



Cisco Unified Communications 500 Series

DISCLAIMER

The attached document is provided as a basic guideline for setup and configuration of Cisco Unified Communications 500 Series IP PBX systems with MegaPath's SIP Trunking service, based on MegaPath's testing and validation process. It does not include advanced configurations to enable features such as voicemail, Find-Me-Follow-Me, etc. MegaPath is not responsible for customer IP PBX configurations. For more information on advanced features, please refer to your IP PBX documentation.

IMPORTANT

DO NOT USE THE DEFAULT MEGAPATH PROFILE THAT SHIPS WITH THE CISCO CALL MANAGER EXPRESS. IT IS THE LEGACY MEGATH PROFILE WHICH NO LONGER FUNCTIONS PROPERLY. CREATE A GENERIC SERVICE PROVIDER PROFILE.

REQUIRED INFORMATION (Provided by MegaPath)

Host: _____

Maximum Number of Calls: _____

SIP username: _____

SIP password: _____



The following screen captures are included as a reference.

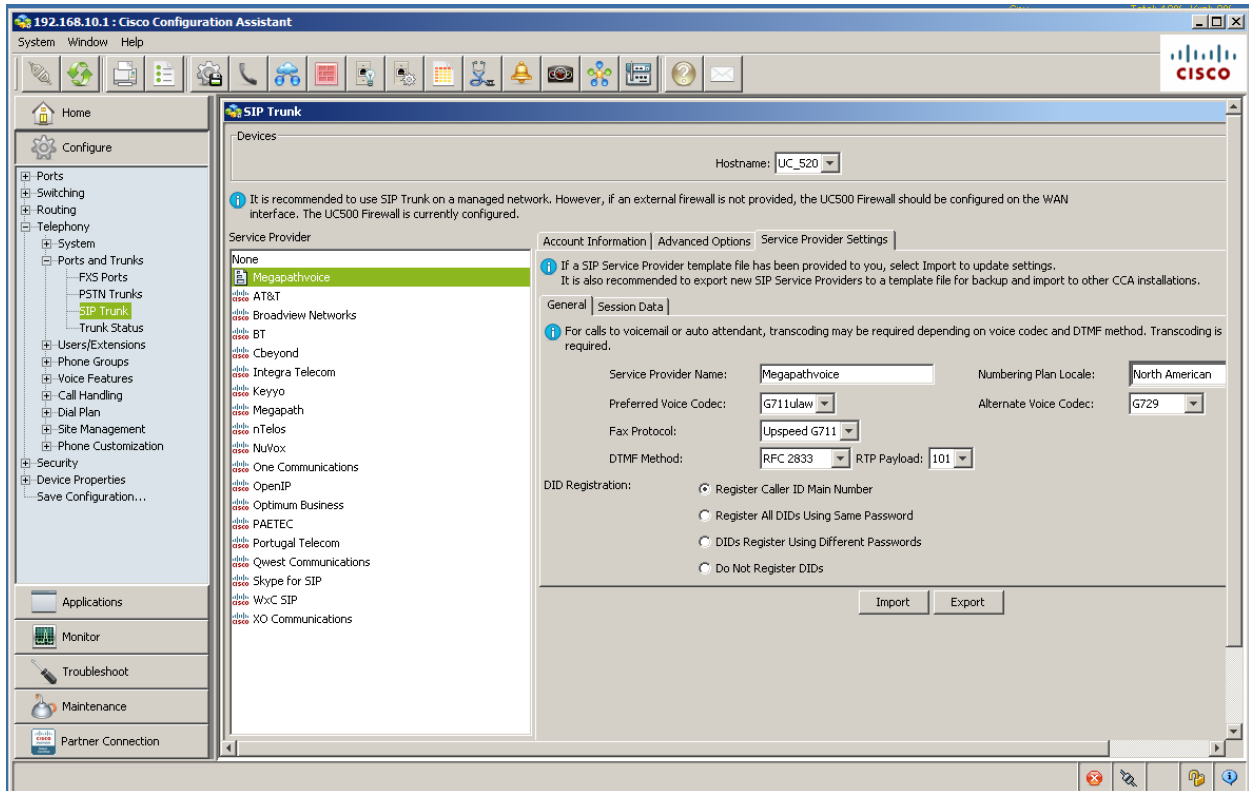


Figure 1: Configuration menu option from the main menu.

From the main configuration menu, select Telephony, Ports and Trunks, SIP Trunk. Create a generic Service Provider profile.

Account Information | **Advanced Options** | Service Provider Settings

Voice Codec:

Proxy Server (primary): Proxy Server (secondary):

Registrar Server:

Outbound Proxy Server:

Maximum Number of Calls: (1 - 40)

Digest Authentication

Username:

Password:

Display Password as Plain Text

Domain Name Service

SIP Domain Name:

DNS Server Address:

See also Routing > IP Addresses > Device Configuration.

User Credentials

Username	Password	Realm
4088906599	*****	192.168.1.254

Figure 2: SIP Trunk Account Information tab.

Under the SIP Trunk Account Information tab, enter or select the following parameters:

- Voice Codec: G711ulaw
- Proxy Server: (Primary): **<Host provided by MegaPath>**
- Proxy Server (Secondary): leave blank
- Register Server: **<Host provided by MegaPath>**
- Outbound Proxy Server: **<Host provided by MegaPath>**
- Maximum Number of Calls: **<Number of Trunks>**
- Username: **<provided by MegaPath>**
- Password: **<provided by MegaPath>**
- SIP Domain Name: leave blank
- User Credentials
 - Username: **<provided by MegaPath>**
 - Password: **<provided by MegaPath>**
 - Realm: **<Host provided by MegaPath>**

NOTE
THE SIP DOMAIN NAME FIELD SHOULD BE LEFT EMPTY. THE EDGEMARC SIP ALG PROVIDES THE SIP DOMAIN INFORMATION TO THE SBC ON BEHALF OF THE CISCO CALL MANAGER EXPRESS.

Account Information | **Advanced Options** | Service Provider Settings

Toll Fraud Protection

i Toll fraud protection prevents unwanted calls but requires the definition of all IP addresses allowed on the VoIP network. Internal networks and all servers defined under Account Information are already permitted. Additional IP addresses from your Service Provider may be entered below. If your Service Provider is unable to provide a complete list, this feature may be disabled.

Enable Toll Fraud Protection (recommended)

Additional Allowed IP Addresses

IP Address
67.103.60.53
67.102.144.83

Total Rows: 2

Timers and Retries

Registrar Server Expiry:	<input type="text" value="3600"/>	(60 - 65535 seconds)
Number of Register Retries:	<input type="text" value="10"/>	(1 - 10)
Number of Invite Retries:	<input type="text" value="2"/>	(1 - 10)
Connect Timer:	<input type="text" value="100"/>	(100 - 1000 milliseconds)
Proxy Server Keepalive Timer (active):	<input type="text" value="100"/>	(10 - 600 seconds)

Figure 3: SIP Trunk Advanced Options

Within the SIP Trunk window, click the "Advanced Options" tab. Click "Add" and input each of the following addresses:

Additional Allowed IP Addresses:

- **<Host provided by MegaPath>**

Account Information | **Advanced Options** | Service Provider Settings

i If a SIP Service Provider template file has been provided to you, select Import to update settings. It is also recommended to export new SIP Service Providers to a template file for backup and import to other CCA installations.

General | Session Data

i For calls to voicemail or auto attendant, transcoding may be required depending on voice codec and DTMF method. Transcoding is currently required.

Service Provider Name: Numbering Plan Locale:

Preferred Voice Codec: Alternate Voice Codec:

Fax Protocol:

DTMF Method: RTP Payload:

DID Registration:

- Register Caller ID Main Number
- Register All DIDs Using Same Password
- DIDs Register Using Different Passwords
- Do Not Register DIDs

Figure 3: Service Provider Settings

Under the Service Provider Settings, apply the following parameters:

- Service Provider Name: **MegapathVoice**
- Preferred Voice Codec: G711ulaw
- Fax Protocol: Up speed G711
- DTMF Method: RFC 2833
- RTP Payload: 101
- Numbering Plan Locale: North American
- Alternate Voice Codec: G729
- DID Registration: Register Caller ID Main Number

NOTE:
IT IS POSSIBLE TO OVER-RIDE THE MAIN NUMBER CALLER-ID BY SPECIFYING THE OUTBOUND CALLER-ID PER USER EXTENSION TO APPLY IN OVER-RIDING THE MAIN NUMBER CALLER-ID. THIS CAN BE ACCOMPLISHED UNDER USER EXTENSIONS.