Main page: Cisco Unified MeetingPlace, Release 7.0

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To find this page, click **System Configuration > Call Configuration > SIP Configuration**.

Table: Field Reference: SIP Configuration Page

Field	Description
Display name	Name that appears on Cisco Unified IP Phone screens during calls placed by Cisco Unified MeetingPlace. Default: Cisco Unified MeetingPlace
Username	Enter the same value that is in the Access phone number 1 field on the Usage Configuration Page. The system uses this value in the "From" address for outgoing calls. Default: 0000
Only accept incoming calls from proxies	Whether to accept incoming calls from only the SIP proxy servers that are configured on the SIP Configuration Page. If you select Yes, then make sure that the configured SIP proxy servers include all Cisco Unified Communications Manager subscriber nodes that might send calls directly to this Cisco Unified MeetingPlace server. Default: No
SIP domain name	Domain name that is used in the "To" address for outgoing calls. If left blank, then the Application Server IP address is used instead. Typically, this field is left blank when the SIP Proxy Server fields are populated with Cisco Unified Communications Manager information. **Caution!** If you enter a value that does not match the domain used by the SIP proxy servers, then Cisco Unified MeetingPlace will fail to dial out. If outgoing calls fail, then configure this field to match the SIP domain used by the SIP Proxy Server or your local Cisco Unified Communications Manager node. In Cisco

	Unified Communications Manager, the SIP domain is specified under System > Enterprise Parameters in the Organization Top Level Domain field. **Note:** For details about any field in Cisco Unified Communications Manager, click the field name, or click **Help > This Page**. Example: cisco.com
SIP Proxy Server	Outgoing SIP calls are directed to these proxies. These proxies are also used to filter incoming calls if the Only accept incoming calls from proxies field is set to Yes. **Note:** At least one SIP proxy server must be configured to enable dial-out calls. If multiple proxy servers are configured when placing a call, Cisco Unified MeetingPlace tries each one successively until it finds one that accepts the call. Note, however, that if a proxy does not respond at all (as opposed to rejecting the invite), the call attempt will probably time out before the next proxy is tried.
Hostname or IP address	Hostname or IP address of the SIP proxy server. In a Cisco Unified Communications Manager environment, the SIP proxy server should be the IP address of a Cisco Unified Communications Manager node. Enter the FQDN (fully-qualified domain name) (or IP address) if the domain of the SIP proxy server differs from the domain that you configured in DNS when you installed Cisco Unified MeetingPlace.
Port	TCP or UDP port on which the proxy listens for SIP calls. Default: 5060
Transport	Whether the port is TCP or UDP. Required: Use UDP for the SIP signaling between Cisco Unified MeetingPlace and Cisco Unified Communications Manager on all Hardware Media Server integrations. Default: UDP

Related Topics

- Configuring Call Control for Cisco Unified MeetingPlace