



How-To Feature Guide

SIP Peering

What is SIP Peering?

Sometimes called SIP Trunking

SIP Peering allows us to deliver your 2talk services to your SIP-based private branch exchange (IP-PBX) and Unified Communications facilities.

SIP Trunking is not for everyone and is primarily used by businesses.



When do I need to setup SIP Trunking?

To setup SIP Trunking you will need the following:

❑ A Static Internet (IP) Address.

Dynamic DNS hostnames are not allowed. You must specify an IP address in 2talk Live to act as your endpoint for inbound calls.

❑ No firewall rules or NAT blocking traffic between 2talk's servers and your IP-PBX.

You may setup firewall rules to limit traffic to 2talk's servers, or setup port forwarding if required for security reasons. See below for more details.

❑ You need a SIP (v2) compatible PBX platform or gateway.

2talk does not support H.323 or other VoIP protocols. A popular example of an IP-PBX platform is Asterisk/Trixbox, although there are a multitude of other hardware versions such as the Epygi gateways for example.

❑



Do I have to use SIP Trunking if I have a IP-PBX?

You have a choice!

Option 1: Login with 2talk on your IP-PBX

Many IP-PBX platforms can register (login) with 2talk as a normal phone would. In fact this is what many Asterisk users do using the 'register' directive in their SIP configuration.

Option 2: Inbound Trunking

If you do not have a static IP address, or your IP-PBX is behind NAT, then you cannot use SIP Trunking and you will need to register your devices with 2talk and use the 'Inbound Trunking' feature on each line instead to achieve a similar goal.





What are the benefits of using SIP Trunking?

Interconnect with 2talk

Many IP-PBX's cannot be configured to register/login to a 2talk as a client, so for these types of devices, SIP Trunking is the only way you can connect to your 2talk account. SIP Trunking has many other benefits.

Quicker Reconnection

Your device does not need to be logged in/registered, so calls will always be routed to your IP-PBX. This means that if your Internet connection was reset, or went down temporarily you would start receiving calls immediately once your connection comes back up.

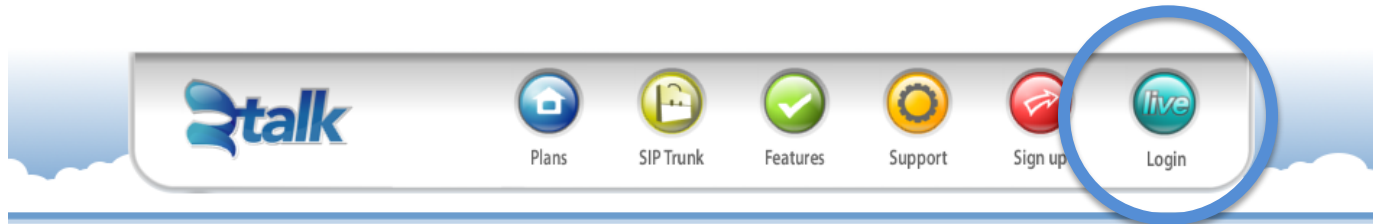
Number Routing Made Easy

You can route all your numbers easily to your IP-PBX with no further configuration required. If you add more numbers to your account then they will automatically be routed down your trunk.



Login to 2talk Live

At 2talk.com, select **LIVE** to Login







\$11.99 /mth

Sign up

Join now and save for the next 3 months

Includes unlimited plan minutes[^] including US and international destinations, unlimited 2talk calling, 2talk features, keep your phone number or add a new USA line, BYOD and save \$3/month, no contracts

[^]\$23.99 after 3 month period expires, includes telephone adapter (usual cost \$3/mth), excludes fees and tax

 Free Trial Try 2talk out now for free. Our trial includes over 50 minutes of calling to 70+ destinations.	 Plans Build a plan to suit your needs with our Business Combos. Lines start at \$12.99. We have plans for personal users too.	 SIP Trunking Make the switch to IP and save with our low cost SIP Trunks. Connect your PBX and pay as little as \$10 per trunk and 1.3 c/min.	 Free Features All services include Custom Call Forwarding, Call Recording, Locate Me, Video Calling, Auto Attendant and more
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2talk Live

Select
SIP PEERING
from the Preferences

Advanced

Call Recording

Setup your call recording options for all your inbound and outbound calls are automatically recorded or not

Group Talk

Use your 2talk number and create a room to talk with others at the same time

Remote Call Back

Call your 2talk number from any phone. When you hear ringing, hang up and you will be called back - so you can make a call from 2talk!

Remote Dial Tone

Make calls from your 2talk account from another phone by remotely dialling in to initiate the call

Call Transfers

During a call you can transfer the other person to a new number by dialing #0 for an attended transfer or ## for a blind transfer

Wake up Call

You can ask 2talk to send you a wake up call

Caller Tunes & Hold Music

Upload your own MP3's to replace ringing when people call you and setup your own music on hold

Preferences

Auto Topup & Notifications

Automatically topup your 2talk account and change account balance and call duration notification thresholds when making calls

Personal Information

Change your personal details including the name and caller ID displayed when making calls.

Extension Dialing

Setup a short extension number for each line on your account to dial your numbers more quickly

Time and Login Options

Set your preferred language, timezone and date options.

Calendar/Task Alarm Calls

If you enable an alarm on a calendar or task event 2talk will call you with a reminder. Set your options for this service here

Time Schedules

Change the default settings for time schedules such as your hours of work and available hours

Voice Quality & Networking

Choose your voice and video call quality preferences and network preferences

SIP Peering

If you have an IP PBX directly connected to the Internet you configure your 2talk account as a SIP peer (Advanced users)



Setting Up SIP Peering

Line: 092804995

2talk Settings for 092804995

Edit options for: Choose Application: [v] GO

SIP Peering << Voice Quality & Networking | Locate Me / One number >>

[Click here to learn more about how SIP peering/trunking works and whether the service is right for you](#)

NOTE: This type of SIP Trunking is a direct peering relationship, so will not work if your PBX is behind a firewall or router and behind NAT on a Private LAN. If your PBX is behind NAT then you need to register your lines and use our Inbound trunking feature instead. Do not change any settings on this page unless you are sure what SIP peering is and how it will affect your account. You cannot login to 2talk normally with a phone once SIP trunking has been enabled on your account.

SIP trunking host address: trunk.2talk.com (Do NOT connect to 2talk.com for SIP trunking)

Enable SIP trunking on this account and make this number my pilot number

Primary Trunk Host IP Address:

Failover Trunk Host IP Address (optional):

Exception route (optional)
(Instead of routing calls to the Primary trunk host IP, you may specify a different IP address for call routing for this one number)

Handle SIP trunk signalling as behind NAT (ignoring Private IP addressing in Contact/SDP and VIA headers etc.)

Do not reset my line settings when enabling my SIP trunk.

NOTE: Unless you tick the box above, the following will happen to each line on your account the very first time (and only the first time) that you enable SIP trunking:

- ALL settings on each line are reset to their default values
- Voicemail is disabled on each line
- Dial by Name feature is disabled on each line
- Call Recording is disabled on each line
- Call transfer features are disabled on each line
- Caller Tunes and Music on Hold are disabled on each line

NOTE: Once enabled, if you then disable SIP trunking again ALL line settings will remain unchanged. You may setup features on a per line basis later if required.

SAVE OPTIONS UNDO CHANGES RETURN TO OPTIONS

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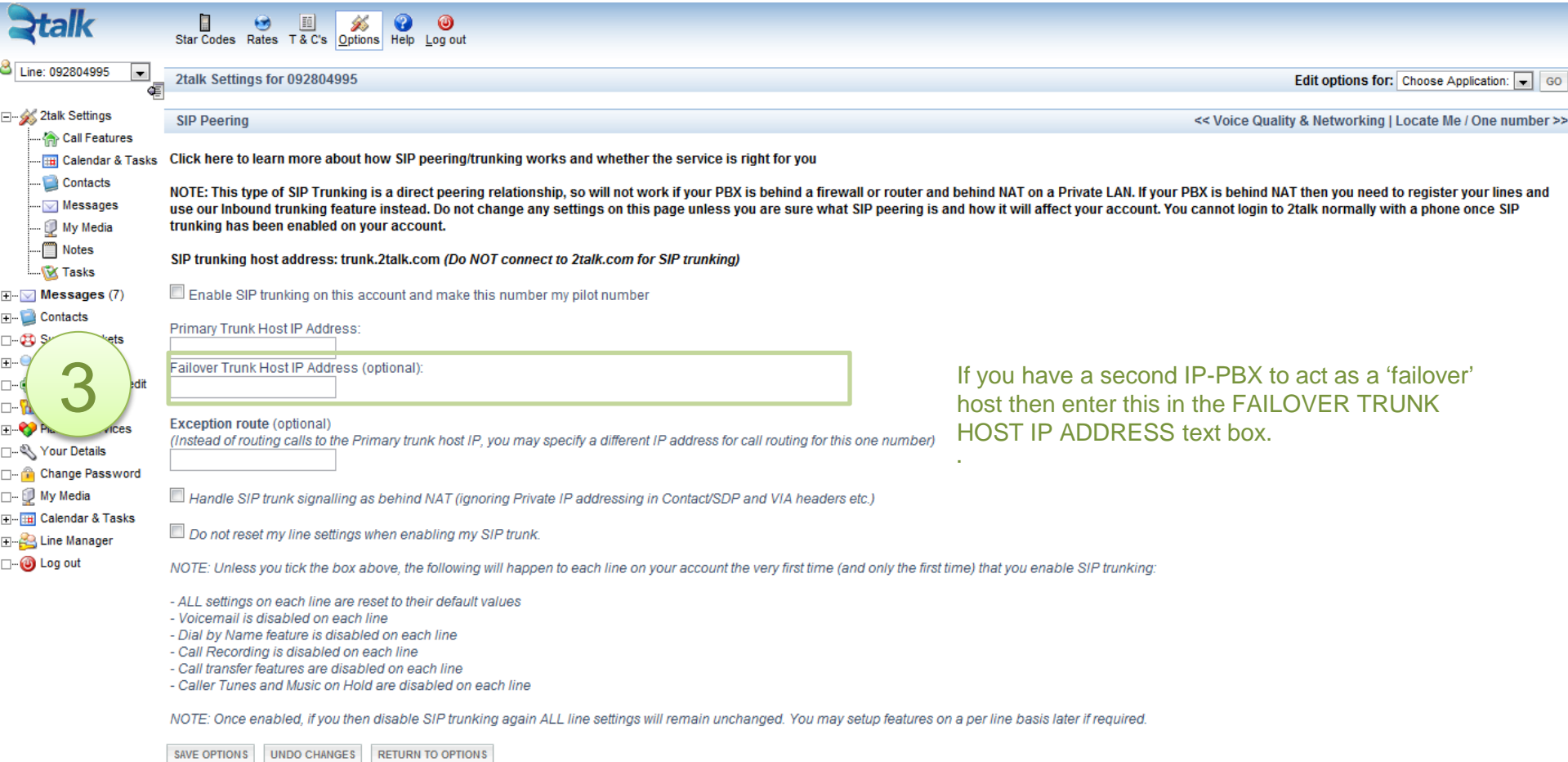
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Setting Up SIP Peering



2talk

Star Codes Rates T & C's Options Help Log out

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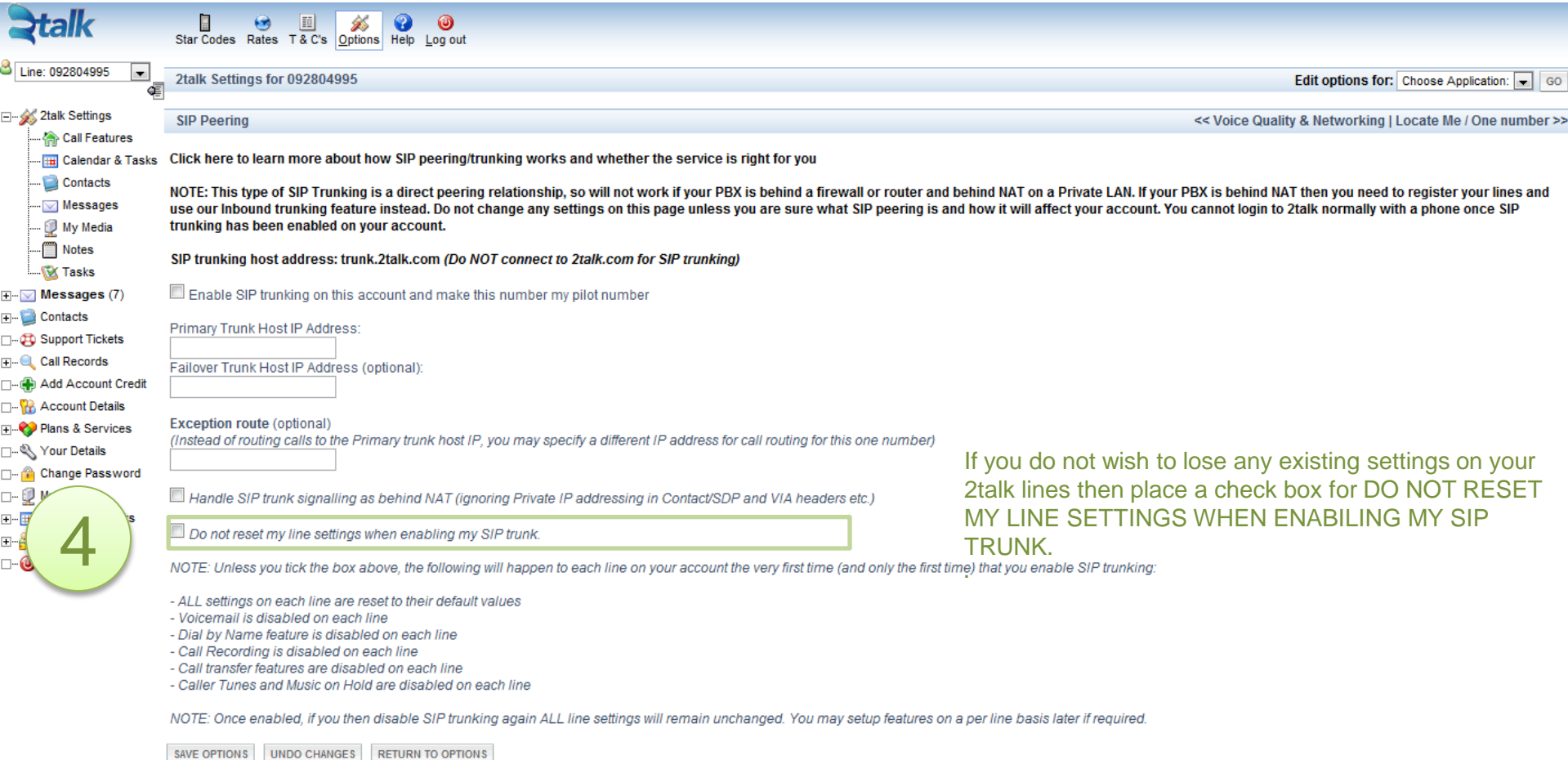
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If you have a second IP-PBX to act as a 'failover' host then enter this in the FAILOVER TRUNK HOST IP ADDRESS text box.

Setting Up SIP Peering



2talk

Star Codes Rates T & C's Options Help Log out

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If you do not wish to lose any existing settings on your 2talk lines then place a check box for DO NOT RESET MY LINE SETTINGS WHEN ENABLING MY SIP TRUNK.

Setting Up SIP Peering

The screenshot shows the 2talk web interface. At the top left is the 2talk logo. The navigation bar includes links for Star Codes, Rates, T & C's, Options, Help, and Log out. A dropdown menu shows 'Line: 092804995'. The main header reads '2talk Settings for 092804995' with an 'Edit options for:' dropdown and a 'GO' button. The left sidebar contains a tree view of settings: 2talk Settings, Call Features, Calendar & Tasks, Contacts, Messages, My Media, Notes, Tasks, Messages (7), Contacts, Support Tickets, Call Records, Add Account Credit, Account Details, Plans & Services, Your Details, Change Password, My Media, Calendar & Tasks, Line Manager, and Log out.

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ALWAYS REMEMBER TO CLICK SAVE OPTIONS.

5

Configuring Your IP-PBX to Connect to 2talk

Your configuration sometimes depends
on the type of PBX you are using.

**The main setting you need to configure
is the host or proxy address of your outbound trunk.**

You should set this to:

trunk.2talk.com



Asterisk Trixbox Users

One of the most common IP-PBX platforms in use today is Asterisk (or one of its variants such as TrixBox).

Below is an example 'SIP peer' definition for Asterisk allowing you to route calls to 2talk via your SIP trunk:

```
[2talk]
type=friend
context=default
host=trunk.2talk.com
dtmfmode=rfc2833
insecure=very
nat=never
qualify=no
canreinvite=no
disallow=all
allow=gsm
allow=alaw
```





For Assistance

Contact support@2talk.com

