Main page: Cisco Unified MeetingPlace Express, Release 2.x

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This page is used to connect Cisco Unified MeetingPlace Express to a supported call-control device through a SIP trunk.

Table: Fields on the SIP Configuration Page

Field	Description	Value
SIP enabled	Whether SIP is enabled. If this field is set to **No**, incoming SIP calls cannot be received. To use SIP for outgoing calls, see the Field Reference: Dial Configuration.	Yes/No Default: Yes
Display name	The name that appears on the Cisco Unified IP Phone screen when a user calls Cisco Unified MeetingPlace Express from a Cisco Unified IP Phone.	Up to 64 characters Default: Cisco Unified MeetingPlace Express
Username	The phone number of the Cisco Unified MeetingPlace Express server. This number should match the Access phone number 1 field. See the Field Reference: Usage Configuration. If Cisco Unified MeetingPlace Express dials out to a Cisco Unified IP Phone, this number appears on the Cisco Unified IP Phone screen.	Up to 64 characters Default: 0000
Local SIP port	UDP port used for incoming SIP calls to Cisco Unified MeetingPlace Express. Restriction: This number must match the port number configured on the call-control device. See the Configuring Call-Control Integration for Cisco Unified MeetingPlace Express with Cisco Unified Communications Manager. The following port settings are automatically configured on Cisco Unified MeetingPlace Express and cannot be modified: • Static UDP port 5060 is used for call setup of outgoing SIP calls from Cisco Unified MeetingPlace Express. • Random UDP ports in the range 5000 to 65535 are used for Real-Time Transport Protocol (RTP) voice streams.	Range: 0 to 65535 Default: 5060

SIP proxy server 1	IP address of the SIP proxy server. The Cisco Unified MeetingPlace Express system directs dial-out calls to this IP address. SIP proxy servers receive SIP messages and forward them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.	Range: 0 to 255 for each field Default: 0
SIP proxy server 2 SIP proxy server 3 SIP proxy server 4 SIP proxy server 5 SIP proxy server 6	IP address of an optional failover SIP proxy server.	Range: 0 to 255 Default: 0