Quality of Service

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Quality of Service Options

Use the following options to configure the quality of service for Cisco Jabber:

Option	Description
Enable Media Assure, on page 1	Configure Media Assure in Cisco Unified Communications Manager.
Supported Codecs, on page 3	Review the supported Codecs for each client.
Define a port range on the SIP profile, on page 4	Configure Port Ranges in Cisco Unified Communications Manager
Define a Port Range in Jabber-config.xml, on page 4	Configure Port Ranges in the jabber-config.xml file.
Set DSCP Values, on page 4	Configure Differentiated Services Code Point (DSCP) values.

Enable Media Assure

Media Assure enhances the quality of real-time media on all network types so that your meetings aren't interrupted because of poor media quality.

Before you begin

Media Assure is supported for video on Cisco Unified Communications Manager Release 10.x or later and for audio and video on Cisco Unified Communications Manager Release 11.5 or later.

Procedure

Step 1 Open the **Cisco Unified CM Administration** interface.

Step 2 Select **Device** > **Device Settings** > **Sip Profile**.

Step 3 Select your profile from the available list.

Step 4 In the **SDP Information** section, select **Pass all unknown SDP attributes** value for **SDP Transparency Profile**.

Step 5 Select **Apply Config.**
All SIP devices using this profile must restart before any changes are applied.

Supported Codecs

Type	Codec	Codec Type	Cisco Jabber for Android	Cisco Jabber for iPhone and iPad	Cisco Jabber for Mac	Cisco Jabber for Windows
Audio	G.711	A-law	Yes Supports normal mode.		Yes	Yes
		µ-law/Mu-law	Yes Supports normal mode.		Yes	Yes
	G.722		Yes		Yes	Yes
	G.722.1	24 kb/s and 32 kb/s	Yes Supports normal mode.		Yes	Yes
	G.729		Does not support Visual Voicemail with G.729; however, you can access voice messages using G.729 and the **Call Voicemail** feature.		No	No
	G.729a		Yes Minimum requirement for low-bandwidth availability. Only codec that supports low-bandwidth mode. Supports normal mode.		Yes	Yes
	Opus		Yes		Yes	Yes
Video	H.264/AVC		Yes		Yes	Yes
Voicemail	G.711	A-law	Yes		Yes	Yes
		µ-law / Mu-law (default)	Yes		Yes	Yes
	PCM linear		Yes		Yes	Yes

If users have issues with voice quality when using Cisco Jabber for Android or Cisco Jabber for iPhone and iPad, they can turn low-bandwidth mode on and off in the client settings.

Define a port range on the SIP profile

The client uses the port range to send RTP traffic across the network. The client divides the port range equally and uses the lower half for audio calls and the upper half for video calls. As a result of splitting the port range for audio media and video media, the client creates identifiable media streams. You can then classify and prioritize those media streams by setting DSCP values in the IP packet headers.

Procedure

Step 1 Open the **Cisco Unified CM Administration** interface.

Step 2 Select **Device** > **Device Settings** > **SIP Profile**.

Step 3 Find the appropriate SIP profile or create a new SIP profile.

The **SIP Profile Configuration** window opens.

Step 4 Specify whether you want common or separate port ranges for audio and video. If you are separating your audio and video port ranges, provide audio and video ports. Specify the port range in the following fields:

- **Start Media Port** — Defines the start port for media streams. This field sets the lowest port in the range.
- **Stop Media Port** — Defines the stop port for media streams. This field sets the highest port in the range.

Step 5 Select **Apply Config** and then **OK**.

Define a Port Range in Jabber-config.xml

This topics applies to Cisco Jabber for Windows.

Procedure

To specify a port range to use when users share their screen from a chat window in Cisco Jabber for Windows, see "SharePortRangeStart" in the *Cisco Jabber Parameters Reference Guide*.

Set DSCP Values

Set Differentiated Services Code Point (DSCP) values in RTP media packet headers to prioritize Cisco Jabber traffic as it traverses the network.

Set DSCP Values on Cisco Unified Communications Manager

You can set DSCP values for audio media and video media on Cisco Unified Communications Manager. Cisco Jabber can then retrieve the DSCP values from the device configuration and apply them directly to the IP headers of RTP media packets.

Restriction For later operating systems such as Microsoft Windows 7, Microsoft implements a security feature that prevents applications from setting DSCP values on IP packet headers. For this reason, you should use an alternate method for marking DSCP values, such as Microsoft Group Policy.

For more information on configuring flexible DSCP values, refer to Configure Flexible DSCP Marking and Video Promotion Service Parameters.

Procedure

Step 1 Open the **Cisco Unified CM Administration** interface.

Step 2 Select **System** > **Service Parameters**.

The **Service Parameter Configuration** window opens.

Step 3 Select the appropriate server and then select the **Cisco CallManager** service.

Step 4 Locate the **Clusterwide Parameters (System - QOS)** section.

Step 5 Specify DSCP values as appropriate and then select **Save**.

Set DSCP Values with Group Policy

If you deploy Cisco Jabber for Windows on a later operating system such as Microsoft Windows 7, you can use Microsoft Group Policy to apply DSCP values.

Complete the steps in the following Microsoft support article to create a group policy: http://technet.microsoft.com/en-us/library/cc771283%28v=ws.10%29.aspx

You should create separate policies for audio media and video media with the following attributes:

Attributes	Audio Policy	Video Policy	Signaling Policy
Application name	`CiscoJabber.exe`	`CiscoJabber.exe`	`CiscoJabber.exe`
Protocol	UDP	UDP	TCP
Port number or range	Corresponding port number or range from the SIP profile on Cisco Unified Communications Manager.	Corresponding port number or range from the SIP profile on Cisco Unified Communications Manager.	5060 for SIP 5061 for secure SIP
DSCP value	46	34	24

Set DSCP Values on the Client

For some configurations, there is an option to enable differentiated services for calls in the Cisco Jabber for Mac client and Cisco Jabber for mobile clients.

Important This option is enabled by default. Cisco recommends not disabling this option unless you are experiencing issues in the following scenarios:

- You can hear or see other parties, but you cannot be heard or seen
- You are experiencing unexpected Wi-Fi disconnection issues

Disabling differentiated service for calls may degrade audio and video quality.

Note If EnableDSCPPacketMarking is configured as true or false, then the user cannot see **Enable Differentiated Service for Calls** in the Cisco Jabber clients.

Procedure

Step 1 In Cisco Jabber for Mac, go to **Jabber > Preferences > Calls > Advanced** and select **Enable Differentiated Service for Calls**.

Step 2 In Cisco Jabber for mobile clients, go to **Jabber > Settings > Audio and Video** and select **Enable Differentiated Service for Calls**.

Set DSCP Values on the Network

You can configure switches and routers to mark DSCP values in the IP headers of RTP media.

To set DSCP values on the network, you must identify the different streams from the client application.

- Media Streams — Because the client uses different port ranges for audio streams and video streams, you can differentiate audio media and video media based on those port range. Using the default port ranges in the SIP profile, you should mark media packets as follows:
  - Audio media streams in ports from 16384 to 24574 as EF
  - Video media streams in ports from 24575 to 32766 as AF41
- Signaling Streams—You can identify signaling between the client and servers based on the various ports required for SIP, CTI QBE, and XMPP. For example, SIP signaling between Cisco Jabber and Cisco Unified Communications Manager occurs through port 5060.

  You should mark signaling packets as AF31.