Psip

Introduction

Psip is a very simple Voice Over Internet Protocol (VOIP) application. It is base on Benny Prijono PJSUA, the engine which has made this project possible. This project is a work in progress and is likely to be continually improved over time. This help file has been written to help get you up and running quickly. The main benefit of this application is it small size, around 600k.

This package was developed specifically for Puppy Linux, the small distro with the huge functionality.

There are a number of other VIOP programs that will run on Puppy Linux, Skype and Gismo to name two. Both are very good applications and have more bells and whistles than Psip however you will find Psip voice quality is excellent and the lag time is barely noticeable.

Functionality

Ok, lets take it for a spin, what can it do?

Make free voice calls from PC to PC

Make voice calls from PC to landlines or mobiles (You need an account at a SIP provder such as proxy01.sipphone.com)

Instant messaging

Leave and retrieve free voice mail messages (using proxy01.sipphone.com)

Conference calls

Listen to news and sports services

Video

There is no video functionality in Psip and there no plans to include it. Psip was designed to be small and efficient so it would be viable to include in future releases of Puppy Linux.

Interface

The interface was writtin in the GTKdialog3 scripting language. Although the language has limitation a very functional interfaces has been developed. You may notice the look will change from time to time and additional functionality will be added. We will try and keep this help file current but as you can imagine it takes a lot of work.

It is difficult to meet the requirements and personal taste of every one however suggestions are welcomed. Please use the Psip forum thread to do this.

Below is an example of one of many interfaces.

At the time of writing, this interface had not been released however, the functions are the same for all current versions. The above interface was designed to run on the EEEPC.

You will notice three colourful button, Call, Answer, Text Chat and Hangup. I don't expect these need further explanation.

There is a dial pad. You will need to use this occasionally to enter numbers when required just like a telephone.

Underneath the dial pad is a button that says Reload Buddies. The pupose of this button is to refresh the online status of your buddies. The buddies list is automatically refreshed every few mintues but if you are impatient you can manually force it to update.

You will also notice a status line that runs across the bottom of the interface/menu. This will display the last status change of your buddies list and other system information.

On the right of the buttons you will notice your buddies list. Initially this will be blank until you added buddies. It might also be blank when you first login, if so press the reload buddies button. If you know you have buddies and the list is not populated PJSUA is probably not running. More about this later.

The "?" indicates the buddy is Offline or not logged into a SIP server. This status defaults to Online When online or On The Phone when making or receiving calls. Other statuses can be selected from the configure menu.

How to make a Call

Click on the address of the person you wish to call then click the green call button. To make a call the person you are calling must be online. If they are not online and you are logged into the proxy01.sipphone.com sip server you should get the opportunity to leave a voice mail message. You need to set up an account on the server to use it.

How to answer a Call

You will hear some incoming beeps and a little box will pop up on your screen. The box has two buttons, answer and hang up. You may also answer the call from pressing the button on the main interface. Wait about a second and start your voice conversation.

You can close the popup window if you like with the x in the top right hand corner of the box. The reason this function is duplicated is to allow you to have Psip minimized while working on something else. If a call comes in you simply press the answer button. It saves you having to maximize Psip to answer the call.

How to Hangup

Press the red Hangup button

How to Text Chat

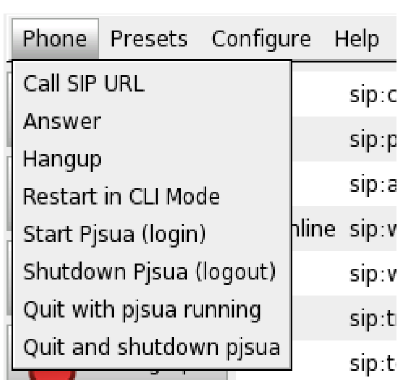
This button allows you to send instant messages to anyone in your buddies list. They have to be online to receive them.

To send an IM or text message, click on the person in the buddies list then click on the Text Chat button. You will notice two windows will pop up. One to type into and the other to receive information in. This will also pop up the same windows on your buddies computer. You may chat to more than one person at a time however all chats are private. You only have one receiving window and all of the information will be displayed in that window. It does identify who the message is coming from at the start of each message.

Only 128 characther can be typed into each message.

Drop Down Menus

At the top of the interface you will see four drop down menus.



The first one, Call SIP URL, allows you to manually make calls when the number or address is not

in your buddy list.
You can also answer and hangup calls from here if you wish.

Restart in Command Line Interface (CLI) mode, this allows you some additional freedom. The interface will not respond to PJSUA and all commands need to me entered via the CLI. This can be a little tricky if you are not used to working this way thats why the interface was developed.

The next four commands may disappear in future releases but they are very useful.

I guess this would be a good time to explain the relationship between PJSUA and Psip. As I said earlier in the document, PJSUA is the engine and can be run from the CLI. Psip is the graphical interface that make it easier to drive PJSUA. As you can see from the above menu it is possible to run PJSUA and Psip independantly.

The main command to remember here is the last command, Quit and shut down pjsua. This command closes both PJSUA and Psip. If you close the Psip interface with the x in the top right corner only Psip will be closed and PJSUA will continue to run. Apparently this is a limitaion of GTKdialog3. The simplest way to close PJSUA if you have already closed Psip is to start Psip again from the start menu and this time select Phone> Quit and shut down pjsua. It will report that PJSUA is already running and won't start a second session.

The next menu item is Presets.

The Gizmo Voicemail option allows you to leave and retrieve voicemails. This is where you will need to use you dial/key pad.

The second option is conference. This option will allow you to initiate or join a conference call on the Gizmo SIP server. As stated earlier you need an account to do this.

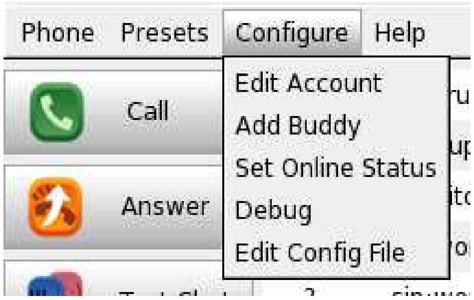
Click on conference and follow the directions.

The maximum number of people experienced in a conference thus far is 4. This may be a limitation of the server.

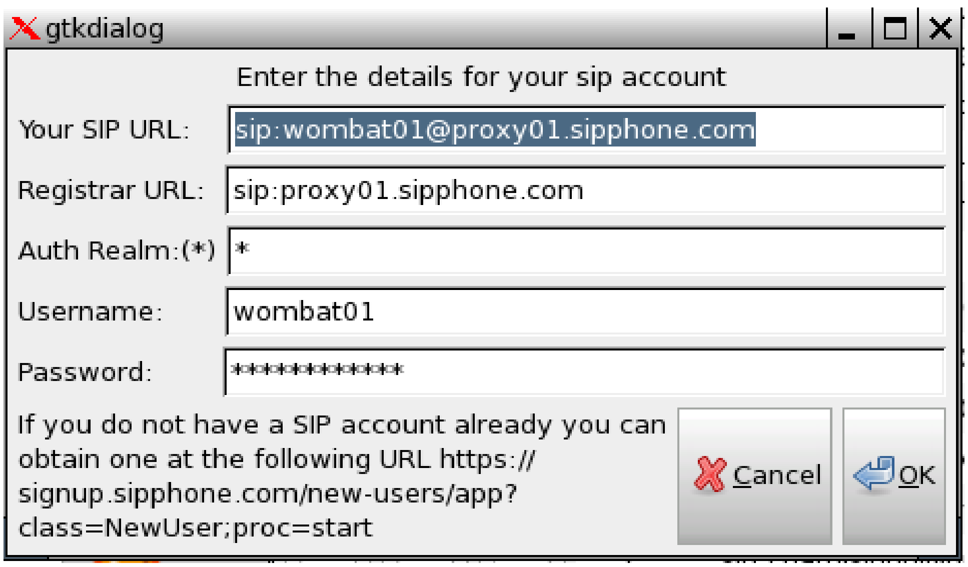
The third option is a news service. It allows you to hear daily news and sports results ETC.

The commands are voice activated and it can sometimes be difficult for the system to recognise you voice commands. It's pretty cool nevertheless.

The next menu items is Configure.



Edit Account allows you to edit your account. This is the account you use to log into your sip server.



Add Buddy allows you to add Buddy's to your Buddy list. The results of both of these functions are saved in you config file.

Set Online Status allows you to display how you would like to be seen.

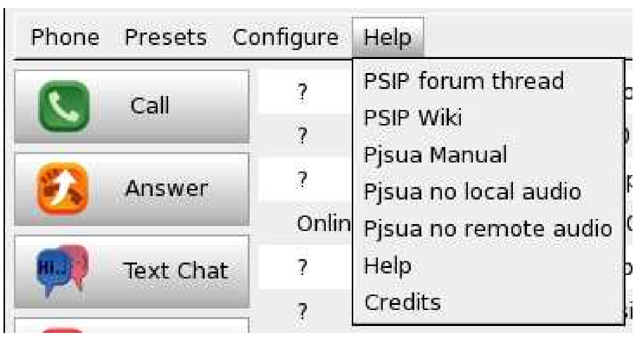
Click on your choice the one you would like to display.

Debug is really designed to help us solve problems.

The menu has a number of items that are useful but the Build Crash Report is the most useful to the developers.

The Edit Config File allows you to make amendments to you configuration file. The file will be preserved when upgrading to later versions of Psip. You need to close PJSUA and restart it again before the changes will take effect.

The final menu item is Help. You need to be connected to the internet to use Help which is not really a problem as you also need to be connected to use Psip.



The first option takes you to the Puppy Forum and the Psip thread. This is a good place to gets updates on new releases. It's also a good place to ask questions.

The Psip Wiki is another useful link to gather information.

The Pjsua Manual option will take you to another web site. It's worth a look if you would like to learn more about Pjsua.

The next two option provide advice on issues with audio.

The help option displays Help.

The Credits option will give you an idea who developed and tested this baby.

This VOIP package was put together in under a month.

Have fun.