

# Configuring Managed and Unmanaged Gateways

## Configuring Plug & Play Managed PSTN Gateways (FXO)

Configuring managed gateways using the sipX Configuration Server is similar to configuring plug & play managed phones. A new gateway is created by selecting the respective model from a drop-down list and then by specifying both its MAC address as well as its IP address. Gateways are stand-alone devices with their own IP address. sipX supports any SIP compliant gateways and they can be located where needed. The least-cost routing mechanism by default selects the best possible gateway. If a gateway is unavailable or all its lines are busy, sipX will automatically fall-back to the next best available gateway.

Name	Address	Description
Paris Gateway	10.0.1.211	Main PSTN Gateway
Rome Gateway	10.0.2.11	PSTN Gateway in Ro

Buttons: Send Profiles, Send All Profiles, Delete

Dropdown menu options:  
Add new gateway...  
Acme 1000  
AudioCodes MP1X4 4 Port FXO  
AudioCodes MP1X8 8 Port FXO  
AudioCodes TP260 1 Span T1/E1  
AudioCodes TP260 2 Span T1/E1  
AudioCodes TP260 4 Span T1/E1  
AudioCodes TP260 8 Span T1/E1  
Unmanaged gateway  
SIP trunk

The MAC address is required so that the Configuration Server can generate a profile. The file name of the generated profile is typically identified using the gateway's MAC address. When the gateway boots it is then able to load its profile typically using (T)FTP (this is gateway specific, so consult the respective HowTo document on this Wiki).

**Gateways do not register with sipXecs** - only end points (phones and terminal adaptors) do. Therefore, the sipXecs system has to know the gateway's IP address so that calls can be properly routed to the gateway.

For every incoming PSTN line a local extension number or a SIP URI have to be programmed into the gateway so that incoming calls are routed properly. By default incoming calls are routed to the default auto-attendant at extension *100* or SIP URI *sip.operator@domain.com*. If you are using [Direct Inbound Dialing](#) (DID) then a local dial plan has to be configured in the gateway that allows the mapping of incoming DNIS identifiers to local extension numbers or SIP URIs.

**Note:** The implementation of the Audiocodes gateways as plug & play managed devices is not yet completed and should not be used as is. Please configure them manually for now.

## Configuring a Prefix for Outgoing Calls

sipX allows configuring a prefix per gateway that is added to the dialed number for all outgoing calls. This is especially useful if gateways are deployed in different countries or connecting to different carriers that have different requirements with respect to dialed digits.

Configuration: Prefix

Dial Plan

Caller ID

Prefix that will be added to all numbers for calls connected through this gateway.

Buttons: OK, Apply, Cancel

## Configuring Caller ID

A very flexible mechanism is available for the definition of outgoing Caller ID. Refer to the [Caller ID HowTo](#) for further information.

## Configuring Unmanaged PSTN Gateways (FXO)

Unmanaged gateways have to be created in the sipX Configuration Server the same way a plug & play managed gateway is. It's MAC address and IP address are required. Select *Unmanaged gateway* from the drop down menu. Once the gateway exists in sipX it can be used to configure [dial plans](#).

### Gateway Details

[Configuration](#)  
[Dial Plan](#)  
[Caller ID](#)

Name	<input type="text" value="Paris Gateway"/>
Address	<input type="text" value="10.0.1.211"/> IP address (example: 10.1.1.1) or a fully qualified hostname (example: gateway.example.com)
Serial Number	<input type="text" value="0086bc34009a"/> Serial number contains 12 hexadecimal digits (0-9 and a-f), for example: 0040214131fa. Usually the serial number is set to the device's MAC address. For an unmanaged gateway, the serial number is only displayed, not used.
Description	<input type="text" value="Main PSTN Gateway in Paris Office (E1)"/>

The gateway then has to be manually configured. Some gateways provide a Web based interface (Audiocodes and Vegastream). Mediatix gateways are configured using SNMP and Mediatix provides a simple SNMP MIB management tool.



Gateways do not register with sipXecs - only end-points (phones and terminal adapters) do. An FXO gateway therefore does not register; however, an FXS gateway does.