

Configure a Provider

A provider represents a trunk or external service used by MiRTA PBX for outbound calls, inbound call handling, SMS delivery, or other carrier-side functions. The screenshots on this page use fictional values and should be replaced with the values supplied by the carrier or service provider.

Open the Provider List

Open **Admin > Providers**. The list shows each configured provider, its peer name, technology, host, penalty, status, current inbound and outbound channel counters, and available actions.

Name	Peer Name	Tech	Host	Penalty	Status	# IN	# OUT	Actions
Docs Demo SIP Provider	docs_demo_sip_provider	SIP	dynamic	99	x			

Provider list filtered to the documentation example provider.

Select **New Provider** to create a carrier trunk or SMS gateway. Existing providers can be opened from the provider name or peer name links.

Information

Start with the provider identity. Use a descriptive **Name** that operators recognize, and use a stable **Peer Name** that matches the SIP peer, PJSIP endpoint, IAX peer, local route, or SMS gateway identifier used by the system. Choose the technology and set the status.

Name: Example Telecom SIP Trunk

Peer Name: example_telecom_sip

Technology: SIP

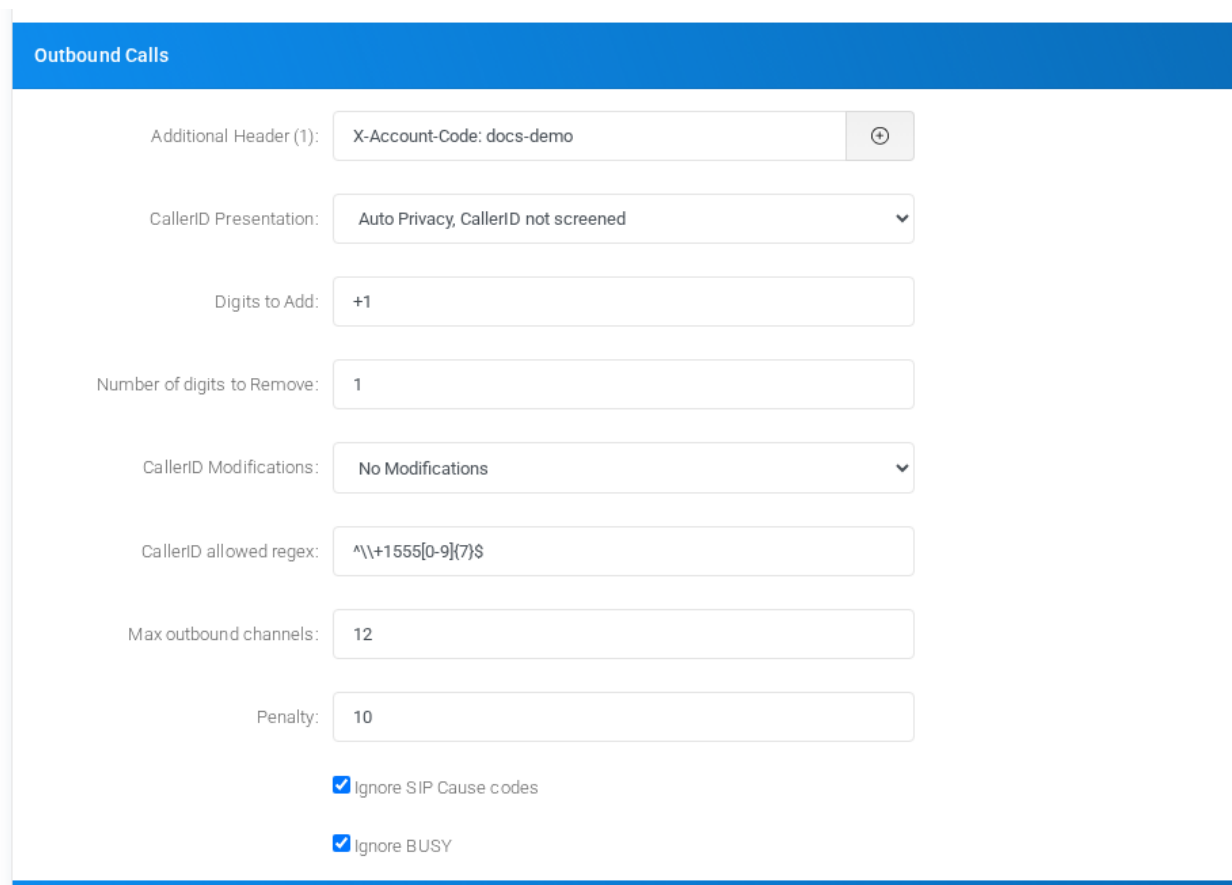
Status: Enabled

Provider Information block filled with fictional SIP trunk values.

- **Enabled** allows inbound and outbound use.
- **Disabled** keeps the provider configured but unavailable.
- **Only Inbound** hides outbound routing options and is useful when the carrier should only deliver calls into the PBX.

Outbound Calls

Use outbound settings to normalize calls before sending them to the provider. Additional headers can be sent to SIP, PJSIP, IAX2, or SMS providers. Digits to add and digits to remove are applied during outbound routing. Caller ID presentation, caller ID modification, and caller ID regex control which caller identity the provider receives.



The screenshot shows a configuration panel titled "Outbound Calls" with a blue header. The settings are as follows:

- Additional Header (1):** X-Account-Code: docs-demo
- CallerID Presentation:** Auto Privacy, CallerID not screened
- Digits to Add:** +1
- Number of digits to Remove:** 1
- CallerID Modifications:** No Modifications
- CallerID allowed regex:** ^\\+1555[0-9]{7}\$
- Max outbound channels:** 12
- Penalty:** 10
- Ignore SIP Cause codes
- Ignore BUSY

Outbound Calls block with fictional caller ID and routing controls.

Max outbound channels limits concurrent outbound calls through the provider. **Penalty** is used by routing decisions and round-robin selection. The ignore options let routing continue to another provider when the current provider returns SIP cause codes or a BUSY status that should not stop the route search.

Inbound Calls

Inbound options normalize numbers before DID matching. Use these fields only when the carrier sends the destination number in a format that does not match the DIDs configured in the tenants.

The screenshot shows the 'Inbound Calls' configuration page. It features a blue header with the title 'Inbound Calls'. Below the header, there are several configuration fields:

- Inbound Digits to Add:** A text input field containing '+1'.
- Inbound Number of digits to Remove:** A text input field containing '0'.
- DID Modifications:** A dropdown menu with 'No Modifications' selected.
- Same provider IN and OUT:** A dropdown menu with 'Yes' selected.
- Allow custom extensions
- Normalize number
- Get number from:** A dropdown menu with 'SIP To: Header' selected.
- Variable name (1):** A text input field containing 'USR- ProviderName'.
- Variable value (1):** A text input field containing 'Example Telecom'.
- Variable name (2):** A text input field containing 'USR- SupportTier'.
- Variable value (2):** A text input field containing 'Gold'.
- Variable name (3):** A text input field containing 'USR-'.
- Variable value (3):** An empty text input field.

Inbound Calls block with fictional DID normalization and user variables.

DID Modifications can rewrite inbound numbers before matching. **Same provider IN and OUT** controls whether a call that arrived from this provider may also leave through the same provider. The variable fields create user variables prefixed with `USR-`, which can later be used by routing logic and dialplan actions.

Realtime Account

For SIP, PJSIP, and IAX2 providers, enable **Use Realtime Account** when the provider should be represented by realtime Asterisk configuration. The fictional example below uses documentation IP and domain values.

Realtime Account

Use Realtime Account

Transport:

RTP Encryption (SRTP):

Host:

Port:

Outbound Proxy:

Username:

Password:

From User:

From Domain:

Send RPID:

Trust RPID:

Trust Outbound for CallerID Presentation:

Qualify:

Qualify Frequency:

Codecs:

Session Timers:

Session Expires:

DTMF Mode:

Progress inband:

NAT:

Call response timer (ms) - T1:

Call setup timer (ms) - B:

RTP Keepalive:

Can Reinvite:

Realtime Account block with fictional SIP registration and media settings.

Use **Host** and **Port** for the provider address. For inbound calls, an IP address is preferred because Asterisk realtime matching cannot always rely on a hostname. Username, password, From User, From Domain, RPID/PAI handling, qualify settings, codecs, DTMF mode, NAT, timers, and RTP keepalive should match the carrier interconnection requirements.

SMS Provider Options

When the technology is **SMS**, the outbound section changes to SMS delivery settings. Choose the SMS protocol and complete the fields required by the provider. The JSON Web URL example shows a fictional REST endpoint and a JSON body built from MiRTA PBX SMS variables.

Outbound Calls

Additional Header (1):	<input type="text" value="X-Account-Code: docs-demo"/>	<input type="button" value="⊕"/>
Additional Header (2):	<input type="text"/>	<input type="button" value="⊕"/>
Additional Header (3):	<input type="text"/>	<input type="button" value="⊕"/>
Additional Header (4):	<input type="text"/>	<input type="button" value="⊕"/>
Additional Header (5):	<input type="text"/>	<input type="button" value="⊕"/>
Additional Header (6):	<input type="text"/>	
Digits to Add:	<input type="text" value="+1"/>	
Number of digits to Remove:	<input type="text" value="1"/>	
CallerID Modifications:	<input type="text" value="No Modifications"/>	<input type="button" value="▼"/>
CallerID allowed regex:	<input type="text" value="^\+1555[0-9]{7}\$"/>	
Max outbound channels:	<input type="text" value="12"/>	
Penalty:	<input type="text" value="10"/>	
SMS Protocol:	<input type="text" value="Web URL (JSON)"/>	<input type="button" value="▼"/>
SMS URL/AGI Script Name:	<input type="text" value="https://sms.example.net/v1/messages"/>	
HTTP Content Type:	<input type="text" value="application/json"/>	
User:	<input type="text" value="docs-demo-user"/>	
Password:	<input type="text" value="FictionalSmsSecret"/>	
JSON data:	<pre>sender.number:\${SMSSOURCENUM} receiver.number:\${SMSDESTNUM} message.text:\${SMSTEXT}</pre>	
	<input checked="" type="checkbox"/> Ignore SIP Cause codes	
	<input checked="" type="checkbox"/> Ignore BUSY	

SMS provider options shown with fictional JSON Web URL values.

Common SMS variables include `${SMSTEXT}`, `${SMSDESTNUM}`, `${SMSSOURCENUM}`, and `${SMSNAME}`. For JSON payloads, dot notation in the field name creates nested JSON objects.

Save and Test

1. Save the provider only after replacing the fictional values with production values from the carrier.
2. Confirm the provider appears in the provider list.
3. Create or update a routing profile so outbound calls can select the provider.
4. Use provider status, channel counters, and call history to validate registration, dialing, inbound DID matching, and caller ID presentation.

Revision #3

Created 2026-06-02 21:56:05 UTC by Admin

Updated 2026-06-03 11:11:18 UTC by Admin