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

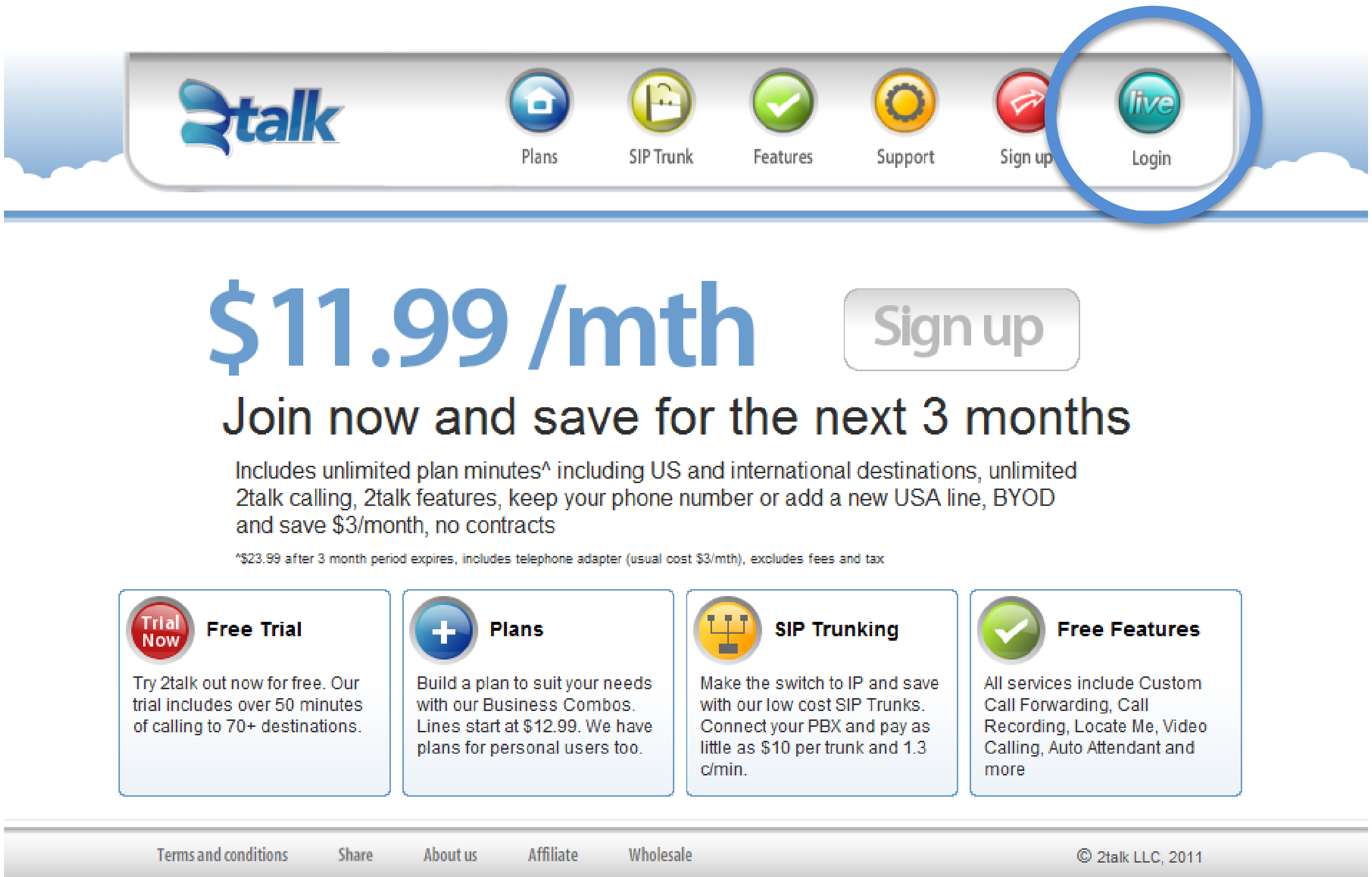
Your device does not need to be logged in/registered, so calls will always be routed to your IP-PBX. This means that is your Internet connection was reset, or went down temporarily you would start receiving calls immediately once your connection comes back up.

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



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 acount 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.

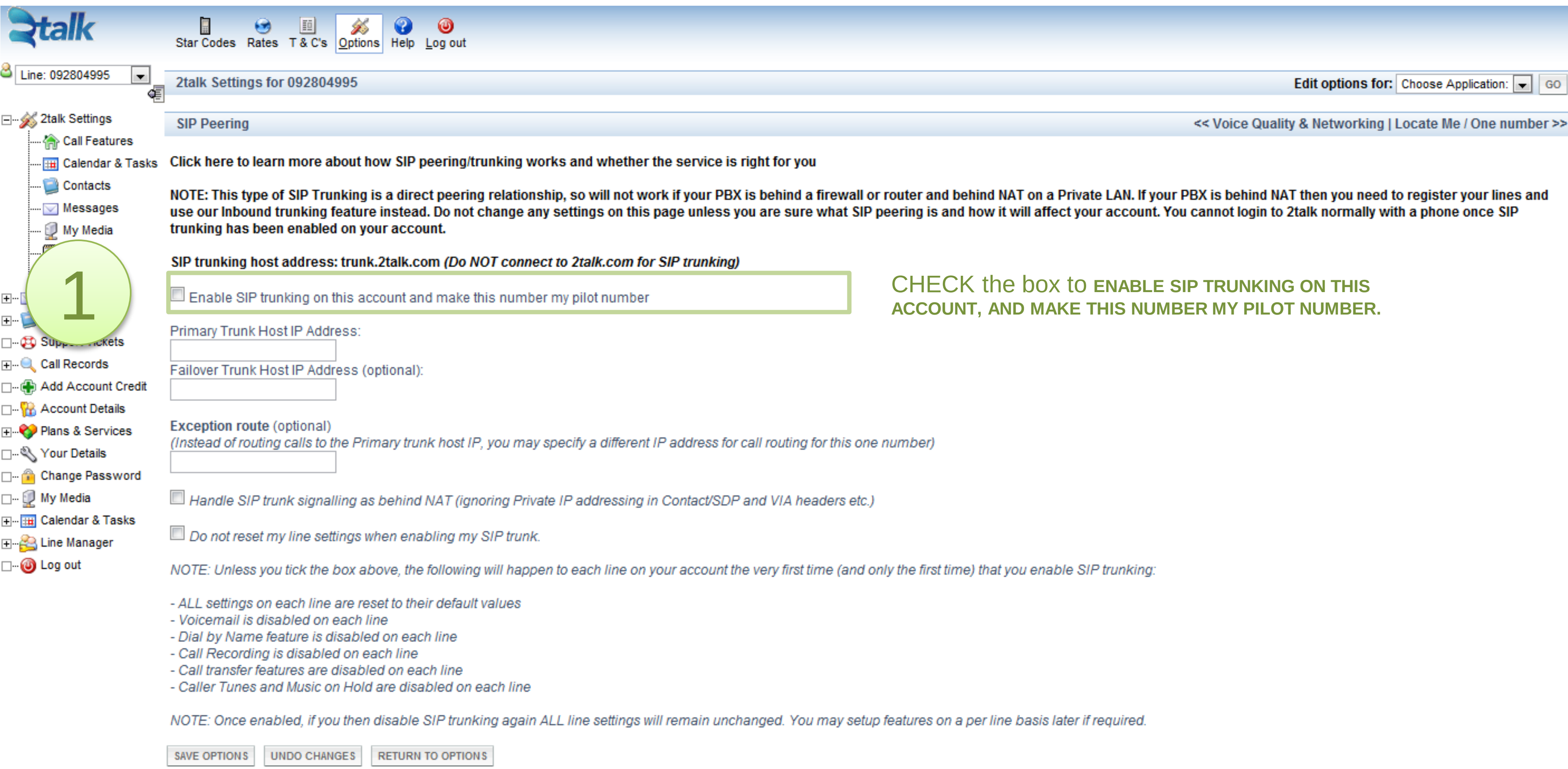
Calendar/Task Alarm Calls
If you enable an alarm on a calendar or task event 2talk will call you with a reminder. Set your options for this service here

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

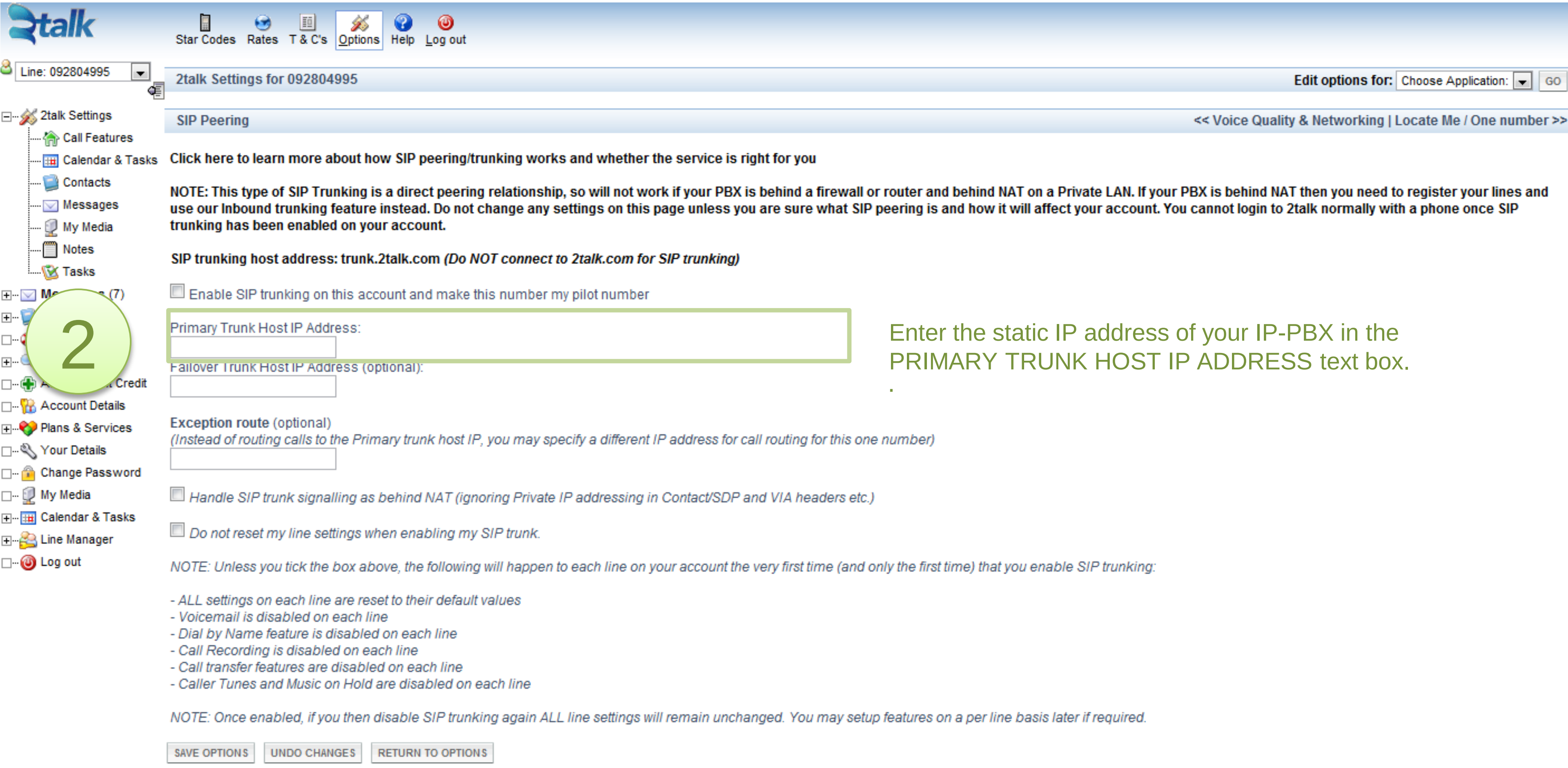
SIP Peering
If you have an IP PBX directly connected to the Internet you configure your 2talk account as a SIP peer (Advanced users)

Setting Up SIP Peering



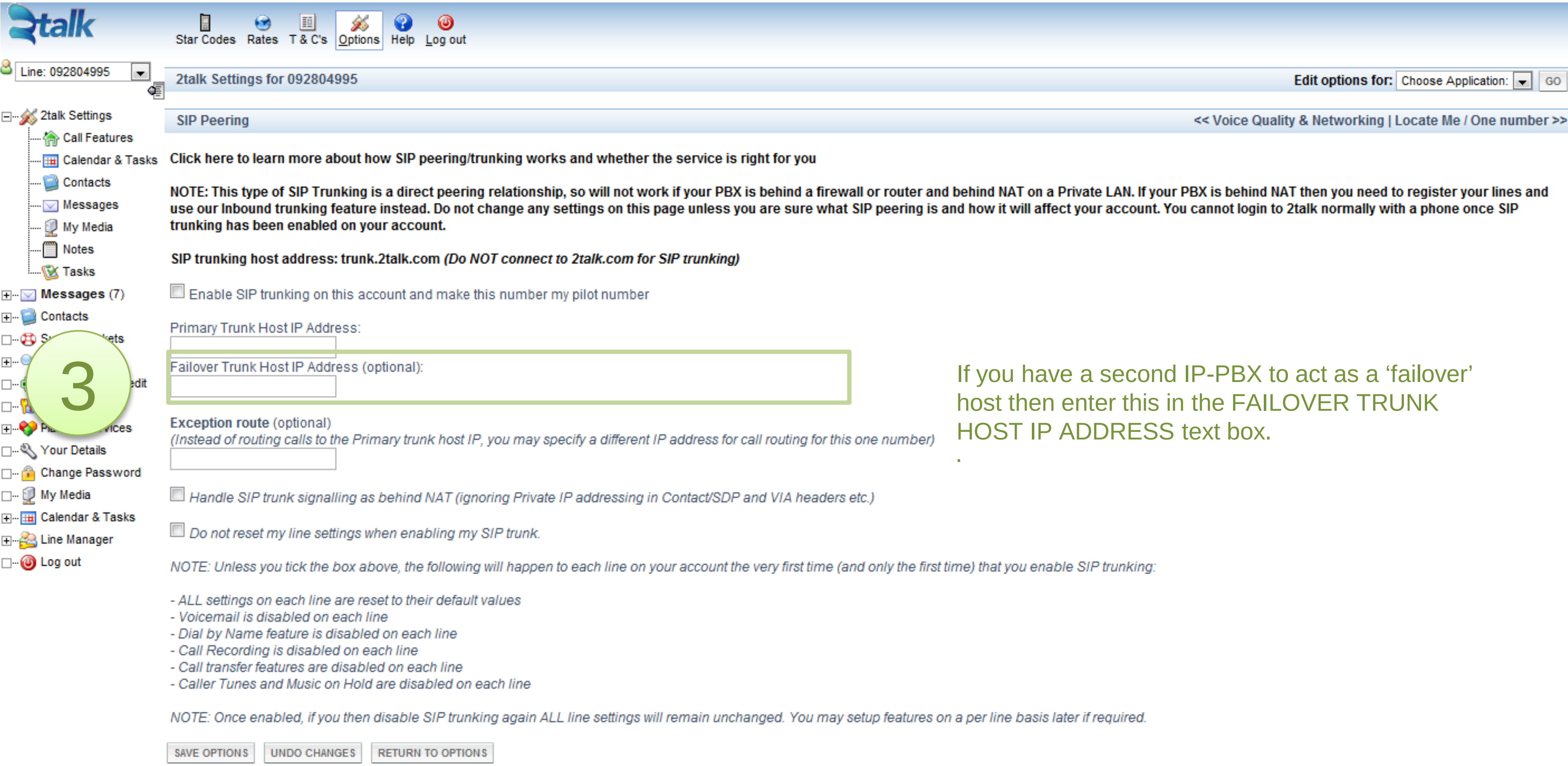
CHECK the box to ENABLE SIP TRUNKING ON THIS ACCOUNT, AND MAKE THIS NUMBER MY PILOT NUMBER.

Setting Up SIP Peering



Enter the static IP address of your IP-PBX in the PRIMARY TRUNK HOST IP ADDRESS text box.

Setting Up SIP Peering



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.

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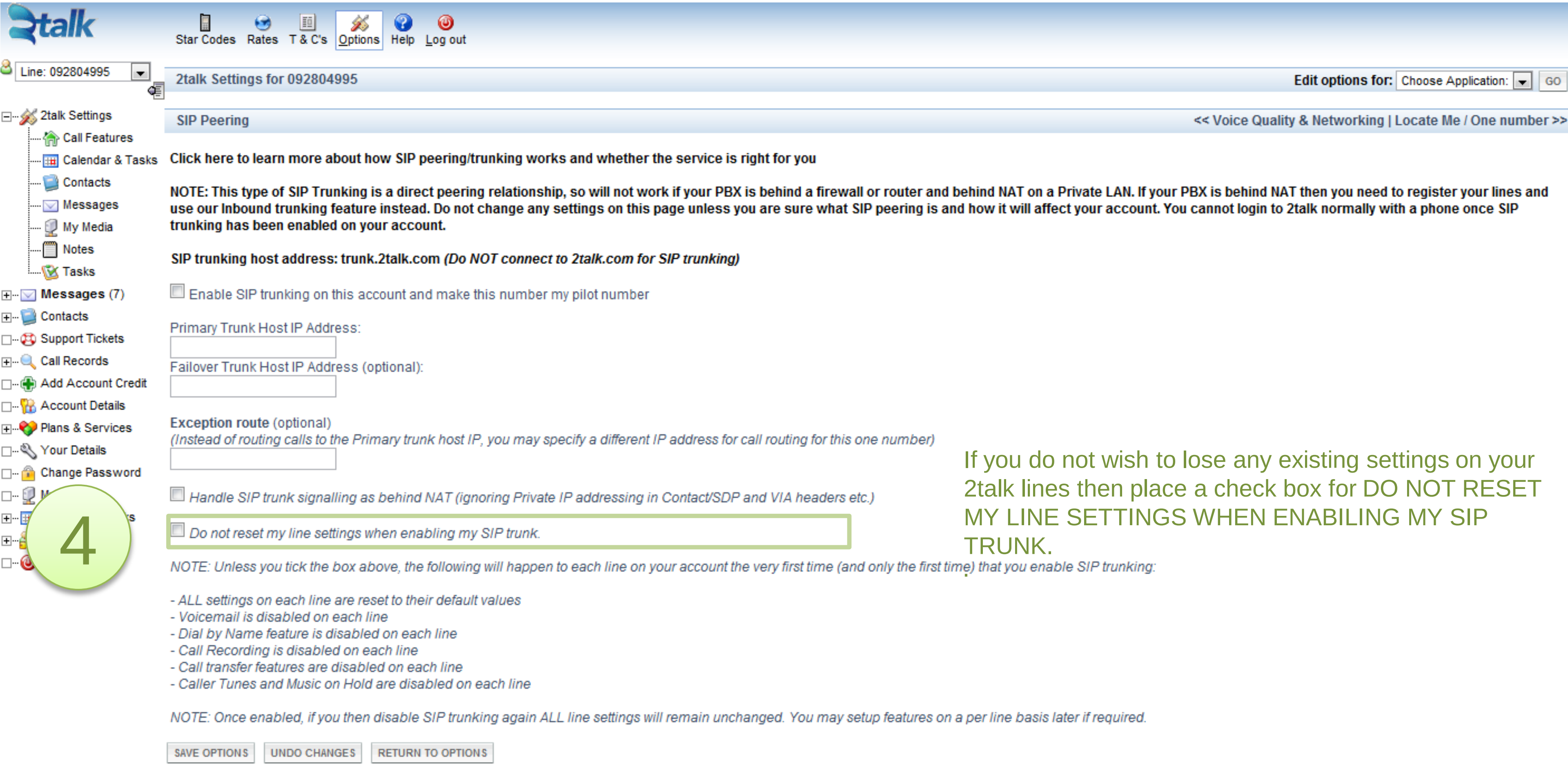
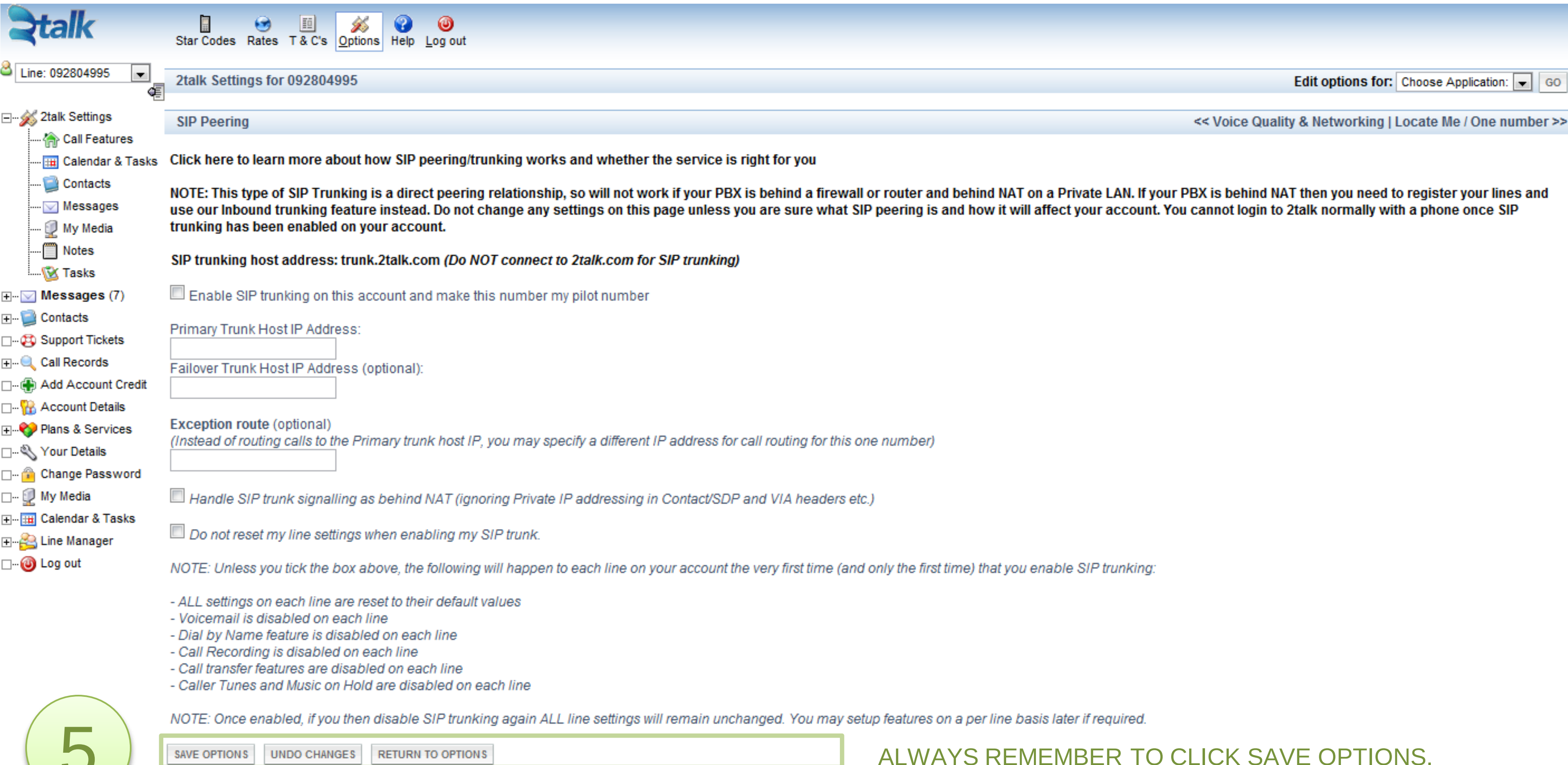
Setting Up SIP Peering



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 ENABILING MY SIP TRUNK.

Setting Up SIP Peering



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ALWAYS REMEMBER TO CLICK SAVE OPTIONS.

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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 it's 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