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Cisco Unified MeetingPlace H.323/SIP IP Gateway Requirements

- One primary IP address for communication with the Cisco Unified MeetingPlace Audio Server.

Hardware Requirements

One of the following Cisco Media Convergence Server (MCS) 7800 series servers with the Cisco Systems version of the Microsoft Windows 2003 server operating system installed:

- Cisco MCS 7835
- Cisco MCS 7845

The H.323/SIP IP Gateway can be installed on a server with other Cisco Unified MeetingPlace components. However, we recommend that you install the H.323/SIP IP Gateway on a dedicated server for installations larger than 96 ports.

Cisco Unified MeetingPlace Audio Server Requirements

- Cisco Unified MeetingPlace 8100 series server with a compatible version of Cisco Unified MeetingPlace Audio Server. See the Compatibility Matrix.
- You must configure the Multi Access (MA) Blade for IP.
- There must be one primary IP address for communication with the H.323/SIP IP Gateway server and other gateways and clients.
- You must assign one additional IP address for every 240 IP user licenses.

Cisco Unified CallManager Requirements

Cisco Unified CallManager (also known as Cisco Unified Communications Manager) Release 3.22 through 6 must be installed and configured to work with your network. For Cisco Unified MeetingPlace Release 6.0 MR5 only, we also support Cisco Unified Communications Manager Release 7.1.2.

Endpoint Requirements

- Cisco Unified IP Phone configured to work with one of the following applications:
    - Cisco Unified CallManager Release 3.22 or later. For Cisco Unified MeetingPlace Release 6.0 MR5 only, we also support Cisco Unified Communications Manager Release 7.1.2.
    - Cisco SIP Proxy Server Release 1.0.
- H.323 endpoints.
- PSTN phone connected to a configured voice gateway.

Network Requirements

For the H.323/SIP IP Gateway server to communicate with the Audio Server, confirm that your network meets the following requirements:

- The Audio Server and the H.323/SIP IP Gateway server must be located on the same subnet.
- The H.323/SIP IP Gateway server must be inside the corporate firewall and must be able to access the softswitch (for example, Cisco Unified CallManager, H.323 gatekeeper, or Cisco SIP Proxy Server) by using H.323 or SIP.
- The H.323/SIP IP Gateway server must be able to open a TCP connection on port 5003 when connecting to an Audio Server.
- The secondary Ethernet connections-those that are connected to the Multi Access Blade and not the second Ethernet port on the CPU card-on the Audio Server must be connected to the same subnet as the corporate network to which your Cisco Unified IP Phone is connected.
- The Ethernet switch port or any other network devices to which the Multi Access Blade connects directly must be set to fixed 100Base-TX full duplex; if not, you may experience decreased voice quality.
- The Audio Server must be connected to a network switch port that is configured for autonegotiate for duplex and speed.
- If G.711 codec is configured, a dedicated 100-Mbps subnet is required for each MA-16 connection; if not, the voice quality may be degraded.
- From the H.323/SIP IP Gateway server standpoint, the IP address of the Audio Server must not be translated by using a Network Address Translation (NAT) routing scheme.
- Connectivity between the H.323/SIP IP Gateway server and the Audio Server must be high quality and not subject to interruptions because of traffic congestion. Any time the round-trip latency exceeds 100 milliseconds or there is more than 1-percent packet loss, expect a noticeable reduction in service quality.