Complete the following worksheets:

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Installation Worksheet

Before you install Cisco Unified MeetingPlace H.323/SIP IP Gateway, complete the following worksheet.

You need to supply these values when you install and configure Cisco Unified MeetingPlace H.323/SIP IP Gateway.

Description	Value	Enter Your Value Here
Hostname or IP address of the IP-gateway server.	host name IP address	____________ ____________
Number of the IP-gateway server.	dialable number	____________
Hostname of the Cisco Unified MeetingPlace Audio Server system.	hostname IP address	____________ ____________
Additional IP addresses of the Cisco Unified MeetingPlace Audio Server system. Up to four additional IP addresses are needed for the Multi Access blade. If a TP1610 Multi Access blade is in use but only 240 VoIP or fewer are deployed, then you must specify the lower address; the upper address can be set to 0.0.0.0. You must also set the Ethernet switch port or any other network devices to which the Multi Access blade connects directly to fixed 100Base-TX Full Duplex.	hostname IP address hostname IP address	____________ ____________ ____________ ____________

Note: Do not set the lower address to 0.0.0.0.	hostname IP address hostname IP address	______________ ______________ ______________ ______________
Hostname or IP address of one of the following: • Cisco Unified Communications Manager server or IP PBX that runs standard H.323 or SIP call control • Cisco SIP Proxy Server	hostname IP address	______________ ______________
Host name or IP address of the Cisco Unified MeetingPlace Web Conferencing server if running on the same server as Cisco Unified MeetingPlace H.323/SIP IP Gateway. **Note:** If you use a hostname, DNS must be enabled to resolve the hostname to an IP address.	hostname IP address	______________ ______________

Dial Plan Worksheet

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About the Dial Plan for the Cisco Unified MeetingPlace H.323/SIP IP Gateway

A dial plan ensures that IP and PSTN calls to and from the Cisco Unified MeetingPlace Audio Server system are directed to the proper endpoints on their respective network. Each type of call has a dial pattern that specifies its call flow to and from the MeetingPlace Audio Server system.

For example, if your Cisco Unified MeetingPlace Audio Server system has both IP and PSTN interfaces, you may want to configure their outdial patterns so that outdials to a PSTN phone will go through the Cisco Unified MeetingPlace Audio Server system PSTN interface. This ensures an outdial to a PSTN phone does not go through the IP network first and then to the PSTN.

For Cisco Unified MeetingPlace Audio Servers systems that have both PSTN and IP interfaces, a dial plan should account for rollover from PSTN to IP ports and vice versa. For example, if you have a Cisco Unified MeetingPlace Audio Server system with 96 IP user licenses and 192 PSTN user licenses, the 97th caller to IP is automatically forwarded to a PSTN port by Cisco Unified Communications Manager through a voice gateway, rather than producing a fast busy signal.

For additional information about mixed-mode configuration, see About Telephony Configurations for Mixed Cisco Unified MeetingPlace Systems and Mixed Cisco Unified MeetingPlace System Configuration Examples.

Dial Plan Worksheet

Use the following worksheet to create a a dial plan

MeetingPlace IP call flow	Value
From an IP phone to the IP-gateway server. If the IP-gateway server is busy, Cisco Unified Communications Manager can forward calls to Cisco Unified MeetingPlace system PSTN through a voice gateway. You must configure Cisco Unified Communications Manager and the voice gateway to route this type of call.	dial pattern____________________ A 4-digit number that does not conflict with a corporate phone extension number scheme.
From a PSTN phone to Cisco Unified MeetingPlace system PSTN. If Cisco Unified MeetingPlace system PSTN is busy, the PBX or CO can forward calls to the IP-gateway server through Cisco Unified Communications Manager. You must configure the PBX or CO to route this type of call.	dial pattern____________________ A 7- or 10-digit phone number that does not conflict with a corporate phone numbering scheme.
From Cisco Unified MeetingPlace system IP to an IP phone.	dial pattern____________________ Typically, the last four digits of the phone number.
From Cisco Unified MeetingPlace system PSTN to a PSTN phone.	dial pattern____________________ Typically **9**, if needed for an outside line, followed by either the 7- or 10-digit phone number.

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