Cisco Jabber Reference Information

Client Availability

Users can define whether their availability reflects their calendar events by setting an option to let others know they are in a meeting from the **Status** tab of the **Options** window from the client. This option synchronizes events in your calendar with your availability. The client only displays **In a meeting** availability for supported integrated calendars.

The client supports using two sources for the **In a meeting** availability:

Note Cisco Jabber for mobile clients don't support this meeting integration.

- Microsoft Exchange and Cisco Unified Communication Manager IM and Presence Integration — Applies to on-premises deployments. The **Include Calendar information in my Presence Status** field in Cisco Unified Presence is the same as the **In a meeting** option in the client. Both fields update the same value in the Cisco Unified Communication Manager IM and Presence database.

  If users set both fields to different values, then the last field that the user sets takes priority. If users change the value of the **Include Calendar information in my Presence Status** field while the client is running, the users must restart the client for those changes to apply.

- Cisco Jabber Client — Applies to on-premises and cloud-based deployments. You must disable Cisco Unified Communication Manager IM and Presence and Microsoft Exchange integration for the client to set the **In a meeting** availability. The client checks if integration between Cisco Unified Communication Manager IM and Presence and Microsoft Exchange is on or off. The client can only set availability if integration is off.

The following deployment scenarios describe how availability is created:

Deployment Scenario	You select In a meeting (according to my calendar)	You do not select In a meeting (according to my calendar)
You enable integration between Cisco Unified Communication Manager IM and Presence and Microsoft Exchange.	Cisco Unified Communication Manager IM and Presence sets availability status	Availability status does not change
You do not enable integration between Cisco Unified Communication Manager IM and Presence and Microsoft Exchange.	Client sets availability status	Availability status does not change
Cloud-based deployments	Client sets availability status	Availability status does not change

Additionally, the following table describes availability that is supported differently by each deployment scenarios:

Availability Enabled in the Client	Availability Enabled by Integrating Cisco Unified Communication Manager IM and Presence with Microsoft Exchange
Offline in a meeting availability is not supported.	**Offline in a meeting** availability is supported.
In a meeting availability is supported for non-calendar events.	**In a meeting** availability is not supported for non-calendar events.

Note Offline in a meeting availability refers to when the user is not logged in to the client but an event exists in the user's calendar.

Non-calendar events refer to events that do not appear in the user's calendar, such as instant meetings, **Offline**, or **On a call**.

Related Topics

Calendar Integration

Multiple Resource Login

All Cisco Jabber clients register with one of the following central IM and Presence Service nodes when a user logs in to the system. This node tracks availability, contact lists, and other aspects of the IM and Presence Service environment.

- On-Premises Deployments: Cisco Unified Communications Manager IM and Presence Service.
- Cloud Deployments: Cisco Webex.

This IM and Presence Service node tracks all of the registered clients associated with each unique network user in the following order:

1. When a new IM session is initiated between two users, the first incoming message is broadcast to all of the registered clients of the receiving user.
2. The IM and Presence Service node waits for the first response from one of the registered clients.
3. The first client to respond then receives the remainder of the incoming messages until the user starts responding using another registered client.
4. The node then reroutes subsequent messages to this new client.

Note If there is no active resource when a user is logged into multiple devices, then priority is given to the client with the highest presence priority. If the presence priority is the same on all devices, then priority is given to the latest client the user logged in to.

Protocol Handlers

Cisco Jabber registers the following protocol handlers with the operating system to enable click-to-call or click-to-IM functionality from web browsers or other applications:

- XMPP: or XMPP://

  Starts an instant message and opens a chat window in Cisco Jabber.

- IM: or IM://

  Starts an instant message and opens a chat window in Cisco Jabber.

- TEL: or TEL://

  Starts an audio or video call with Cisco Jabber.

  Note TEL is registered by Apple native phone. It cannot be used to cross launch Cisco Jabber for iPhone and iPad.

- CISCOTEL: or CISCOTEL://

  Starts an audio or video call with Cisco Jabber.

- SIP: or SIP://

  Starts an audio or video call with Cisco Jabber.

Registry Entries for Protocol Handlers

To register as a protocol handler, the client writes to the following locations in the Microsoft Windows registry:

- `HKEY_CLASSES_ROOT\tel\shell\open\command`
- `HKEY_CLASSES_ROOT\xmpp\shell\open\command`

- `HKEY_CLASSES_ROOT\im\shell\open\command`

In the case where two or more applications register as handlers for the same protocol, the last application to write to the registry takes precedence. For example, if Cisco Jabber registers as a protocol handler for XMPP: and then a different application registers as a protocol handler for XMPP:, the other application takes precedence over Cisco Jabber.

Protocol Handlers on HTML Pages

You can add protocol handlers on HTML pages as part of the `href` attribute. When users click the hyperlinks that your HTML pages expose, the client performs the appropriate action for the protocol.

TEL and IM Protocol Handlers

Example of the TEL: and IM: protocol handlers on an HTML page:

```
<html>
  <body>
    <a href="TEL:1234">Call 1234</a><br/>
    <a href="IM:msmith@domain">Send an instant message to Mary Smith</a>
  </body>
</html>
```

In the preceding example, when users click the hyperlink to call 1234, the client starts an audio call to that phone number. When users click the hyperlink to send an instant message to Mary Smith, the client opens a chat window with Mary.

CISCOTEL and SIP Protocol Handlers

Example of the CISCOTEL and SIP protocol handlers on an HTML page:

```
<html>
  <body>
    <a href="CISCOTEL:1234">Call 1234</a><br/>
    <a href="SIP:msmith@domain">Call Mary</a><br/>
    <a href="CISCOTELCONF:msmith@domain;amckenzi@domain">Weekly conference call</a>
  </body>
</html>
```

In the preceding example, when users click the *Call 1234* or *Call Mary* hyperlinks, the client starts an audio call to that phone number.

XMPP Protocol Handlers

Example of a group chat using the XMPP: protocol handler on an HTML page:

```
<html>
  <body>
    <a href="XMPP:msmith@domain;amckenzi@domain">Create a group chat with Mary Smith and
Adam McKenzie</a>
  </body>
</html>
```

In the preceding example, when users click the hyperlink to create a group chat with Mary Smith and Adam McKenzie, the client opens a group chat window with Mary and Adam.

Tip Add lists of contacts for the XMPP: and IM: handlers to create group chats. Use a semi-colon to delimit contacts, as in the following example:

```
XMPP:user_a@domain.com;user_b@domain.com;user_c@domain.com;user_d@domain.com
```

Add Subject Lines and Body Text

You can add subject lines and body text to any of the protocol handlers so that when users click on the hyperlink to create a person-to-person or group chat, the client opens a chat window with pre-populated subject line and body text.

Subject and body text can be added in any of the following scenarios:

- Using any supported protocol handler for instant messaging on the client
- For either person-to-person chats or for group chats
- Including a subject and body text, or one or the other

In this example, when users click on the link below it opens a person-to-person chat window with a pre-populated body text of **`I.T Desk`**:

```
xmpp:msmith@domain?message;subject=I.T.%20Desk
```

In this example, when users click on the link below it opens a **Start Group Chat** dialog box with a topic of **I.T Desk**, and the input box for the chat window is pre-populated with the text **`Jabber 10.5 Query`**:

```
im:user_a@domain.com;user_b@domain.com;user_c@domain.com?message;subject=I.T%20Desk;body=Jabber%2010.5%20Query
```

Audio and Video Performance Reference

Attention The following data is based on testing in a lab environment. This data is intended to provide an idea of what you can expect in terms of bandwidth usage. The content in this topic is not intended to be exhaustive or to reflect all media scenarios that might affect bandwidth usage.

Audio Bit Rates for Cisco Jabber Desktop Clients

The following audio bit rates apply to Cisco Jabber for Windows and Cisco Jabber for Mac.

Codec	RTP (kbits/second)	Actual bit rate (kbits/second)	Notes
G.722.1	24/32	54/62	High quality compressed
G.711	64	80	Standard uncompressed
G.729a	8	38	Low quality compressed

Audio Bit Rates for Cisco Jabber Mobile Clients

The following audio bit rates apply to Cisco Jabber for iPad and iPhone and Cisco Jabber for Android.

Codec	Codec bit rate (kbits/second)	Network Bandwidth Utilized (kbits/second)
g.711	64	80
g.722.1	32	48
g.722.1	24	40
g.729a	8	24

Video Bit Rates for Cisco Jabber Desktop Clients

The following video bit rates (with g.711 audio) apply to Cisco Jabber for Windows and Cisco Jabber for Mac. This table does not list all possible resolutions.

Resolution	Pixels	Measured bit rate (kbits per second) with g.711 audio
w144p	256 x 144	156
w288p This is the default size of the video rendering window for Cisco Jabber.	512 x 288	320
w448p	768 x 448	570
w576p	1024 x 576	890
720p	1280 x 720	1300

Note The measured bit rate is the actual bandwidth used (RTP payload + IP packet overhead).

Video Bit Rates for Cisco Jabber for Android

The client captures and transmits video at 15 fps.

Resolution	Pixels	Bit Rate (kbits per second) with g.711 audio
w144p	256 x 144	290
w288p	512 x 288	340
w360p	640 x 360	415

Video	Resolution	Bandwidth
HD	1280 x 720	1024
VGA	640 x 360	512
CIF	488x211	310

Note To send and receive HD video during calls:

- Configure the maximum bit rate for video calls higher than 1024 kbps in Cisco Unified Communications Manager.
- Enable DSCP on a router to transmit video RTP package with high priority.

Video Bit Rates for Cisco Jabber for iPhone and iPad

The client captures and transmits at 20 fps.

Resolution	Pixels	Bit rate (kbits/second) with g.711 audio
w144p	256 x 144	290
w288p	512 x 288	340
w360p	640 x 360	415
w720p	1280 x 720	1024

Presentation Video Bit Rates

Cisco Jabber captures at 8 fps and transmits at 2 to 8 fps.

The values in this table do not include audio.

Pixels	Estimated wire bit rate at 2 fps (kbits per second)	Estimated wire bit rate at 8 fps (kbits per second)
720 x 480	41	164
704 x 576	47	188
1024 x 768	80	320
1280 x 720	91	364
1280 x 800	100	400

Maximum Negotiated Bit Rate

You specify the maximum payload bit rate in Cisco Unified Communications Manager in the **Region Configuration** window. This maximum payload bit rate does not include packet overhead, so the actual bit rate used is higher than the maximum payload bit rate you specify.

The following table describes how Cisco Jabber allocates the maximum payload bit rate:

Audio	Interactive video (Main video)
Cisco Jabber uses the maximum audio bit rate	Cisco Jabber allocates the remaining bit rate as follows: The maximum video call bit rate minus the audio bit rate.

Bandwidth Performance Expectations for Cisco Jabber Desktop Clients

Cisco Jabber for Mac separates the bit rate for audio and then divides the remaining bandwidth equally between interactive video and presentation video. The following table provides information to help you understand what performance you should be able to achieve per bandwidth:

Upload speed	Audio	Audio + Interactive video (Main video)
125 kbps under VPN	At bandwidth threshold for g.711. Sufficient bandwidth for g.729a and g.722.1.	Insufficient bandwidth for video.
384 kbps under VPN	Sufficient bandwidth for any audio codec.	w288p (512 x 288) at 30 fps
384 kbps in an enterprise network	Sufficient bandwidth for any audio codec.	w288p (512 x 288) at 30 fps
1000 kbps	Sufficient bandwidth for any audio codec.	w576p (1024 x 576) at 30 fps
2000 kbps	Sufficient bandwidth for any audio codec.	w720p30 (1280 x 720) at 30 fps

Cisco Jabber for Windows separates the bit rate for audio and then divides the remaining bandwidth equally between interactive video and presentation video. The following table provides information to help you understand what performance you should be able to achieve per bandwidth:

Upload speed	Audio	Audio + Interactive video (Main video)	Audio + Presentation video (Desktop sharing video)	Audio + Interactive video + Presentation video
125 kbps under VPN	At bandwidth threshold for g.711. Sufficient bandwidth for g.729a and g.722.1 .	Insufficient bandwidth for video.	Insufficient bandwidth for video.	Insufficient bandwidth for video.

Upload speed	Audio	Audio + Interactive video (Main video)	Audio + Presentation video (Desktop sharing video)	Audio + Interactive video + Presentation video
384 kbps under VPN	Sufficient bandwidth for any audio codec.	w288p (512 x 288) at 30 fps	1280 x 800 at 2+ fps	w144p (256 x 144) at 30 fps + 1280 x 720 at 2+ fps
384 kbps in an enterprise network	Sufficient bandwidth for any audio codec.	w288p (512 x 288) at 30 fps	1280 x 800 at 2+ fps	w144p (256 x 144) at 30 fps + 1280 x 800 at 2+ fps
1000 kbps	Sufficient bandwidth for any audio codec.	w576p (1024 x 576) at 30 fps	1280 x 800 at 8 fps	w288p (512 x 288) at 30 fps + 1280 x 800 at 8 fps
2000 kbps	Sufficient bandwidth for any audio codec.	w720p30 (1280 x 720) at 30 fps	1280 x 800 at 8 fps	w288p (1024 x 576) at 30 fps + 1280 x 800 at 8 fps

Note that VPN increases the size of the payload, which increases the bandwidth consumption.

Bandwidth Performance Expectations for Cisco Jabber for Android

Note that VPN increases the size of the payload, which increases the bandwidth consumption.

Upload speed	Audio	Audio + Interactive Video (Main Video)
125 kbps under VPN	At bandwidth threshold for g.711. Insufficient bandwidth for video. Sufficient bandwidth for g.729a and g.722.1.	Insufficient bandwidth for video.
256 kbps	Sufficient bandwidth for any audio codec.	Transmission rate (Tx) — 256 x 144 at 15 fps Reception rate (Rx) — 256 x 144 at 30 fps
384 kbps under VPN	Sufficient bandwidth for any audio codec.	Tx — 640 x 360 at 15 fps Rx — 640 x 360 at 30 fps
384 kbps in an enterprise network	Sufficient bandwidth for any audio codec.	Tx — 640 x 360 at 15 fps Rx — 640 x 360 at 30 fps

Note Due to device limitations, the Samsung Galaxy SII and Samsung Galaxy SIII devices cannot achieve the maximum resolution listed in this table.

Bandwidth Performance Expectations for Cisco Jabber for iPhone and iPad

The client separates the bit rate for audio and then divides the remaining bandwidth equally between interactive video and presentation video. The following table provides information to help you understand what performance you should be able to achieve per bandwidth.

Note that VPN increases the size of the payload, which increases the bandwidth consumption.

Upload speed	Audio	Audio + Interactive Video (Main Video)
125 kbps under VPN	At bandwidth threshold for g.711. Insufficient bandwidth for video. Sufficient bandwidth for g.729a and g.722.1.	Insufficient bandwidth for video.
290 kbps	Sufficient bandwidth for any audio codec.	256 x144 at 20 fps
415 kbps	Sufficient bandwidth for any audio codec.	640 x 360 at 20 fps
1024 kbps	Sufficient bandwidth for any audio codec.	1280 x 720 at 20 fps

Video Rate Adaptation

Cisco Jabber uses video rate adaptation to negotiate optimum video quality. Video rate adaptation dynamically increases or decreases video bit rate throughput to handle real-time variations on available IP path bandwidth.

Cisco Jabber users should expect video calls to begin at lower resolution and scale upwards to higher resolution over a short period of time. Cisco Jabber saves history so that subsequent video calls should begin at the optimal resolution.

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**.

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.

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.

Procedure

In Cisco Jabber for Mac, go to **Jabber** > **Preferences** > **Calls** > **Advanced** 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.

Cisco Jabber Reference Information