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.

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).

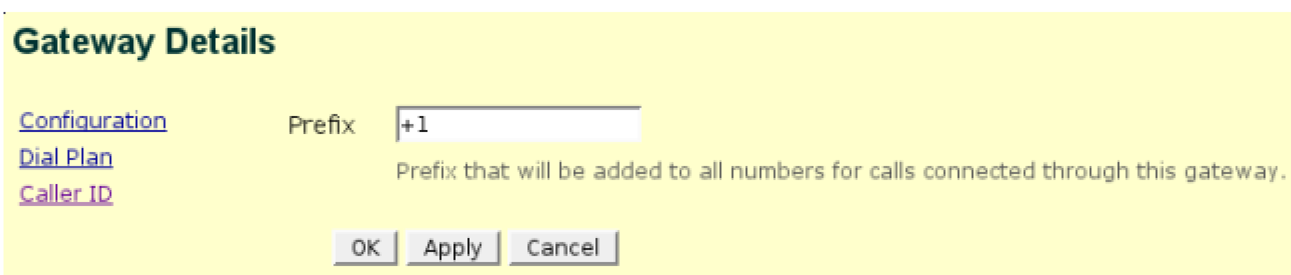
Gateways do not register with sipXecs - only end points (phones and terminal adaptors) do. Therefore, the sipXecs system has to know the gateway's IP address so that calls can be properly routed to the gateway.

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.

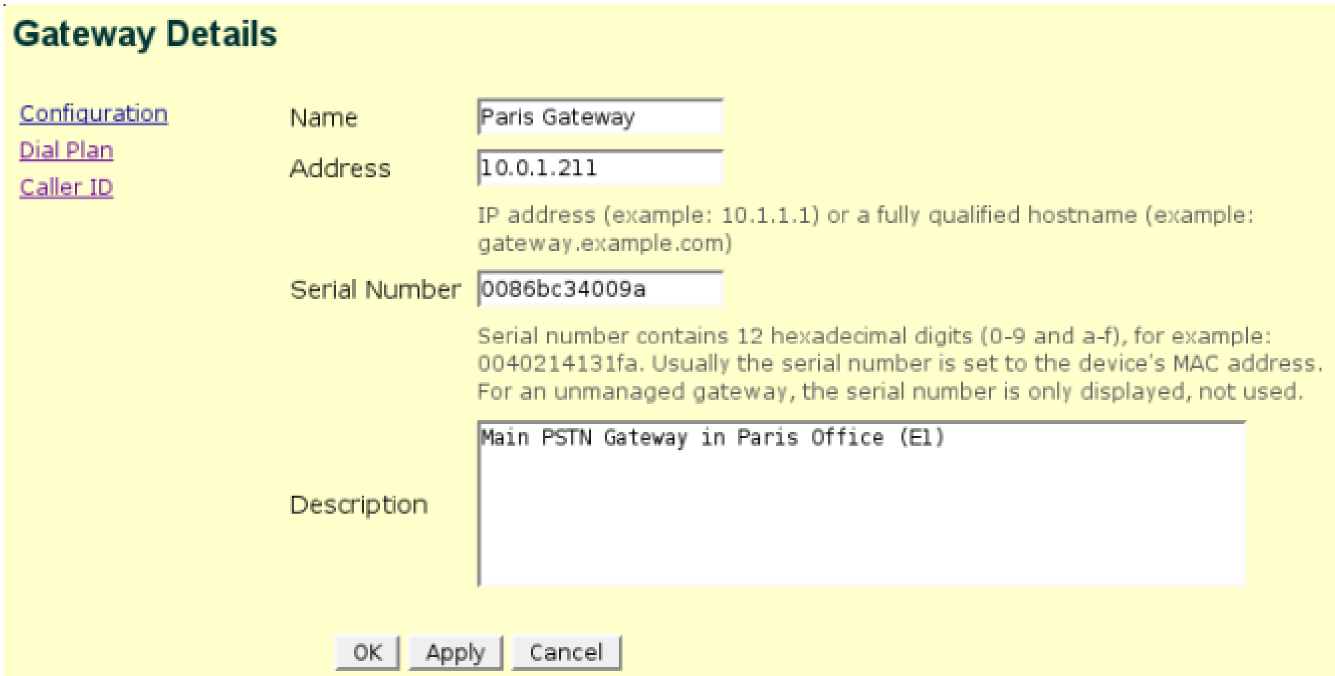


Configuring Caller ID

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.



The gateway then has to be manually configured. Some gateways provide a Web based interface (Audiocodes and Vegastream). Mediatrix gateways are configured using SNMP and Mediatrix 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.