics, including call quality and call setup, by Location and then by component. Metrics categories are based on threshold levels: Normal, Degraded, Excessive. The overview provides a metric average that is based on the number of observations.

Call performance metrics are combined in the Performance by Location, by Media Device, and by Call Server views to show call performance on the network. The call performance rating is a reflection of end-user experience from the time they pick up the telephone until they hang up.

Data from both legs of a call is available from the reports in the Related Reports section of the Overview.

The data that you see was taken from call legs traveling toward the indicated Locations. That data helps you understand the user experience while interacting with the system by emphasizing the call quality that the listener perceived. The term *sending* is used to label drill-down views that show data traveling toward a selected Location from one Location or device.

Performance by All

The Performance by All view summarizes the performance of your entire unified communications system. Data from all monitored Locations and components is rolled up into a Performance bar chart. If you registered UC Monitor as a data source with CA Performance Center, this view is a rollup of data from all groups. Otherwise, this view is a rollup of data from all Locations.

This view presents the call volume of the system during the indicated time frame. The Calls Originated column shows the number of calls from all endpoints in the monitored system.

Performance by Group

The Performance by Group view is available only when custom groups are defined in CA Performance Center. You can drill down into individual group members and their data by clicking a group name.

The Performance by Group view examines call performance by group. The view rates call performance in the incoming direction to gauge the listening quality for VoIP and video users in that group. Groups are sorted by worst call performance.

Click a Location link to see ratings for performance metrics and for the components participating in calls to the Locations listed in this view.

Important: The Performance by Group view displays calls between Locations in the selected group. The view also identifies all associated call servers, including those not explicitly part of the group definition. For example, a user has permission to view Location1, Location2, CallServer1, and CallServer2. Group A contains Location1, Location2, and CallServer1. When viewing calls for Group A, the user sees all calls between Location1 and Location2, including those associated with CallServer2 even though CallServer2 is not explicitly part of Group A. In general, the Performance by Group view respects the permissions associated with Locations, but not the permissions associated with call servers.

Performance by Location

The Performance by Location view examines call performance by individual Location. This view lists all monitored Locations where endpoints had call activity. This view rates call performance in the incoming direction to gauge the listening quality for VoIP and video users.

Data that is unrated can indicate that the Minimum Observations threshold for that Location was not met during the selected time frame.

To see ratings for performance metrics and for the components participating in calls to the Locations listed in this view, click a Location link.

You can hide this view. Click the Settings link and clear the Performance by Location check box.

Performance by Media Device

In this view of media device performance, the perspective is of calls that the device handled: incoming from the PSTN, and outgoing from the IP network.

The media device category includes voice gateways and other devices that support call routing and processing. The metrics available for each device vary according to the device type and the environment. In a Microsoft environment, SNMP polling of media devices is not possible. As a result, fewer metrics are available for Microsoft media devices than for Cisco voice gateways.

The Calls Originated column provides the total number of calls that originated at points in the PSTN.

You can hide this view. Click the Settings link and clear the Performance by Media Device check box.

Performance by Call Server

The call servers shown in the Performance by Call Server view handled calls during the selected time frame. Their performance ratings are derived from the Locations they served. The name of each server is a link that allows further investigation.

The Call Performance category includes both call quality and call setup metrics. Where available, video metrics are included. The bar graph represents the number of calls originated or call minutes that were rated Normal, Degraded, or Excessive.

- Calls Originated: Calls from endpoints that are registered to a call server.
- Call Minutes: Minutes of call activity that were reported by a call server.

The Calls Originated column provides the total number of calls that were set up by the indicated call servers. Calls not included in the Calls Originated total were routed by the call servers in this view but were set up by a different call server.

You can hide this view. Click the Settings link and clear the Performance by Call Server check box.

Drilldown Views

The UC Monitor reports are designed to help you investigate performance degradation. You can *narrow*, or drill-down into, reports to compare the ratings among Locations, metrics, media devices, and call servers. You facilitate troubleshooting by narrowing report views so that the charts of interest are presented in proximity.

Drill down into the Call Performance Overview (see page 16) to expose one or more of the following views.

Narrow by Audio Metric

This view rates performance for the selected component, group, or Location: MOS, packet loss, latency, Network MOS, call setup failures, ACOM, delay to dial tone, post-dial delay.

Note: The ACOM metric is a gateway-only metric. ACOM data is unavailable when no voice gateway is used for the calls whose performance is reported in a particular view. ACOM metrics are shown only for Locations that made or received PSTN calls during the selected time frame.

Narrow by Call Server

This view rates the performance of the call server associated with the selected Location or media device.

Narrow by Sending Location

This view rates the performance of the Locations or devices that sent data to the selected Location.

Narrow by Sending Media Device

This view rates the performance of the media devices, such as voice gateways, that sent data to the selected Location.

Narrow by Video Metric

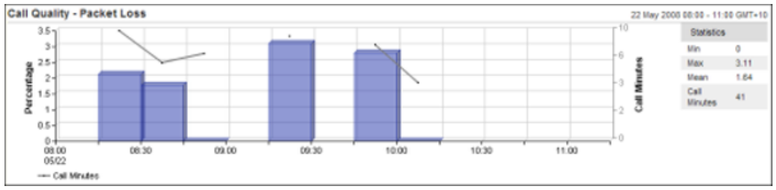
This view rates video performance for the selected component, group, or Location. Video performance includes such metrics as frozen video, video frame loss, video packet loss, and video latency. The view also rates performance for video streams that were reported by medianet.

Metric Details

The Metric Details page provides detailed charts of the collected data for each performance metric.

In the Metric Details charts, collected data is graphed over an X-axis showing the time frame. A line highlights the number of applicable data observations used in the ratings: calls originated or call minutes. The total number of observations is shown in the legend for each data view.

A logarithmic scale on the right Y-axis shows the number of observations. The Y-axis on the left side of each view plots the metric itself. In the following example, data loss rose above two percent when call activity spiked at around ten call minutes:



The legends provide more statistics that were calculated when analyzing data for each metric.

In a few cases, older or lower-end IP telephones do not support the collection of some performance statistics. For example, the Cisco model 7902 telephone returns only values for received packets and lost packets. Therefore, UC Monitor reports only packet loss for this model.

Call Leg Details

The Call Leg Details page provides data about completed calls. The data is useful with a drill-down view, such as Sending Location, to help you determine which calls experienced performance problems, and in which direction.

The Call Leg Details views rate performance per call leg and identify the directory numbers of the endpoints (see definition on page 112) in each Location. You can also use this report to find the IP address of an endpoint that appears as <Unassigned> in reports.

You can add or remove columns from the tables on this page. Click Settings to customize the page.

The Call Leg Details view lets you access detailed, per-call information from individual call legs. In tables that have detailed call data, the information icon (the letter "i" on a blue circle) provides a link to the Call Details report.

More information:

Performance by All (see page 21)
Performance Audio Call Leg Details (see page 21)
Performance Video Call Leg Details (see page 22)
Call Details (see page 51)
Report Metrics (see page 97)

Performance by All

This table is a rollup of call setup and call quality metrics for all Locations or for all groups, when custom groups are defined. A bar chart provides an indication of the times when calls were made and the quality of their performance. The Call Leg Details table breaks down individual call performance by metric.

Performance Audio Call Leg Details

This table provides data about when calls were made and by which specific endpoints. The table also identifies the Location to which each endpoint is assigned, and lists raw performance metrics for each call.

Some columns in this view appear only when the data is not included in the top-level view. For example, you access Call Leg Details from a drill-down view in which two Locations are selected. The Location/Media Device column is only shown in the Selected Location view and is not included in the Call Leg Details.

Note: Some metrics are vendor-specific and are not available from all telephones. Some older or lower-end IP telephones do not support the collection of all performance statistics.

Call Server

The DNS host name of the call server that handled the call.

Call Setup Failure Code

The code that the call server returns when a call fails during the setup phase. The code indicates the type of failure. For more information, see the list of call termination cause codes on the Cisco website.

Codec

The codec in use during setup of the call.

Duration

The length of the call.

Receiving IP Address

The IP address of the endpoint that received the call data.

Receiving IP Domain

The IP domain of the endpoint that received the call data. This field is available only when UC Monitor is registered to a CA Performance Center instance where at least one custom IP domain is defined.

Receiving Location/Media Device

The assigned Location of the endpoint that received the call data, or the DNS host name of the media device through which the call was sent.

Receiving Number

The directory number or SIP URI of the endpoint in the Location that experienced call quality problems.

Receiving Port

The port that was used on the endpoint that received the call data.

Sending IP Domain

The IP domain of the endpoint in the sending Location. This field is available only when UC Monitor is registered to a CA Performance Center instance where at least one custom IP domain is defined.

Sending Location/Media Device

The assigned Location of the endpoint that sent the call data or the DNS host name of the media device that routed the call to the receiving endpoint.

Sending Number

The directory number of the endpoint in the sending Location.

Time

The time and date when the call was made.

Call performance metrics

Raw data for each metric that contributed to the call performance rating. A colored icon indicates the rating with respect to the relevant threshold setting.

More information

Call Termination Cause Codes (see page 105)

Performance Video Call Leg Details

This table is available only when you monitor a Microsoft environment.

Call Server

The DNS host name of the call server that handled the call.

Codec

The codec in use during setup of the call.

Duration

The length of the call.

Receiving IP Address

The IP address of the endpoint that received the call data.

Receiving IP Domain

The IP domain of the endpoint that received the call data. This field is available only when UC Monitor is registered to a CA Performance Center instance where at least one custom IP domain is defined.

Receiving Location/Media Device

The assigned Location of the endpoint that received the call data, or the DNS host name of the media device through which the call was sent.

Receiving Number

The directory number or SIP URI of the endpoint in the Location that experienced call quality problems.

Receiving Port

The port that was used on the endpoint that received the call data.

Sending IP Domain

The IP domain of the endpoint in the sending Location. This field is available only when UC Monitor is registered to a CA Performance Center instance where at least one custom IP domain is defined.

Sending Location/Media Device

The assigned Location of the endpoint that sent the call data or the DNS host name of the media device that routed the call to the receiving endpoint.

Sending Number

The directory number of the endpoint in the sending Location.

Time

The time and date when the call was made.

Call performance metrics

Raw data for each metric that contributed to the call performance rating, as shown in the Performance by All view. A colored icon indicates the rating with respect to the relevant threshold.

More information:

Video Metrics (see page 101)

Incidents Reports

The Incidents report category provides three sets of reports:

Incidents Overview

The Incidents Overview shows a list of incidents that were raised in response to threshold violations. The Overview provides information about when incidents were reported and where, and lets you drill down into individual incident reports.

Collector Incidents

The Collector report shows provides information about each incident that is reported against collector performance. Collector incidents identify problems that are local to a specific collector. They do not apply to Microsoft deployments. In addition, only the Abnormal Termination incident applies to Avaya-only deployments.

Incident Investigations

The Investigations report identifies investigations that ran in response to incidents or that were started manually. The Investigations report provides information about when investigations were launched and where they ran. The report lets you drill down into individual Traceroute Investigation Details reports.

More information:

Incidents Overview (see page 24)
Collector Incidents (see page 40)
Incident Investigations (see page 43)

Incidents Overview

The Incidents Overview shows information about incidents that were raised in response to threshold violations. The overview provides a list of recent call performance-related incidents, with information about when the incident was reported and which network locations were affected.

The Incidents Overview provides the following information for network operators and engineers:

- Whether troublesome network conditions exist.
- Whether call servers have experienced a failover.
- Whether unauthorized devices have accessed the network.

Network personnel can also use this report to access more information about a recent incident. Details about a particular incident are available by clicking the incident ID number link.

In a Cisco environment, a Call Server Incidents table is available in the Incidents Overview. The Call Server Incidents table provides information about each of the call server incident types. The different types of call server incident apply either to individual call servers or to call server groups:

- Call server: Registration Failures, Poor Call Quality
- Call server group: Phone Status Changes

The following list describes the fields in the Incidents Overview, in alphabetical order:

Acknowledged

Select the check box next to the incident that you want to acknowledge and click Apply. Acknowledging an incident lets other operators know that you are aware of the associated condition.

Call Server

The name of the call server or call server group, usually representing a call server cluster, that is associated with the metrics that triggered the incident.

Call Server/Call Server Group

The DNS host name or IP address of the call server that is associated with the metrics that triggered the incident. Can also be the name of the call server group that reported the metrics.

Device

The IP address of the endpoint that reported the incident to the call server. Can be a link to the endpoint web page.

Duration

The length of time that the performance condition persisted.

ID

The sequential number that is assigned to this incident. Also a link to more information.

Location/Media Device

The Location of the endpoint that reported the incident condition, or the IP address or DNS host name of the device that sent the call from the PSTN.

Sending Location/Media Device

The Location of the endpoint that sent the call leg that triggered the incident. Can be the IP address or DNS host name of the device when the call involves the PSTN.

Severity

The severity of the incident: Degraded or Excessive.

State

Whether the incident is Open or Closed.

Time

The date and time when the incident was reported.

Type

The type of incident:

- Phone Status Changes. Includes information about missing phones, moved phones, and new or found phones. This type of incident can indicate a failover situation, where a call server within a call group has become unreachable.
- Poor Call Quality (QRT)
- Quality (call quality). Call quality incidents, which include video quality incidents, take precedence over call setup incidents when the incidents occur during the same incident interval, and the severity of the call quality incidents is equal to or greater than the severity of the call setup incidents.
- Registration Failures. Includes information about devices that attempted to register with the call servers where the incident was reported. This type of incident can indicate a configuration or security problem.
- Setup (call setup)

Filter the Incidents Overview

The Incident Overview provides filters for narrowing the scope of the data in the lists. These filtering options are associated with your user account and persist across login sessions.

Follow these steps:

1. Select an option in the Incident State field to see incidents in the selected state, or status:
   - Open and Closed. This default option shows all incidents.
   - Open. Select this option to show only incidents in the Open state.
   - Closed. Select this option to show only incidents in the Closed state.
2. Select an option in the Minimum Severity field to see incidents for the selected severity level:
   - **Degraded**. This default option shows all incidents, Degraded and Excessive.
   - **Excessive**. Select this option to show only Excessive incidents.
3. Select a time value in the Minimum Duration field to see incidents that were open for no less than the selected time.

   Duration does not apply to some Poor Call Quality incidents. An incident is considered instantaneous when the QRT key is pressed while a call is in progress. Select Any to see these types of incidents.

Call Performance Incident Details

You can drill down from a Call Performance incident to review the related Incident Details report. The Incident Details report provides details of performance degradation for the selected incident. Bar charts break out the performance for each direction of the degraded calls. The information in the detailed report is narrowed to show the affected Location or pair of Locations and a media device or call server.

The Incident Details report provides the following views:

Incident Details

A summary of the incident details that are presented on the Incidents Overview (see page 24).

Performance by Location

Bar charts illustrate call performance for each direction of a call:

- By Location of the origination and destination parties
- By the call server or media device involved in the call

The Call Performance category includes averages of the call setup and call quality data that contributed to this incident. Bar charts are aligned over a time frame indicator to help you compare performance ratings. The actual time that the incident report was created is shown below the bar charts, where an incident icon marks the time in question.

When custom groups are defined in CA Performance Center, the applicable group and subgroups for the selected incident are indicated with indentation to indicate hierarchy. The last nested item is the call server.

Performance by Audio Metric, Performance by Video Metric

Bar charts illustrate how data is rated for each applicable call performance metric.

Metric Details

You can drill down from a Call Performance incident to review the related Metric Details report. The Metric Details report provides detailed charts of the collected data for each performance metric that contributed to the selected incident.

Data is graphed over an X-axis showing the time frame. The background uses different colors to show how the data compared to the thresholds for expected normal, degraded, and excessive (excessively degraded) performance.

A line highlights the number of observations that were used in the ratings: calls (originated calls) or call minutes. The total number of observations is shown in the legend for each data view. A logarithmic scale on the right Y-axis shows the number of observations. The left Y-axis plots the metric itself. The legends also report other statistics that are calculated for each rated metric.

More information:

Units of Observation (see page 13)
Report Metrics (see page 97)

Call Leg Details

You can drill down from a Call Performance incident to review the related Call Leg Details report, which provides data about poorly performing calls. Use the data to determine which calls, and which direction of each call, experienced performance problems. Incident-specific information is included in the views, such as performance ratings per-call and the directory numbers of the endpoints (see definition on page 112) within each Location.

The Call Leg Details report provides the following views.

Incident Details

A summary of the incident details that are presented on the Incidents Overview (see page 24).

Performance by Location

Bar charts illustrate call performance for the call legs in the incident.

Incident Audio Call Leg Details/Incident Video Call Leg Details

- **Time**. The time and date when the call was made.
- **Receiving Number**. The directory number or SIP URI of the endpoint in the Location that experienced call quality problems.
- **Sending Number**. The directory number of the endpoint in the sending Location.
- **Codec**. The codec used during the call.
- **Duration**. The length of time that the performance condition persisted.

- **Audio or Video Metrics**. Raw data for each metric that contributed to the call performance rating, as shown in the Performance by Location view. A colored icon indicates the rating with respect to the relevant threshold setting. For definitions of these metrics, see Report Metrics (see page 97).
- **Call Setup Failure Code**. The code that the call server returns when a call fails during the setup phase. The code indicates the type of failure.

Tip: The presence of call setup metrics indicates the origination endpoint.

Traceroute Investigation Details

You can drill down from a Call Performance incident to review the related Investigation Details report, which provides information about a selected traceroute investigation and its associated incident. The Investigation Details report provides the following views.

Incident Details

A summary of the incident details that are presented on the Incidents Overview (see page 24).

Traceroute Investigation Details

The Traceroute Investigation Details view provides information about the route, router hops, and delay from the traceroutes that were performed. Expand the Traceroute Chart and the Traceroute Table to see the data in graphical and tabular formats.

- **ID**. Unique number that identifies the traceroute investigation that was performed.
- **Device/Location/Media Device**. The target of the investigation: Location, Media Device, or Device. Device indicates that an IP address was selected for a traceroute investigation.
- **Collector**. The name of the collector that performed the traceroute. Under certain conditions, more than one collector sends traceroutes to the same target device. The baseline path results are less accurate in such a case because different collectors get different results.
- **Completed On**. The date and time when the traceroute was completed. Depending on the settings for the number of retries and timeout length, this time is different from the time the investigation was first attempted.

- **Trigger**. The reason the traceroute was performed:
  - **Incident**: A traceroute action was launched in response to a threshold violation: a call setup or call server group incident. In some cases, traceroute investigations can be launched in response to other types of incidents, but the results are rarely helpful.
  - **Manual**: A UC Monitor user launched the investigation from the Launch Traceroute page.
  - **Scheduled**: The traceroute was performed as part of baseline threshold monitoring.
- **Name**. The name of the affected endpoint.
- **Address**. The IP address of the affected endpoint.
- **Paths**. The number of unique network paths that were found during this investigation.
- **Avg Delay**. The average number of milliseconds that it took for the traceroute to complete.
- **Protocol**. The protocol that was used for the traceroute.
- **Path**. The sequential number that is assigned to this network path.
- **Hops**. The number of hops found in the network path.
- **Min Delay**. The smallest amount of path delay of all path samples for this unique path.
- **Avg Delay**. The average amount of path delay of all path samples for this unique path.
- **Usage**. The percentage of time that this path was taken during the baseline date range. The value varies based on the number of different paths found.

Baseline Traceroute Details

The Baseline Traceroute Details view lets you compare the routes that call traffic takes through the network. Expand the Traceroute Chart and the Traceroute Table to see the data in graphical and tabular formats.

- **Date Range**. The time period during which the baseline path data was collected. By default, the 14 days before and including the Completed On date. Baseline traceroute data is collected from every voice gateway and every key phone at four-hour intervals.
- **Path**. The sequential number that is assigned to this network path.
- **Hops**. The number of hops found in the network path.
- **Min Delay**. The smallest amount of path delay of all path samples for this unique path.

- **Max Delay**. The greatest amount of path delay of all path samples for this unique path.
- **Avg Delay**. The average amount of path delay of all path samples for this unique path.
- **Usage**. The percentage of time that this path was taken during the baseline date range.
- **Name**. The host name of the target device (destination) for the investigation.
- **Hop Delay**. The average delay for all hops in all paths that were detected during the traceroute.
- **Total Delay**. The cumulative delay for all hops in all paths that were detected during the traceroute.

More information:

Traceroute Investigations (see page 70)
Baseline Traceroutes (see page 72)

Registration Failures Incident Details

A Registration Failures incident is triggered when the number of times that a device attempts unsuccessfully to register with a call server exceeds the threshold. The threshold is 15 registration failures per call server, per 15-minute reporting interval.

This type of incident can indicate a configuration or security problem. Review the Call Performance Incident Details (see page 27) report for call setup failure incidents, which help identify a call server issue.

The Registration Failures Incident Details table contains the following information:

Acknowledged

Whether the incident is acknowledged.

Call Server

The host name and IP address of the call server where the indicated endpoint (see definition on page 112) is trying to register.

Current Location

The Location definition for the subnet of the endpoint.

ID

The identifier for the selected incident. Matches the ID in the Incident Details Overview summary table. UC Monitor assigns the ID.

IP Address

The IP address of the endpoint that reports registration failures.

Name

The name of the endpoint.

Phone Number

The directory number of an endpoint that has registration failures on the call server.

Protocol

The protocol that the call server uses for call setup.

Registration Failures

The number of times the indicated endpoint tried unsuccessfully to register with the call server during the interval when the incident was reported.

Status

Whether the incident is Open or Closed.

Timeframe

The date and time when the incident was reported.

Type

The type of call server incident (Registration Failures).

Poor Call Quality (QRT) Incident and Phone Details

You can drill down from a Poor Call Quality (QRT) incident to review the related Incident Details and Phone Details reports. The views in these reports provide the following information.

Call Server Incident Details

- **ID**. The sequential number that is assigned to this incident. Also a link to more information. Matches the ID in the Call Server Incident table on the Incident Overview (see page 24).
- **Phone Number**. The directory number of the affected endpoint.
- **Name**. The name of the affected endpoint.
- **IP Address**. The IP address of the affected endpoint.
- **Current Location**. The Location definition for the subnet of the endpoint.
- **Call Server/Call Server Group**. The name of the call server or call server group, usually representing a call server cluster, that is associated with the metrics that triggered the incident.
- **Protocol**. The protocol that the endpoint uses for call setup signaling.

- **Trigger**. The type of the trigger for this incident.
  - **In Progress**: The QRT key was pressed while the call was active.
  - **On Hook**: The QRT key was pressed while the telephone was on hook, usually after the call was completed.
- **Time QRT pressed**. The time at which a user pressed the QRT key on a telephone.
- **Type**. Always Poor Call Quality (QRT).
- **QRT Count**. The number of times the QRT key was pressed to trigger the incident.
- **Acknowledged**. Whether the incident is acknowledged.
- **Status**. Whether the incident is Open or Closed.

Origination and Destination Phone Information

Details can include the following items relevant to the phone that placed or received the call:

- **Name**. The name of the affected endpoint.
- **Phone Number**. The directory number of the affected endpoint.
- **IP Address**. The IP address of the affected endpoint.
- **Port**. The port number used on the endpoint.
- **Location**. The Location of the endpoint that reported the incident condition, or the IP address or DNS host name of the device that sent the call from the PSTN.
- **Model/Type**. The vendor-assigned model and type for the endpoint.
- **Call Server**. The name of the call server or call server group, usually representing a call server cluster, that is associated with the metrics that triggered the incident.
- **Codec**. The codec used during the call.
- **Firmware Version**. The firmware name and version number on the endpoint.
- **Serial Number**. The vendor-assigned serial number of the endpoint.
- **Switch Address**, **Name**, **Port**. Network information for the endpoint.

For calls involving an endpoint in the PSTN, the following items are identified:

- The gateway media device that handled the call.
- The interface and voice channel through which the data passed.

Call Leg Details

Below the Call Leg Details table are charts that contain call quality metric details. Examine these metrics to help determine why the user initiated the QRT report. The table can include the following items:

- **Call ID**. The identifier for a call. UC Monitor automatically assigns the ID.
- **Origination Time**. The date and time the call began.
- **Receiving Phone**. The directory number of the endpoint that received the call data. Call quality is rated from the perspective of the listener.
- **Sending Phone**. The Location of the endpoint that sent the call leg that triggered the incident. Can be the IP address or DNS host name of the device when the call involves the PSTN.
- **Type**. Whether the Call Watch data was triggered by QRT or an existing Call Watch definition.
- **Delay to Dial Tone**. The amount of time it takes for a user to hear a dial tone after picking up the receiver of an IP telephone. During the call setup phase of a VoIP call, the device receives messages from the call server to play a dial tone. Users can think that the system is not working when dial tone is delayed.
- **Post Dial Delay**. The amount of time from when a user enters the last digit of a telephone number to when the user hears a ring or busy signal.
- **Mean Opinion Score (MOS)**. The Mean Opinion Score (MOS) is an industry standard method for gauging call quality. MOS is an estimation of how impairments to a voice signal affect listener perception of call quality.

Phone Details

- **ID**. UC Monitor assigns an ID to each endpoint that is discovered during monitoring.
- **Name**. The name of the affected endpoint.
- **Phone Number**. The directory number of the affected endpoint.
- **IP Address**. The IP address of the affected endpoint.
- **Location**. The Location of the endpoint that reported the incident condition, or the IP address or DNS host name of the device that sent the call from the PSTN.
- **Model/Type**. The vendor-assigned model and type for the endpoint.
- **Call Server**. The name of the call server or call server group, usually representing a call server cluster, that is associated with the metrics that triggered the incident.
- **Previous Call Server**. The previous call server, if any, to which the endpoint was registered.
- **Protocol**. The protocol that the endpoint uses for call setup signaling.
- **Last Activity**. The date and time of the last call activity from the endpoint.

- **Status**. Whether the phone is registered.
- **Previous Status**. The previous status, if any.
- **Status Time**. The time at which the current status was verified.
- **Firmware Version**. The firmware name and version number on the endpoint.
- **Serial Number**. The vendor-assigned serial number of the endpoint.
- **Switch Address**, **Name**, **Port**. Network information for the phone.

Phone Call Details

- **Call ID**. The identifier for a call. UC Monitor automatically assigns the ID.
- **Origination Time**. The date and time the call began.
- **Origination Number**. The directory number or SIP URI of the origination endpoint.
- **Origination Location/Media Device**. The Location of the origination endpoint, or the gateway device that forwarded the call from the PSTN.
- **Destination Number**. The directory number or SIP URI of the destination endpoint.
- **Destination Location/Media Device**. The Location of the destination endpoint, or the gateway device that forwarded the call to the PSTN.
- **Duration**. The length of the call.
- **Media Type**. The type of data in the call, either audio or video, or both.
- **Origination MOS**. The Mean Opinion Score at the origination endpoint.
- **Destination MOS**. The Mean Opinion Score at the destination endpoint.
- **Call Setup Failure Code**. The code that the call server returns when a call fails during the setup phase. The code indicates the type of failure.

Phone Status Changes Incident Details and Investigation List

You can drill down from a Phone Status Changes incident to review the related Incident Details and Investigation List reports. The views in these reports summarize the changes to endpoint (see definition on page 112) registration status over the duration of the incident, from Open state to Close state. Changes to status can include such events as failover and branch outages.

The Incident Details and Investigation List reports provide the following views.

Call Server Incident Details

- **ID**. The sequential number that is assigned to this incident. Also a link to more information. Matches the ID in the Call Server Incident table on the Incident Overview (see page 24).
- **Call Server/Call Server Group**. The name of the call server or call server group, usually representing a call server cluster, that is associated with the metrics that triggered the incident.
- **Time Frame**. The date and range of time when the incident was reported.
- **Type**. Always Phone Status Changes, which includes Currently Missing Phones, Recently Moved Phones, or New/Found Phones.
- **Acknowledged**. Whether the incident is acknowledged.
- **Status**. Whether the incident is Open or Closed.

Currently Missing Phones

The Currently Missing Phones view lists all endpoints whose call server registration status is unknown. These endpoints are considered missing. They were previously known to the servers that reported the incident. However, during the last 15-minute reporting interval, these endpoints did not send keepalives to their call servers for a period of 5 minutes. Endpoints that had normal shutdowns are not considered missing. They and their accompanying deregistrations are excluded from this list.

The Currently Missing Phones view provides the following information:

- **Device**. The endpoint that is reported as missing, moved, or new.
- **Phone Number**. The directory number of the affected endpoint.
- **Name**. The name of the affected endpoint.
- **Location**. The Location definition that is associated with the last known subnet of the endpoint.

- **Previous Call Server**. The host name and IP address of the call server where the endpoint was registered before being registered to the current call server.
- **Current Call Server**. The host name and IP address of the call server where the endpoint was most recently registered. This is the call server from which the endpoint is considered missing.
- **Time**. The date and 15-minute time range when the endpoint lost contact.

Recently Moved Phones

The Recently Moved Phones view lists all endpoints that registered to a different call server during the specified 15-minute reporting interval. The table provides the following information:

- **Device**. The endpoint that is reported as missing, moved, or new.
- **Phone Number**. The directory number of the affected endpoint.
- **Name**. The name of the affected endpoint.
- **Location**. The Location definition for the subnet of the endpoint.
- **Previous Call Server**. The host name and IP address of the call server where the endpoint was previously registered.
- **Current Call Server**. The host name and IP address of the call server where the indicated endpoint was registered during the specified reporting interval.
- **Time**. The date and 15-minute time range when the endpoint changed call servers.

New/Found Phones

The New/Found Phones view lists all endpoints that registered or reregistered to a monitored call server group during the specified 15-minute reporting interval.

  - A *new* endpoint has not registered to this call server group since monitoring began.
  - A *found* endpoint lost contact with this call server group in the past, but registered with the same group during the reporting interval.

The New/Found Phones view provides the following information:

- **Device**. The endpoint that is reported as missing, moved, or new.
- **Phone Number**. The directory number of the affected endpoint.
- **Name**. The name of the affected endpoint.
- **Location**. The Location definition for the subnet of the endpoint.
- **Previous Call Server**. If applicable, the host name and IP address of the call server where the endpoint was registered in previous reporting intervals.

- **Current Call Server**. The host name and IP address of the call server where the new or found endpoint was registered during the specified interval.
- **Time**. The date and 15-minute time range during which the endpoint was discovered.

Phone Status Changes

The Phone Status Changes view summarizes the information in the other views on this page. As you read the data in each row from left to right, the resulting statement is similar to the following example:

```
During the specified reporting interval, there were X active endpoints for
this call server group. Y of these endpoints were new, or were found during
the interval. Z of these endpoints lost contact with all call servers in the
group during the interval. And U of these devices moved from one call server
to another during the interval. Therefore, the percentage of device status
changes for that interval is calculated as (Y + Z + U)/X.
```

- **Active Phones**. The total number of endpoints that were active at the beginning of the reporting interval or that were new or found during the interval. Active endpoints were in an active contact status with a call server in the call server group.
- **New/Found Phones**. The number of new or found endpoints that were registered to servers in the call server group during the reporting interval. A *new* endpoint has not registered to this call server group since monitoring began. A *found* endpoint lost contact with this call server group in the past, but registered with the same group during the reporting interval.
- **Missing Phones**. The number of endpoints that were previously registered to servers in the call server group that did not send keepalives during the reporting interval. This value does not include endpoints that had normal shutdowns.
- **Moved Phones**. The number of endpoints that registered to a different call server in this group during the specified reporting interval.
- **Time**. The date and 15-minute time range during which the status changed.

Investigations

The Investigations view identifies the underway or completed investigations that are associated with the selected incident.

- **ID**. The sequential number that is assigned to this incident. Also a link to more information.
- **Trigger**. Identifies the action that launched the investigation. One of the following:
  - **Incident**: A traceroute action was launched in response to a threshold violation: a call setup or call server group incident. In some cases, traceroute investigations can be launched in response to other types of incidents, but the results are rarely helpful.
  - **Manual**: A UC Monitor user launched the investigation from the Launch Traceroute page.
  - **Scheduled**: The traceroute was performed as part of baseline threshold monitoring.
- **Target Type**. The target of the investigation: Location, Media Device, or Device. Device indicates that an IP address was selected for a traceroute investigation.
- **Target Name**. The host name of the target device (destination) for the investigation.
- **Target Address**. The IP address of the target device (destination) of the investigation.
- **Collector**. The name of the collector that performed the traceroute. Under certain conditions, more than one collector sends traceroutes to the same target device. The baseline path results are less accurate in such a case because different collectors get different results.
- **Completed On**. The date and time when the traceroute was completed. Depending on the settings for the number of retries and timeout length, this time is different from the time the investigation was first attempted.

Collector Incidents

The Collector Incidents page provides information about each incident reported against collector performance. Collector incidents identify problems that are local to a specific collector. The incidents help you detect and troubleshoot problems with SPAN port configuration and alert you to collector performance issues.

Collector incidents are not reported for Lync collectors that monitor Microsoft unified communications deployments. Only the Abnormal Termination incident applies to Avaya deployments.

Tip: Review the Collector Incidents page regularly to verify that SPAN ports are configured correctly and that the collectors in your system perform as expected.

The following information is provided in the Collector Incidents table. Detailed information is available when you click the linked incident ID number.

Acknowledged

Select the check box next to the incident that you want to acknowledge and click Apply. Acknowledging an incident lets other operators know that you are aware of the associated condition.

Collector

The host name or IP address of the collector where the incident was reported.

Duration

The length of time that the performance condition persisted.

ID

The sequential number that is assigned to this incident. Also a link to more information.

State

Whether the incident is Open or Closed.

Time

The date and time when the incident was reported.

Type

The type of collector incident. Corresponds to the collector threshold that was exceeded.

- **Abnormal Termination**. Indicates that a service on the collector logged a fatal exception. The path to the collector crash dump log file is provided.
- **Discarded Packets**. The percentage of discarded packets that exceeds the threshold. Traffic bursts can cause the collector to discard packets.

- **Duplicate Packets**. The percentage of traffic that was composed of duplicate packets exceeds the threshold. Usually indicates a SPAN port issue.
- **Lost Bytes**. The percentage of lost traffic that exceeds the threshold. Bytes of data are lost due to traffic bursts that exceed UC Monitor capacity for analysis. Some messages are too large to fit in one packet and are split into separate packets. Packets can also arrive out of order. To handle these situations, the collector reassembles packets using their sequence numbers. Missing sequence numbers indicate that expected bytes were not received. These bytes are counted as lost.

  A high number of lost bytes usually indicates a one-way spanning problem. The collector sees only the packets from one side of a conversation. Therefore, the collector counts the data flowing in the other direction as lost because it cannot see the data.

Filter Collector Incidents

The Incident Overview provides filters for narrowing the scope of the data that is displayed in the Collector Incidents list. These filtering options are associated with your user account and persist across login sessions.

Follow these steps:

1. Select an option from the Incident State field to see incidents for the selected state or status:
   - Open and Closed. Select this default option to show all incidents.
   - Open. Select this option to show only incidents in the Open state.
   - Closed. Select this option to show only incidents in the Closed state.
2. Select a time value in the Minimum Duration field to see incidents that remained open for no less than the time you select.

   Duration does not apply to Abnormal Termination incidents, which are instantaneous. Select “15 minutes” or “Any” to see Abnormal Termination incidents in the list.

Collector Incident Details

Two tables appear when you drill down from the Collector Incidents (see page 40) list:

- The Collector Incident Details table summarizes the incident-related information from the Collector Incidents list.
- The Collector Incident Metric Details table provides more information about the type of incident identified in the Type field in the Collector Incident Details table.

The following fields are available in the Collector Incident Metric Details table. The fields are specific to a particular type of collector incident.

Abnormal Termination Incident

- **Dump File on Collector**. The file name and path to a crash dump file that was created automatically in response to the collector service fatal exception event.
- **Packet Capture on Collector**. If available, the file name and path to a packet capture file that was created automatically in response to the collector service fatal exception event.

Discarded Packets Incident

Packets are discarded due to traffic bursts that exceed UC Monitor capacity for analysis. Packets are dropped when they arrive for processing but the collector is too busy to receive them.

- **Total Number of Packets**. The number of packets that were processed during the reporting interval when the incident was triggered.
- **Discarded Packets (%)**. The amount of data that the collector discarded, expressed as a percentage of all packets that were processed during the reporting interval.

Duplicate Packets Incident

A problem at the monitored SPAN port causes packet duplication. Duplication can occur when a misconfigured SPAN sends packets to the collector twice.

Duplicate packets also indicate network problems. Packets are retransmitted when a response to a previous transmission is not received in time.

- **Total Number of Packets**. The number of packets that were processed during the reporting interval.
- **Duplicate Packets (%)**. The amount of data that the collector treated as duplicated, expressed as a percentage of all packets that were processed during the reporting interval.
- **Duplicate Packets**. The amount of data that the collector treated as duplicated, expressed as a whole number. The total number of duplicate packets for the reporting interval.

Lost Bytes Incident

Bytes of data are lost due to traffic bursts that exceed UC Monitor capacity for analysis. Some messages are too large to fit in one packet and are split into separate packets. Packets can also arrive out of order. To handle these situations, the collector reassembles packets using their sequence numbers. Missing sequence numbers indicate that expected bytes were not received. These bytes are counted as lost.

A high number of lost bytes usually indicates a one-way spanning problem. The collector sees only the packets from one side of a conversation. Therefore, the collector counts the data flowing in the other direction as lost because it cannot see the data.

- **Total Number of Bytes**. The number of bytes that were processed during the reporting interval when the incident was triggered.
- **Lost Bytes (%)**. The amount of data that the collector recorded as lost, expressed as a percentage of all bytes processed during the reporting interval.

Incident Investigations

The Investigations page describes all traceroute investigations that ran in response to incidents during the selected time frame. The table lets you drill down into individual Traceroute Investigation Details reports. Filtering by group is available when custom groups are defined in CA Performance Center. Click the Settings link for filtering options.

Investigations are linked to the Incident Overview because many investigations are launched automatically, in response to an Incident report of a call setup threshold violation. However, investigations can also be launched in the following ways:

- Regularly to establish a baseline of data, such as the most popular route for call setup flows between certain Locations and call servers.
- On demand by users with the appropriate permissions to launch investigations manually.

The Investigations table contains the following information:

Collector

The name of the collector that performed the traceroute. Under certain conditions, more than one collector sends traceroutes to the same target device. The baseline path results are less accurate in such a case because different collectors get different results.

Completed On

The date and time when the traceroute was completed. Depending on the settings for the number of retries and timeout length, this time is different from the time the investigation was first attempted.

ID

Unique number that identifies the traceroute investigation that was performed.

Target Address

The IP address of the target device (destination) of the investigation.

Target Name

The host name of the target device (destination) for the investigation.

Target Type

The target of the investigation: Location, Media Device, or Device. Device indicates that an IP address was selected for a traceroute investigation.

Trigger

The reason the traceroute was performed:

- Incident: A traceroute action was launched in response to a threshold violation: a call setup or call server group incident. In some cases, traceroute investigations can be launched in response to other types of incidents, but the results are rarely helpful.
- Manual: A UC Monitor user launched the investigation from the Launch Traceroute page.
- Scheduled: The traceroute was performed as part of baseline threshold monitoring.

More information:

Traceroute Investigations (see page 70)

Quality Reports

The Monitoring, Quality category provides the following reports:

Worst Locations

The Worst Locations report is an overview of the worst-performing pairs of Locations, according to their performance data and the associated performance thresholds.

Worst Phones

The Worst Phones report is an overview of the worst-performing endpoints (see definition on page 112), according to their MOS or Network MOS and associated performance thresholds.

Worst Locations

The Worst Locations view shows the pairs of talkers (Locations) that had the lowest-quality metrics for the data traveling between them. The Severity Breakdown bar chart shows a comparison of data ratings by severity. Each bar represents 100 percent of the rated quality data for the type of metric you select. Colored portions indicate the relative percentages of data that received each severity rating: Normal, Degraded, Excessive, or Unrated.

The Worst Locations view provides the following information:

Call Minutes

The number of minutes that calls were active between this pair of Locations.

Call Server

The call server that handled the calls whose severity breakdown is shown.

Calls

The number of distinct calls that ran between this pair of Locations and that also contributed to the “worst” performance metric displayed in the table.

The number is a link to the Calls Overview table, which is filtered to show calls that ran between the selected pair of Locations or media devices.

The number does not always match the number of calls shown in the Calls Overview table. The Worst Locations table defines a call as a unique combination of Origination Location, Destination Location, and originating call server. Endpoints in the same Location can be registered to different call servers. Calls for these endpoints are counted separately in the Worst Locations report. But all calls appear in the Calls Overview table.

Metric Type

Select the type of data to display in the report. Select from all available call setup and call quality performance metrics.

Receiving Name

The name of the Location or media device that received the data stream with poor performance metrics. Or, the name of the origination Location when the view is filtered by a call setup metric.

Sending Name

The name of the Location or media device that sent the low-performing call data to the other Location in the pair.

Severity Breakdown

A stacked bar chart that adds up to 100 percent. Each applicable severity rating is shown as a portion of that 100 percent and color-coded to match the severity indicators in other reports. Click a bar to drill down to the call traffic. From this filtered view, a helpful next step is to review the Call Leg Details (see page 20) report. Click the link from the Related Reports section.

Severity Percentages

The actual severity percentages for the selected metric type. Percentages always total 100 percent:

- Unrated
- Normal
- Degraded
- Excessive

Tip: An Unrated metric indicates that a threshold is disabled, or that the threshold for minimum observations is too high for typical levels of call traffic. Set a lower minimum for observations to allow all metrics to be rated or assign different thresholds to the Location pairs in question.

More information:

Calls Overview (see page 49)

Worst Phones

The Worst Phones view identifies the endpoints (see definition on page 112) with the lowest MOS or Network MOS for the selected time frame. The report provides MOS by default. Network MOS is available only when the report contains data from a monitored Microsoft environment.

The Worst Phones view provides the following information:

Average Metric

The average MOS or Network MOS for all calls to and from this endpoint during the selected time frame. Your selection in the Metric Type field determines the contents of this column.

Call Minutes

The number of minutes that calls were active on this endpoint during the selected timeframe.

Calls

The number of distinct calls that were placed or received. The number is a link to the Calls Overview (see page 49) table. All calls from the time frame of the Worst Phones table are included in the Calls Overview table when the endpoint is the Origination or Destination Number.

IP Address

The IP address of the endpoint.

Metric Type

The selection in this field, MOS or Network MOS, determines the contents of the Average Metric and Severity Breakdown columns. This field appears only when the report contains data from a monitored Microsoft environment.

Name

The name, MAC address, or DNS host name of the endpoint.

Phone Number

The directory number or SIP URI of the endpoint. The number is a link to the Phone Details (see page 74) and Phone Call Details tables for the selected endpoint.

Severity Breakdown

A stacked bar chart that adds up to 100 percent. Each severity rating is shown as a portion of that 100 percent and color-coded to match the severity indicators in other reports.

Show breakdown values as

The selection in this field, Calls or Percentages, determines the contents of the Unrated, Normal, Degraded, and Excessive columns.

Unrated, Normal, Degraded, Excessive

Depending on your selection in the "Show breakdown values as" field, these columns display one of the following formats:

- The percentage of calls in a severity category. Percentages always total 100 percent.
- The number of calls in a severity category

More information:

Phones (see page 73)
Calls Overview (see page 49)
Call Quality Metrics (see page 97)

Chapter 2: Reports for Troubleshooting

The Troubleshooting reports help network engineers explore performance metrics, plan for network changes, and identify performance issues at a particular VoIP endpoint. Data from SPAN and from endpoints (see definition on page 112) while calls are in progress help you track statistics such as:

- When calls are made
- Where calls come from
- How calls are routed
- How calls perform

This section contains the following topics:

Calls Overview

The Calls Overview summarizes recent call activity. Call IDs are links to information about the origination and destination endpoints or phones, and to metrics used in calculating MOS and deriving averages.

For Cisco video call legs or a Microsoft environment, the Media Type list lets you filter the list of calls.

- **All**: All calls are displayed. Calls without video streams have the Audio media type. Calls with accompanying video streams usually have the Audio/Video media type and provide correlated call leg statistics in the Session Details view.
- **Audio**: Audio-only calls are displayed, and for calls with an accompanying video stream, only metrics from the audio stream are displayed.
- **Video**: All calls that include a video component are displayed, but only the metrics from the video stream are displayed.
- **Audio/Video**: (*Microsoft only*) All calls that include a video component are displayed, and their audio and video components are correlated. Separate audio and video streams are represented together in a Session Details (see page 59) data view. A Session ID represents calls correlated in this way.

The Calls Overview provides the following information about calls from the selected time frame.

Abandoned Calls

A call attempt in which an endpoint initiated a call that was disconnected before the call was answered. Abandoned calls have an origination time and a disconnect time, but no connect time. This field is not displayed by default. Click the Settings link to display the field.

Call ID

The identifier for a call. UC Monitor automatically assigns the ID. The call ID is a link to details about each call that was made during the selected time frame.

Call Setup Failure Code

The code that the call server returns when a call fails during the setup phase. The code indicates the type of failure. For more information, see the list of call termination cause codes on the Cisco website.

Destination IP Address/Port

The IP address and port number of the destination endpoint. These fields are hidden by default. Click the Settings link to display the fields.Video Packet Loss

The packet loss of the data stream, as perceived at the origination and destination endpoints. This field is hidden by default. Click the Settings link to display the field.

Destination Location/Media Device

The Location of the destination endpoint, or the gateway device that forwarded the call to the PSTN.

Destination MOS

The MOS of the data stream, as perceived at the destination endpoint.

Destination Number

The directory number or SIP URI of the destination endpoint. For a medianet environment, a dash appears in place of the directory number. Medianet provides only the origination IP address and port.

Duration

The length of the call.

Media Type

The type of data in the call, either audio or video, or both.

(*Microsoft only*) The Audio/Video media type refers to calls with correlated call leg data that is displayed in a Session Details view.

Origination IP Address/Port

The IP address and port number of the origination endpoint. These fields are hidden by default. Click the Settings link to display the fields.

Origination Location/Media Device

The Location of the origination endpoint, or the gateway device that forwarded the call from the PSTN.

Origination MOS

The MOS of the data stream, as perceived at the origination endpoint.

Origination Number

The directory number or SIP URI of the origination endpoint.

Origination Time

The date and time the call began.

Session ID

(*Microsoft only*) The session identifier that UC Monitor assigns to identify call legs for audio and video data streams that are part of the same call. The session ID is a link to the Session Details page, which provides detailed data about each separate leg in the correlated session.

More information:

Mean Opinion Scores (see page 102)
Report Metrics (see page 97)
Call Termination Cause Codes (see page 105)

Call Details

The Call Details report provides in-depth metrics for the calls in the Calls Overview (see page 49) and the Watched Calls (see page 65) reports.

This report contains data views that provide information about the selected call and the endpoints (see definition on page 112) involved in that call, plus call quality metrics for each side of a bi-directional call. The raw metrics presented in these views are useful for scenarios such as:

- Understanding how other quality metrics, such as the MOS, were derived.
- Looking closely at individual metrics, such as jitter.
- Tracking metrics that are not used in calculating the MOS, such as echo, noise, video metrics, and burst.
- Understanding the performance of traffic flowing through medianet-enabled devices.

Call Information Table

The Call Information table provides the following information.

ID

The identifier for a call. UC Monitor automatically assigns the ID.

Origination Number

The directory number or SIP URI of the origination endpoint.

Origination Called Number

The phone number or SIP URI that was dialed.

Destination Number

The directory number or SIP URI of the destination endpoint.

Origination Time

The date and time the call began.

Duration

The length of the call.

Media Type

The type of data in the call, either audio or video, or both.

(*Microsoft only*) The Audio/Video media type refers to calls with correlated call leg data that is displayed in a Session Details view.

Origination and Destination Information Tables

These tables provide information about the origination or destination endpoint (see definition on page 112):

- Phone number or SIP URI
- IP address, which can be a link to the endpoint web page.
- Port through which the data passed
- Endpoint make and model
- Call server
- Codec
- Firmware version
- Serial number

- Connection information
- For calls that involve endpoints in the PSTN:
  - Voice gateway
  - Interface and voice channel through which the data passed

Availability of this information depends on the type of endpoint, or how UC Monitor learned of the data. When a call server is unknown, the label <Unknown Call Server> is used.

Call Quality Metrics Tables

The Origination and Destination Call Quality Metrics tables provide the following information, depending on the environment you are monitoring. Video metrics are reported in the following circumstances:

- When monitoring video over IP in a Microsoft environment.
- When medianet data is available.
- When monitoring Cisco video calls.

802.1p Tag

(*Avaya only*) The QoS priority setting for voice packets at Layer 2.

ACOM

(*Cisco PSTN calls only*) The total echo return loss on the network. ACOM measures how significantly the voice gateway reduced the echo. ACOM includes echo reduction that occurs with or without the activity of an echo cancellation device.

Burst Density

The percentage of packets within burst periods that are lost or discarded.

Burst Duration

The average duration of all high-loss periods in a data stream.

Concealment Ratio (Cumulative or Maximum)

The percentage of frames in a data stream that are concealment frames, which the endpoints generate to conceal packet loss. Includes both early and late packets. Either the cumulative value (the total since the beginning of the stream) or the maximum value (the highest since the beginning of the stream).

Conversational MOS

The Mean Opinion Score (MOS) based on metric factors from both directions of data flow.

Differentiated Services Code Point

The Differentiated Services Code Point setting of the incoming RTP packets.

Echo Tail Length

The “length” of echo cancellation processing. Based on the distance between a voice gateway and the endpoint. Typical values range from 8 milliseconds to 32 milliseconds.

Frozen Period

The average length of frozen video instances.

Frozen Video Frequency

The frequency of long and noticeable frozen video periods for an entire session. Expressed as a percentage of session time.

Gap Density

The percentage of lost or discarded packets in the gaps between bursts in a data stream.

Gap Duration

The average duration of periods of good performance (low loss) between periods of data loss in a data stream.

Jitter, Jitter (Max)

Packet delay that distorts the quality of a voice conversation.Either the average or the maximum value.

The definition of Max Jitter varies by monitored environment:

- Avaya environment: The maximum jitter per interval
- Cisco environment: The maximum jitter thus far

Jitter Buffer Delay

Delay that the jitter buffer introduces while it holds one or more packets to reduce variations in packet arrival times.

Jitter Buffer Loss

The packets that are lost when jitter hinders the caching capacity of the jitter buffer.

Jitter Buffer Over Runs

The number of times that jitter exceeded the maximum size setting of the jitter buffer. Packets arrive too quickly to be contained by the jitter buffer. Over runs usually result in packet loss.

Jitter Buffer Under Runs

The number of times that the jitter buffer became empty. Packets arrive too slowly to be contained by the jitter buffer. Under runs usually indicate that delays are too lengthy for the buffer setting.

Latency

One-way delay. Calculated from the origination party to the destination party. Includes propagation delay, network delay, and packetization delay.

Listening MOS, Listening MOS (Min)

The Mean Opinion Score, which is based on call legs traveling toward the endpoint to reflect listener perception of quality. Either the average or minimum value.

Time-To-Live (Minimum or Maximum)

A counter embedded in data to prevent a data packet from circulating through the network indefinitely. The counter decrements each time that the packet passes through a router or a switch. Either the minimum or maximum value in a data stream.

MOS, MOS (Min)

The Mean Opinion Score (MOS) is an industry standard method for gauging call quality. MOS is an estimation of how impairments to a voice signal affect listener perception of call quality. Either the average or the minimum value.

Network MOS, Network MOS (Min)

MOS listening quality value that is based only on network factors, such as codec, packet loss, packet reordering, packet errors, and jitter. Either the average or the minimum value.

Network MOS Degradation

(*Microsoft only*) A breakdown of the metrics used in calculating the Network MOS. Degradation refers to an impairment factor that decreased the MOS.

- **Network MOS Degradation**. Average network MOS degradation for the data stream.
- **Network MOS Degradation (Max)**. Highest degradation observed for the stream.
- **Jitter and Packet Loss Degradations**. Percentage of the overall Network MOS Degradation metric that is attributed to the individual metrics.

Noise Level

(*Microsoft only*) The average portion of an audio signal that is noise and not actual voice data. Measured in decibels.

Packet Loss, Packet Loss (Max)

The percentage of data packets that were lost in transit. These packets were sent but never received at the destination. Either the average or the maximum value.

Packets Lost

The number of data packets that were lost in transit. These packets were sent but never received at the destination.

Packets Received

The number of data packets in a stream.

Resolution

The width and height of the video image, in pixels. Higher resolutions require more bandwidth and more CPU resources. Support for resolution settings varies by endpoint.

Sequence Falls, Maximum Sequence Falls

The number of times that at least one packet arrived out of order. Either the actual or the maximum number of sequence falls in a stream.

Sequence Jumps, Maximum Sequence Jumps

The number of times that at least one consecutive packet was lost. Either the actual or the maximum number of sequence jumps in a stream.

Severely Concealed Seconds

The number of call seconds that had more than 5 percent concealment events from the start of the data stream.

Signal Level

The average audio signal level in decibels (dBm0).

Video Bit Rate, Video Bit Rate (Max)

The number of bits sent per second for an entire video stream. Bit rates provide a gauge of codec performance. Either the average or the maximum value.

Video Consecutive Packet Loss

The percentage of consecutive video packets that were lost in transit.

Video Frame Decoding Time

The average amount of time for decoding frames in a stream. A slower decoding rate can be the result of conditions on the endpoint, such as lack of CPU resources, and can affect call quality.

Video Frame Loss

The average number of unique consecutive images, or video frames, lost due to corruption and error concealment for the entire system. Video frames can span multiple packets.

Video Frame Rate

The average number of frames that were sent or received per second for an entire stream.

Video Jitter, Video Jitter (Max)

The variation in delay among video packets in the same stream. Either the average or the maximum value.

Video Latency

The maximum time for a video packet to travel between the calling parties. Measured from end-to-end in one direction. Calculated by taking the average round-trip time for a call leg in a given video call and dividing it in half.

Video Packet Loss, Video Packet Loss (Max)

The percentage of video packets that were lost in transit. These packets were sent but never received at the destination. Either the average or the maximum value.

Video Packets Received

The number of video packets in a stream.

Video Packets Lost

The number of video packets that were lost in transit. These packets were sent but never received at the destination.

VLAN ID

(*Avaya only*) The VLAN that carried the RTP packets.

Call Setup Metrics Tables (Audio)

Call setup metrics are provided for audio-only calls that use Cisco Unified Communications Manager for call setup. The Call Setup Failure metric is also available from Microsoft environments.

Values for delay to dial tone and post-dial delay are collected only from the origination endpoint.

The server returns a call termination cause code when the call fails during the setup phase. The code indicates the type of failure. For more information, see the list of call termination cause codes on the Cisco website.

More information

Call Termination Cause Codes (see page 105)

Midstream Details Tables

The Midstream Details tables provide in-depth data about the performance of RTP traffic flows (or stream legs) on the medianet-enabled devices on your network.

Device

The name of the medianet-enabled (midstream) device that is associated with the stream leg. Can be the DNS name or the IP address.

DSCP

The Differentiated Services Code Point setting of the incoming RTP packets.

Egress Interface

The interface where traffic exits a device.

Forwarding Status

When applicable, this field explains why a medianet-enabled device did not forward packets as expected. For example, when the device drops packets, this field provides a Status of “Dropped” and a Reason such as “Bad TTL.”

The device manufacturer provides the Status and Reason descriptions. Two other descriptions provided in this field, Value and Extended, are not defined, and can vary depending on the Status and Reason.

When a forwarding status is available, this field contains a blue “i” icon. Position your mouse pointer over the “i” to review the information.

Ingress Interface

The interface where traffic enters a device.

Jitter, Max Jitter

Packet delay that distorts the quality of a voice conversation. Either the average or the maximum value for the stream leg.

Packet Loss

The percentage of data packets that were lost in transit. These packets were sent but never received at the destination.

Packet Rate

The number of data packets that are received per second. UC Monitor uses this value to determine whether an RTP stream is audio or video for medianet-enabled devices that do not report a codec.

Packets Dropped

The number of data packets that reached the destination, but then were discarded.

Packets Lost

The number of data packets that were lost in transit. These packets were sent but never received at the destination.

Packets Received

The number of data packets in a stream.

TTL

(Time to Live) A counter embedded in data to prevent a data packet from circulating through the network indefinitely. The counter decrements each time that the packet passes through a router or a switch.

Video Jitter, Video Max Jitter

The variation in delay among video packets in the same stream. Either the average or the maximum value.

Video Packet Loss

The percentage of video packets that were lost in transit. These packets were sent but never received at the destination.

Video Packet Rate

The number of video packets that were received per second.

Video Packets Dropped

The number of video packets that reached the destination, but then were discarded.

Video Packets Lost

The number of video packets that were lost in transit. These packets were sent but never received at the destination.

Video Packets Received

The number of video packets in a stream.

Session Details

Calls that Microsoft devices process can consist of simultaneous audio and video data streams with distinct call legs:

- Legs that traveled between a user and a conferencing server.
- Legs that carried the audio stream.
- Legs that carried video data.

UC Monitor uses an identifying session ID to correlate call data for such calls. The collector assigns the ID when it detects similarities in endpoint identities and timestamps among audio and video data streams.

When session information is available for a call, the Session ID in the Calls Overview table provides a link to correlated session details.

When the collector identifies data from call legs as being from the same call, a separate data view shows statistics for each correlated data stream. A shared session ID identifies the correlated streams. The following table summarizes the data in the Session Details table:

Call ID

The identifier for a call. UC Monitor automatically assigns the ID.

Call Setup Failure Code

The code that the call server returns when a call fails during the setup phase. The code indicates the type of failure. For more information, see the list of call termination cause codes on the Cisco website.

Destination Location/Media Device

The Location of the destination endpoint, or the gateway device that forwarded the call to the PSTN.

Destination MOS

(*Audio streams only*) The MOS of the data stream at the destination endpoint.

The Mean Opinion Score (MOS) is an industry standard method for gauging call quality. MOS is an estimation of how impairments to a voice signal affect listener perception of call quality.

Destination Number

The directory number or SIP URI of the destination endpoint.

Duration

The length of the call.

Media Type

The type of data in the call, either audio or video, or both.

Origination Location/Media Device

The Location of the origination endpoint, or the gateway device that forwarded the call from the PSTN.

Origination MOS

(*Audio streams only*) The MOS of the data stream at the origination endpoint.

Origination Number

The directory number or SIP URI of the origination endpoint.

Origination Time

The date and time the call began.

More information:

Mean Opinion Scores (see page 102)
Call Termination Cause Codes (see page 105)

Session Information Table

The Session Information table provides the following information.

ID

The session identifier that UC Monitor assigns to identify call legs for audio and video data streams that are part of the same call.

Origination Number

The directory number or SIP URI of the origination endpoint.

Origination Called Number

The phone number or SIP URI that was dialed.

Destination Number

The directory number or SIP URI of the destination endpoint.

Origination Time

The date and time the call began.

Duration

The length of the call.

Media Type

The type of data in the call, either audio or video, or both.

Origination and Destination Information Tables

These tables provide detailed information about the origination and destination endpoints (see definition on page 112). The following information is provided, when available.

Endpoint Name

The host name of the endpoint, or the name that is associated with the device.

User Agent

The firmware version and build of the software on the endpoint.

URI

The SIP URI of the logged-in user who is associated with the endpoint.

Other Information Table

This table provides information to identify conference calls, such as the Conference URI. The Conferencing Server generates the URI and usually includes the URI of the user who established the conference call.

Session Details Table

When call data is available from call legs in the same call, a separate data view shows statistics for each correlated data stream. A shared session ID identifies the correlated streams.

The Session Details table provides the following information:

Call ID

The identifier for a call. UC Monitor automatically assigns the ID.

Call Setup Failure Code

The code that the call server returns when a call fails during the setup phase. The code indicates the type of failure. For more information, see the list of call termination cause codes on the Cisco website.

Destination Location/Media Device

The Location of the destination endpoint, or the gateway device that forwarded the call to the PSTN.

Destination MOS

(*Audio streams onl*y) The MOS of the data stream, as perceived at the destination endpoint.

Destination Number

The directory number or SIP URI of the destination endpoint.

Duration

The length of the call.

Media Type

The type of data in the call, either audio or video, or both.

Origination Location/Media Device

The Location of the origination endpoint, or the gateway device that forwarded the call from the PSTN.

Origination MOS

(*Audio streams only*) The MOS of the data stream, as perceived at the origination endpoint.

Origination Number

The directory number or SIP URI of the origination endpoint.

Origination Time

The date and time the call began.

More information:

Call Quality Metrics (see page 97)
Mean Opinion Scores (see page 102)
Call Termination Cause Codes (see page 105)

Midstream Metric Details

When the Call Details (see page 51) page contains Midstream Details tables, you can drill down to the Midstream Metric Details page for per-metric information.

The Origination and Destination Information sections provide details about the endpoints (see definition on page 112) at the beginning and end of the stream.

The Midstream Details tables (see page 57) are the same as the tables on the Call Details page.

The charts on the Midstream Metric Details page are graphical representations of the metrics from the Midstream Details tables: one chart per metric per direction of the stream leg. Each chart graphs the metric over the length of the stream leg for the ingress and egress interfaces of medianet-enabled devices.

Important: When Call Watch and midstream data exist for a call, the Midstream Metric Details (see page 63) charts are part of the Call Watch Details report.

Understanding Uncorrelated Call Legs

When intermediate devices handle a call, that call is displayed in the Calls Overview table as multiple call legs. The following scenario is an example:

First Leg

Call ID 6381

Origination Time: 06/12/2009 18:23:18

Origination Number: 15225678910

Origination Location/Media Device: 172.12.3.45

Destination Number: 1718

Destination Location/Media Device: mediator01.ca

Duration: 37 seconds

Media Type: Audio

Second Leg

Call ID 6382

Origination Time: 06/12/2009 18:23:18

Origination Number: 15225678910

Origination Location/Media Device: mediator01.ca

Destination Number: Jack.Smith@ca.com

Destination Location/Media Device: Austin HQ

Duration: 37 seconds

Media Type: Audio

The separate legs represent one call, as you can see from the identical Origination Times, Origination Numbers, and Durations. They are shown in separate table rows because the call created distinct legs:

- One leg traveled between 5225678910 (the Origination Number in the PSTN) and the Mediation Server (mediator01), which translated it for the VoIP network.
- The other leg traveled between the Mediation Server and Jack (the destination endpoint).

The separate metrics are useful for determining the location of a performance issue, at the Mediation Server, for example.

The presence of quality metrics in a Call Quality Metrics table depends on information that is available from the endpoints that handled a call leg.

Watched Calls

The Watched Calls report presents the metrics that were collected from watched calls. The time frame selector lets you view data from calls that were watched previously. You can select a fifteen-minute or one-hour time frame.

The report provides the following information. For each call, two MOS bar charts are shown, one for each call leg.

Note: Older or lower-end IP phones do not support the collection of all Call Watch statistics. The Call Watch reports show data only for supported statistics.

Call ID

The call identifier that UC Monitor assigns for each call that a watched device made or received during the call watch period. The call ID is a link to the Call Watch Details (see page 66), which provides detailed information about a watched call.

(*Avaya only*) An asterisk (*) appears next to the call ID when the origination party cannot be determined from collected data. Usually indicates that CDR collection is not enabled on an Avaya system.

Delay to Dial Tone

(*Cisco only*) The amount of time it takes for a user to hear a dial tone after picking up the receiver of an IP telephone. During the call setup phase of a VoIP call, the device receives messages from the call server to play a dial tone. Users can think that the system is not working when dial tone is delayed.

Destination Location/Media Device

The Location of the destination endpoint, or the gateway device that forwarded the call to a point in the PSTN. These fields are not displayed by default. Click the Settings link to display the fields.

Destination Phone

The directory number or SIP URI of the destination endpoint.

MOS

The Mean Opinion Score (MOS) is an industry standard method for gauging call quality. MOS is an estimation of how impairments to a voice signal affect listener perception of call quality.

Origination Location/Media Device

The Location of the origination endpoint, or the gateway device that forwarded the call from a point in the PSTN. These fields are not displayed by default. Click the Settings link to display the fields.

Origination Phone

The directory number or SIP URI of the origination endpoint.

(*Avaya only*) The origination and destination parties are identified after the call completes. An asterisk next to the Call ID indicates that the origination endpoint was not determined from the collected data. A common cause for this problem is when CDR collection is not enabled.

Origination Time

The date and time the call began.

Post-Dial Delay

The amount of time from when a user enters the last digit of a telephone number to when the user hears a ring or busy signal.

Type

The type of Call Watch:

- **Automatic**. Applies to all Avaya calls, which are automatically watched.
- **Defined**. Applies to a Call Watch you set up manually, or to a Call Watch that was launched in response to a user QRT event. QRT events are specific to Cisco phones.

Call Watch definitions are created in the Troubleshooting, Call Watch, Definitions area.

More information:

Mean Opinion Scores (see page 102)

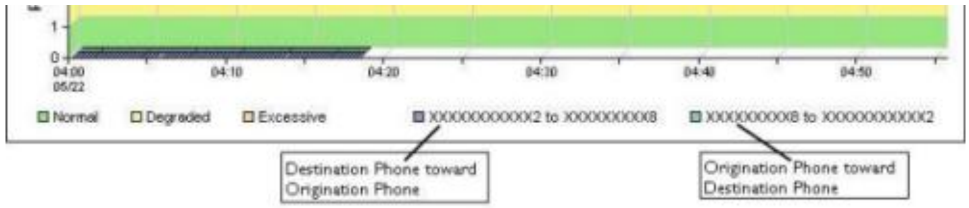
Call Watch Details

The Call Watch Details page is a subset of the Watched Calls (see page 65) report and displays information specific to the Call ID you selected. The following details are available, depending on the monitored equipment. Some metrics are specific to a hardware vendor, and some endpoints provide different metrics.

Important: When Call Watch and midstream data exist for a call, the Midstream Metric Details (see page 63) charts are part of the Call Watch Details report.

When number-masking is used, the charts in the Call Watch Details can be difficult to interpret. Chart legends identify each call leg by the directory number. Masking limits the display to specific digits of the directory number, such as only the first digit. In this scenario, the legends are useless because often the first digits in a dial plan are identical.

To decipher the Call Watch Details charts when masking is applied, understand that legend entries are positional. From left to right, the first pair of masked numbers indicates Destination endpoint to Origination endpoint. The second pair indicates Origination endpoint to Destination endpoint, as shown in the following example:



The Call Watch Details page provides the following information:

Call Leg Details

Summarizes the metrics shown in the Watched Calls (see page 65) report with the MOS rating broken out per call leg, or directional flow of the conversation.

Conference ID

Avaya identifier for a voice gateway call.

Destination Phone Information

Information about the destination endpoint, including the directory number, make and model, call server, voice gateway, codec, firmware version, serial number, and switch connection type. Both the availability and reliability of this information depends on the type of endpoint and your configuration.

Origination Phone Information

Information about the origination endpoint, including the directory number, make and model, call server, voice gateway, codec, firmware version, serial number, and switch connection type.

Availability of this information depends on the type of endpoint.

In an Avaya environment, information about the origination endpoint requires CDR data from the Communication Manager and is available after the call has completed. While the call is in progress, either endpoint can be the origination or destination party. The origination and destination parties can be identified only when CDRs are enabled. When the origination and destination parties are not identified, the Call ID and directory number fields contain an asterisk (*).

Switch

The name, IP address, and port of the switch to which the endpoint is connected. This information comes from the web page of the endpoint, which uses Cisco Discovery Protocol (CDP) to collect the information. If the switch does not support CDP, the next CDP-enabled device in the path of the watched call is shown.

Trunk Group

Trunk group number for a voice gateway call. While the call is still in progress, the directory number of the endpoint sending data through the gateway cannot be identified. The Trunk Group ID is substituted for the directory number, and displayed in the Phone Number field, until this information is received.

Call Quality metrics

For definitions of these metrics, see Call Quality Metrics (see page 97).

More information:

Mean Opinion Scores (see page 102)
Report Metrics (see page 97)

Call Path

The Call Path report is available only for Avaya Communication Manager environments. This report uses the call path data from the RTCP packets that were collected from Avaya endpoints during a call. The Call Path report compares path data from the present call with historical data from the same endpoints to provide a baseline call path.

The Call Path report shows current and baseline path information for each direction of call data flow. More information is also provided about the route, router hops, and per-hop delay.

By default, baseline path data is available when calls between the same endpoints, in the same Locations, were observed in the past. This information is kept in the database for up to 14 days and is used to determine a base path. This information lets you compare the routes that calls take through the network and gauge whether calls are routed properly. Expand the Path Chart or Path Table areas to see the data in the format you prefer, either graphical or tabular. Click the Settings link to remove the Baseline Call Path Details chart and table from the report.

The Call Path report provides the following information for individual paths and baseline call paths:

Address

The IP address of the destination endpoint.

Avg Delay

One of the following:

- The average delay for all hops in all paths that were detected between these endpoints.
- The average amount of path delay of all path samples for this unique path.

Completed On

The date and time when the call was completed.

Date Range

The time period during which the baseline path data was collected. The data comes from that were made calls between the endpoint Locations during the 14 days before (and including) the Completed On date.

For the first call observed between these two Locations, the Date Range field is blank.

Hops

The number of hops found in the network path.

ID

The identifier for a call. UC Monitor automatically assigns the ID.

Max Delay

The greatest amount of path delay of all path samples for this unique path.

Min Delay

The smallest amount of path delay of all path samples for this unique path.

Name

The DNS host name of the destination endpoint.

Path

The sequential number that is assigned to this network path.

Paths

The number of unique paths that were detected between these endpoints. Includes paths reported in the Baseline Call Path chart and table.

Usage

The percentage of time that this path was taken during the baseline date range. Varies based on the number of different paths found.

Traceroute Investigations

To help you collect extra information about the sources of poor call performance, UC Monitor offers traceroute investigations.

Traceroute investigations are available for Cisco devices, and also for Avaya with some limitations. Avaya IP phones support traceroute investigations that use only ICMP. However, Avaya voice gateways and Communication Managers support traceroute investigations that use either TCP or ICMP. The baseline route shown in the report for an Avaya call path is calculated differently from the baseline for a Cisco call path.

An administrator can configure a traceroute investigation to run automatically in response to violations of call setup thresholds and call server group thresholds. An operator with the necessary user account permissions can launch an investigation on demand. To instruct UC Monitor to launch a traceroute automatically, the administrator enables a traceroute investigation as an incident response action.

The results of a traceroute investigation are displayed in a Traceroute Investigation Details (see page 29) report.

If broader access to this feature is desired, the administrator must grant the role for each user account permission to launch an investigation. This ability is enabled by default for user accounts with the IT Manager role. User accounts with other roles lack the necessary area access.

More information:

Incident Investigations (see page 43)

Manually Launch Traceroute Investigation

Take the following steps to launch a traceroute investigation.

Follow these steps:

1. Click Troubleshooting, Launch Investigation in the navigation bar.

   The Launch Traceroute page opens.

2. Complete the following fields:

   - **Target Type**. Use this field to select the type of target you want to investigate. Select Location, Media Device, or Device. Your selection determines the options that are available in the Target menu and in the Investigation Options section.
     Tips:
     - If the target is an IP address or DNS host name, select Device.
     - If the target is a voice gateway, select Media Device.
   - **Target**. Use this field to identify the target device you want to investigate. This device is the destination of the traceroute. The available targets depend on the Target Type you select.
     - If the Target Type is Location, the target Location must have a Cisco key phone defined.
     - If the Target Type is Device, enter the device IP address or DNS host name.
     - If the Target is an Avaya IP phone, select ICMP in the Protocol field.
   - **Investigate From**. This field identifies the collector that launches the traceroute. The management console selects the appropriate collector for the Target unless you selected Device as the Target Type.
   - **Protocol**. This field identifies the protocol to use for the traceroute.

     If you select Location or Media Device as the Target Type, the collector uses TCP over port 80.

     If you select Device as the Target Type, select the protocol for the call server or gateway device:
     - ICMP. ICMP is the only supported protocol for Avaya IP phone targets.
     - TCP over port 80
     - TCP for SIP (port 5060)
     - TCP for MGCP (port 2428)
     - TCP for H.323 (port 1720)

     For all traceroutes, including baseline traceroutes, the DiffServ ToS bits in the header are set for CS3 handling, the standard for call setup traffic.

- **Retries**. Specify the number of probes, from 0 to 10, to send to each component in the route when a target endpoint does not respond. The default is 2 retries.
- **Route Searches**. Specify the number of times, from 0 to 20, to attempt to find other routes for the selected target. Multiple traceroutes are attempted to see whether alternate routes are used between the collector and the target endpoint. The default is 0.

  All attempted unique routes are included in the Investigation Report.
- **Timeout**. Specify the number of seconds, from 1 to 10, that must elapse before UC Monitor considers the traceroute timed out because the target is unreachable. The default is 2 seconds.

3. Click Launch.

   Results are displayed when they are available.

More information:

Incident Investigations (see page 43)

Baseline Traceroutes

UC Monitor calculates baseline routes by running regular traceroutes to selected Cisco IP endpoints. The route information is included for route comparison in the Traceroute Investigation Details (see page 29) report for Locations where you defined a key phone. The key phone serves as the target device for the traceroutes.

Routine traceroutes to find a baseline are a Cisco-only feature. Routine traceroute monitoring is performed regularly on all Cisco voice gateways, including gateways that UC Monitor discovers and gateways that you add. An administrator can disable the option to “Perform routine traceroutes for baseline” when adding or editing a voice gateway definition.

Tip: Endpoints in Microsoft and Avaya deployments do not support routine traceroutes to derive a baseline route. To find out more about a particular Avaya call path, perform a Traceroute Investigation with the phone, gateway, or Communication Manager as the target.

Routine traceroute tests are launched every four hours, starting at 2:00 a.m. The time zone of the collector determines the schedule, but the times that appear in reports reflect the time zone of the management console.

In general, only one collector runs a routine traceroute to a target, but under certain circumstances, multiple collectors run traceroutes to the same target. In such a case, baseline path results are less accurate. Different collectors get different results, and baselines are calculated based on the frequency of path results.

Traceroute Rules

Baseline traceroute information is collected from your network unless one or more of the following scenarios is true:

- Key phones are not defined for your Locations. To enable traceroutes to Locations, an administrator must define key phones when adding or editing Location definitions.
- Key phones have not sent or received call traffic. The investigation fails when you launch a traceroute investigation to a Location whose key phone did not send or receive call traffic. An error message indicates that traceroutes are initiated only when collectors are found for a Location. The collector must observe the key phone in collected traffic before it can run a traceroute to it. As a work-around, select Device as the Target Type and enter the IP address of the key phone.
- *Routine* traceroutes are disabled for voice gateways. The “Perform routine traceroutes for baseline” parameter is enabled by default for all Cisco voice gateways. An administrator can disable it.

Phones

The Phones table provides preliminary details about each endpoint that is known to the call servers in your system. You can use this information to troubleshoot a hardware issue or a Call Server incident.

The Phones table provides the following information:

Call Server

The call server to which the endpoint is registered.

ID

UC Monitor assigns an ID to each endpoint that is discovered during monitoring.

IP Address

The IP address of the endpoint.

(*Cisco only*) Click the IP address to view logs, device information, network setup information, network statistics, and streaming statistics.

Last Activity

The date and time of the last call activity from the endpoint.

Location

The Location definition for the subnet of the endpoint. Matches the Location definition for the last active call.

Name

The name, MAC address, or DNS host name of the endpoint.

Phone Number

The directory number or SIP URI of the endpoint.

Status

(*Cisco only*) Status of the endpoint.

Phone Details and Phone Call Details

You can drill down from the Phones report to review the Phone Details and Phone Call Details views for a selected endpoint. Both view contain details from the last 24 hours of data in the UC Monitor database.

Phone Details

- **Call Server**. The call server to which the endpoint is registered.
- **Firmware Version**. The version of the firmware that is running on the endpoint.
- **ID**. UC Monitor assigns an ID to each endpoint that is discovered during monitoring.
- **IP Address**. The IP address of the endpoint.

  (*Cisco only*) Click the IP address to view logs, device information, network setup information, network statistics, and streaming statistics.
- **Last Activity**. The date and time of the last call activity from the endpoint.
- **Location**. The Location definition for the subnet of the endpoint.
- **Model/Type**. The endpoint hardware model and type.
- **Name**. The name, MAC address, or DNS host name of the endpoint.
- **Phone Number**. The directory number or SIP URI of the endpoint.
- **Previous Call Server**. (*Cisco only*) The call server where the endpoint was previously registered.
- **Previous Status**. (*Cisco only*) Status of the endpoint.
- **Protocol**. The protocol that the endpoint uses for setting up calls.
- **Serial Number**. (*Cisco only*) The serial number that the endpoint manufacturer assigns.

- **Status**. (*Cisco only*) Status of the endpoint.
- **Status Time**. (*Cisco only*) The date and time that the status was updated in the UC Monitor database.
- **Switch Address**, **Name**, **Port**. (*Cisco only*)
  - The IP address and host name of the switch that the endpoint uses to send data to its call server.
  - The switch port number through which traffic passes.

Phone Call Details

You can select the types of calls to view and the call direction. The default direction is To/From Phone. You can filter the Phone Call Details table so that only one direction is included in the report. Although all fields are described in the following list, some information is excluded from the report so that the table can be viewed without scrolling. You can change these settings with the Display On Page options in the Settings dialog.

- **Abandoned Calls**. A call attempt in which an endpoint initiated a call that was disconnected before the call was answered.
- **Call Setup Failure Code**. The code that the call server returns when a call fails during the setup phase. The code indicates the type of failure.
- **Call ID**. The identifier for a call. UC Monitor automatically assigns the ID. The ID is useful for identifying the call in other reports. The ID number is a link to the Call Details (see page 51) report for the call.
- **Destination IP Address/Port**. The IP address and port number that handled the call at the destination endpoint.
- **Destination MOS**. The MOS of the call leg traveling toward the destination endpoint.
- **Destination Number**. The directory number or SIP URI of the destination endpoint. Call quality statistics are for the data that the endpoint received.
- **Destination Location/Media Device**. The Location of the destination endpoint, or the gateway device that forwarded the call to the PSTN.
- **Duration**. The length of the call.
- **Media Type**. The type of data in the call, either audio or video, or both.
- **Origination IP Address/Port**. The IP address and port number that handled the call at the origination endpoint.
- **Origination Location/Media Device**. The Location of the origination endpoint, or the gateway device that forwarded the call from the PSTN.
- **Origination MOS**. The MOS of the call leg traveling toward the origination endpoint.

- **Origination Number**. The directory number or SIP URI of the origination endpoint.
- **Origination Time**. The date and time the call began.
- **Video Packet Loss**. The percentage of video packets that were lost in transit. These packets were sent but never received at the destination.

More information:

Call Performance Overview (see page 16)

Midstream Devices

The Midstream Devices page provides an overview of the performance of RTP traffic flows (or stream legs) on the medianet-enabled devices on your network. A *stream leg* is a unidirectional stream of packets.

ID

A link to the Calls Overview (see page 49), where you can drill down to more information.

IP Address

The IP address of the medianet-enabled device that is associated with the stream leg.

Last Activity

The date and time of the most recent activity for the medianet-enabled device.

Max Jitter

Packet delay that distorts the quality of a voice conversation.

Max Packet Loss

The percentage of data packets that were lost in transit. These packets were sent but never received at the destination.

Name

The name of the medianet-enabled device that is associated with the stream leg. Can be the DNS name or the IP address. If UC Monitor is a registered data source for CA Performance Center, the name is a link to the CA Performance Center context page.

Packets Dropped

The number of data packets that reached the destination, but then were discarded.

Stream Count

The number of active and completed stream legs that are used to calculate the metrics on the Midstream Devices page. The number is a link to the Midstream Legs (see page 77) page.

Video Max Jitter

The variation in delay among video packets in the same stream.

Video Max Packet Loss

The percentage of video packets that were lost in transit. These packets were sent but never received at the destination.

Video Packets Dropped

The number of video packets that reached the destination, but then were discarded.

Midstream Legs

The Midstream Legs page identifies the RTP traffic flows (or stream legs) on the medianet-enabled devices on your network. A *stream leg* is a unidirectional stream of packets. An active stream leg is not yet associated with the end-of-call record for a device.

You can filter the information in the Midstream Leg List with the Media Type and Status lists.

The Midstream Leg List provides the following information:

Average MOS

The Mean Opinion Score (MOS) is an industry standard method for gauging call quality. MOS is an estimation of how impairments to a voice signal affect listener perception of call quality.

MOS calculated from medianet data considers only the codec and any packet loss metrics. Other impairment metrics traditionally associated with a MOS, such as latency and jitter buffer loss, are unknown in medianet streams. Therefore, the MOS for medianet streams may be higher (better) than MOS reported from the endpoints, where all impairment metrics are known.

Codec

The codec in use for the stream leg, or one of the following descriptions:

- "n/a" indicates that a codec exists, but UC Monitor does not recognize it. Or that the router did not send codec information.
- "Dynamic Payload" indicates a video stream leg.
- A dash (-) or "unavailable" indicates that no codec existed for the stream leg.

This field is disabled by default. Click the Settings link to include the field in the Midstream Leg List.

Destination IP Address

The IP address of the device that received the stream leg.

Destination Location

The Location of the device that received the stream leg.

Destination Port

The port number through which the device that received the stream leg. This field is disabled by default. Click the Settings link to include the field in the Midstream Leg List.

DSCP

The Differentiated Services Code Point setting of the incoming RTP packets. Two or more values in this field indicate that the DSCP changed over the course of the stream.

Duration

The length of the stream leg.

ID

A link to the Midstream Leg Information (see page 79) page, and to charts representing MOS, packet loss, average jitter, and maximum jitter.

Last Activity

The last time data was processed for the stream leg. The end time appears on the Midstream Leg Information (see page 79) page.

Media Type

The type of media in the stream leg: Audio or Video.

Packet Loss

The percentage of data packets that were lost in transit. These packets were sent but never received at the destination.

Source IP Address

The IP address of the device that sent the stream leg.

Source Location

The Location of the device that sent the stream leg.

Source Port

The port through which the device sent the stream leg. This field is disabled by default. Click the Settings link to include the field in the Midstream Leg List.

SSRC

The contents of the SSRC field in the RTP header. Also a unique identifier of the source of the stream leg. This field is disabled by default. Click the Settings link to include the field in the Midstream Leg List.

More information:

Mean Opinion Scores (see page 102)

Midstream Leg Information

The tables on the Midstream Leg Information page provide some or all of the following information. The charts below the tables present MOS, packet loss, average jitter, and maximum jitter for the stream leg.

Average MOS, Minimum MOS

The Mean Opinion Score (MOS) is an industry standard method for gauging call quality. MOS is an estimation of how impairments to a voice signal affect listener perception of call quality. Either the average or minimum value for the midstream leg.

MOS calculated from medianet data considers only the codec and any packet loss metrics. Other impairment metrics traditionally associated with a MOS, such as latency and jitter buffer loss, are unknown in medianet streams. Therefore, the MOS for medianet streams may be higher (better) than MOS reported from the endpoints, where all impairment metrics are known.

Call ID

The ID number of the call to which the midstream leg becomes associated. The ID is a link to the call details on the Calls Overview (see page 49) page.

The call ID is "Unavailable" when the midstream leg is not associated with a call. For example, the caller dials a number, which generates stream legs. The callee does not answer, so the call is incomplete. Midstream legs are not associated with incomplete calls.

Codec

The codec in use for the stream leg, or one of the following descriptions:

- "n/a" indicates that a codec exists, but UC Monitor does not recognize it. Or that the router did not send codec information.
- "Dynamic Payload" indicates a video stream leg.
- A dash (-) or "unavailable" indicates that no codec existed for the stream leg.

Duration

The length of the stream leg.

Egress Interface

The interface where traffic exits a device.

End

The date and time that the stream entered the destination port.

Forwarding Status

When applicable, this field explains why a medianet-enabled device did not forward packets as expected. For example, when the device drops packets, this field provides a Status of “Dropped” and a Reason such as “Bad TTL.”

The device manufacturer provides the Status and Reason descriptions. Two other descriptions provided in this field, Value and Extended, are not defined, and can vary depending on the Status and Reason.

When a forwarding status is available, this field contains a blue “i” icon. Position your mouse pointer over the “i” to review the information.

ID

The ID number of the stream leg. Matches the ID number from the Midstream Legs page.

Ingress Interface

The interface where traffic enters a device.

IP Address

The IP addresses of the devices that sent and received the stream leg, and a link to more information.

Jitter, Max Jitter

Packet delay that distorts the quality of a voice conversation. Either the average or the maximum value for the midstream leg.

Location

The Locations of the devices that sent and received the stream leg. Can be <Unassigned> or another defined Location.

Media Type

The type of media in the stream leg: Audio or Video.

Packet Loss

The percentage of data packets that were lost in transit. These packets were sent but never received at the destination.

Packet Rate

The number of data packets that are received per second. UC Monitor uses this value to determine whether an RTP stream is audio or video for medianet-enabled devices that do not report a codec.

Packets Dropped

The number of data packets that reached the destination, but then were discarded.

Packets Lost

The number of data packets that were lost in transit. These packets were sent but never received at the destination.

Packets Received

The number of data packets in a stream.

Port

The port number through which the devices sent and received the stream leg.

SSRC

The contents of the SSRC field in the RTP header. Also a unique identifier of the source of the stream leg.

Start

The date and time the stream left the source port.

TTL

(Time to Live) A counter embedded in data to prevent a data packet from circulating through the network indefinitely. The counter decrements each time that the packet passes through a router or a switch.

Chapter 3: Reports for Capacity Planning

Network engineers must plan for network growth and track usage statistics for their unified communications systems. UC Monitor can help engineers plan for capacity needs by determining the current operating levels of key unified communications components.

Capacity Planning reports focus on quality, volume, and usage:

- Quality reports provide a view of the effects of call volumes on call performance.
- Volume reports provide information about the usage rates and performance of call servers and gateway media devices and help you track call setup failure rates. The Top Volume report lets you locate the areas of the network with the highest call volume.
- Utilization reports are especially useful for understanding telephony resource usage and negotiating contracts with service providers. These reports include per-gateway, per-interface, or per-trunk group statistics.

This section contains the following topics:

Units of Observation

The observation counts shown in all UC Monitor reports gauge the reliability of the statistics and also measure system activity during a particular period. Stated simply, more observations equal greater system activity and more valid results.

The units of observation for Call Performance reports do not generally apply to Capacity Planning reports, which measure volume and usage metrics.

The following observation units are used in Capacity Planning reports:

Call Minutes

The number of minutes when calls ran between a pair of Locations or groups, or between a Location and another Location, in both directions.

Calls

The number of calls that ran between a pair of Locations or groups, or between a Location and another Location, in both directions.

Calls Active

Calls that were *active* during the selected time frame, but not necessarily *initiated* during that time frame.

Calls Established

Calls that were successfully initiated during the selected time frame.

- Calls from endpoints inside of the monitored system.
- Calls from outside of the system to endpoints inside of the system.

Total Calls

The number of calls in the system during the selected time frame.

Total Minutes

All minutes when calls of any type ran anywhere on the network.

Audio Call Quality

The Audio Call Quality page provides quality ratings that help you track audio-only call quality, systemwide. MOS values are graphed over an hourly scale showing the percentage of all VoIP calls on the network that were rated Normal, Degraded, or Excessive. On the chart, a secondary axis indicates the number of call minutes that contributed to the ratings.

By default, VoIP call activity to and from all groups and all Locations is graphed, using colors to show quality ratings.

You can use the MOS Type field to filter the page by MOS or Network MOS.

Call minutes are graphed with a line graph to show the number of observations for the Call Quality bar chart.

The Audio Call Quality page provides the following information. Several of the following fields appear on the Call Quality Details table, which is not displayed by default. Use the Settings link to display the table.

Call Minutes

The number of minutes that calls were active during the selected time period.

Degraded (%)

The percentage of call minutes that have a MOS in the Degraded range of the assigned threshold.

Degraded Minutes

The number of call minutes that have a MOS in the Degraded range of the assigned threshold.

Excessive (%)

The percentage of call minutes that have a MOS in the Excessive range of the assigned threshold.

Excessive Minutes

The number of call minutes that have a MOS in the Excessive range of the assigned threshold.

Normal (%)

The percentage of call minutes that have a MOS in the Normal range of the assigned threshold.

Normal Minutes

The number of call minutes that have a MOS in the Normal range of the assigned threshold.

Time

The hour of the day when calls were active on the network.

Unrated (%)

The percentage of call minutes that did not have a threshold applied. The performance threshold for MOS was disabled for these Locations.

Unrated Minutes

The number of call minutes that did not have a threshold applied. The performance threshold for MOS was disabled for these Locations.

Call Volume Audio

The Call Volume Audio page highlights the volume of audio-only calls and identifies call setup failures. This page provides information that is useful for tracking call volumes for the following items:

- Locations or groups of Locations
- Media devices
- Call servers

This report groups call setup failures by cause. You can use this report to determine busy-hour call attempts (BHCA) and busy-hour call completions (BHCC) for audio-only calls on your network. The complementary Call Volume Total (see page 87) report shows volume levels for all calls, including audio-only and video calls.

The Call Volume data view provides the following information:

Calls completed

The number of audio-only calls that were successfully completed during the selected time frame. Includes calls from endpoints within the monitored system and calls from *outside of* the system to endpoints *inside of* the system.

Calls attempted

All calls that the monitored system tried to place, either successfully or unsuccessfully. This metric is the primary unit of measurement for the Call Volume Audio (see page 86) views.

Setup failures

Calls that failed during the setup phase.

Grade of Service

(*Cisco only*) An estimation of the probability that a VoIP call receives a busy signal. The GoS value (a decimal fraction) is always expressed with reference to the busy hour when the traffic intensity is the greatest. GoS is reported from the perspective of the origination Location or gateway device (the outgoing direction).

Call failures that occurred during the reporting period are displayed in a pie chart in the Call Setup Failure Breakdown view. The Call Setup Failure Breakdown view shows a distribution of the most commonly observed call setup error codes.

Notes:

- In an Avaya-only environment, GoS and call setup failures do not appear in the Call Volume and the Call Volume Details views.
- In an environment that includes Avaya and Cisco or Microsoft equipment, the GoS and call setup failure values are "n/a" in the Call Volume and Call Volume Details views.
- Click the Settings link to hide or display the following views: Grade of Service, Call Setup Failure Breakdown, Call Volume Details.

Call Volume Total

The Call Volume Total page shows traffic volume for all calls, including audio-only (VoIP) and video calls. You can use the following filters to determine the calls that are included on the page.

Include volume

Select the type of calls to include: Audio and Video (calls of all types), Audio, Video

Show calls

Select the state of the calls to include. Your selection determines the metric that is graphed on the left Y-axis.

- Established: Initiated during the selected monitoring interval, or
- Active: Ongoing during the selected monitoring interval, but possibly initiated at another time

The Call Volume Total page provides the following information. Several of the following fields appear on the Total Volume Details table, which is not displayed by default. Use the Settings link to display the table.

Audio Calls

The number of audio-only calls in the system during the indicated period.

Audio Minutes

All minutes during the indicated period when audio-only calls were running.

Calls Active

The number of calls of all types that were active during the selected time frame, but not necessarily initiated during that period. Shows the time frame when calls were active. The bar graph indicates the number of calls along the left Y-axis, and the number of minutes when calls were active along the right Y-axis.

Calls Established

The number of calls of all types that were successfully initiated during the selected time frame. Includes calls from endpoints inside the monitored system and calls from outside of the system to endpoints inside of the system.

The bar graph indicates the number of calls along the left Y-axis, and the number of minutes when calls were active along the right Y-axis.

Time

The time of day when calls were placed. Usually one hour per table row.

Total Calls

The number of calls in the system during the indicated period.

Total Minutes

All minutes when calls of any type ran on the network. The Total Minutes line graph can appear with no corresponding bar graph. In such a situation, the line graph identifies calls that were established during a previous reporting interval but still active during the graphed interval.

Video Calls

The number of calls that contain audio and video streams during the indicated period.

Video Minutes

All minutes during the indicated period when calls containing both audio and video streams were running.

Top Volume

The Top Volume page compares the call volume of the groups, Locations, or endpoints (see definition on page 112) with the highest usage during the selected time frame. Think of it as a list of the “top talkers” at a particular point in time. By default, all calls are considered when compiling the list of top call volumes.

The Volume bar chart shows a comparison of call volumes among the groups, Locations, or endpoints with the highest volumes. Each bar represents a relative activity level so that you can easily compare call volumes among busy Locations.

You can filter the Locations, groups, and endpoints on the page.

Media Type

Select the type of calls to include in the Top Volume data view. By default, calls of all types are displayed.

- Audio and Video (calls of all types)
- Audio
- Video

Calculate using

Select the observation units to use when selecting Locations or groups to include in the list of top talkers in the data view. Your selection determines the metric that is graphed on the left Y-axis.

- Calls: Initiated during the selected monitoring interval
- Call Minutes: Ongoing during the selected monitoring interval, but possibly initiated at another time.

The values in the Calls columns are links to the Calls Overview (see page 49). The Calls Overview shows only calls from the same time frame as the current Top Volume report.

In the Top Phones Volume table, the Phone Number column provides links to the Phones (see page 73) table.

If no directory number is available for an endpoint, its IP address is provided in the IP Address column. Where available, the IP address is a link to the web page of the endpoint. The endpoint name is also shown in the Name column.

Top Trunk Groups

The Top Trunk Groups table helps you understand trunk group usage and provides individual trunk usage statistics. Use this page to find underutilized trunk groups and overburdened trunks, where performance can deteriorate.

Statistics for all known trunk groups are included. The most heavily used trunks are shown first.

Call Minutes

The number of call minutes used to calculate usage statistics. This value provides a sense of the scope of activity and helps determine the significance of the data as it relates to sample size.

Call Minutes Capacity

The number of call minutes that were supported during the selected time period. The capacity is discovered during monitoring or is manually supplied during configuration.

Grade of Service

(*Cisco only*) An estimation of the probability that a VoIP call receives a busy signal. The GoS value (a decimal fraction) is always expressed with reference to the busy hour when the traffic intensity is the greatest. GoS is reported from the perspective of the origination Location or gateway device (the outgoing direction).

Maximum Utilization

The highest recorded usage during the selected time period, expressed as a percentage of capacity. Maximum usage is a good indicator of potential capacity issues. Maximum usage is significant because it often indicates the usage during the "busy hour" of the day. Maximum usage is computed as capacity multiplied by the time frame.

- (*Avaya only*) The capacity is based on information from the Avaya Communication Manager.
- (*Cisco only*) The capacity is discovered when voice gateway devices are discovered.

Name

The name of the trunk group. Click to drill down into an hourly breakdown of usage for this trunk group. The trunk group name is based on information from the monitored call traffic and from the naming convention of the trunking equipment.

Note: Administrators typically use the default names when using the Avaya Site Administration (ASA) interface to configure the system. Using the defaults can lead to redundant trunk group names, which then appear identical in UC Monitor reports. We recommend using the ASA interface to change the names and make each trunk group easily distinguishable in UC Monitor reports. No data is lost when you use the ASA to change trunk group names.

Utilization (%)

The average usage during the time period. Expressed as a percentage of capacity. Average usage is computed as capacity divided by the time frame.

- (*Avaya only)* The trunk group capacity is discovered from SNMP polling of the Avaya Communication Manager. The computation for Avaya trunk group usage is channel capacity divided by the CCS (Centum Call Seconds) for all active channels. The result is then divided by the selected time frame.
- (*Cisco only*) Usage is the percentage of total voice interface capacity in use during the selected time frame. Total interface capacity is derived from the known capacity of each voice interface in the group.

Tip: To identify underused trunk groups, sort this column to see the least used trunk group first.

Notes:

- An administrator can organize gateway voice interfaces into groups and then copy those groups (as subgroups) into container groups. Therefore, an interface can appear twice in the Top Trunk Groups table, once in its own group, and once in its container.
- (*Avaya only*) The collector retrieves trunk group statistics from an Avaya Communication Manager that is configured to allow SNMP polling. The Top Trunk Groups table is not available when the collector cannot retrieve information from Communication Manager.
- (*Cisco only*) Trunk groups are created in CA Performance Center. They are not automatically discovered. The capacity of each Cisco trunk group is based on voice interface capacity. These values are discovered when voice gateways are discovered. However, capacity values are not updated in the gateway MIB when the interface capacity changes. You can verify voice interface capacity on the Voice Gateway Properties page.

Trunk Group Utilization

The Trunk Group Utilization page provides information about usage levels on voice trunk groups. This information helps you determine the usage for voice and media gateways by revealing per-trunk group capacity and volume statistics. This report is useful for assessing trunk group usage and capacity.

This page is available when at least one trunk group is monitored.

- (*Avaya only*) Trunk groups determine capacity and usage for Avaya voice gateways and channels. Configure the Avaya Communication Manager to allow SNMP polling reports such metrics as queue overflows and the All Channels Busy percentage.
- (*Cisco only*) Trunk groups are not discovered from the call servers or from network traffic, but are created as groups of voice interfaces in CA Performance Center. The metrics that are available for Cisco trunk groups are similar to metrics provided in the Voice Interface reports. For Cisco, the device MIB does not always report voice interface capacity accurately. Verify interface configuration in the Administration, Data Collection, Media Devices, Voice Gateways page, and change it when necessary.

The Trunk Group Utilization page contains the following views and data points. Some metrics are vendor-specific. Some metrics appear in the Trunk Group Utilization Details view, which is not displayed by default. Click the Settings link to display the view.

% Utilization

The usage for all known trunk groups. Shown as a percentage of capacity over a day, a week, or a month.

- (*Avaya only*) The usage for a trunk group is computed as channel capacity divided by the CCS (Centum Call Seconds) observed for all active channels. The result is then divided by the selected time frame.
- (*Cisco only*) Usage is the percentage of total channel capacity in use during the selected time frame. Total channel capacity is derived from the known capacity of each voice interface in the group.

All Channels Busy (%)

(*Avaya only*) The percentage of the reporting interval for which all active channels in a trunk group carried traffic.

Call Minutes

The number of call minutes used to calculate usage statistics. This value provides a sense of the scope of activity and helps determine the significance of the data as it relates to sample size.

Call Minutes Capacity

The number of call minutes that were supported during the selected time period. The capacity is discovered during monitoring or is manually supplied during configuration. For Avaya, statistics for inactive channels are subtracted from the total capacity.

Channels Out-of-Service

(*Avaya only*) The number of channels in a trunk group that are out of service.

Queue Abandons

(*Avaya only*) Calls that were removed from the Trunk Group Queue for one of the following reasons:

- The system automatically removes calls that have been in the queue for the maximum allowed time, typically 30 minutes.
- Calls are forced out of the queue when users cancel the auto-callback setting to move the call to the queue when all channels are busy.

Group Overflows

(Avaya only) The number of outgoing calls that were presented to the trunk but not carried. These *overflow* calls arrived when all trunks in the trunk group were busy, and, therefore, were not queued on the trunk group. This value does not include calls that were denied service on the trunk group because of authorization failures.

Queue Overflows

(*Avaya only*) The number of calls that arrived when all slots in the Trunk Group Queue were busy.

Time

The hour from which the detailed data was collected.

Utilization (%)

The usage during the time period. Expressed as a percentage of available capacity.

Top Voice Interfaces

The Top Interfaces report helps you understand voice gateway usage and provides individual gateway voice interface statistics. Use this report to find underused interfaces and look for overburdened interfaces, where performance can deteriorate.

Note: The Top Interfaces report supports Cisco environments only.

Verify that all known gateway voice interfaces have the number of channels correctly configured. UC Monitor uses the known channel capacity for each interface to calculate usage as a percentage of capacity. This report is less accurate when the device MIB incorrectly reports the capacity of the gateway voice channel.

The Voice Interface Top Interfaces report contains the following information. Some of the metrics appear in the Interface Utilization view, which is not displayed by default. Click the Settings link to display the view.

Call Minutes

The number of call minutes used to calculate usage statistics. This value provides a sense of the scope of activity and helps determine the significance of the data as it relates to sample size.

Call Minutes Capacity

The number of call minutes that were supported during the selected time period. The capacity is discovered during monitoring or is manually supplied during configuration.

Grade of Service

An estimation of the probability that a VoIP call receives a busy signal. The GoS value (a decimal fraction) is always expressed with reference to the busy hour when the traffic intensity is the greatest. GoS is reported from the perspective of the origination Location or gateway device (the outgoing direction).

Maximum Utilization

The highest recorded usage during the selected time period, expressed as a percentage of capacity. Maximum usage is a good indicator of potential capacity issues. Maximum usage is significant because it often indicates the usage during the “busy hour” of the day. Maximum usage is computed as capacity multiplied by the time frame.

Name

The name of the interface. Click to drill down into an hourly breakdown of usage for this interface. By default, the interface name is based on information from the gateway and the naming convention for the trunking equipment. However, a UC Monitor administrator can change the name of a discovered gateway voice interface during voice gateway configuration.

Top Interfaces

The usage of all gateway voice interfaces on all monitored gateway devices. All known interfaces are listed. Interfaces with the highest usage are listed first.

Utilization

The average usage during the time period. Expressed as a percentage of capacity. Average usage is computed as capacity divided by the time frame.

To see whether any interfaces are underused, sort this column to see the least used interfaces first.

Voice Interface Utilization

The Voice Interface Utilization report provides information about usage levels on voice gateway-type media devices. Information in this report is useful for assessing:

- Gateway device usage and capacity
- Voice gateway interface usage

Note: The Voice Interface Utilization report supports Cisco environments only.

Verify that all known gateway voice interfaces have the number of channels correctly configured. UC Monitor uses the known channel capacity for each interface to calculate usage as a percentage of capacity. This report is less accurate when the device MIB incorrectly reports the capacity of the gateway voice channel.

Interfaces that are deleted from the system do not appear in the report. However, some deleted interfaces have data. You can use the Settings link to include deleted interfaces in the report.

The Voice Interface Utilization page provides the following information.

Interface Utilization

% Utilization

The usage of all gateway voice interfaces on all voice gateway devices. Shown as a percentage of capacity over a day, a week, or a month.

Usage is computed by multiplying the time frame by the number of channels in a gateway or interface. For one hour, a single channel can provide 60 call minutes. Thus, a full PRI interface (23 channels) can support the following usage:

`23 channels x 60 minutes/hour = 1380 call minutes/hour = 33120 call minutes/day`

Call Minutes

The number of call minutes used to calculate usage statistics. This value provides a sense of the scope of activity and helps determine the significance of the data as it relates to sample size.

Interface Utilization Details

The Interface Utilization Details table is not displayed by default. You can use the Settings link to display the table.

Time

The hour from which the detailed data was collected.

Call Minutes

The number of minutes that calls were active during the selected time period. By default, applies to all call minutes logged on all gateways in the system.

Total Capacity

The number of call minutes that were supported during the selected time period. The capacity is discovered during monitoring or is manually supplied during configuration.

Utilization (%)

The usage during the time period. Expressed as a percentage of available capacity.

Appendix A: Report Metrics

This section contains the following topics:

Call Quality Metrics

The following metrics are available from UC Monitor reports.

ACOM

(*Cisco PSTN calls only*) The total echo return loss on the network. ACOM measures how significantly the voice gateway reduced the echo. ACOM includes echo reduction that occurs with or without the activity of an echo cancellation device.

Burst

(*Microsoft only*) The points in a data stream when a high percentage of packets is lost or discarded due to packets arriving late.

- **Burst Density**. The percentage of packets within burst periods that are lost or discarded.
- **Burst Duration**. The average duration of all high-loss periods in a data stream.

Concealment Ratio

A technique for masking the effects of packet loss in VoIP communications. Also known as packet loss concealment (PLC).

- **Cumulative Concealment Ratio**: The percentage of all call seconds that had concealment events due to lost data.
- **Maximum Concealment Ratio**: The highest concealment ratio value during the call.
- **Severely Concealed Seconds**: The number of call seconds that had more than 5 percent concealment events from the start of the audio stream.

Echo Tail Length

(*Avaya only*) The “length” of echo cancellation processing. Based on the distance between a voice gateway and the endpoint. Typical values range from 8 milliseconds to 32 milliseconds.

Gap

(*Microsoft only*)

- **Gap Density**. The percentage of lost or discarded packets in the gaps between bursts in a data stream.
- **Gap Duration**. The average duration of periods of good performance (low loss) between periods of data loss in a data stream. Occasionally, gap duration exceeds the call duration.

Jitter

Packet delay that distorts the quality of a voice conversation.In charts, maximum jitter values are graphed as data points and indicated on the right Y-axis.

The definition of Maximum Jitter varies by monitored environment:

- Avaya environment: The maximum jitter per interval
- Cisco environment: The maximum jitter thus far

Jitter Buffer

- **Jitter Buffer Delay**. (*Avaya only*) Delay that the jitter buffer introduces while it holds one or more packets to reduce variations in packet arrival times. Acceptable jitter buffer delay is two RTP datagrams or less. Because most codecs have a datagram size of 20 to 30 milliseconds, a good jitter buffer delay is no more than 40 to 60 milliseconds.
- **Jitter Buffer Loss**. (*Cisco only*) The packets that are lost when jitter hinders the caching capacity of the jitter buffer.
- **Jitter Buffer Over Runs**. (*Avaya only*) The number of times that jitter exceeded the maximum size setting of the jitter buffer. Packets arrive too quickly to be contained by the jitter buffer. Over runs usually result in packet loss.
- **Jitter Buffer Under Runs**. (*Avaya only*) The number of times that the jitter buffer became empty. Packets arrive too slowly to be contained by the jitter buffer. Under runs usually indicate that delays are too lengthy for the buffer setting.

Note: Avaya endpoints have a limitation that affects the way the Over Runs and Under Runs metrics are reported. The maximum value is 255. In a report, this cumulative value means that the value was greater than 255, and the actual maximum value cannot be reported. These values, reported every 15 seconds during a Call Watch, are additive. After the maximum is reached, Jitter Buffer Over Runs and Under Runs charts show 0 values for successive intervals until the call is completed.

Latency

One-way delay. Calculated from the origination party to the destination party. Includes propagation delay, network delay, and packetization delay. Latency has a severe effect on VoIP call quality.

Mean Opinion Score (MOS)

The Mean Opinion Score (MOS) is an industry standard method for gauging call quality. MOS is an estimation of how impairments to a voice signal affect listener perception of call quality.

- MOS is the average MOS listening quality (LQK) score observed for the bi-directional voice stream.
- MOS (Min) is the lowest LQK score observed.
- (*Microsoft only*) Conversational MOS is based on MOS values from both directions of data flow.
- Listening MOS is based on call legs traveling toward the endpoint (see definition on page 112) to reflect listener perception of quality.

Network MOS

MOS listening quality value that is based only on network factors, such as codec, packet loss, packet reordering, packet errors, and jitter.

Noise Level

(*Microsoft only*) The average portion of an audio signal that is noise and not actual voice data. Measured in decibels.

Packet Loss

The percentage of data packets that were lost in transit. These packets were sent but never received at the destination.

- **Packet Loss**. The average loss rate.
- **Packet Loss (Max)**. (*Avaya only*) The maximum loss rate.
- **Packets Received**. A means of gauging the size of the data stream.
- **Packets Lost**. The difference between the number of packets that were sent and the number of packets that were received.

Sequence

(*Avaya only*)

- **Sequence Jumps**. The number of times that at least one consecutive packet was lost.
- **Sequence Falls**. The number of times that at least one packet arrived out of order.

Signal Level

(*PSTN calls only*) The average audio signal level in decibels. The dBm0 abbreviation refers to decibels relative to a power level of one milliwatt (dBm) measured at a zero transmission level.

- **Signal in**. The signal level of the data traveling into the echo canceler.
- **Signal out**. The signal level of the data traveling out of the echo canceler toward the IP network.

Time-To-Live

A counter embedded in data to prevent a data packet from circulating through the network indefinitely. The counter decrements each time that the packet passes through a router or a switch. The minimum and maximum TTL values are provided.

Call Setup Metrics

The term *call setup* refers to the connections that occur between a device trying to make a VoIP call and the active call server. The call server signals the device to play a dial tone and initiate the call. The call server also establishes a connection to the destination device. The call setup protocol defines the messages that are passed among the endpoints (see definition on page 112).

Only endpoints that initiate a call can generate call setup metrics, which are shown in Call Performance reports.

Call Setup Failures

The calls that fail to connect during the setup phase. Expressed as a percentage of all calls that were attempted during the monitoring interval.

Call Setup Failure Code

The code that the call server returns when a call fails during the setup phase. The code indicates the type of failure. For more information, see the list of call termination cause codes on the Cisco website.

Delay to Dial Tone

The amount of time it takes for a user to hear a dial tone after picking up the receiver of an IP telephone. During the call setup phase of a VoIP call, the device receives messages from the call server to play a dial tone. Users can think that the system is not working when dial tone is delayed.

Post-Dial Delay

The amount of time from when a user enters the last digit of a telephone number to when the user hears a ring or busy signal.

Video Metrics

Maintaining user Quality of Experience (QoE) is immensely challenging for video applications because it is difficult to measure success in delivering high-quality video. Video applications do not have a widely accepted video quality standard equivalent to the MOS for audio. Video quality is more subjective than audio quality, and it is more complicated to implement.

Note: Video metrics are available only from monitored Microsoft Lync environments or from medianet-enabled devices.

Frozen Period

The average length of frozen video instances.

Frozen Video

The frequency of long and noticeable frozen video periods for an entire session. Expressed as a percentage of session time.

Video Bit Rate

Bit rates provide a gauge of codec performance.

- **Video Bit Rate**. Average number of bits sent per second for an entire stream.
- **Video Bit Rate (Max)**. Maximum number of bits sent per second for and entire stream.

Video Frame Decoding Time

The average amount of time for decoding frames in a stream. A slower decoding rate can be the result of conditions on the endpoint, such as lack of CPU resources, and can affect call quality.

Video Frame

- **Frame Loss**: The average number of unique consecutive images, or video frames, lost due to corruption and error concealment for the entire system. Video frames can span multiple packets. Video frames can span multiple packets, so this metric is useful when compared with the video packet loss metric.
- **Frame Rate**: The average number of frames that were sent or received per second for an entire stream.

Video Jitter

- **Video Jitter**. The variation in delay among video packets in the same stream.
- **Video Jitter (Max)**. The highest observed jitter level for the call.

Video Latency

The maximum time for a video packet to travel between the calling parties. Measured from end-to-end in one direction. Calculated by taking the average round-trip time for a call leg in a given video call and dividing it in half.

Video Packet

The percentage of video packets that were lost in transit. These packets were sent but never received at the destination.

- **Packet Loss**. The average loss rate. The maximum rate is also provided.
- **Packets Received**. A means of gauging the size of the data stream.
- **Packets Lost**. The difference between the number of packets that were sent and the number of packets that were received.
- **Consecutive Packet Loss**. The percentage of all packets that were lost consecutively. This value provides a gauge of loss burstiness.

Mean Opinion Scores

The Mean Opinion Score (MOS) is an industry standard method for gauging call quality. MOS is an estimation of how impairments to a voice signal affect listener perception of call quality. The MOS scale ranges from 5.0 to 1.0.

- 5.00 represents an audio signal of the highest quality, free from impairment.
- 1.00 represents the lowest quality.

The MOS provided in UC Monitor reports is an average that is derived from multiple samples, unless otherwise specified.

The G.107 standard provides a scale that relates MOS values to user satisfaction:

MOS Range	Likely User Opinion of Call Quality
4.3-5.0	Users are very satisfied
4.0-4.3	Users are satisfied
3.6-4.0	Some users are dissatisfied
3.1-3.6	Many users are dissatisfied
2.6-3.1	Nearly all users are dissatisfied
1.0-2.6	Not recommended

The MOS value for a VoIP call leg partly depends on the codec that is used to packetize the audio signal. Different codecs advertise different *theoretical maximum* MOS values, which are the highest possible score they can achieve in the absence of other impediments.

Cisco MOS Calculation

UC Monitor uses information from Cisco IP endpoints (see definition on page 112) to report MOS for the calls that do not leave the monitored network. Each endpoint tells its call server about the MOS of the last call that was made or received. The collector inspects the packets sent to the call server to find the MOS information.

For VoIP calls from endpoints in the PSTN, the collector polls the gateway for MOS information. For these PSTN calls, the collector uses the G.107 standard to calculate MOS from gateway call legs that use MGCP, H.323, or SIP.

Cisco uses a proprietary algorithm that estimates voice listening quality by taking into account the following factors:

- Perceptual weighting factors.
- Quality impairment factors that affected the audio stream, such as the type of codec used.

Avaya MOS Calculation

Avaya endpoints (see definition on page 112) send metrics every few seconds. The collector uses these metrics to calculate and report MOS during an active call. The collector uses an algorithm similar to the Cisco voice gateways, using the G.107 standard. The scores are thus based on MOS listening quality.

Microsoft MOS Calculation

Endpoints that support VoIP and video quality metrics in a Microsoft system provide the following types of MOS:

- **Listening quality MOS** (MOS-LQ): Isolates the listening quality of audio by excluding bidirectional effects, such as delay and echo.
- **Conversational quality MOS** (MOS-CQ). Consists of the listening quality in each direction of the call, taking into account impairments from delay and echo.
- **Network MOS**: MOS listening quality value that is based only on network factors, such as codec, packet loss, packet reordering, packet errors, and jitter. Generally, the Lync instances report these values and other quality metrics at the end of each call. The Network MOS therefore helps you isolate network impediments on audio quality.

The proprietary Microsoft codecs, RTAudio and Siren (used for conference calls), can operate in two bandwidth modes. Therefore, MOS-LQ and Network MOS are reported on a wideband scale.

Appendix B: Call Termination Cause Codes

UC Monitor reports identify the Call Setup Failure Code, which the call server returns when a call fails during the setup phase. The code indicates the type of failure. For more information, see the list of call termination cause codes on the Cisco website.

UC Monitor uses the following call termination cause codes.

Code	Description
0	No error.
1	Unallocated (unassigned) number
2	No route to specified transit network
3	No route to destination
4	Send special information tone
5	Misdialed trunk prefix
6	Channel unacceptable
7	Call awarded and being delivered in an established channel
8	Preemption
9	Preemption -- circuit reserved for reuse
16	Normal call clearing
17	User busy
18	No user responding
19	No answer from user
20	Subscriber absent
21	Call rejected
22	Number changed
26	Non-selected user clearing
27	Destination out of order
28	Invalid number format (address incomplete)
29	Facility rejected
30	Response to STATUS ENQUIRY
31	Normal, unspecified

Code	Description
34	No circuit/channel available
38	Network out of order
39	Permanent frame mode connection out of service
40	Permanent frame mode connection operational
41	Temporary failure
42	Switching equipment congestion
43	Access information discarded
44	Requested circuit/channel not available
46	Precedence call blocked
47	Resource unavailable, unspecified
49	Quality of Service not available
50	Requested facility not subscribed
53	Service operation violated
54	Incoming calls barred
55	Incoming call barred within Closed User Group (CUG)
57	Bearer capability not authorized
58	Bearer capability not presently available
62	Inconsistency in designated outgoing access information and subscriber class
63	Service or option not available, unspecified
65	Bearer capability not implemented
66	Channel type not implemented
69	Requested facility not implemented
70	Only restricted digital information bearer capability is available
79	Service or option not implemented, unspecified
81	Invalid call reference value
82	Identified channel does not exist
83	A suspended call exists, but this call identity does not
84	Call identity in use
85	No call suspended

Code	Description
86	Call having the requested call identity has been cleared
87	User not member of Closed User Group (CUG)
88	Incompatible destination
90	Destination number missing and DC not subscribed
91	Invalid transit network selection
95	Invalid message, unspecified
96	Mandatory information element is missing
97	Message type nonexistent or not implemented
98	Message is not compatible with the call state, or the message type is nonexistent or not implemented
99	An information element or parameter does not exist or is not implemented
100	Invalid information element contents
101	The message is not compatible with the call state
102	Call terminated when time expired; a recover routine executed to recover from the error
103	Parameter nonexistent or not implemented -- passed on
110	Message with unrecognized parameter discarded
111	Protocol error, unspecified
122	Precedence Level Exceeded
123	Device not Preemptable
125	Out of bandwidth (Cisco specific)
126	Call split (Cisco specific)
127	Interworking, unspecified
129	Precedence out of bandwidth
131	Call Control Discovery PSTN Failover (Cisco specific)

Glossary

ACOM
: The total echo return loss on the network. ACOM measures how significantly the voice gateway reduced the echo. ACOM includes echo reduction that occurs with or without the activity of an echo cancellation device.

all channels busy
: The percentage of the reporting interval for which all active channels in a trunk group carried traffic.

analog telephone adapters (ATA)
: A device used to connect a standard telephone to a computer or network so that the user can make calls over the internet. ATAs are typically cheaper than specialized VoIP phones that connect directly to a computer's USB port. An ATA typically supports one or two ports.

Answer Seizure Ratio (ASR)
: The number of successfully answered calls compared to the number of call attempts.

Application Enablement Services (AES)
: The Avaya application server that provides system management APIs.

audio/visual conferencing server
: A server in a Microsoft Lync environment that enables audio and video (multi-party) conference calls. Also referred to as an A/V MCU.

Automatic Number Identification (ANI)
: A feature of telephony that lets subscribers display or capture the telephone numbers of calling parties.

burst
: The points in a data stream when a high percentage of packets is lost or discarded due to packets arriving late.

burst density
: The percentage of packets within burst periods that are lost or discarded.

burst duration
: The average duration of all high-loss periods in a data stream.

busy-hour call attempts (BHCA)
: The number of calls attempted at the busiest (peak) hour of the day.

busy-hour call completions (BHCC)

The number of calls completed at the busiest (peak) hour of the day. BHCC is a measure of the throughput capacity of a VoIP network.

call detail record (CDR)

Storage of information about the endpoints of a call and other aspects of call control and routing.

call leg

A discrete segment of a call connection in a VOIP network. A logical connection between a router and an endpoint.

call management record (CMR)

Storage of information about the quality of the streamed audio of a call.

call minutes

The number of minutes that calls were active during the selected time period.

call path

The path, or route, a call takes between the origination and destination endpoints in a network.

call setup

A series of connections that occur between a telephone placing a VoIP call and the active call server. The call server is responsible for certain signaling to the telephone that allows it to play a dial tone and make the call. The call server also establishes a connection to the destination endpoint in the PSTN. The call setup protocol defines the messages that are passed among the call server, gateway, and endpoints.

call setup failures

The calls that fail to connect during the setup phase. Expressed as a percentage of all calls that were attempted during the monitoring interval.

call setup protocol

Protocols involved in the call setup process: SIP, SCCP, H323, and MGCP.

calls attempted

All calls that the monitored system tried to place, either successfully or unsuccessfully. This metric is the primary unit of measurement for the Call Volume Audio (see page 86) views.

calls completed

The number of audio-only calls that were successfully completed during the selected time frame. Includes calls from endpoints within the monitored system and calls from *outside of* the system to endpoints *inside of* the system.

channels out-of-service

The number of channels in a trunk group that are out of service.

Cisco CallManager cluster

A group of physical servers, running Cisco Unified Communications Manager (CallManager), to work together as an IP PBX system.

Cisco IP Communicator

A Microsoft Windows-based softphone application for making voice and video calls.

Cisco Performance Monitor

A feature of Cisco routers and switches that enables reporting of quality metrics for a medianet environment.

codec

Codecs (the term is short for coder-decoder) convert an audio signal into compressed digital form for transmission and then back into an uncompressed audio signal for replay.

concealment

A technique for masking the effects of packet loss in VoIP communications. Also known as packet loss concealment (PLC).

concealment ratio

The percentage of frames in a data stream that are concealment frames, which the endpoints generate to conceal packet loss. Includes both early and late packets.

conference ID

Identifier for a voice gateway call.

connection attempts

The number of times a connection to the server is attempted before timing out.

controller LAN board (C-LAN)

G650 voice gateways can have C-LANs defined and running on the device as separate call servers. Each C-LAN has a dedicated IP address, which appears in UC Monitor reports as a call server. However, the actual call server is the Communication Manager, which is usually installed on a separate media server.

conversational MOS

The Mean Opinion Score (MOS) based on metric factors from both directions of data flow.

currently missing phones

The percentage of endpoints that were registered to a server in the group, but are no longer registered to any server in the group, and were never observed to have been formally unregistered.

delay

see latency (see page 115)

delay to dial tone

The amount of time it takes for a user to hear a dial tone after picking up the receiver of an IP telephone. During the call setup phase of a VoIP call, the device receives messages from the call server to play a dial tone. Users can think that the system is not working when dial tone is delayed.

Differentiated Services Code Point (DSCP)

The Differentiated Services Code Point setting of the incoming RTP packets.

digital telephone

Digital telephones convert analog sound into digital format at the handset. Digital telephones do not include web browsers or more advanced applications generally available from IP telephones.

directory number (DN)

A telephone number.

echo

The phenomenon of your voice coming back to you, as if you were repeating yourself. In a VoIP network, echo is accentuated by the amount of delay in the network.

Echo Return Loss (ERL)

Reduction in the echo level produced in the circuit without an echo canceler. The degree or amount of loss reflects the volume of the echo that remains, and a measurement of how significantly echo was reduced.

Echo Return Loss Enhancement (ERLE)

An enhancement in the echo return loss that an echo canceler produces. An echo canceler removes the echo portion of a VoIP call signal as it exits the tail circuit and heads into the WAN. Also referred to as *cancellation loss*.

echo tail length

The “length” of echo cancellation processing. Based on the distance between a voice gateway and the endpoint. Typical values range from 8 milliseconds to 32 milliseconds.

edge server

In a Microsoft Lync environment, a server running in the perimeter network to provide connectivity for external users and public instant messaging connections. The edge server ensures that users outside the firewall are authorized before they obtain access to the Lync deployment. The edge server also provides media relay for audio/visual streams where direct connection is not possible.

egress interface

The interface where traffic exits a device.

endpoint

An endpoint is any device where a media stream begins or ends, such as telephone, softphone, telepresence, voice gateway, media device, and video camera.

erlang

In telephony, a statistical measure of the volume of telecommunications traffic. Traffic of one Erlang refers to a single resource being in continuous use, or two channels being at fifty percent use, and so on.

failover

Failover is the process of switching to a backup server or system when the primary server or system fails, is offline, or becomes unavailable.

Flexible NetFlow

The next generation in flow technology from Cisco. Flexible NetFlow enables the delivery of medianet data to UC Monitor.

front-end server

In a Microsoft Lync environment, a server that typically performs call processing functions. Microsoft Lync supports a pool of one or more front-end servers working together to perform functions such as call processing.

frozen period

The average length of frozen video instances.

frozen video frequency

The frequency of long and noticeable frozen video periods for an entire session. Expressed as a percentage of session time.

G.107

An ITU-T standard for reporting VoIP conversational call quality. UC Monitor uses this standard to calculate MOS from the voice gateway's perspective at the end of an IP-PSTN call.

G.711

A high-performance, high bit-rate codec (64 Kbps) often used for its excellent voice quality. Because it does not use compression, G.711 requires more bandwidth than some other common codecs.

gap density

The percentage of lost or discarded packets in the gaps between bursts in a data stream.

gap duration

The average duration of periods of good performance (low loss) between periods of data loss in a data stream.

gatekeeper

An optional component of a VoIP network that provides services such as endpoint registration, address resolution, admission control, and user authentication.

gateway

A device that provides the conversion interface between the PSTN and an IP network.

Grade of Service (GoS)

An estimation of the probability that a VoIP call receives a busy signal. The GoS value (a decimal fraction) is always expressed with reference to the busy hour when the traffic intensity is the greatest. GoS is reported from the perspective of the origination Location or gateway device (the outgoing direction).

group overflows

The number of outgoing calls presented to the trunk but not carried. *Overflow* calls arrived when all trunks in the trunk group were busy, but were not queued on the trunk group. This value does not include calls that were denied service on the trunk group because of authorization failures.

H.323

An ITU standard protocol for call setup. UC Monitor supports gateways that use this protocol to communicate with Cisco Unified Communications Manager.

ingress interface

The interface where traffic enters a device.

Interactive Connectivity Establishment (ICE)

A mechanism for SIP-based VoIP clients to successfully traverse the variety of firewalls that may exist between a remote user and a network.

jitter

Packet delay that distorts the quality of a voice conversation.

jitter buffer

Buffers that attempt to reduce or eliminate network jitter by caching packets. If jitter exceeds caching capacity, packets are lost (jitter buffer loss).

jitter buffer delay

Delay that the jitter buffer introduces while it holds one or more packets to reduce variations in packet arrival times.

jitter buffer loss

The packets that are lost when jitter hinders the caching capacity of the jitter buffer.

jitter buffer over runs

The number of times that jitter exceeded the maximum size setting of the jitter buffer. Packets arrive too quickly to be contained by the jitter buffer. Over runs usually result in packet loss.

jitter buffer under runs

The number of times that the jitter buffer became empty. Packets arrive too slowly to be contained by the jitter buffer. Under runs usually indicate that delays are too lengthy for the buffer setting.

keepalive

A message sent by one device to another to verify that the connection between the two is operating, or to prevent the connection from breaking.

latency

One-way delay. Calculated from the origination party to the destination party. Includes propagation delay, network delay, and packetization delay.

Listening MOS

The Mean Opinion Score, which is based on call legs traveling toward the endpoint to reflect listener perception of quality.

Mean Opinion Score (MOS)

The Mean Opinion Score (MOS) is an industry standard method for gauging call quality. MOS is an estimation of how impairments to a voice signal affect listener perception of call quality.

mean time to repair (MTTR)

Time required to repair a failed component or device. MTTR is also defined as "mean time to recovery," which is the amount of time required for a device to recover from a failure.

media device

Specialized devices to route calls from the PSTN, handle conference calls, or transcode media streams. Examples include voice gateways, mediation servers, conferencing servers, and unified messaging servers.

Media Gateway Control Protocol (MGCP)

Signaling and call control protocol used in a distributed VoIP system.

media processor

The IP termination point for audio. It performs the conversion between time-division multiplexing (TDM) and IP. The audio payload is encapsulated in RTP, then UDP, then IP.

media relay

An edge server function used with interactive connectivity establishment to provide end-to-end delivery of media streams where direct connectivity between two IP endpoints is not possible.

medianet

A medianet is an IP architecture that enhances the performance of video, voice, and data, and automates many aspects of configuration.

mediation server

Handles calls from the PSTN and interoperates with media devices that are outside the Microsoft Lync environment, such as other IP telephony environments.

midstream device

A medianet-enabled device, such as a router or switch, that sends NetFlow data to UC Monitor to report on the quality of audio or video streams.

narrowband codec

Compresses and decompresses traditional speech, covering frequencies 300 to 3400 Hz, to more easily fit over an IP network.

NetFlow

Developed by Cisco, this network protocol collects IP traffic information.

network congestion

Occurs when a network device carries so much data that the QoS deteriorates. Network congestion can result in packet loss and delay.

network delay

Transport delay produced by intervening network equipment, such as routers and switches.

Network MOS

MOS listening quality value that is based only on network factors, such as codec, packet loss, packet reordering, packet errors, and jitter.

noise level

(*Microsoft only*) The average portion of an audio signal that is noise and not actual voice data. Measured in decibels.

origination/destination

The origination phone initiates the call. The destination phone receives the call.

packet loss

The percentage of data packets that were lost in transit. These packets were sent but never received at the destination.

packet rate

The number of data packets that are received per second. UC Monitor uses this value to determine whether an RTP stream is audio or video for medianet-enabled devices that do not report a codec.

packetization delay

Delay introduced by a codec.

port mirroring

On a network switch, the port mirroring function sends copies of network packets from one port to another switch or port for analysis. The port mirroring function on Cisco switches is named Switched Port Analyzer (SPAN).

post-dial delay

The amount of time from when a user enters the last digit of a telephone number to when the user hears a ring or busy signal.

POTS

Plain Old Telephone System. The voice-grade telephone service that is the basic form of residential and small business service connection to the telephone network in most parts of the world.

presence

In a unified communications environment, the ability for users to know the status and availability of other users.

propagation delay

Delay produced by the physical distance that packets travel in a data transmission.

PSTN

Public Switched Telephone Network. The network of the world's public circuit-switched telephone networks, in much the same way that the internet is the network of the world's public IP-based packet-switched networks.

Publisher

Required member of the Cisco Unified Communications Manager cluster that publishes database (config) updates to other members of the cluster. In failover situations, the Publisher can take over call processing functions from the Subscriber.

Quality of Service (QoS)

QoS provides different priorities, or throughput levels, for different applications, users, or data flows on a packet-switched telecommunications network.

Quality Report Tool (QRT)

A problem-reporting tool for Cisco IP phones, which allows users to easily report audio and other general problems with their IP phone. Many Cisco phones have a QRT softkey.

queue abandons

Calls that were removed from the Trunk Group Queue.

queue overflows

The number of calls that arrived when all slots in the Trunk Group Queue were busy.

Real-Time Control Protocol

An IETF standard for providing out-of-band statistics and control information for an RTP flow. Generally sent over the next highest odd-numbered port as the corresponding RTP flow. RTCP provides feedback on the QoS in media distribution.

Real-Time Transport Protocol (RTP)

An IETF standard for delivering audio and video over the Internet. Generally sent on an even-numbered UDP port.

R-value

R-value is a number, or score, that is used to quantitatively express the subjective quality of speech in a VoIP network. The R-value can range from 1 (worst) to 100 (best), and is based on the percentage of users who are satisfied with the quality of a test voice signal after it passed through a network from a source (transmitter) to a destination (receiver). In many cases, an R-value is mapped to a MOS, which is used most frequently when referring to VoIP call quality.

sending/receiving

All endpoints that are involved in a call are, at some point, a sender and a receiver. During a call, all endpoints send data (talk) and receive data (listen).

sequence falls

The number of times that at least one packet arrived out of order.

sequence jumps

The number of times that at least one consecutive packet was lost.

Session Initiation Protocol (SIP)

A signaling protocol for setting up and tearing down multimedia communication sessions such as voice and video calls.

severely concealed seconds

The number of call seconds that had more than 5 percent concealment events from the start of the data stream.

signal level

The average audio signal level in decibels (dBm0).

Simple Network Management Protocol (SNMP)

Protocol for managing devices on IP networks. A network that is managed by SNMP consists of a managed device, an agent on the managed device, and a network management system on the manager.

SIP Enablement Services

The SIP proxy server for Avaya SIP endpoints.

SIP trunking

A service offered by an Internet Telephony Service Provider that permits businesses with a PBX to use VoIP outside the enterprise network by using the same connection as the Internet connection.

Skinny Call Control Protocol (SCCP)

A proprietary Cisco messaging protocol that is used between clients (phones) and the Cisco Unified Communications Manager in a VoIP environment. SCCP passes messages using TCP and port 2000.

stream leg

Unidirectional stream of packets.

Subscriber

A server in the Cisco Unified Communications Manager cluster that typically performs call-processing functions.

successful call ratio

Number of successful call completions divided by the number of call attempts.

Time to Live (TTL)

A counter embedded in data to prevent a data packet from circulating through the network indefinitely. The counter decrements each time that the packet passes through a router or a switch.

traceroute

A diagnostic tool that displays the route (path) and measures the transit delays of packets across an IP network

trunk group

A group of trunks serving the same special purpose. The term commonly is applied to voice Private Branch Exchange (PBX) trunks.

trunk group ID

The trunk group number for a voice gateway call in a Call Watch. The phone number of the endpoint that sends data through the gateway cannot be identified while the call is in progress. The trunk group ID is substituted for the phone number and displayed in the Phone Number field until the phone number is identified.

unified communications

The convergence of multiple modes of communication (such as phone, video, and email) within applications and infrastructure to allow people, teams, and organizations to communicate more effectively.

Uniform Resource Identifier (URI)

A user's SIP phone number. A SIP URI may resemble an email address, such as sip:john.smith@ca.com.

VG 224 gateway

A Cisco gateway device that lets analog phones connect to an IP PBX, which typically supports 24 analog phones. All phones are assigned the same IP address with different port numbers.

video bit rate

The number of bits sent per second for an entire video stream. Bit rates provide a gauge of codec performance.

video frame decoding time

The average amount of time for decoding frames in a stream. A slower decoding rate can be the result of conditions on the endpoint, such as lack of CPU resources, and can affect call quality.

video frame loss

The average number of unique consecutive images, or video frames, lost due to corruption and error concealment for the entire system. Video frames can span multiple packets.

video frame rate

The average number of frames that were sent or received per second for an entire stream.

video jitter

The variation in delay among video packets in the same stream.

video latency

The maximum time for a video packet to travel between the calling parties. Measured from end-to-end in one direction. Calculated by taking the average round-trip time for a call leg in a given video call and dividing it in half.

video packet loss

The percentage of video packets that were lost in transit. These packets were sent but never received at the destination.

VLAN ID

The ID of the virtual local area network (VLAN) that carries RTP packets.

voice gateway

A router or switch with a specialized card that enables VoIP calls to and from the PSTN.

Voice over IP (Voip)

A set of technologies, protocols, and transmission techniques for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet.

VoIP trunk

A large bandwidth channel that handles multimedia data and forms the backbone of a network. In telephone exchanges, a trunk simultaneously transmits data and voice packets from one point to another.

weighted average

A computation for an average value that takes into consideration the number of observations.

For example, to compute the value of average jitter across sites in your network, use the following calculation:

```
(site1 avg + site2 avg + site3 avg + site N avg) divided by N
```

where *N* is the number of sites.

To compute weighted average, use the following calculation, which allots more weight to sites with more jitter:

```
(o1(s1) + o2(s2) + o2(s3)+ oN(sN)) divided by (o1 + o2 + o3)
```

where *N* is the number of sites and *o* is the number of observations.

wideband codec

Compresses and decompresses wideband speech, such as high-definition voice, to more easily fit over an IP network.

Index

O

P

Q

R

S

T

U

V

W