Phone Terminal Provisioning

This page contains instructions on how to provision a Phone terminal extension.

Overview

By using VoipNow's provisioning feature you can set and maintain identical configurations for a large number of equipment. SIP devices associated with a Phone Terminal extension can be automatically configured using either customized provisioning templates added by the extension's parent accounts or the default configuration files, specific to each device model.

At extension level, in order to configure the required device, the Phone terminal account owner is **only** allowed to use the default provisioning template.

After you have successfully added your Phone Terminal extension, VoipNow displays the following message:

```
Extension John Smith (0027*001) has been successfully added. Click here if you want to provision this extension now.
If you want to provision every newly added extension, click here.
```

Clicking the first link will redirect you to the **SIP Devices** page, where you can add a new SIP device or assign the extension to the devices from the **SIP Devices Inventory** table.

If you click the second link, you will be redirected to the extension's management page. Click on the **SIP Preferences** icon under the **Telephony** area and you will be able to customize the SIP options for your extension and/or enable the Extension virtualization feature.

SIP Preferences

The account owners that log in to the VoipNow interface using a service provider, organization or user have complete access to the **SIP Preferences** page only if the **Phone extension SIP management** permission is enabled.

The **Phone extension SIP management** permission can be enabled from the **Roles and Phone Numbers for <account_type> <account_name>** page, when the account is created.

If this permission is disabled, then the user of the Phone terminal extension will only be able to see the **Equipment description** details in the SIP Preferences page. The rest of the SIP settings will not be displayed.

Option	Details
Media encryption	This option allows media (voice or video calls) to be encrypted. VoipNow supports the following crypto standards: SDES and DTLS-SRTP. SDES is a crypto standard we use for voice and video calls over mobile networks. DTLS-SRTP is a crypto standard we use for voice and video calls through WebRTC. From the drop-down list, you can choose among the following options: SDES, DTLS-SRTP, SDES/DTLS-SRTP (and/or). By default, this option is set to **None**. If you want to use any of these crypto standards, you must first ensure that your client (IP phone or softphone) supports it. If the crypto standard you have selected is not supported by your client, calls will not work.
DTMF	Choose the DTMF mode. Default: rfc2833. If you choose the auto option, Asterisk automatically detects whether the channel supports `rfc2833` or not. If not, Asterisk uses the `inband` mode.
A PBX is connected to this extension	This option allows the system to direct an incoming call made to a public phone number to a particular extension on the PBX server connected to the extension for which the current setting is enabled. When the call is sent to the PBX server, the public phone number that was called is saved and thus the call can be directed to the chosen extension on the PBX server. This setting is not available unless your license supports SIP trunking (the Maximum number of SIP trunking channels must be bigger than 0). If the license supports it, this option can only be enabled if the extension's **Maximum public concurrent calls** value has not been previously set to **Unlimited**; if the value is set to **Unlimited**, the line is still available, but you will not be allowed to select the checkbox and you'll see warning message displayed next to it: To make it available, setup the **Maximum public concurrent calls** to a limited number (now it is unlimited).
Ping the extension to check its status	When enabled, the server sends ping SIP messages to the extension regularly. Usually, this option is used for extensions behind NAT.

Allow re-invites from this extension	If enabled, your extension will be allowed to send re-INVITES.
Extension is on private network (<ip_address>/<network_mask>)	This option is available only if you choose the **NAT** (Network Address Translation) or **Public/Private Networks Routing** infrastructure types from **Cloud Management Infrastructure Properties**. Enable this option if your extension is located inside the private network of your SIP server, in order to prevent unnecessary NAT processing.
Extension publishes its own state	Enable this option if you do not want the server to send presence notifications to the phones watching this extension for presence. If enabled, the server will no longer send any notification events unless this extension explicitly publishes its presence by sending PUBLISH messages to the SIP server.
Force enable of MWI	Enable this option if you want to receive **Message Waiting Indicator** notifications and your phone does not send explicit subscriptions for MWI. Most phones do not need this option.
Allowed codecs	Select the codecs supported by the phone device. The list of possible codecs displayed in the **Allowed codecs** section can be modified from the **Unified Communications Zero Priority General** page.
Phone does not register, is located on IP <> Port <> and <has to/does not have to> authenticate	All incoming calls from this IP/Port require/do not require authentication. More than one extension can use the same IP, as long as every single one authenticates. If the same IP is being used by another extension that does not have to authenticate, an error message will be displayed, telling you that this configuration cannot be saved. The drop-down list is disabled until an IP address is filled in.
SIP Signaling Transport	Select the network protocol used on channels: UDP or TCP. Available for extensions that do not register, located on a fixed IP. Default: UDP. The drop-down list is disabled until an IP address is filled in the **Phone does not register, is located on IP** field.
Allow extension SIP connection only from IP <IP_address> (maximum class C (/24)	Limit the extension usage to an IP or a network. Only the IP addresses specified here will be allowed to receive and make calls from this extension. Registration on the phone is still needed in order to receive calls. You can add several IP addresses by using the **+/-** buttons.
Equipment description	Briefly describe your device.

Extension virtualization

Allow virtualization on this extension: If enabled, any other member of the organization can use the phone device where the extension is provisioned. By default, it is **unchecked**.

Related topics
Manage equipment templates