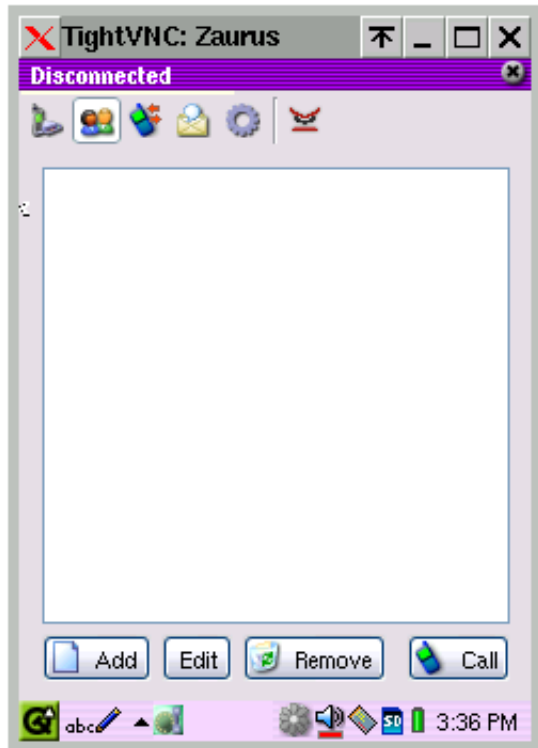








TkcPhone:

There are a few different ways to set up tkcPhone, depending on your own requirements, so I'll try to document a few possible scenarios to help you get started. VOIP is quite complex and there are many variations/implementations which are impossible to cover completely here, but this should get you started. If you are in an Enterprise situation, you may need to work with your Voice over IP specialist to get relevant information.

When you first open tkcPhone, you'll come to the Address Book screen:



## Button Explanations – Top Bar:


1.  Telephone Screen: Make calls, view logs, set volume
2.  Telephone Book: Add contact information
3.  Call Log: Incoming and Outgoing call information
4.  Instant Messenger: Send instant messages to your contacts
5.  Options: Set your configuration options
6.  Splash Screen

## Configuration Options:

### Scenario 1: Zaurus behind NAT router to Zaurus/PC behind NAT router

If your Zaurus is using NAT, you will need to configure tkcPhone to use a STUN server and Registrar/Proxy. It's best if the person you are calling uses the same STUN server and Registrar. For this example, theKompany is hosting the STUN server and Registrar.

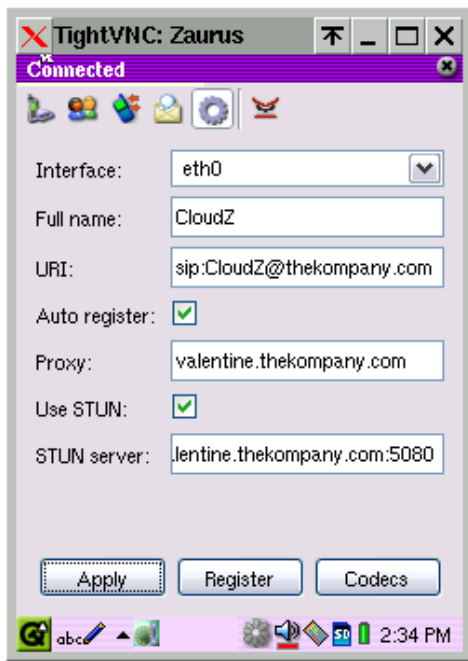
### Zaurus #1 (Your Zaurus): CloudZ

1. Tap on the  icon to go to the Configuration Options screen.
2. Tap on the drop-down box on the Interface box and change to Eth0.
3. Choose a name to use. Mine is CloudZ
4. Set your URI  
sip:Cloudz@thekompany.com (this can also be a telephone number in this format:  
sip:CloudZ@4035551212 or your ip address, sip:CloudZ@24.24.24.24)
5. Check Auto Register box to automatically register when you start tkcPhone
6. Set proxy (Registrar) to: valentine.thekompany.com (or the Registrar/Proxy of your choice.
7. Check Use Stun box if you are using NAT. Note that this may not work with some implementations of NAT. In general, if your NAT type allows SIP/RTP connection (and the NAT type of the person you're calling) - you'll get media for conversation.
8. Dec 15 12:41:17 <gonzo> for conversation I am using a linksys router and use STUN.
9. Enter STUN server URL: valentine.thekompany.com:5080 (or the STUN server of your choice. There are also other public STUN servers available).
10. Codecs: tap the Codecs button. There are 3 choices, G.729a, GSM and PCMU. At the bottom you will see an Up and a Down button. Tap on your preferred Codec and tap the Up button to move it up in the list. The top choice is the most preferred choice. If you choose PCMU, you will need to have a fairly fast connection to the Internet. If you and the person you are calling are on the same

LAN, this would be your preferred choice. If you are on a dial-up connection or slow WAN link, this wouldn't be your Codec of choice. G.729a is a good choice for both quality and bandwidth. Tap OK to return to the Configuration screen.

11. Tap the Register button to connect to the Registration/Proxy server.

Your configuration screen should look similar to this:



Zaurus #2 (The person you are calling): Configuration would be similar to Zaurus #1 configuration, with the person you're calling using the username of his/her choice and the same Registrar/Proxy and STUN server.

Notes:

For the use of tkcPhone customers, theKompany has set up a Registrar and STUN server. Choose your Username and your URI would be:  
sip:YourUserName@thekompany.com.

You can use any STUN server (there are other public ones) and SIP Registrar. tkcPhone supports Digest auth scheme and Basic auth scheme, so any SIP-compatible registrar/proxy could be use for registration.

If you use a Registrar/Proxy server and you change your location, when you open tkcPhone it will connect to the Registrar and identify your IP address and Port Number. It will also locate the person you're calling if they are also using the same Registrar. This allows you to call people without knowing exactly where they are, and they could call you without knowing where you are and without having to know your IP Address.

There are some SIP soft phones available, which are software applications that run on PC's which would allow you to call someone with tkcPhone who isn't using VOIP in an enterprise or who doesn't have a SIP phone. There are also hardware SIP telephones that would plug into a switch with an Ethernet cable.

Pingtel, [net@phone](#) windows client, eStrata (GET MORE INFO)

Scenario 2: Zaurus to Telephone (PSTN):

To call from your Zaurus to a regular telephone, you will need to use a service like Net2Phone, which would do the switching from IP to PSTN. There is a cost involved for this, but it is generally lower in cost, about 2-3 cents per minute.


Scenario 3: Zaurus to IP Phone

Scenario 4: IP Phone to Zaurus (LAN)

If the Zaurus is part of the Enterprise VoIP system, it **should** be possible to call from an IP phone to the Zaurus, but it is out of the scope of this document and you would need to work with your VoIP specialist to set this up.

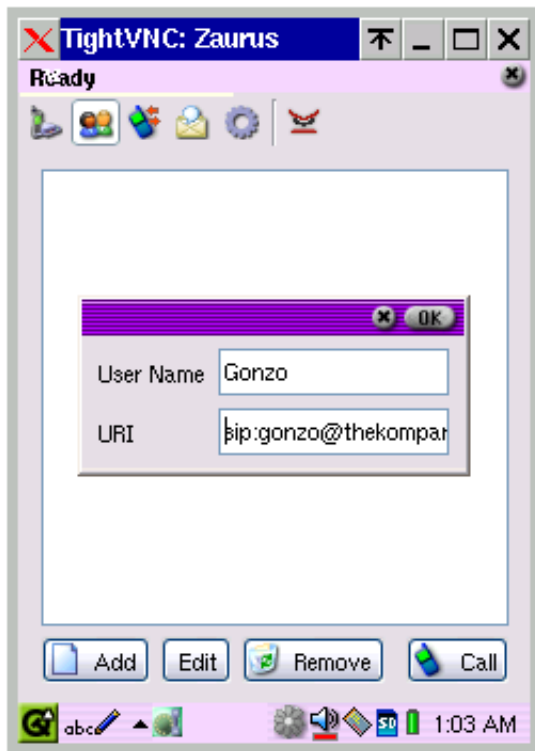
Scenario 5: IP Phone to Zaurus over Internet:

Beyond the scope of this document, but if your IP phone is a SIP phone, and there is a place to enter a URI, this should work if the Zaurus and the IP phone use the same Registrar and if there are no NAT issues.

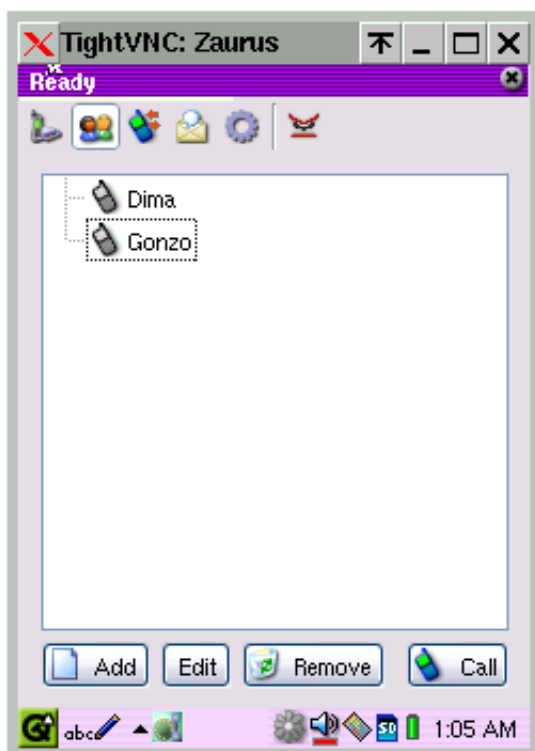
Once you have set your Configuration Options, tap the  icon to go to the Address Book screen.


Adding Contacts:

1. Tap the Add button. A small window will pop-up. Enter the contact name and URI of the person you want to call:




Tap OK. Your Contact will appear in the big white box:



Note that both contacts' icons are grey. This indicates neither contact is available for calls. The telephone icon will turn blue if a contact is online/available to receive a call, like this: 

You can initiate a call from this screen by tapping on the name of the person in your list that you want to call, then tap the Call button on the bottom right of this screen.

If you tap on the contact name, then tap the  icon, you will see the URI of the selected contact in the box at the top. Tap Call to initiate a call with that contact. This screen has 2 tabs, the Log tab and the Mixer tab.

The Log window shows call information:

The Mixer tab allows you to adjust the volume and the microphone:

o General configuration issues

'General' tab

Contains your Fullname information and allows you to create configuration profiles.

'Network' tab

All settings which affects you connection - network interface which will be used to bind on

STUN server info and SOCKS proxy server info. STUN server requires only if you're behind NAT/Firewall

Because of SIP is bidirectional protocol othe UserAgent should know your external address. It could be

obtained using STUN. You can use public STUN service provided by theKompany. Just enter into 'STUN server'

field 'stun.thekompany.com:5080' (without quotes) And check 'use stun' checkbox. If you have socks proxy

installed in your LAN - it is recomended to use it. It could be entered like 'proxyip:port'. STUN \*should\* be

enabled if you're using Socks proxy.

'iConnectHere' tab

See next paragraph "Using iConnectHere service"

'Sip' tab

Every SIP User Agent can be identified by its SIP URI it looks like sip:username@realm.here

(sip:gonzo@thekompany.com). You can use theKompany's SIP registrar service. To make it work enter

sip.thekompany.com as regsitrar and check 'auto register' checkbox. In the field URI enter yournick@thekompany.com

iLeave password field blank. Usualy this tab is filled with information which you obtain from your

SIP service provider (iConnectHere, Net2Phone etc..).

'Codecs' tab

manage your codecs here. You can check which codecs should be used and set priorities for them. Priorities

will be used for codec negotiation with your called party. Recomend ed codecs for low-speed connections are speex & G729.

Use PCMU codec if you have a LAN connection it has no compression but has better quality than the others.

'Devices' tab

Here is where you can specify devices which will be used to play/record audio. Ususally autodetected settings are ok but

some cards/drivers could cause problems or maybe you'll want to use one of two sound cards installed on your PC (e.g.

onboard instead of pluggable) Refer to your driver/card documentation for device-specific info.

o Using iConnectHere service

- a) Fill out fields on 'iConnectHere' tab with information provided by iConnect Account number & password

b) restart phone

c) enter phone number in the way it should be dialed from USA e.g.

01144XXXXXXXXX

for UK. Without spaces/plus signs - only digits.

o Using Net2phone service

Enter sip.net2phone.com as registrar and set your URI to

10-digit-code@net2phone.com. After registering you will be prompted for

your

login/password 10-digit N2P code acts as login 4-digit PIN code acts as

password(also, it could be entered in Password field in 'Sip' tab). To call

on land phone you should call to sip:9123456789@net2phone.com(for

example).

Or if it is international call sip:+38YYYXXXXXXXX@net2phone.com where

+38YYYXXXXXXXX - phone with country and city code.