Add PRI Channel

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

Overview

VoipNow automatically determines the available (i.e. unassigned) physical PRI channels installed on the server. If no PRI channels are available, the **Add PRI Channel** icon is greyed out and unavailable. On mouse over, an explanation gets displayed.

The Primary Rate Interface channel is a type of ISDN designed for larger companies, typically used to connect a PBX to a telephone exchange.

- It includes 30 B-Channels (23 in USA) and one D-Channel.
- It is usually transmitted through E1 lines (T-1 lines in USA).

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

Enter channel details

- **Clone settings from channel:** Replicates all the settings of an existing channel.
- **Name:** Requires a descriptive name used to identify the channel.
- **Use channels From ID <number> To ID <number>:** Select the channels that belong to the previously selected node you want to group. The lists display only the channels that were not used in other configurations. You can add/remove several channel intervals using the **+/-** icons.
- **Group ID:** Select the ID of the channels group previously defined.
- **Channel capabilities:** Select the channel functionality: Incoming – Only incoming calls can be received through the channel. Outgoing – Only outgoing calls can be sent to the channel. Both – The channel can both receive and send calls.
- **Charging plan: Specify the channel type.** Paid – The channel charges a fee every time it is used. Free - The channel can be used for free to call any destination.
- **Default language:** The channel will have a default language code that determines the language of the sound files played on this channel. The default language is English. Select your language using the drop-down list.
- **Channel notes:** Add any channel details you want.

Define signaling and voice quality options

Option	Description
Signaling method	Choose the signaling method used on the channel. Default method: fxo_ls. Please note that, before adding the channel, VoipNow verifies if the method you have selected is actually supported by the physical channel. If the method is not supported, VoipNow will display an error message informing you which methods are allowed. Please see the Appendix to read more on each of the signaling methods available.
Echo cancellation	If set to Yes, VoipNow will apply an echo cancellation algorithm to the voice transmitted over this channel.
Echo cancellation when bridged	If set to Yes, VoipNow will apply an echo cancellation during pure TDM bridging.
Enable echo training	Trains the echo canceling algorithm. If you select a value other than `No`, VoipNow will mute the channel, send an impulse, and use the impulse's response to pre-train the echo canceler. The values available are: Yes, No, or a number of milliseconds the training to be delayed with.
Relax DTMF detection	Useful if you are having problems with the DTMF tones detection. If set to `Yes`, the DTMF detector is more likely to experience "talk off" (e.g. the detector identifies some voice signals as tones).
Receive gain	Set a receive gain using the drop-down list. Drop-down list values: dB.
Transmit gain	Set a transmit gain using a drop-down list. Drop-down list values: dB.

Answer the channel immediately	Specify if the channel should be answered to immediately or the simple switch should provide dial tone, read digits, etc.
Tones are played on the channel for	Specify for how long the generated tones (DTMF and MF) will be played on the channel. Accepts values: 1 to 9,999 ms. Default value: 100 ms.
Distinctive ringing on FXO lines	Indicate if an attempt at recognizing distinctive ring styles on incoming calls should be made or not.
Enable ADSI	If the phone device connected to this channel has ADSI (Analog Display Services Interface) capabilities, set **Enable ADSI** to Yes. Default value: No.
Enable busy detection	If set to Yes, the incoming audio signals during a call or a dial attempt will be analyzed in order to recognize busy signals. This setting is useful on analog trunk interfaces to identify a busy signal when dialing out to detect when the caller has hung up. You need to make sure that you don't use this setting on digital interfaces such `QuadBri` cards, otherwise you will have "broken calls" problems.
How many busy tones to wait before hangup	Specifies how many busy tones the system waits for before hanging up. Accepts values: 2 to 10. Default value: 4 tones. A higher number of busy tone means a higher time needed to hang up a channel and a lower probability to experience random hangups.
Busy tone cadence	Allows you to specify the pattern of the busy signal. Default pattern: 500 ms on, 500 ms off. The system will check the length of the sound (tone) and silence, which will reduce the probability of hanging up randomly. Accepts values: 50 to 9,999 ms.
Use polarity reversal to mark answered remote calls	Specifies whether the system must check for a polarity reversal to know when an outgoing call is answered by the remote party. Default value: `Yes`.
Hangup on polarity reversal	Specifies whether a polarity reversal is used to signal the disconnection of a phone line. If set to Yes, a call will be considered "hung up" on a polarity reversal. Default value: Yes.
Detect call progress	If set to `Yes`, the system attempts to determine answer, busy and ringing events on the channel.
Call progress zones	Defines the timing and frequencies for call progress detection. The default value is `United States/North America`.
Ring timeout	Specify how long the system should take before considering to hung up on a non-ringing line. FXO (FXS signaled) devices must have a timeout to determine if there was a hangup before the line was answered. Accepted values: 1,000 to 99,999 ms. Default value: 8,000 ms.
Dial pulse	For FXO (FXS signaled) devices, this setting specifies whether to use pulse dial instead of `DTMF`. Default value is No.
Fax detection	Specifies whether the system will try to detect incoming, outgoing, both types of faxes or it will not detect faxes at all.
Jitter buffer	Specify the length of the jitter buffer. The jitter buffer is used to compensate for jitter (the variation between packet delays) and out of order delivery of packets. Default value: 4 ms.

Choose CallerID preferences

Option	Description
Use CallerID name	Specify the CallerID name that will be sent for all outgoing calls routed through this channel.
Use CallerID number	Specify the CallerID number that will be sent for all outgoing calls routed through this channel. You can fill in a number in the format `+39203991739`.
Use CallerID	In the case of phones, if set to `Yes`, this option allows the system to send the CallerID data to the phone when ringing it. In the case of trunk lines, if set to `Yes`, this option allows the system to look for the CallerID on the incoming calls.
CallerID signaling	Select the CallerID signaling method using the drop-down list. Default method is `Bell` (used in the United States). You can also choose among `V23` (used in the UK) and `DTMF` (used in Denmark, the Netherlands, and Sweden).

What signals the start of callerID	Specify the type of signal that marks the start of the CallerID. `Ring` means that a ring signals the start of the caller ID, while `Polarity` means that a polarity reversal signals the start of the CallerID.
Hide callerID	In the case of FXS phones, if set to `Yes`, this option allows the system to stop sending this channel's CallerID information to the called party. Default setting: `No`.
Call waiting	If set to Yes, the system will activate the call waiting function for all calls routed through the channel. Default setting: No. If the channel has Call waiting set to Yes, a user can deactivate the call waiting function temporarily for the duration of a call by dialing *70 on their phone keypad.
Restrict callerID presentation	If set to `No`, the system will restrict the destination phone device from accessing the CallerID information of the user that initiated the call. This setting applies only to PRI channels.
Use callerID presentation	If set to `No`, the system will not send CallerID information to the destination phone device. This setting only applies to PRI channels.
Send callerID after [X] rings	Some countries (e.g. UK) have ringtones with different ring patterns. A ring may be made up of a sequence of simple rings. This means that the CallerID must be sent after this sequence and not after the first ring. Specify the number of rings in the sequence using the drop-down list. Default value: 1 ring.
CallerID on call waiting	Specifies whether Asterisk will send CallerID data to the handset during call waiting indication or not.

Set the channel functions

Set the channel functions in the **Channel Features** section.

- **Transfer to busy extensions:** If set to Yes, the system performs flash-hook transfers even if the destination channel is busy. If set to No and the destination channel is busy, the system returns the call to the user that performed the transfer.
  Flash-hook transfers are used in the call waiting feature. If a new call rings while the called party is involved in a conversation, Instead of producing a busy signal, the new caller hears the ringing sound and the called party hears a special beep tone. If he chooses so, the called party can depress send a flash-hook by briefly pushing the phone's hookswitch. The original caller is put on hold, and the new caller is connected. Repeated flash-hooks will swap between the two callers. On a PBX, a flash-hook will produce a new dial-tone, allowing a three-way conference or call transfer.
- **Support three-way calling:** If set to Yes, the channel supports conferences made up of three participants.
- **Support flash-hook call transfer:** If set to Yes, the channel supports call transferring by performing a flash-hook and then dialing the transfer destination's number.
- **Support call forward:** If set to Yes, the channel supports call forwarding features.
- **Support call parking:** If set to Yes, the channel supports call parking/unparking features.

Customize channel preferences

Option	Description
Switch type	Choose the PRI switch type your PRI line is connected to. The default type is National. The other available values are dms100, 4ess, 5ess, EuroISDN, ni1, qsig.
Network specific facility	Some PRI switches (AT&T for example) require that you provide the network specific facility. The default value is none. Other values possible include sdn, megacom, accunet.
PRI dial plan (Rarely used!)	Rarely used. Controls the Type of Number (TON) and the Numbering Plan Index (NPI) associated with the called party information element. The values available are: • **Unknown** - The receiving switch will interpret the digits according to the standard used by the PSTN in that country. e.g. A switch in Sweden will interpret the number "00461234567" as the national Swedish number "1234567". • **Private** • **Local** • **National** - The receiving switch will interpret the digits as a national number (e.g. with an area code at the beginning); escape digits, such as leading zero for the area code are not interpreted. • **International** • **Dynamic** - The receiving switch will parse the dialed number and try to find a matching prefix in the settings international prefix, national prefix, local prefix, private prefix, unknown prefix. Default value is National. This is the recipient of the call. This parameter is required only by a few types of PRI switches. If your switch does not require this parameter, it will simply be ignored.

PRI local dial plan (Rarely used!)	Rarely used. Controls the Type of Number (TON) and the Numbering Plan Index (NPI) associated with the calling party information element. The available values are: • **Unknown** - The receiving switch will interpret the digits according to the standard used by the PSTN in that country. e.g. A switch in Sweden will interpret the number "00461234567" as the national Swedish number "1234567". • **Private** • **Local** • **National** - The receiving switch will interpret the digits as a national number (e.g. with an area code at the beginning). Escape digits, like such as zero for the area code, are not interpreted. • **International** • **Dynamic** - The receiving switch will parse the dialed number and try to find a matching prefix in the settings international prefix, national prefix, local prefix, private prefix, unknown prefix. Default value is National.
PRI callerID international prefix	Specifies the international prefix part of the dialed number. The default value is 00. Another example of a valid international prefix is +.
PRI callerID national prefix	Specifies the national prefix part of the dialed number. The default value is 0. Another example of a valid national prefix is +49.
PRI callerID local prefix	Specifies the local prefix part of the dialed number. This text box can be empty. Examples of valid local prefixes: 0711 or +49711.
PRI callerID private prefix	Specifies the private prefix part of the dialed number. This text box can be empty. Examples of valid private prefixes: 07115678 or +497115678.
PRI callerID unknown prefix	This text box can be empty.
PRI reset interval	Specifies the time interval between resetting unused channels. Default value is 3,600 s. Some PBXes have problems with channel resetting. In these cases, you can set the time interval to a larger value (e.g. 86,400 s).
Send overlap digits	Specifies whether a dialed phone number's digits should be sent to the switch by using the overlap sending procedure or the en-bloc procedure. If set to Yes, overlap sending will be used. Default value is Yes. The overlap sending procedure means that the system sends each dialed digit in its own ISDN information message. The en-bloc procedure means that all digits are sent in the same ISDN setup message.
PRI indications	Specifies the type of signaling used for indicating events such as busy and congestion. When the setting is In band, the system signals busy or congestion using in-band tones and does not answer. When the setting is Out of band, the system disconnects with busy or congestion information code and the switch will play the indication tones to the caller.
Force all PRI channels to be marked exclusively selected	If set to Yes, the system will override the existing channels selection routine. As a result, all PRI channels will be marked as exclusively selected.

Set ISDN parameters

- **Transmit facility-based ISDN:** If set to `Yes`, the transmission of facility-based ISDN supplementary services is enabled.
- **Set ISDN timer t<numeric_value_1> to <numeric_value_2> ms:** Set ISDN timers. <numeric_value_1> and <numeric_value_2> accept values between 1 and 99,999. You can create several ISDN timers. Use the + and - buttons to add more text boxes.

Set timing parameters

Set the timing paramters used for non-PRI T1 lines. All text boxes accept numeric values that range between 1 and 9999ms:

- Pre-wink time
- Pre-flash time
- Wink time
- Flash time
- Start time
- Receiver wink time
- Receiver flash time
- De-bounce timing

Assign channel to channel groups

Assign the channel to one or several channel groups in the **Channel Group** section:

- **All groups:** Lists all the groups currently in the system. To assign the channel to a specific group, click the desired channel and it will be automatically moved into the **Assigned groups** pool.

- **Assigned groups:** Lists the groups the channel is assigned to. To remove the channel from a group, click the desired channel and it will be automatically moved back into the **All groups** pool.

Configure public phone numbers and DIDs

Configure the public phone numbers and DIDs as sent by the channel provider. Select the **Set up the public phone numbers assigned to the channel** checkbox. 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 Costs** section.

Click **OK** to add the new channel. If you want to add the new channel without configuring its costs, deselect the **Setup channel costs** checkbox and click **OK** to save the channel costs.

Related Topics
Manage Channels

Manage Public Numbers

Manage Channel Costs