



IX System Network and Bandwidth Requirements

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This chapter describes the network and bandwidth requirements to ensure the immersive video quality of the IX system. This chapter contains the following sections:

- [Quality of Service, page 1-1](#)
 - [Packet Loss, page 1-2](#)
 - [Jitter, page 1-2](#)
 - [Latency, page 1-4](#)
- [Bandwidth Management for the IX System, page 1-4](#)



Note

The bandwidth requirements in this section are based on network testing in environments without jitter or packet loss. Due to the extremely high-definition of the video, a small amount of packet loss or jitter can cause noticeable disruption to the user experience. To maintain the real-time interaction of human conversations, latency must be kept to an absolute minimum. Networks that have not been configured with proper QoS or inefficient routing over the WAN may cause loss of video quality.

Quality of Service

This section briefly describes Quality of Service (QoS) considerations for the IX5000 and IX5200. The immersive quality of the IX system requires that QoS mechanisms available on Cisco switches and routers throughout the network be configured to reduce packet loss, latency and jitter. Redundant devices and network links that provide quick convergence after network failures or topology changes are also important to ensure a highly available infrastructure. The following sections describe the packet loss, jitter and latency requirements for the IX system.

- [Packet Loss, page 1-2](#)
- [Jitter, page 1-2](#)
- [Latency, page 1-4](#)

For more detailed information, see the *Cisco Collaboration System 11.x Solution Reference Network Designs (SRND)* document at:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11.html

Packet Loss

Because the IX system is highly sensitive to packet loss, your network should be configured so that packet loss does not exceed a target of 0.05%. Cisco TelePresence codecs transmit at approximately 5 Mbps (max) per 1080pdisplay, which translates to more than 99% compression. The overall effect of packet loss is proportionally magnified, and dropping even one packet in 2000 (0.05% packet loss) becomes noticeable to end users.

For more detailed information about packet loss requirements for TelePresence systems, see the *Cisco TelePresence Network Systems 1.1 Design Guide* at:

https://www.cisco.com/c/dam/en/us/td/docs/solutions/TelePresence_Network_Systems_1-1_DG.pdf

Jitter

- [Understanding Jitter and Defining Jitter Thresholds, page 1-2](#)
- [How Your IX System Measures Jitter, page 1-3](#)

Understanding Jitter and Defining Jitter Thresholds

Jitter is defined as the difference between the time the video frame is expected to arrive at the endpoint, and the actual time that it arrives. There are two types of jitter: packet jitter, and video frame jitter. The video frames are enclosed in packets.

For packet jitter, you should follow the guidelines in the “[Network Infrastructure](#)” section of the *Cisco Collaboration System 11.x Solution Reference Network Designs (SRND)*. Currently the jitter level at the packet level is set at a maximum of 100 milliseconds (ms).

You should measure jitter at the video frame level (application layer) for Cisco TelePresence systems, rather than the packet level (network layer). A network with 0 ms of packet frame jitter can still have jitter at the video frame level if the RX buffer of the codec is overwhelmed with a large number of packets. Therefore, you should measure the arrival time of the entire video frame vs. the expected arrival time of that frame, based on the clock rate of video frame intervals. The clock rate for 30 fps is 33ms, while the clock rate for 60 fps is 16.5ms. Use the information in [Table 1-1](#) as the guideline for jitter in your network.

Measure jitter from the Ethernet port of the system codec on the far end to the Ethernet port of the system codec on the near end.

Table 1-1 Jitter Value

Metric	Target	Thresholds			
		1st	2nd	3rd	4th
Video Jitter	50 ms	85 ms	125 ms	165 ms	245 ms

Your system monitors jitter in the following ways:

- During a call, you can view the jitter information on the Touch device by tapping **More > Status > Call Status** and viewing the Jitter field. If the jitter value is Good, the jitter rate is lower than 125 ms. A value of Marginal represents jitter rates between 125 and 165 ms. A value of Poor represents jitter rates above 165 ms.

- You can monitor the jitter rate for your system by logging into the IX Administration Console for your system and navigating to **Monitoring > Call Statistics**, then clicking the **AV Call Video Stream Statistics** tab and checking the jitter rates in both the Transmit and Receive areas.

The jitter rates are color-coded. Jitter rates that are less than 125 ms is marked in green. Jitter rates between 125 and 165 ms are marked in yellow, and rates above 165 ms are marked in red.

Although no system actions are performed when jitter levels are exceeded, jitter at the video frame level is closely related to dropped video frames. If the RX buffer is exceeded, the system starts dropping frames. The Cisco TelePresence system changes the call quality based on the number of dropped frames.

- You can use the status bars on the IX system screen to monitor packet drop, which is closely related to jitter. For more information, see the [“Checking IX Bandwidth Quality On the System Display” section on page 1-12](#).



Note The status bars are not shown on calls that use a Cisco TelePresence Server; however, the same steps are taken, whether or not the bars are shown.

How Your IX System Measures Jitter

The IX system measures jitter upon the arrival of each frame and reports the jitter based on per 10-second and per call averages. The jitter period report provides the jitter measurement for the last 10-second period. The jitter call report shows the average jitter measurement per call. You can access both reports in the **Monitoring > Call Statistics** page in the IX Administration interface.



Note The IX system measures jitter between video frames and not packet frames, which applies to network devices. A video frame is a compressed picture that is used to update the screen.

The IX system calculates jitter as the sum of the maximum deviation (both late and early packets) from the expected arrival time as given by the frame period. (A late packet is a packet delivered after the picture has been reassembled and sent to the display. These packets are not lost but have the same impact as lost packets because they cannot be used by the video decoder.) The IX system computes frame jitter based on the arrival time of the last packet of a frame.

For example, for a 30 fps video stream with a measurement period of 165 ms or 5 frames (instead of 10 seconds and 300 frames), the IX system performs 5 jitter calculations. The Jitter (Period) would be reported as 5 ms (or 1 ms per frame [5 ms/5 frames = 1 ms jitter per frame]).

```
Frame Actual Arrival Time(ms) = 0 33 70 99 131
Frame Expected Arrival Time(ms) = 0 33 66 99 132
Offset = 0 0 +4 0 -1
Max Late = 4ms (absolute value)
Min Late = 1ms (absolute value)
Jitter/Period = 5ms (for this 165ms period)
```

If there were only 2 jitter periods in this call, the first period jitter measurement would be 5 msec and the second period jitter measurement would be 10 msec. The Jitter/Call would be reported as 7.5 msec.

```
Jitter/Call = (((PerCallJitter * (NumMeasurementIntervals - 1)) + CurrentJitter) /
(NumMeasurementIntervals)
Where: PerCallJitter = 5msecs
NumMeasurementIntervals = 2 (1-relative)
CurrentJitter = 10 msec
Jitter/Call = 7.5msecs = ((5msecs * (2-1)) + 10msecs) / 2
```

Latency

The IX system, like other Cisco TelePresence systems, has a network latency target of 150ms. This target includes network flight time but does not include codec processing time. For more detailed information about latency requirements for TelePresence systems, see the *Cisco TelePresence Network Systems 1.1 Design Guide* at:

https://www.cisco.com/c/dam/en/us/td/docs/solutions/TelePresence_Network_Systems_1-1_DG.pdf

Bandwidth Management for the IX System

This section describes bandwidth management considerations for the IX5000 and IX5200. The following sections describe the enhanced bandwidth control mechanisms and how they are implemented on the IX5000 and IX5200:

- [Bandwidth Management Overview, page 1-4](#)
- [Bandwidth Provisioning Guidelines, page 1-5](#)
- [Bandwidth Requirements for 1080p 60-fps Main Video, page 1-7](#)
- [High-Definition Presentation, page 1-14](#)
- [Required Configuration For HD Presentation, page 1-15](#)
- [Video Bandwidth Allocation Weights, page 1-17](#)



Note

This section was previously included in the “[IX Software Features](#)” section on page 4-1.

Bandwidth Management Overview

To provide the immersive video experience for the IX5000 and IX5200, the system requires careful planning for peak bandwidth requirements. Make sure to provision enough bandwidth from your service provider to account for total traffic requirements. The IX supports the following bandwidth management mechanisms:

- [Gradual Decoder Refresh \(GDR\), page 1-5](#)
- [Encoder Pacing, page 1-5](#)

Due to the nature of the video compression, the amount of TelePresence traffic may fluctuate with microbursts. Peak rates during a video conference generally occur at the following times:

- At the beginning of the call (or when a held call is resumed).
- When repair frames are generated (for example, recovering from packet loss).
- During periods of high motion in the video (for example, when participants all stand up or walk about the room).
- When there are details and high motion in the content, or when there is a complete scene change in the presentation video.

When microbursts happen, it may cause network packet loss and trigger repair frames to be created, which introduce further traffic variability in the bandwidth utilization profile. The IX system uses enhanced encoder rate control and packet pacing to maintain constant average bandwidth usage and reduce microburst in network data traffic. See the “[Gradual Decoder Refresh \(GDR\)](#)” section on

[page 1-5](#) and the “Encoder Pacing” section on [page 1-5](#) for more information.

Gradual Decoder Refresh (GDR)

Gradual Decoder Refresh is used to replace IDR (Instantaneous Decoder Refresh) to fulfill frame repair requests. Because IDR frames are much larger than other frames, IDR is usually the source of microbursts in video traffic. GDR gradually refreshes the picture over several frames, providing a smoother, less bursty bitstream. The IX system supports GDR for H.264 Main video on point-to-point calls between the IX system and the following systems:

- Another IX system
- TX Series systems
- CTS Series systems

GDR is supported for H.264 point-to-point Presentation video on calls between an IX system and the following systems:

- Another IX system
- TX Series systems

For more conceptual information about GDR, see the “Bandwidth Management” chapter of the [Cisco Collaboration System 11.x Solution Reference Network Designs \(SRND\)](#) document.

Encoder Pacing

Encoder pacing is a bandwidth management technique used to spread packets as evenly as possible, smoothing out the peaks of the bursts of bandwidth. Network packet pacing is used in the IX5000 after the video encoder.

Encoder pacing is supported in all call scenarios. For more conceptual information about encoder pacing, see the “Bandwidth Management” chapter of the [Cisco Collaboration System 11.x Solution Reference Network Designs \(SRND\)](#) document.

Microbursts are observed more frequently when bandwidth is measured in smaller intervals such as 100 milliseconds. Microbursts are observed less frequently when bandwidth is measured in intervals of one second or longer over the duration of the video call.

When measuring bandwidth statistics for how the IX system performs encoder pacing, note the following:

- Perform measurements at the network interface.
- Do not rely on statistics taken from the Touch 10 device or the Administrator user interface.

Statistics shown on the Touch 10 device show only instantaneous measurements on the encoder and decoder behavior, and the jitter buffer state. These statistics do not necessarily represent accurate measurements of the bandwidth used. These statistics should be interpreted as the bandwidth prior to the encoder pacing mechanism.

Bandwidth Provisioning Guidelines

To ensure the immersive video experience of the IX system, make sure to provision sufficient bandwidth from your service provider. Like other systems, the IX system requires bandwidth consumption headroom on top of average bandwidth usage to account for network overhead and microburst absorption.

Table 1-2, Table 1-3, Table 1-4, and Table 1-5 show the recommended amount of bandwidth to provision from a service provider to handle different call scenarios.

The following information applies to the recommended bandwidth guidelines:

- The bandwidth provisioning guidelines add 20 percent over normal bandwidth requirements to account for Layer 2-to-Layer 4 and other overhead. The calculations also include allowance for bursts and microbursts.
- The guidelines shown are for best motion quality only. Bandwidth requirements for good and better motion quality are less.

For information on calculating bandwidth requirements, see the “[Understanding How Endpoints Determine fps and Video Quality](#)” section on page 1-8.

- Calls from one IX system to another include two 1080p30 presentation streams (H.264).

Table 1-2 shows the bandwidth provisioning guidelines for 1080p30 content over H.264.

Table 1-2 *Bandwidth Provisioning Guidelines: H.264 for 1080p30 Content*

Call Scenario	Total Bandwidth Consumption		
	Resolution	30 fps	60 fps
		Best	Best
IX system to IX system	1080p	24.4 Mbps	31.6 Mbps
	720p	18.1 Mbps	22.1 Mbps
IX system to CTS or TX system ¹	1080p	19.5 Mbps	26.7 Mbps
	720p	13.2 Mbps	17.3 Mbps
IX system to Cisco TelePresence Server ²	1080p	16.8 Mbps	NA
	720p	13.2 Mbps	NA
IX system Native SIP	1080p	9.7 Mbps	12.1 Mbps
	720p	7.6 Mbps	8.9 Mbps

1. 60 fps supported for TX systems only.

2. Cisco TelePresence Server supports “Better” motion quality only in TIP/MUX calls.

Table 1-3 shows the bandwidth provisioning guidelines for 1080p5 content over H.264.

Table 1-3 *Bandwidth Provisioning Guidelines: H.264 for 1080p5 Content*

Call Scenario	Total Bandwidth Consumption		
	Resolution	30 fps	60 fps
		Best	Best
IX system to IX system	1080p	17.2 Mbps	24.4 Mbps
	720p	10.9 Mbps	14.9 Mbps
IX system to CTS or TX system ¹	1080p	15.9 Mbps	23.1 Mbps
	720p	9.6 Mbps	13.7 Mbps

1. 60 fps supported for TX systems only.

Table 1-4 shows the bandwidth provisioning guidelines for 1080p30 content over H.265.

Table 1-4 *Bandwidth Provisioning Guidelines: H.265 for 1080p30 Content*

Call Scenario	Total Bandwidth Consumption		
	Resolution	30 fps	60 fps
		Best	Best
IX system to IX system	1080p	18.6 Mbps	22.9 Mbps
	720p	14.8 Mbps	17.3 Mbps
IX system Native SIP	1080p	6.5 Mbps	NA
	720p	5.4 Mbps	6.1 Mbps

Table 1-5 shows the bandwidth provisioning guidelines for 1080p5 content over H.265.

Table 1-5 *Bandwidth Provisioning Guidelines: H.265 for 1080p5 Content*

Call Scenario	Total Bandwidth Consumption		
	Resolution	30 fps	60 fps
		Best	Best
IX system to IX system	1080p	11.4 Mbps	15.7 Mbps
	720p	7.6 Mbps	10.1 Mbps

Bandwidth Requirements for 1080p 60-fps Main Video

The Cisco TelePresence IX5000 System can send and receive main video at 60 fps (frames per second) with 1080p quality (1080p 60) during a point-to-point call. The following sections describe the 1080p 60-fps Main Video functionality:

- [Required Main Video Configuration, page 1-7](#)
- [Understanding How Endpoints Determine fps and Video Quality, page 1-8](#)
- [Checking IX Bandwidth Quality On the System Display, page 1-12](#)

Required Main Video Configuration

The following configuration is required to enable the 60-fps main video feature in your Cisco TelePresence environment:

- For IX endpoints, make the following changes in Unified CM version 10 or later:
 - Set Main Display Frames Per Second in the Phone Configuration page to “60 fps main”. For more information, refer to the “[Product Specific Configuration Layout Area](#)” section of the *Configuring Cisco Unified Communications Manager for the IX System* document.

- (Optional) Set Video Bandwidth Allocation Weights on the Phone Configuration page appropriately. For more information about this parameter, see the “[Video Bandwidth Allocation Weights](#)” section on page 1-17.

After performing the required configuration, the IX endpoints can send and receive main video at a maximum frame rate of 60 fps.

Understanding How Endpoints Determine fps and Video Quality

During Cisco TelePresence call setup, the sending and receiving endpoints determine the fps (30 or 60 fps) and video quality (1080p or 720p) for the sent and received video streams.

The determination is made as a result of the following factors:

- The amount of Transport Independent Application Specific (TIAS) bandwidth that is negotiated between the sending and receiving endpoints.

The minimum amount of bandwidth is determined by the settings of Main Display Frames Per Second and Quality (Per Display) in the Phone Configuration page of Unified CM. [Figure 1-1](#) and [Table 1-6](#) show the bandwidth requirements for H.265 based on the Unified CM configuration for 30 fps and 60 fps calls. [Figure 1-2](#) and [Table 1-7](#) show the corresponding bandwidth requirements for H.264.

- The video compression standard (H.264 or H.265) that is used.
- The maximum frame size that the network and system can accommodate.
- The negotiation of the video resolution and frame rate (in fps) by the sending and receiving endpoints.
- The maximum bit rate allowed in the Region settings for your device in Unified CM. These settings are applied to the Device Pool, which in turn are applied to your device.

To find your region settings, log in to the Cisco Unified CM Administration GUI and navigate to **System > Region**. The maximum rate is the value shown in the Max Video Call Bit Rate (Includes Audio) field.

- The packet loss that is detected during a call. This loss is shown as status bars that appear on the main display. If the rate changes, the new rate is shown on the main display. [Table 1-11](#) shows the bars and provides a description.

[Table 1-6](#) and [Table 1-7](#) show the required criteria in the first three columns of the table. The resulting video stream that can be sent is shown in the fourth column of the table.

If any factors do not meet the minimum requirements, the system attempts to send and receive video at the next lowest rate as shown in [Table 1-10](#).

For example, if the network cannot meet the minimum requirements to send a video stream with the maximum video quality of 1080p 60, the system attempts to negotiate a video stream of 720p 60. If the system cannot meet the requirements to send a video stream of 720p 60, it attempts to negotiate a video stream of 720p 30 as shown in [Table 1-10](#).

If a call is put on hold, then resumed, the amount of bandwidth is renegotiated using the same factors.



Note

These rates are given per video stream, and do not include the presentation stream. Since the IX system has three video streams for the three displays, multiply this number by three. Then add the bandwidth required for the presentation given in [Table 1-15](#) to obtain the required network bandwidth.

Table 1-6 Possible Values of Network Parameters and Resulting Resolution/Frame Rate (H.265)

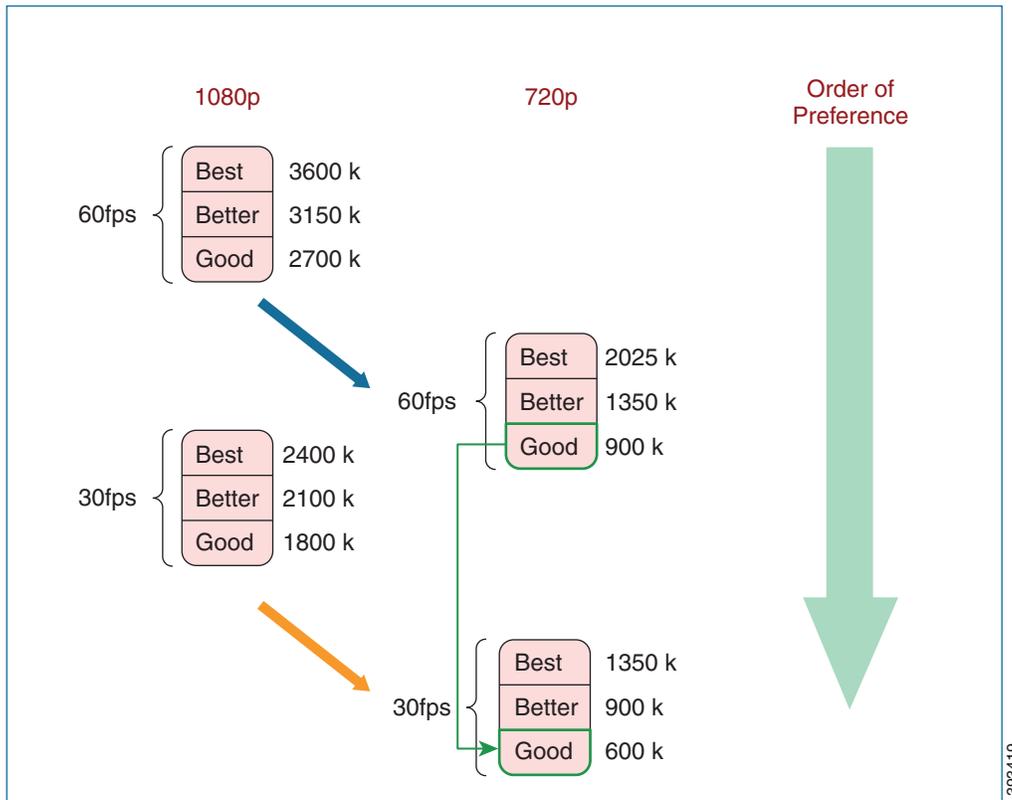
Minimum Preferred Bit Rate (Kbps)	Minimum Frame Size	Minimum fps	Resulting Resolution and fps
2700	8100	60	1080p 60
900	3600	60	720p 60
1800	8100	30	1080p 30
600	3600	30	720p 30

Table 1-7 Possible Values of Signaling Parameters and Resulting Resolution/Frame Rate (H.264)

Minimum Preferred Bit Rate (Kbps)	Minimum Frame Size	Minimum fps	Resulting Resolution and fps
4500	8100	60	1080p 60
1500	3600	60	720p 60
3000	8100	30	1080p 30
1000	3600	30	720p 30

Figure 1-1 shows the bandwidth required when using the H.265 video compression format, while Figure 1-2 shows the bandwidth required when using the H.264 format.

Figure 1-1 H.265 Bandwidth Requirements Per Unified CM Quality (Per Display)



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Figure 1-2 H.264 Bandwidth Requirements Per Unified CM Quality (Per Display)

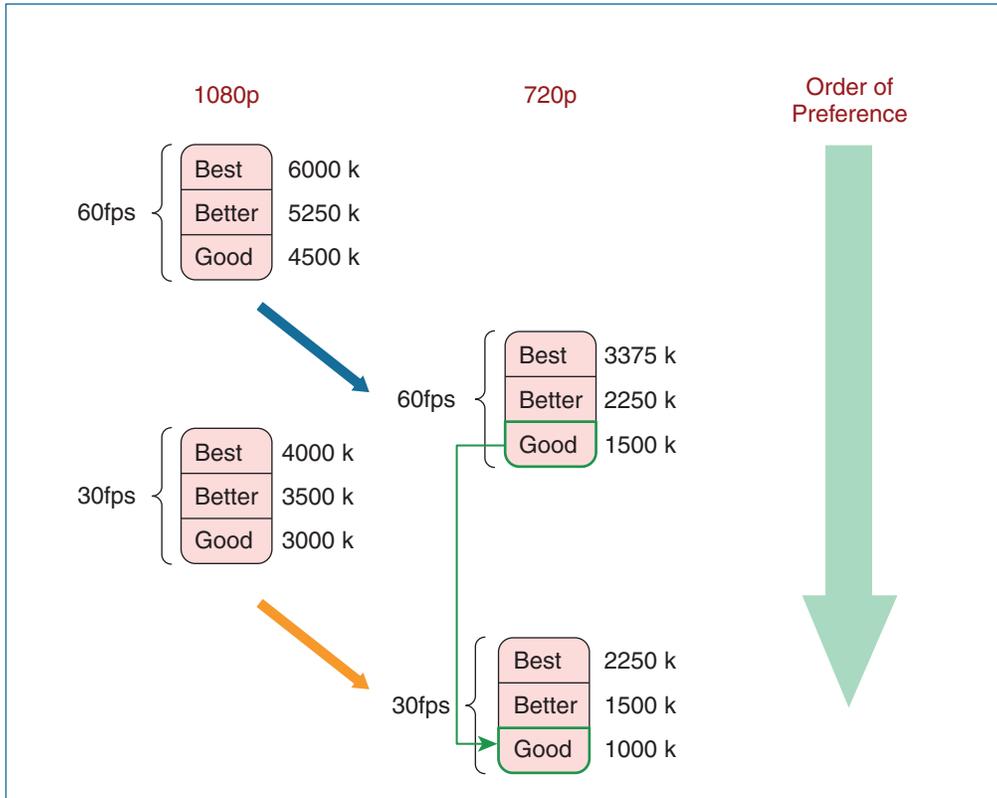


Table 1-8 provides the bandwidth requirements for H.265 for 30 fps and 60-fps calls; Table 1-9 provides the same information for H.264.

Table 1-8 H.265 Bandwidth Requirements Based On Unified CM Quality (Per Display)

Main Display Frames Per Second Setting	Quality (Per Display) Setting		
	Best	Better	Good
1080p Bandwidth Requirements (kbps)			
60 fps main	3600	3150	2700
30 fps main	2400	2100	1800
720p Bandwidth Requirements (kbps)			
60 fps main	2025	1350	900
30 fps main	1350	900	600

Table 1-9 H.264 Bandwidth Requirements Based On Unified CM Quality (Per Display)

Main Display Frames Per Second Setting	Quality (Per Display) Setting		
	Best	Better	Good
1080p Bandwidth Requirements (kbps)			
60 fps main	6000	5250	4500
30 fps main	4000	3500	3000
720p Bandwidth Requirements (kbps)			
60 fps main	3375	2250	1500
30 fps main	2250	1500	1000

Table 1-10 Negotiated Downgrade Paths

Initial Negotiated Resolution and Frame Rate	Downgrade Path During Call
Initially Negotiated 60 fps Calls	
1080p@60 fps	720p@60 fps, then 720p@30 fps
720p@60 fps	720@30 fps
Initially Negotiated 30 fps Calls	
1080p@30 fps	720@30 fps
720p@30 fps	720@30 fps

The 60 fps-capable Cisco TelePresence endpoints and device send the main video stream at 30 fps under the following circumstances:

- When in a call with an endpoint or device that supports a maximum frame rate of 30 fps.
- When in a call with a Cisco TelePresence endpoint that is registered with a Cisco Unified CM version that does not support 60 fps as a setting for Main Display Frames Per Second in the Phone Configuration page.
- When in a call with a Cisco TelePresence endpoint that is registered with a Unified CM version that supports 60 fps, but “30 fps main” is the setting for Main Display Frames Per Second in the Phone Configuration page.

Checking IX Bandwidth Quality On the System Display

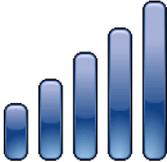
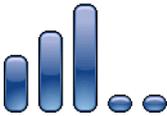
When the IX software detects changes in network quality in the network, an icon is displayed on the main display screen. When connection quality reaches the poor state, the call is terminated.

Table 1-11 describes the call connection status icons that display on the main display.

**Note**

Five, four, and three bars are reserved to show the video rates (1080p, 720p, or CIF) for the call. The remaining bars are reserved to show packet loss. A call dropping to one or two bars indicates that your network is having excessive packet loss at the rates shown in [Table 1-11](#).

Table 1-11 Call Connection Network Status Bars

Status Bars	Description
Five Bars 	<ul style="list-style-type: none"> All received streams are 1080p with no packet loss above the 1% warning threshold, and The received presentation (if active) has no packet loss above 2%.
Four Bars 	<ul style="list-style-type: none"> The lowest Resolution of received HD Streams is 720p with a packet loss less than or equal to 1%, and The received presentation (if active) has a packet loss less than or equal to 2%. <p>Note In cases where a 1080P call at 30 fps is made with Cisco TelePresence Server (TS), the system shows only four bars even though there is no packet loss.</p>
Three Bars 	<ul style="list-style-type: none"> The lowest Resolution of received HD Streams is Common Intermediate Format (CIF) with a packet loss less than or equal to 1%, and The received presentation (if active) has a packet loss less than or equal to 2%.
Two Bars 	<ul style="list-style-type: none"> The highest percentage packet loss is above the 1% warning threshold, but less than 10%, or The received presentation, if active, has a loss between 2% and 10%.
One Bar 	<ul style="list-style-type: none"> The highest percentage packet loss of all received HD streams is more than 10%, or The received presentation has loss above 10% if active

High-Definition Presentation

This section provides you with information about the supported presentation resolutions and presentation audio and video cables and includes the following sections:

- [HD Presentation Overview, page 1-14](#)
- [Supported Presentation Devices and Tested Adapters, page 1-14](#)
- [Resolution Support, page 1-14](#)
- [Required Configuration For HD Presentation, page 1-15](#)
- [Scaling HD Presentation Video Resolution, page 1-16](#)
- [Bandwidth Requirements for the HD Presentation Feature, page 1-16](#)

HD Presentation Overview

Systems running IX software support high-definition (HD) presentations up to a maximum of 1080p resolution at 30 frames per second (1080p 30). The Cisco TelePresence IX5000 and IX5200 systems support the HD presentation feature.

Supported Presentation Devices and Tested Adapters

The presentation cable for the IX5000 system includes the following digital connectors: DisplayPort, Mini-DisplayPort, and HDMI.

Various third-party presentation devices are supported by connectors on the IX system's presentation cable. Supported presentation devices include:

- Laptop (PC) using the HDMI connector
- MacBook Air and MacBook Pro using the following connectors:
 - HDMI connector
 - DisplayPort connector
 - Mini DisplayPort connector



Note Some Apple devices show flickering on the screen when being used for presentation sharing. If you encounter this issue, change the resolution quality to a lower resolution.

- iPad products using a proprietary HDMI adapter available from Apple

If an adapter is required for VGA presentations, Cisco recommends the following tested adapter:

- SYBA USA VGA to HDMI converter

Resolution Support

The proportional relationship between the width and height (also known as *aspect ratio*) of the HD presentation signal is 16:9, whereas an analog presentation signal is 4:3.

[Table 1-12](#) outlines the common resolutions shared by the presentation devices, the corresponding aspect ratios, and the presentation digital cables that support the resolutions. While the IX system supports all of the presentation resolutions indicated in the table, EDID (Extended Display Identification Data) only supports 1080p, 720p, and VGA.

Table 1-12 Aspect Ratios For Resolutions Shared By Presentation Device

Resolutions Shared By Presentation Device	Aspect Ratio
1920x1080 (1080p) @ 30, 15, or 5 fps	16:9
1280x720 (720p) @ 30, 15, or 5 fps	16:9
1024x768 (XGA) @ 30, 15, 5, or 1 fps	4:3
640x480 (VGA) @ 30, 15, 5, or 1 fps	4:3

**Note**

Some presentation resolutions do not take up the full space of the presentation display area. For example, if a laptop that is sending the presentation is set to a resolution of 1600x900, and the presentation display is set to a resolution of 1920x1080, the image shown on the display is 1600x900 pixels with a black border around it to make a total pixel size of 1920x1080.

To eliminate this black border around the presentation display when using a PC, open the PC's Control Panel, navigate to the laptop's Change Desktop Background, and change the picture position to "Fill."

Required Configuration For HD Presentation

If both Binary Floor Control Protocol (BFCP) and TelePresence Interoperability Protocol (TIP) are negotiated for a call, TIP takes precedence, and BFCP is not used to control presentation. BFCP is the preferred protocol for controlling the presentation for systems that run IX software.

[Table 1-13](#) describes the Unified CM configuration that enables HD presentation to function on IX endpoints. No additional configuration is required.

Table 1-13 Unified CM Configuration Required For Each IX Endpoint

Unified CM Configuration	Notes
For each IX endpoint, use the Standard SIP Profile (not the Standard SIP BFCP Profile) for the SIP profile, even if you use BFCP.	If you specify the Standard SIP BFCP profile, calls might be dropped and BFCP might not function properly.
Make sure that you check the Allow Presentation Sharing Using BFCP check box in the Protocol Specific Information area of the Phone Configuration window.	For more information about configuring BFCP, refer to the "Configuring the BFCP over UDP Collaboration Feature" section of the <i>Configuring Cisco Unified Communications Manager for the IX System</i> document.
In the Phone configuration, the default value of Video Bandwidth Allocation Weights is 8 Main/2 Presentation. Adjust this setting if desired.	For more information about this parameter, see the "Video Bandwidth Allocation Weights" section on page 1-17.

For more details about the Unified CM configuration, see the [Configuring Cisco Unified Communications Manager for the IX System](#) document.

Scaling HD Presentation Video Resolution

For HD presentation, the system running IX software that is sharing content can automatically change resolutions for the content being shared, such as a slide presentation, document, or video. [Table 1-14](#) outlines common presentation device display resolutions, and the higher or lower resolutions to which content can be scaled and sent by the IX endpoint.

In general, HD presentation supports the following scaling schemes:

- Scaling to a higher resolution is not supported except for 1280x720 (720p), which can be scaled to XGA.
- Scaling to lower resolutions of XGA is supported.

Scaling is done to accommodate lower network bandwidth. Before the system running IX software scales the presentation, it lowers the frame rate of the presentation, which preserves the video clarity.

Table 1-14 Common Device Display Resolutions and Scaled Resolutions

Resolutions Shared By Presentation Device	Scaled Resolutions			
	1920x1080 (1080p)	1280x720 (720p)	1024x768 (XGA)	640x480 (VGA)
1920x1080 (1080p)	Yes	No	Yes	Yes
1280x720 (720p)	No	Yes	Yes	Yes
1024x768 (XGA)	No	No	Yes	Yes
640x480 (VGA)	No	No	No	Yes

Bandwidth Requirements for the HD Presentation Feature

The resolutions and frame rates at which each type of video can be sent is based on the available bandwidth.

[Table 1-8 on page 1-11](#) and [Table 1-9 on page 1-12](#) display the bandwidths required for supported 60 fps main video send resolutions and frame rates, while [Table 1-15](#) displays the bandwidth required for supported HD presentation send resolutions and frame rates.



Note

Regardless of how much network bandwidth is actually used as described in [Table 1-15](#), the IX system always negotiates the maximum amount of bandwidth for an HD presentation (4000Kbps).

Table 1-15 Bandwidth Required For Supported HD Presentation Send Resolutions and Frame Rates

Send Resolution	30 fps (Kbps)	15 fps (Kbps)	5 fps (Kbps)	5 fps (minimum) ¹ (Kbps)
1920x1080 (1080p)	4000	2500	1000	500
1680x1050	3700	2200	900	450

Table 1-15 *Bandwidth Required For Supported HD Presentation Send Resolutions and Frame Rates*

Send Resolution	30 fps (Kbps)	15 fps (Kbps)	5 fps (Kbps)	5 fps (minimum) ¹ (Kbps)
1440x900	2900	1750	725	350
1280x800	2450	1450	600	300
1280x720 (720p)	2250	1350	550	250
1024x768 (XGA)	2000	1200	500	250
800x600	1400	825	350	175
640x480 (VGA)	1000	600	250	100

1. These bandwidth rates are the lowest possible rates required. The higher bandwidths are the recommended minimum bandwidth for best performance.

For calls that support more than one presentation stream, the required bandwidth is multiplied for each additional stream. When two presentation streams are used in a P2P call between IX endpoints, the required bandwidth is doubled. For example, two presentation streams at 1080p60 would require 8 Mbps.

For more information about calculating bandwidth for video and presentation content, see the [“Sample Bandwidth Calculations”](#) section on page 1-18,



Tip

If, in a limited bandwidth scenario, you want to send a presentation with a higher fps but a lower resolution, you can change the resolution of the presentation at the source of the presentation. For example, given a maximum rate of 1000 Kbps, if you are sending a 1920x1080 presentation at 5 fps, you can instead send 640x480 at 30 fps by changing the resolution of your presentation to 640x480.

Multiple Presentation Streams

On an IX-to-IX point-to-point conference, you can share up to two presentation streams - for example, one presentation content source and one whiteboard source. If hosting a meeting locally with no video conferencing, you can share up to three presentation streams.

Video Bandwidth Allocation Weights

The Video Bandwidth Allocation Weights parameter allows you to balance the bandwidth ratio for main video and presentation video during a conference.

Use this feature when the amount of session bandwidth that is used by a Cisco TelePresence endpoint to send audio, main video, and presentation video media streams exceeds the amount of available session bandwidth.

You add this value in the Bandwidth Allocation Weights field in the Product Specific Configuration Layout Area of the Unified CM administration console.

The weight is based on a total number of 10m and the default value of this parameter is a weight of 8 for main video and a weight of 2 for presentation video (8 main / 2 presentation).

The Cisco TelePresence System IX5000 supports the bandwidth allocation feature.

The following values are supported for this feature. The first value is the weight for main video, and the second value is the weight for the presentation video.

9 main / 1 presentation

8 main / 2 presentation (**default**)

6 main / 4 presentation

4 main / 6 presentation

3 main / 7 presentation



Note

If you see bandwidth allocation weights used in the System Operations Log, check the Unified CM region settings and ensure enough video bandwidth is allowed for video and immersive video calls. The Unified CM region settings frequently can trigger the bandwidth allocation weights, which can significantly reduce call quality.

Sample Bandwidth Calculations

Bandwidth for a full 1080p60 “best” quality call with a 1080p30 presentation would require the following bandwidth:

H.265: 8.1 Mbps main video + 4 Mbps (presentation video@1080p 30) = 12.1 Mbps

H.264: 13.5 Mbps main video + 4 Mbps (presentation video@1080p 30) = 17.5 Mbps



Note

The main video rates are derived from the rate per display multiplied by the number of screens (three). For H.265, given the rates in [Table 1-8](#), the rate is 2.7 Mbps per screen for 1080p60 (Good), and for H.264, given the rates in [Table 1-9](#), the rate is 4.5 Mbps per screen for 1080p60 (Good). Multiplying these numbers gives you the video rates of 8.1 and 13.5 Mbps, respectively.

Bandwidth Calculations

To fit the available bandwidth, the endpoint performs calculations based on these general formulas, which include values from the Video Bandwidth Allocation Weights parameter:

Definitions:

Session Video Bandwidth (**SVB**) = Total session bandwidth - Audio bandwidth

Main Video Weight (**Mwt**) = Configured weight for main video stream

Total Weight for main video (**T_Mwt**) = Number of Streams x Main Video weight

Presentation Video Weight (**Pwt**) = Configured weight for presentation video stream

Total Weight for presentation video (**T_Pwt**) = Number of streams x Presentation video weight

Total Weight (**TW**) = T_Mwt + T_Pwt

Formula to allocate main video bandwidth:

SVB x (T_Mwt/ TW) = Mwt

Formula to allocate presentation video bandwidth:

SVB x (T_Pwt/ TW) = Pwt

Rate Calculation Example

Given a total available network bandwidth of 6.0 megabits per second (Mbps), and a single presentation stream:

$$\mathbf{T_Mwt} = (3 \text{ (number of main video streams)} \times 8) (\mathbf{Mwt}) = \mathbf{24}$$

$$\mathbf{T_Pwt} = (1 \text{ (number of presentation video streams)} \times 2) (\mathbf{Pwt}) = \mathbf{2}$$

$$\mathbf{TW} = \mathbf{26}$$

Allocated main video bandwidth = $6 \times (24/26) = \mathbf{5.53 \text{ Mbps (1.84 Mbps per display)}}$

Allocated presentation video bandwidth = $6 \times (2/26) = \mathbf{0.47 \text{ Mbps}}$

For H.265, the 1.84 Mbps per display is sufficient to support either 720p60 (better) or 1080p30 (good), per the rates in [Table 1-8](#), depending on whether the initial call was negotiated at 60 or 30 fps.

For H.264, the 1.84 Mbps per display is sufficient to support 720p30 (better), per the rates in [Table 1-9](#).

Both H.265 and H.264 encoding support a resolution stream of 720p @ 5 fps if receiving, and 640x480 @ 5 fps if sending, per the rates in [Table 1-15](#).

**Note**

This example assumed a single presentation stream. Additional content streams change the calculation in that the available bandwidth stream is divided by 2 for each content.
