Trunks Module - User Guide

Overview

The Trunks module is where you control connectivity to the PSTN and your VoIP provider(s). This is where you also control to interconnect other PBX's for multi-site applications. The most common trunks are SIP and DAHDi (or Zap). Other than the Extensions module, the Trunks module is one of the most critical modules on the system and allows for a great deal of flexibility.
Logging in

- From the top menu click **Connectivity**
- In the drop down click **Trunks**

Adding a Trunk

You will want to click on the trunk type you wish to create. Please note in this guide we will cover the generic settings that are universal to all trunk types first. Following this we will cover technology specific settings.

### Add a Trunk

- Add SIP (chan_sip) Trunk
- Add DAHDI Trunk
- Add IAX2 Trunk
- Add ENUM Trunk
- Add DUNDI Trunk
- Add Custom Trunk

### General Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Trunk Name</td>
<td>Set a descriptive name for the trunk.</td>
</tr>
<tr>
<td>Outbound CallerID</td>
<td></td>
</tr>
<tr>
<td>CID Options</td>
<td>Allow Any CID</td>
</tr>
<tr>
<td>Maximum Channels</td>
<td></td>
</tr>
<tr>
<td>Asterisk Trunk Dial Options</td>
<td>Check to always try next trunk</td>
</tr>
<tr>
<td>Continue If Busy</td>
<td></td>
</tr>
<tr>
<td>Disable Trunk</td>
<td></td>
</tr>
</tbody>
</table>

### Trunk Name

Set a descriptive name for the trunk.

### Outbound CallerID
Use this field to specify caller ID for calls placed out of this trunk with the \(<NXXNXXXXXX>\) format. You can also use the format: “hidden” \(<NXXNXXXXXX>\) to hide the caller ID sent out over digital lines, if supported (E1/T1/J1/BRI/SIP/IAX2).

**CID Options**

This setting determines what CIDs will be allowed out of this trunk. Please NOTE that Emergency CIDs defined on an extension or device will ALWAYS be used if this trunk is part of an emergency route regardless of these settings.

**Allow Any CID**

All CIDs, including foreign CIDs from forwarded external calls, will be transmitted.

**Block Foreign CIDs**

This will block any CID that is the result of a forwarded call from off the system. CIDs that are defined for an extension or device will be transmitted.

**Remove CNAM**

This will remove the CNAM (Name) from any CID sent out of this trunk.

**Force Trunk CID**

This will always use the CID defined for the trunk, except if the trunk is part of an emergency route with an emergency CID defined for the extension or device. Intra-Company routes will always transmit an extension’s internal number and name.

**Maximum Channels**

Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. To count inbound calls against this maximum, use the auto-generated context: from-trunk-[trunkname] as the inbound trunk’s context (see extensions_additional.conf). Leave blank to specify no maximum.

**Continue if Busy**

Normally the next trunk is only tried upon a trunk being ‘Congested’ in some form, or unavailable. Checking this box will force a failed call to always continue to the next configured trunk or destination even when the channel reports BUSY or invalid NUMBER. This should normally be unchecked.

**Disable Trunk**

Check this to disable this trunk in all routes where it is used.

**Dial Pattern Manipulation Rules**

These rules can manipulate the dialled number before sending it out of this trunk. If no rule applies, the number is not changed. The original dialled number is passed down from the route where some manipulation may have already occurred. This trunk has the option to further manipulate the number. If the number matches the combined values in the prefix plus the match pattern boxes, the rule will be applied and all subsequent rules ignored. Upon a match, the prefix, if defined, will be stripped. Next, the prepend will be inserted in front of the match pattern and the resulting number will be sent to the trunk. All fields are optional.
### Pattern Description Table

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>X</td>
<td>Matches any digit from 0-9</td>
</tr>
<tr>
<td>Z</td>
<td>Matches any digit from 1-9</td>
</tr>
<tr>
<td>N</td>
<td>Matches any digit from 2-9</td>
</tr>
<tr>
<td>[1237-9]</td>
<td>Matches any digit in the brackets (example: 1,2,3,7,8,9)</td>
</tr>
<tr>
<td>. (dot)</td>
<td>Wildcard, matches one or more dialled digits.</td>
</tr>
</tbody>
</table>

#### Prepend

Digits to prepend upon a successful match. If the dialled number matches the patterns in the prefix and match pattern boxes, this will be prepended before sending to the trunk.

#### Prefix

Prefix to remove upon a successful match. If the dialled number matches this, plus the match pattern box, this prefix is removed before adding the optional prepend box and sending the results to the trunk.

#### Match pattern

The dialled number will be compared against the prefix, plus this pattern. Upon a match, this portion of the number will be sent to the trunks after removing the prefix and appending the prepend digits. You can completely replace a number by matching on the prefix only, replacing it with a prepend and leaving the match pattern blank.

### Dialled Rules Wizard

#### Always dial with prefix

This is useful for VoIP trunks, where if a number is dialled as "5551234," it can be converted to "16135551234."

#### Remove prefix from local numbers

This is useful for ZAP and DAHDi trunks, where if a local number is dialled as "6135551234," it can be converted to "555-1234."

#### Setup directory assistance

This is useful to translate a call to directory assistance.

#### Lookup numbers for local trunk

This looks up your local number on [www.localcallingguide.com](http://www.localcallingguide.com) (NA-only), and sets up so you can dial either 7 or 10 digits (regardless of what your PSTN is) on a local trunk (where you have to dial 1 + the area code for long distance, but only “5551234” (7-digit dialling) or “6135551234” (10-digit dialling) for local calls.

#### Upload from CSV

Upload patterns from a CSV file, replacing existing entries. If there are no headers, then the file must have 3 columns of patterns in the same order as in the GUI. You can also supply headers: prepend, prefix and match pattern in the first row. If there are less than 3 recognized headers, then the remaining columns will be blank.

#### Outbound Dial Prefix

The outbound dialling prefix is used to prefix a dialling string to all outbound calls placed on this trunk. For example, if this trunk is behind another PBX or is a Centrex line, then you would put "9" here to access an outbound line. Another common use is to prefix calls with "w" (to add a 500ms wait per w) on a POTS line that needs time to obtain a dial tone to avoid eating digits. Most users should leave this option blank.

### SIP (chan_sip)/IAX2 Specific Settings
Trunk Name
Give the trunk a descriptive name such as “mysiptrunk.”

PEER Details
Here you give the PEER connection parameters supplied by your VoIP provider. You may need to add to the provided default settings and in some cases remove default settings depending on your provider or application.

Order is important, as it will be retained. For example, if you use an “allow/deny” directive, then make sure the “deny” is first (reading top down).

USER Context
This is most often the account name or number your provider expects.

USER Details
Here you supply the USER connection parameters supplied by your VoIP provider. You may need to add to the provided default settings and in some cases remove default settings depending on your provider or application.

Order is important, as it will be retained. For example, if you use an “allow/deny” directive, then make sure the “deny” is first (reading top down).

Register String
Most VoIP providers require your system to register with theirs. If required, you will need to enter the string the provider specifies, such as username:password@some.voipprovider.com.

In some cases you may need to provide a DID and the end of the string:
such as username:password@some.voipprovider.com/7045551212.

**DAHDI Specific Settings**

DAHDI (Digium Asterisk Hardware Device interface) is the driver framework for interfacing with telephony hardware, such as a T1/PRI card. You would refer to the hardware documentation for initial installation. Then turn to the DAHDI Trunk module for integration of the hardware into your dial plan (usually in conjunction with the Inbound Route and/or Extensions module. See their respective user guides for more info).

DAHDI specific configurations are handled by the DAHDI Config module. Available trunks will show up under DAHDI Trunks.

### DAHDI Trunks

When not using the DAHDI Module this will be replaced with DAHDI Trunk Identifier. With the module you get a select box with “Group #” and the hunt scheme. See the table below for explanation. In the table we will reference as Group 0 (Zero)

<table>
<thead>
<tr>
<th>Managed</th>
<th>Unmanaged</th>
<th>Scheme</th>
</tr>
</thead>
<tbody>
<tr>
<td>Group 0 Round Robin Ascending</td>
<td>r0</td>
<td>Take turns using each available Channel on group from lowest to highest</td>
</tr>
<tr>
<td>Group 0 Round Robin Descending</td>
<td>R0</td>
<td>Take turns using each available Channel on group from highest to lowest</td>
</tr>
<tr>
<td>Group 0 Ascending</td>
<td>g0</td>
<td>Use first available channel from lowest to highest</td>
</tr>
<tr>
<td>Group 0 Descending</td>
<td>G0</td>
<td>Use first available channel from Highest to lowest</td>
</tr>
</tbody>
</table>

**SIP TRUNK(PJSIP)**

**Simple**

When the configuration mode is set to simple you will see the following items.
### Permanent Auth Rejection
Determines whether failed authentication challenges are treated as permanent failures.

### Retry Interval
How long between retries

### Expiration
Expiration time for registrations in seconds.

### Forbidden Retry Interval
Interval used when receiving a 403 Forbidden response.

### Max Retries
Maximum number of registration attempts.
Qualify Frequency
Interval at which to qualify.

Username
SIP Username

Secret
SIP Password

SIP Server
SIP Server you register to

SIP Server Port
SIP Port

Outbound Proxy
SIP Proxy

Contact User
Contact User to use in request.

Context
Context to send the Inbound Call to.

Transport
The Transport to use for connection

Codecs
Check the desired codecs, all others will be disabled. Drag to re-order.

Advanced
When the configuration mode is set to Advanced the following fields are added to the above.

Client URI :
Client SIP URI used when attempting outbound registration

Server URI :
SIP URI of the server to register against

AOR Contact
Permanent contacts assigned to AoR

**ENUM TRUNK**

ENUM (E.164 Number Mapping) is a method, which uses a special DNS record to translate a telephone number into a URI (Uniform Resource Identifier). URI’s in turn can translate into an IP address or a specific extension at a specific IP address or perhaps end up back on the PSTN locally from the receiving system. This facility can be very useful for PSTN unification.

ENUM lookups are done automatically on the e164.org domain, therefore little configuration is required. Although it is possible to offer outside access to local resources (PSTN) that would be beyond the scope of this guide. See http://www.e164.org/ for more information.

**DUNDi Trunk**

DUNDi (Distributed Universal Number Discovery) is a trusted, peer-to-peer network that functions somewhat like ENUM, however instead of using centralized service, DUNDi is fully distributed and uses no centralized authority mechanism. See http://www.dundi.com for more information and http://doxygen.asterisk.org/trunk/Config_dun.html for a dundi.conf example.

The PBX offers limited support for DUNDi trunks and additional manual configuration is required. The trunk name should correspond to the [mappings] section of the remote dundi.conf systems. For example, you may have a mapping on the remote system and corresponding configurations in dundi.conf locally, that looks as follows:

```
[mappings]
priv => dundi-extens,0,IAX2,priv:{$SECRET}@218.23.42.26/{$NUMBER},nopartial
```

In this example, you would create this trunk and name it priv. You would then create the corresponding IAX2 trunk with proper settings to work with DUNDi. This can be done by making an IAX2 trunk in PBX or by using the iax_custom.conf file

**DUNDi Mapping**

This is the name of the DUNDi mappings as defined in the [mappings] section of the remote dundi.conf peers. This corresponds to the “include” section of the peer details in the local dundi.conf file.

**Custom Trunk**

Custom trunks are useful when attempting to integrate certain devices like cellular gateways or perhaps when setting up least cost routing schemas.

**Custom Dial String**

Define the custom dial string here. Use the token $OUTNUM$ wherever the number to dial should go. For example:

- Local/$OUTNUM$@leastcost-custom
- SIP/$OUTNUM$@192.168.41.23:5064
- IAX2/$OUTNUM$@10.100.1.11

**Permanent Auth Rejection?**
<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Retry Interval?</td>
<td></td>
</tr>
<tr>
<td>Expiration?</td>
<td></td>
</tr>
<tr>
<td>Forbidden Retry Interval?</td>
<td></td>
</tr>
<tr>
<td>Max Retries?</td>
<td></td>
</tr>
<tr>
<td>Qualify Frequency?</td>
<td></td>
</tr>
<tr>
<td>Username?</td>
<td></td>
</tr>
<tr>
<td>Secret?</td>
<td></td>
</tr>
<tr>
<td>SIP Server?</td>
<td></td>
</tr>
<tr>
<td>SIP Server Port?</td>
<td></td>
</tr>
<tr>
<td>Outbound Proxy?</td>
<td></td>
</tr>
<tr>
<td>Contact User?</td>
<td></td>
</tr>
<tr>
<td>Context?</td>
<td></td>
</tr>
<tr>
<td>Transport?</td>
<td></td>
</tr>
<tr>
<td>Codecs</td>
<td></td>
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</tbody>
</table>