Add SIP Channel

This page describes the SIP Channel and contains instructions on how to add one to the VoipNow system.

Overview

SIP channels allow the VoipNow PBX to communicate with other SIP exchanges and other VoIP networks.

The **Session Initiation Protocol** (SIP) is a signaling protocol widely used for controlling multimedia communication sessions such as voice and video calls over IP. The protocol can be used to create, modify and terminate two-party (unicast) or multi-party (multicast) sessions consisting of one or several media streams.

SIP channels can be configured using predefined provider templates, with the help of which you can instantly set up the channels for certain providers, without the need to manually fill in all the specific parameters. This feature simplifies the configuration process and reduces the time required to define an up and running SIP channel.

To set up a SIP Channel, first go to the **System** menu on the left and click on Channels. In the **Channel Management** page, click the Add SIP Channel icon under the **Trunks** area.

Enter channel details

- **Clone settings from channel:** Helps you create channels with a similar or identical structure more rapidly. You can replicate the settings of an existing SIP channel using the drop-down list.
- **Name:** Fill in a descriptive name to identify the channel.
- **Channel groups:** Click the channel name in the **All groups** list to assign it to one of the channel groups currently in the system. It will get automatically moved into the **Assigned groups** pool. The **Assigned groups** list contains the groups the channel is assigned to.
  To remove the channel from a group, click the desired group and the channel will get automatically be moved back into the **All groups** list. For more information on how to create channel groups, please see the **Channel Groups** section.
- **Channel notes:** Add any channel details you want.
- **I want to configure the channel:** Select how you would like to set up the channel:
  - **Using a known provider** - Set up the channel using the already defined settings included in the XML file specific to the desired provider (default selection). If you go with this option, you will have to fill in the **Select Your Provider** section detailed below.
  - **Manually** - Manually fill in all the details required to set up the SIP channel (recommended for experienced users).

Customize channel preferences

If you set up the channel manually, all available options will be displayed and can be customized. If you configure the channel with a known provider, this section will only display the options for which the visibility level was set to 1 in the selected provider template XML file. In this case, the fieldset will be renamed **Channel Preferences - Reveal All Fields** and will become foldable; all the other options will be displayed with a click. When extended, the fieldset will be renamed **Channel Preferences - Hide Fields**; all the fields not included in the XML file will be hidden with a click.

Option	Description
Send calls to <IP>:<port>	Hostname/IP address and port of SIP provider server. The port is not required unless the hostname cannot be resolved using `DNS SRV`.
Accept calls from IPs /network	IP address(es) from which the system should accept calls. You may add/remove multiple IPs using the **+/-** buttons.
Username	The SIP provider's registrar server deals with the setup of all SIP calls in the SIP provider's network. Fill in the address of the record that is going to be created or updated on the registrar. The specified value will compose the username for the To and From SIP headers. If no Authorization username is specified, this value will be used for the Username SIP header as well. You can read this parameter as "we are registering as." Please check the **Appendix**, to see what character types can the username contain.
Password	Used for authorization with the SIP provider's `registrar` server. You can read this parameter as "our digest authorization password is." Please check the Appendix, to see what character types can the password contain.

Confirm password	Confirm password used to connect to the SIP provider's `registrar` server.
Do not register	When enabled, you will not need to register before connecting.
Authorization username	Overrides the username used for authorization with the SIP provider's `registrar` server. You can read this parameter as "we authenticate as."
Concurrent calls (outgoing+incoming)	Applies if your SIP provider limits the number of concurrent calls through the channel. The channel will no longer be selected by the routing engine when the limit is reached and your provider will no longer accept other calls on that channel.
Charging plan	Specify the channel type: 1. **Paid:** The channel charges a fee every time it is used. 2. **Free:** The channel can be used to call any destination for free.
Channel capabilities	Select the channel functionality: 1. **Incoming:** Only incoming calls can be received through the channel. 2. **Outgoing:** Only outgoing calls can be sent through the channel. 3. **Both:** The channel can be used for both receiving and sending calls.
DTMF mode	Choose the DTMF (Dual-tone multi-frequency signaling) mode from the drop-down list. The default option is rfc2833. If you choose the auto option, Asterisk will automatically detect whether the channel supports rfc2833 or not. If it does not support it, Asterisk will use the inband mode.
Behind NAT	Check this option if the channel is located behind a NAT (Network Address Translation) router.
CallerID for outgoing calls	Specify the number that will be displayed by phones with an active CallerID feature (e.g. John Doe <555890123>). Usually the CallerID is the same as the public phone number.
From user	Some SIP providers require this parameter to identify the user. To verify if the channel requires this parameter, please check with the SIP provider.
From domain	Some SIP providers require this parameter to identify its users. To verify if the channel requires this parameter, please check with the SIP provider. Please note that the domain name can contain characters from any official language script. Domain names that contain these special, so called non-ASCII, characters are called Internationalized Domain Names (IDNs). VoipNow supports IDNs.
Authorization extension	The username part of the Contact SIP header used for creating a registration binding with the service provider's network. You can read this parameter as "we are located at."
Get DID from custom header	If a valid DID number is not present in the request line of the SIP packet, it will be taken from a custom header. This is where you can define the name of that header. For example, if you fill in **Test** and the **DIDnumber** is 12345, then the custom header will be Test: 12345.
Qualify value	The server checks the remote party for presence regularly. Specify a time-frame using the drop-down list. If it does not answer within the time-frame, the device will be considered offline. Can be disabled if you select **None**.
Prefix all calls with	The prefix that will be added to all destination numbers of the calls routed through this channel.
Trusted channel	When enabled, VoipNow considers that all communication through this channel comes from a trusted source (based on IP) and does not authenticate incoming calls.
Trust Remote-Party-ID SIP header	When this parameter is set to 1 (enabled), VoipNow trusts and uses the Remote-Party-ID headers in the SIP messages received on the channel.

Send Remote-Party-ID SIP header	When this parameter is set to 1 (enabled), VoipNow includes Remote-Party-ID headers in the SIP messages sent through the channel.
ICE Support	Select this option if you want to enable Interactive Connectivity Establishment (ICE) support for NAT Traversal in multimedia sessions.
Session timers	The user agents send periodic re-INVITE or UPDATE requests (session refreshm requests) to keep the session alive. The interval for the session refresh requests is determined through a negotiation mechanism. If a session refresh request is not received by the end of the interval, the session is considered terminated. Select how the channel will handle the session timers using the drop-down list: 1. **Accept** - Default value; the channel accepts the session refresh requests sent by user agents; 2. **Originate** - The channel accepts this mechanism only for session refresh requests sent from inside the system; requests sent by user agents outside the system will not be allowed; 3. **Refuse** - The channel sends a 422-rejection response, informing that it does not support session timers.
Session refresh interval	Maximum amount of time (in seconds) between session refresh requests in a dialog before the session will be considered timed out. This time-frame is included in the SIP Session-Expires header field. The user agent server (UAS) obtains this value form the Session-Expires header field in a 2xx response to a session request that it sends. The user agent client (UAC) determines this value from the Session-Expires header field in a 2xx response to a session request that it receives. Default value: 1,800 seconds. Accepted integer values: 10 to 84,000 seconds. Small session intervals can be destructive to the network. They cause excessive messaging traffic that affects both the user agents and the proxy servers, increasing the possibility of glare that can occur when both user agents send a re-INVITE or UPDATE request at the same time. The default value of 1,800 seconds was chosen because 95% of the phone calls are shorter then 30 minutes, however the exact value is dependent on many network factors, including network bandwidths and latencies, computing power, memory and so on. Therefore, you should set a value in concordance with your requirements and system resources.
Minimum session refresh interval	Minimum value (in seconds) that will be accepted by the channel for the session interval. This value is included in the SIP Min-SE header field. Default value: 90 seconds, representing little more than twice the duration that a SIP transaction can take in the event of a timeout. Accepted integer values: 90 to 18,000 seconds. This value allows enough time for a user agent to attempt a refresh at the half-point of the session interval and for that transaction to complete normally before the session expires. If the UAC's Session-Expires is lower than the Min-SE defined for another SIP device, then that device rejects the session refresh request with a 422 response. This response contains a Min-SE header identifying the minimum session interval it is willing to support. The UAC will try again, this time including the specified Min-SE header. This way, the minimum timer meets the constraint.
Session refresh source	Indicate who will handle the SIP headers: 1. **UAC:** the user agent client; 2. **UAS:** the user agent server.
SIP Signaling Transport	Select the network protocol used on channels: UDP or TCP. Default value: UDP.
Channel codecs	Select the codecs for encoding–decoding the voice signal over this channel.
Use MD5	VoipNow encrypts the passwords used in the authentication process. The method used is an MD5 hash function.

Select your provider

This option is only displayed if you have previously selected to configure the channel with a known provider. Providers get displayed in alphabetical order, starting with the Gold certified providers, followed by the Silver ones and ending with those that are not certified at all. Recommended providers are displayed in the **Name (Certification: <level>)** format, e.g. Service Provider (Certification: Gold). If the provider is not certified, it will be displayed as Service Provider (Certification: None).

- **I want to get service in:** Lists all the countries the defined providers are available in. This allows you to target the SIP channel for certain destinations. If you just want to set up a generic channel, stick to the No preference option, which is the default value.
- **Recommended providers:** Lists all the providers you have previously defined for the country. - - is the default selection; for the No preference option, the list displays all the providers in the system, regardless of the country they are available in.
- **Provider website:** The homepage of the service provider. Click the link to open the site in a new window. The link is automatically updated, depending on the selected provider.

Set the SIP server

Set the SIP server that proxies all the SIP traffic for this channel. Use the **SIP node assign** option from the **Calls Distribution** section.

Configure public phone numbers and DIDs

Configure public phone numbers and DIDs as sent by the channel provider. Select the **Set up the public phone numbers assigned to the channel** checkbox and click **OK**. If you want to add the new channel without configuring the public phone numbers, deselect the **Set up the public phone numbers assigned to the channel** checkbox and click **OK**. For more info on how to add public phone numbers, check the **Manage Public Numbers** section.

Set channel costs

Set up the channel costs charged by the channel provider, based on the call destination. Select the **Setup channel costs** checkbox. You won't be able to do this unless you have defined a Paid charging plan in the **Channel setup** page. For more info on how to add channel costs, check the **Add Channel Cost** section.

Click **OK** to add the new channel. If you want to add the new channel without configuring the public phone numbers, deselect the **Set up the public phone numbers assigned to the channel** checkbox and click **OK** to save the channel costs.

Related topics
Manage channels

Manage public numbers

Manage channel costs