Fixing Poor Quality for Room Systems

Recommendations for Solving Quality Issues for Room Systems

If you are having quality issues with a room system such as pixilation, pulsing video, low bitrate or choppy audio. Here are some steps you can take to troubleshoot and resolve these issues.

1) Bandwidth - make sure you have enough bandwidth available to allow for at least HD 720p. Most room systems can achieve 720p at 768 (some even lower if no loss is present). Recommend using a call speed between 1024 and 1600 when dialing BlueJeans. So at a minimum you will need at least 1M in bandwidth in each direction at a minimum to achieve 720p as a general guideline.

Please note that common speed tests are generally not all that useful for a "real-time" application like video conferencing as these are usually just "ping" tests that do not test sustained bandwidth needs.

2) Set speed/duplex settings to 100/full for the endpoint LAN (via the web interface) and the network switch port LAN. "Auto negotiation" unfortunately does not always work as you would expect. With the deployment of 1000/full switches being more common, many video units only do
100/full and leaving them to AUTO/AUTO results many times in speed/duplex mismatches. Speed and duplex mismatches can cause poor quality due to packet loss.

Check the path for the Ethernet connection. Make sure that a switch in the path is not causing issues that result in speed/duplex mismatches or collisions. Look at the all switch port interfaces, and see if there are collision errors.

2) Look for bandwidth spikes in the network (which could be due to competing network traffic from other endpoints or activities causing congestion). You may be able to set up a QoS policy inside your network that can give the video conferencing traffic a high priority.

3) Make sure your firewall ports are allowing the IP/Port range for BlueJeans.

Connections made to BlueJeans cloud server, uses the following TCP and UDP ports.

- **199.48.152.0/22**
- **31.171.208.0/21**
- **103.20.59.0/24**
- **103.255.54.0/24**
- **8.10.12.0/24**
- **165.254.117.0/24**

**Note:** Blue Jeans has several POPs distributed globally. The call will be automatically redirected to the closest/native POP to the endpoint or media egress point. Audio/video traffic will be routed to any of above ip range, based on geo location. Hence it's important that firewall ports are allowed against all the ip ranges shown above. Below are the ports that BlueJeans will use:

**H.323 based Room System:**
TCP Port 1720 - H.225 Signaling for H.323
TCP Ports 5000-5999 - H.245 Call Control for H.323
UDP Ports 5000-5999 - RTP Media

**SIP based Room System (supported over TLS):**
TCP Port 5060 - SIP Signaling
TCP Port 5061 - SIPS (TLS) Signaling
UDP Ports 5000-5999 - RTP Media

Ports on your endpoint side can be set by the room system or your firewall.

**Please Note:** Some firewalls, such as Palo Alto Networks, prefer to filter network traffic based on the Fully Qualified Domain Name (FQDN). If this applies to your firewall configuration please use the following FQDN in order to connect to Blue Jeans: bjn.vc
It is NOT recommended for your firewall to use ALG (Application-level Gateway) for videoconferencing. This can cause dialing and audio loss as well as other issues.

4) Check to make sure there is no filtering or deep packet inspection. Solutions like Riverbed, BlueCoat, WebSense, etc can sometimes cause excessive packet loss if no exception has been added to bypass the video conferencing traffic.

5) Try putting your room system on a separate VLAN or testing outside the firewall if quality issues still persist.

6) Traceroute or MTR tests can be helpful to identify loss or latency over the Internet.

7) Updating the room system software to most recent version whenever possible is usually a good idea.
Room System Signaling to BlueJeans

<table>
<thead>
<tr>
<th>Ports</th>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1720</td>
<td>TCP</td>
<td>H.323 Call Setup H.225</td>
</tr>
<tr>
<td>5060 / 5061</td>
<td>TCP</td>
<td>SIP Call Signalling and TLS</td>
</tr>
<tr>
<td>5000-5999</td>
<td>Dynamic TCP</td>
<td>H.323 Call Control H.245</td>
</tr>
<tr>
<td>5000-5999</td>
<td>Dynamic UDP</td>
<td>RTP Media</td>
</tr>
</tbody>
</table>
Call Initiated to BlueJeans from Room System with Properly Configured NAT

Public

NAT

Network Address Translation

Public IP

Private IP

Private

BlueJeans sees the public IP
as NAT changes IP headers

108.6.25.xx

Internal (non-routable)
IP address

10.10.10.xx

Static one-to-one NAT

H.323/SIP
Room System
has private IP
on LAN

Ports used on user side can be
configured on Firewall or Endpoint

Call Initiated to BlueJeans

Call from Room System NAT to BlueJeans
Properly NAT (Network Address Translation) configured Room System calling BlueJeans

**Call Flow:**
1) Room System IP address 10.10.10.xx (private) dials BlueJeans using 199.48.152.152 - Figure 1
2) Ports used are configured by Room System or customer Firewall (depending on how NAT is configured)
3) Network Address Translation mechanism changes IP headers to the "public" address configured
4) BlueJeans receives call that advertises the "public" address 108.6.25.xx on ports from customer side - Figure 2
5) BlueJeans will send back to the "public" address 108.6.25.xx and ports advertised from customer side FROM BlueJeans IP/Port range

NOTE: If "private" non-routable IP address is advertised due to mis-configured NAT the call could fail, cause one-way audio or video or not allow for proper dual stream content sharing. Properly configured NAT has the Room System has an assigned to a static "private" IP on internal LAN. One-to-one static NAT (usually done at Firewall) configures a "public" IP address to the "private"
address. Ports used on customer side can be configured at Room System or at Firewall. BlueJeans IP/Port range must be allowed. H.323 Inspect can usually be disabled on most firewalls. If properly configured NAT, the Room System can have its NAT setting (located in admin settings on endpoint) be set to NAT Off or Auto in most cases.

### Room System NAT Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Firewall</strong></td>
<td></td>
</tr>
<tr>
<td>Fixed Ports</td>
<td></td>
</tr>
<tr>
<td>TCP Ports</td>
<td>3230 to 3235</td>
</tr>
<tr>
<td>UDP Ports</td>
<td>3230 to 3280</td>
</tr>
<tr>
<td>Enable H.460 Firewall Traversal</td>
<td></td>
</tr>
<tr>
<td>NAT Configuration</td>
<td>Manual</td>
</tr>
<tr>
<td>NAT Public (WAN) Address</td>
<td>108.5.123.210</td>
</tr>
<tr>
<td>NAT is H.323 Compatible</td>
<td></td>
</tr>
<tr>
<td>Address Displayed in Global Directory</td>
<td>Public</td>
</tr>
<tr>
<td>Enable SIP Keep-Alive Messages</td>
<td></td>
</tr>
</tbody>
</table>

Enabling "Fixed" Ports on the room system will generally use the ports set on the room system to advertise to remote endpoint.
Locating the NAT settings on Room Systems

1) Polycom NAT settings (above)
   - Admin settings -> Network -> IP Network -> Firewall settings

2) Cisco NAT settings - (below)
   - System settings -> H.323 -> NAT
Troubleshooting Quality and Low Resolution Issues

Quality of Service

Type of Service:
- IP Precedence

Type of Service Value:
- Video: 4
- Audio: 5
- Control: 3

Maximum Transmission Unit Size:
- Default: 1260 bytes

Enable PVEC:
- Yes

Enable RSVP:
- Yes

Dynamic Bandwidth:
- Off

Maximum Transmit Bandwidth:
- 6144 Kbps

Maximum Receive Bandwidth:
- 6144 Kbps

Turn off Dynamic Bandwidth on Polycom endpoint
This setting will reduce resolution due to packet loss on the network. Sometimes this feature can be too sensitive or aggressive. Disabling this can sometimes help with where the video resolution is staying low - (above)

Call Speed Too Low or Too High

May have to enable call speeds on Polycom endpoints if the desired call speed does not show up in the preferred drop down.

Access Call Speeds on Polycom via Admin settings -> Network -> Call Speeds - (above)
Access Call Speeds on Polycom via Admin settings -> Network -> Call Preference and set the preferred call speed. Recommend 1024 to 1600 to achieve 720p to BlueJeans. Too high may tax the customers bandwidth and raise network issues. To low may affect quality and resolution below the expectations.
Call speed or call rate can be defaulted on Room Systems so the user does not have to specify on each call - (above)
Polycom Basic or Diagnostic Mode

Polycom Room Systems - make sure that your room system does not have Basic or Diagnostic mode enabled. This will have an impact on the video and audio quality as it will only use H.261 video codec and G.711 audio codec. The quality will suffer and it can affect content share displaying - (above)

To turn off this configuration, log into your Polycom web admin interface, and go to:

- Admin Settings
- Network
- Call Settings, and
- Uncheck "enable Basic Mode"

On HDX systems running firmware version 3.0.3 or higher this option is now called "Diagnostic Mode" and can be found in the following menu:

- Admin Settings
- Network
- Call Preference, and
- Uncheck "Diagnostic Mode"

You may need to reboot the system for the new codecs to take effect. Once you reconnect to the meeting your endpoint should begin using the G.722 (audio) and H.264 (video) codecs.

Camera Settings

For maximum resolution the camera settings should be set for "sharpness" as opposed to "motion"
- (above and below)

Change the Camera 1 Video Quality setting to Sharpness and re-test. This can be found on Polycom via Admin Settings > Cameras > Camera 1 > Video Quality = set to Sharpness.

After making the change, try to connect via H.323 and confirm if Transmit video resolution has improved. SIP capabilities could then be confirmed based on the software version in-use, you may see different operational results.
Things that affect quality and resolution:

1) All switch ports should be set for 100/full. Many times an auto/auto does always allow for the desires full duplex and speed due a bug in some older Cisco switches.
2) Network Packet Loss - Results above 1% can lead to poor video quality
3) High Jitter - Results above 100 ms can lead to poor video quality
4) Available bandwidth or limited bandwidth - minimum of 384 Kbps, for both upload and download, is needed to support your video call. For 720p HD resolution the user should have at least 1024Kbps (1Mbps) up/down bandwidth
5) Too low MTU - Lower that 1420 can affect video conferencing
6) Deep Packet Inspection or Filtering for solutions like Websense, BlueCoat, Riverbed, etc.
7) High Round Trip Delay and Latency - Results above 150 ms can lead to poor video quality

Echo Issues

Most Room System use hardware and software echo cancellation. Echo is usually caused by sound coming from a speaker going into a microphone creating a loop. Most times the participant causing the echo does not hear it themselves.
Common cause of echo issues:
1) Customer using a TV/ Monitor connected to Room System for speakers or worse yet a soundbar. These devices have latency issues when using DSPs that can have adverse effects to echo cancellation. Using Surround modes is a bad idea.
   • Try "Game Mode" on TV if it has one as this will reduce latency.
   • They may be better off switching to powered speakers (analog) for audio instead of TV audio.
2) Customer has ceiling mounted speakers and microphones that do not have proper echo cancellation
   • They may have to add or enable echo cancellation or "tune" the room
3) Mic is just too close to a speaker that is too loud that the echo cancellation can not handle this situation.
4) Very loud hollow echoey room. Can be caused by glass, hard surfaces and high ceilings that are not treated for sound.
   • Even a good echo cancellation solution has its limits. May have to "treat" room or move mics, speakers and adjust levels to improve.

Questions?

Contact BlueJeans support via
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