Setting up SIP Peering

Mike Johnstone - 2024-04-24 - PBX, Peering, Ports and IPs

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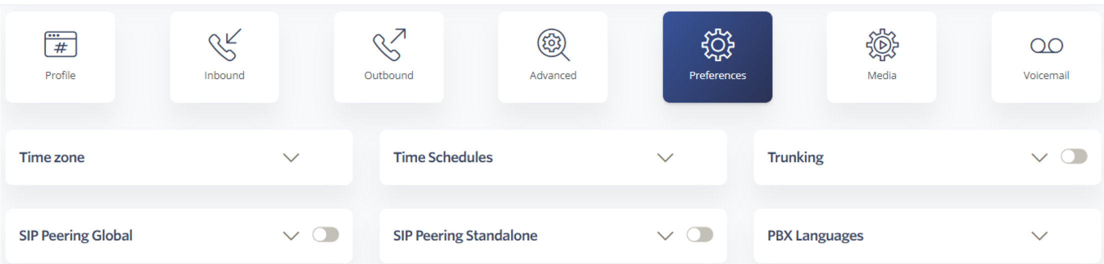
SIP Peering or SIP Trunking enables you to statically connect IP-PBX with our public sip proxy (103.55.116.65).

Note: *Peering differs from Registration which relies on an authenticated UserName, Password to connect to our voice proxy.*

Once you have enabled SIP Peering we whitelist your WAN IP blocking any other IP from communicating with our Voice service but an additional security measure from your side we advise setting a firewall rule restricting access to your SIP port to our **public IP *103.55.116.0/24.**

We support three modes of Peering:

1. SIP Peering Global
2. SIP Peering Standalone
3. Trunking



SIP Peer Global

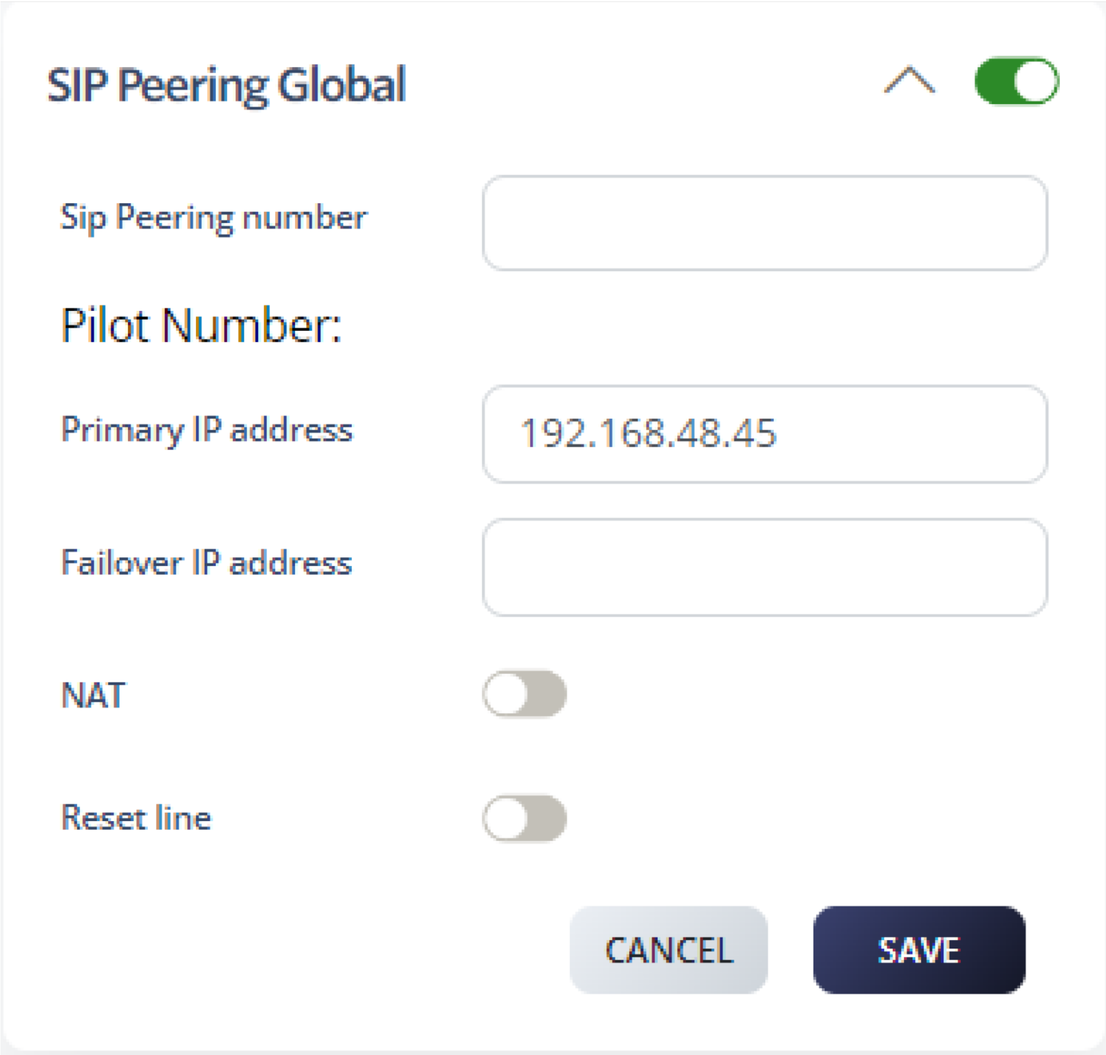
Routes all Inbound and Outbound traffic on an account to a single nominated WAN IP linked to one phone number. With the exception of Call Forwarding for emergency failover global Peering disables all functionality other CloudPBX functionality.

1. Log into your account.
2. Select **Switchboard**.
3. Select the phone number to set **SIP Peering** against.
4. Select Preferences.

5. Select **SIP Peering Global**.

6. Add Primary Trunk Host IP Address, and failover Trunk IP Address (optional).

7. Select SAVE.



Standalone Peer

A Standalone peer is where the network admin connects an IP-PBX to a single number. Standalone peering is a convenient mechanism enabling administrators to connect multiple offices each with its own WAN IP.

1. Log into your account.

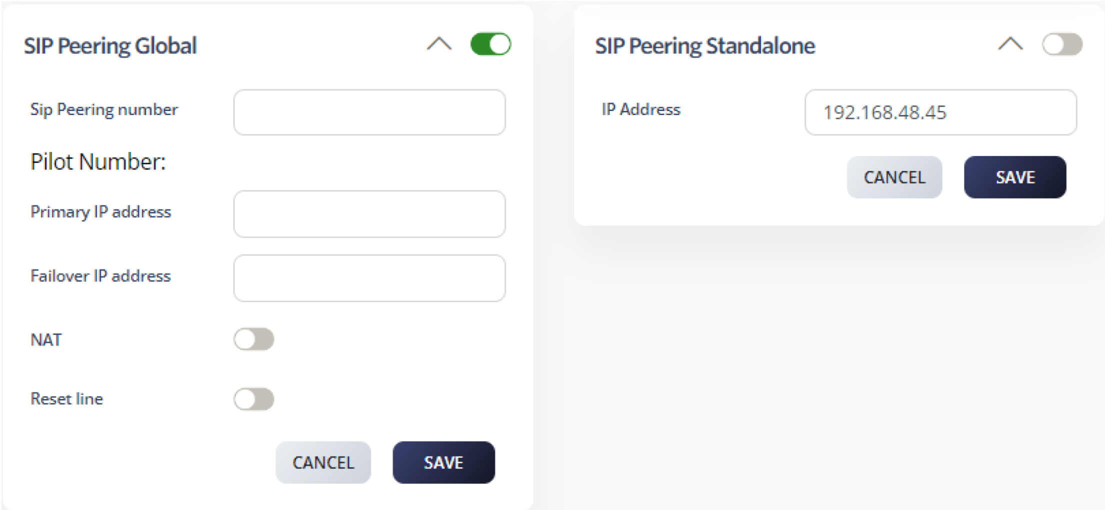
2. Select **Switchboard.**

3. Select Preferences.

4. Select Line SIP Peering Standalone > Enable.

5. Add IP Address.

6. Select **SAVE**



Once Standalone is enabled SIP Peering Global will be disabled.

Trunking

Trunking is a Registration feature that allows you to present the CLI of another number on your account, using a trunking number to remove the onerous task of individually registering large blocks of numbers to preserve CLI.

The Outbound Trunking example below will present 61289707502 as the outbound CLI while using 61289707500 as the registered trunk number:

From: "61289707502" <sip:61289707500@192.168.17.82>;tag=1c1952424
To: <sip:0450301522@192.168.17.82>

1. Log into your account.

2. Select **Switchboard**.

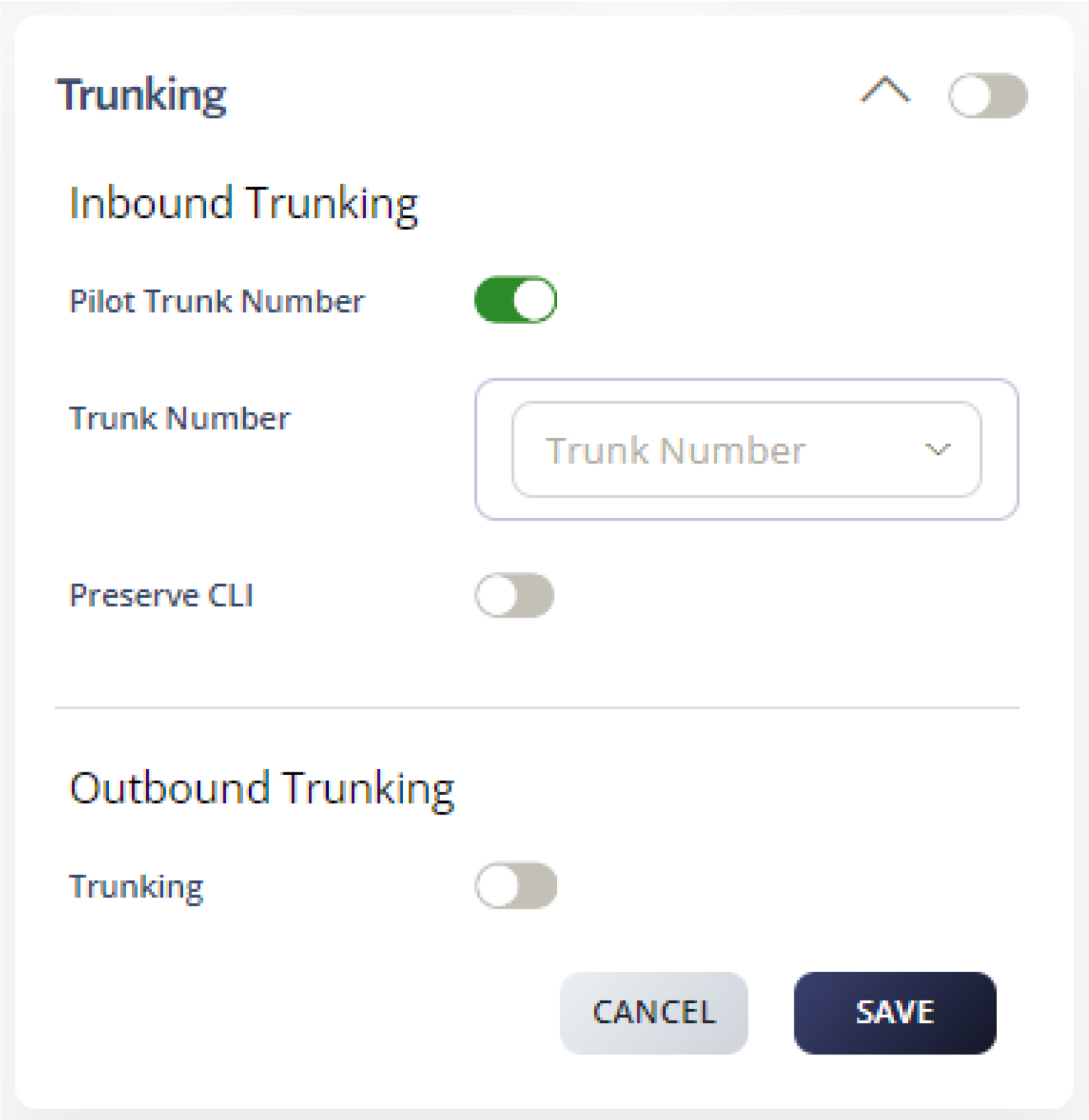
3. Select your **number**.

4. Select **Preferences > Trunking**.

5. Set **Inbound Trunking** > **Choose Number**.

6. Select **Outbound Trunking**.

7. Select Save.

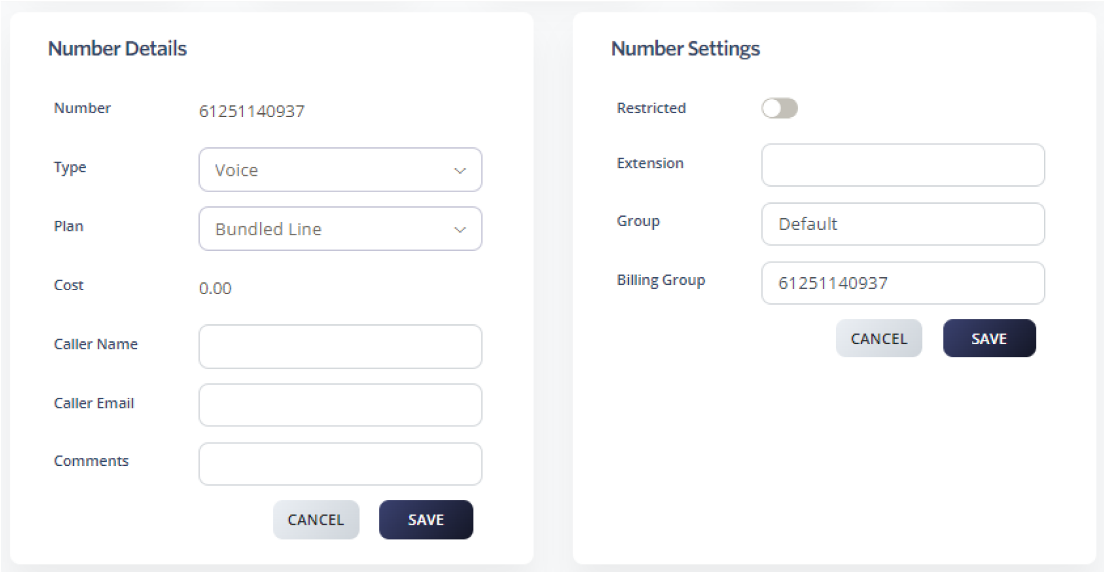


Important!

- **CallerID**: In asterisk-based PBX systems the name part can be set in the SIP configuration with the caller id= field – or if you wish to present it in the dial plan when you use the CALLER ID (name) variable. By changing this name partly to the number you wish to present on the call you can achieve multiple caller ID presentations for each DDI over a single registration or login.

- **P-Asserted-Identity**: see also a P-Asserted-Identity header (RFC 3325) to define the Caller ID as an alternative to manipulating the name field (subject to your system support for RFC 3325).

- **Groups**: Ensure that all FROM numbers are within the same 'Group' as the Outbound Trunk number.


Number Details
Number
61251140937
Type
Voice
Plan
Bundled Line
Cost
0.00
Caller Name
Caller Email
Comments
CANCEL
SAVE
Number Settings
Restricted
Extension
Group
Default
Billing Group
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