



Connecting SIP Devices To The TraiTel Network

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Introduction:

TraiTel Telecommunications provides SIP connectivity for our customers. Connection is free, as are calls from virtual number to virtual number.

Account Set Up:

Each SIP device requires a separate sub-account to connect. You should create as many sub-accounts as you require, these are free and there is no limit set on the number of sub-accounts. When a sub-account is created, it will be automatically assigned a virtual number. If you are only going to be making outbound calls, a virtual number is all that is required for this (as well as account credit). A Real Number is required to receive calls from the PSTN. Sub-accounts are found in your account web-messaging interface under **Settings, Sub-Accounts**.

Configuration:

If establishing a SIP connection (SIP trunk or phone), this is configured in web-messaging under **Voice, Configure SIP device**. Select the appropriate sub account from the dropdown box.

Instructions will provide the initial settings required:

SIP Server / Registrar: sip.traitel.com.au

SIP Server Port: 5060

Username / ID / Phone number: *username.1*

Password: *Your TraiTel password* – this is the main account password.

Realm: sip.traitel.com.au

Codec: G729 and GSM

STUN server: stun.traitel.com.au

In order to use G729, you must disable VAD/Silence Suppression. Failure to do so will result in poor quality calls and call drops.

Local and National Settings details dialling instructions and number set up. Changes will not take effect until your SIP device re-registers with TraiTel. Depending on the configuration of your device, this may take anywhere from a

few minutes to a few hours. We recommend rebooting your SIP device for settings to take effect immediately.

Call Timeout and Diversion

Using this form, you may configure the way inbound calls are sent to your SIP device.

The call timeout field defines the number of seconds during which your SIP device will ring when an inbound call is received. After this time limit is reached, there are three possible user-defined actions: Reject the call, send to voicemail or divert to another number. **Note:** The diversion number is entered as you would dial it on your normal telephone.

Offline Diversion: If your SIP phone is offline, you can divert inbound calls to a specific number as well. Leave this field blank to send calls to voicemail when your device is offline.

On Hold Music

Use this form to upload an audio file that will be played to callers while your SIP device is on hold. Please ensure that the uploaded file is a Windows WAV file, and that the recording bitrate is set to 8000bps, 16 bit mono. Other wave files will be converted, but audio quality might degrade as a result.

Alternatively, you may select an internet radio station from ShoutCast.org to play as on-hold music. Once you have found a radio station on ShoutCast, copy the 'Tune-In' URL to the field provided.

Checking Status:

The online status can be checked in web-messaging under **Voice, View SIP device status**.