

## Softphones for SIP, H.323 and IAX2

# Give me a virtual call

In order to enter into the world of Internet telephony, you don't need to start out with extensive hardware. A softphone is enough. A small program allows Voice over IP (VoIP) conversations and is happy with a sound card, microphone and external speakers. The only thing to worry about is the supported protocols. The two most important ones are H.323 and SIP, which are also used by VoIP providers, who route their customers to conventional or mobile telephone networks.

Up to now, no softphone exists that supports H.323 as well as SIP. However, there are programs that don't know what to do with either, although they are designed for VoIP telephony. For example, the PBX software Asterisk uses two of its own protocols: IAX and IAX2.

Linux magazine took a look at seven different VoIP user agents and put each to the test. Software for H.323, SIP and IAX2 was considered. An overview of all softphones, including download links, is provided in Table 1.

We used two slightly ancient PCs for our lab environment (Celeron-433, Pentium III 800). As is the case in many companies, machines of this type are still faithfully fulfilling their duty as desktop workstations. The machines had a SoundBlaster 32 PnP, and AC97-compatible on-board chip as sound devices. We used a cheap microphone as an input device and ran the output through a standard PC headset. This configuration is typical of many office and home PCs.

## Bonephone

Bonephone, the IP telephone, is still at the alpha stage, although this is not immediately apparent when running the application. The SIP client's GUI front-



original photo: www.photocase.de

Softphones are the inexpensive way to make a phone call using the Internet. The hardware is already sitting in each PC and besides that, virtual telephones are simple to configure. **BY JÖRG REITTER**

end is Java-based. It is kept simple and lets the user make only a few configuration changes. One outstanding thing is the ability to specify a certain bandwidth for up or downstream in order to optimally configure the VoIP communication for your existing Internet connection. For this, Bonephone supports data rates of 33.6 KBit/s for modem connections up to a comfortable 1 MBit/s (see Figure 1).

The Bonephone install is quite fast. However, you need to be aware that the application assumes version 1.4 or later of the Java development kit (JDK). In addition, the configuration package, which is available as a separate

download, needs to be put into the correct place in the user's home directory. There are two README files to help with the installation and configuration. These are available in English, but stick to the basics and lead the user step-by-step through the necessary tasks.

Four files need to be edited for the configuration, all of which are in the user's home directory. At no time are root privileges required.

The user first needs to set the paths to the JDK in the file `~/JAVAHOME`. Then the main settings can be defined in `~/bonephone/bonerc`. Here is where you configure the location of the `rat-4.2.20` media engine, which takes care of Bonephone audio processing. In addition, `bonerc` includes personal data like the SIP identity (user name in SIP URLs) and the IP address of the local computer. All the other settings can remain untouched for the time being.

Next, we continue in the main directory `~/bonephone` with the file `configfile`. In this file, the Fully Qualified Domain Name (FQDN) is initially set. The FQDN is composed of the hostname and domain name, for example `pc1.domain.local`. The local IP address likewise needs to be set. All other settings should remain untouched.

For most users the configuration is now complete and the IP telephone can be started from the main directory with the `./run` command. If you have two network cards with a public and a private IP address, or use IPv6, you will need to modify `/etc/hosts`. This is important, since Java

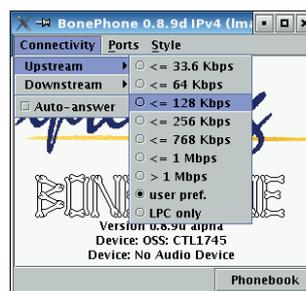


Figure 1: Bonephone allows users to individually configure the Internet connection bandwidth.

```

joe@lmag:~/softphones/Cornfed/cornfedsipua> ./sip
Jun 10 08:41:25 Cornfed SIP User Agent
Jun 10 08:41:25 Version 0.2.6
Jun 10 08:41:25 Copyright (C) 2004 Cornfed Systems
Jun 10 08:41:25 Written by Frank W. Miller
Jun 10 08:41:25 full-duplex soundcard operation
Jun 10 08:41:25 local URI: [sip:100@192.168.1.103]
Jun 10 08:41:25 userinfo [100]
Jun 10 08:41:25 user [100]
Jun 10 08:41:25 hostport [192.168.1.103]
Jun 10 08:41:25 host [192.168.1.103]
Jun 10 08:41:25 port [5060]
Jun 10 08:41:25 remote URI: [sip:613@fwd.pulver.com]
Jun 10 08:41:25 userinfo [613]
Jun 10 08:41:25 user [613]
Jun 10 08:41:25 hostport [fwd.pulver.com]
Jun 10 08:41:25 port [5060]
Jun 10 08:41:25 registrar URI: [sip:276140@fwd.pulver.com]
Jun 10 08:41:25 userinfo [276140]
Jun 10 08:41:25 user [276140]
Jun 10 08:41:25 hostport [fwd.pulver.com]
Jun 10 08:41:25 domain [fwd.pulver.com]
Jun 10 08:41:25 host [192.246.69.223]
Jun 10 08:41:25 port [5060]
Jun 10 08:41:25 echo IP address: [192.168.10.6]
Jun 10 08:41:25 echo subnet mask: [255.255.255.0]
Jun 10 08:41:25 www.myipaddress.com: [12.148.220.166]
Jun 10 08:41:25 visible IP address: [217.232.160.137]
sip>

```

**Figure 2:** By means of the detour to *myipaddress.com* (next to last line), Cornfed determines the official IP address of the the NAT gateway (the visible address).

might otherwise use the wrong IP address for VoIP communication. In addition, Suse users should change the localhost entry in the *hosts* file according to the instructions in the *README.config* file. Otherwise, the SDP transmits the local host address as the official IP address, which makes communication impossible.

Bonephone is a good place to get started, as it is easily set up just by following the the instructions. The individual bandwidth specification feature additionally opens up SIP telephony to modem owners. Contacts are stored in a simple telephone book using URL notation. Bonephone logs all calls in a text file, which, by default, is in the working directory of the SIP user agent. Bonephone is provided for download under the GPL as binary or source packages.

## Cornfed

Cornfed comes as both a GUI version and a command-line client, which only supports SIP. A user manual in pdf format shows the configuration as well as how to call using the console.

Cornfed is available only as a binary package, which, in addition to the documentation and the license, contains only the executable. This can be launched directly from the directory in which it is unpacked. One outstanding function of the Cornfed user agent is the automatic determination of the official IP address, in case the user is behind a NAT gate-

way. For this, the client contacts the website *myipaddress.com* (see Figure 2).

However this is not the only address with which Cornfed connects itself unmasked. The SIP Phone contains a Trojan horse, which independently sends information to the Cornfed developers. The developers nevertheless refer to the existence of the data collector in the User Guide and the FAQs on their web site. According to these, the user agent transmits only warning and error messages to a Cornfed web server. However, this function cannot be deactivated.

The first time we launched the GUI on our lab system, it failed as Cornfed was missing the *libpopt.so.0* library. The newer version *libpopt.so.1* was already in */usr/lib* and thus a softlink fixed the problem with *ln -s*. The GUI showed up, but now the user was missing a few things. Since a GUI is only available with the current version 0.3.0, it does not yet contain all the options that the command-line client has. However, the first test with the trimmed-down GUI ran positively.

In general, the softphone is stingy on configuration options. You can disable NAT, or select the desired network interface, but not much more. Currently the only audio codec Cornfed supports is G.711-muLaw. On the upside, the user can produce DTMF tones, which can be used, for example, to query an answering machine.

The documentation, which mainly deals with the details of the command-line interface is very good. The GUI is easy to use as it is extremely concise, lacking only and address book. Thus, users will need to create their own contact database.

Cornfed is designed for users who do not want to waste time on the configuration, but do have experience with the command-line tools. Very positive is the support both for ALSA and for OSS compatible sound cards. On the downside, there is only

one audio codec. Considering the fact that Cornfed can only be had as a binary, which happens to contain a Trojan, we would hesitate to recommend the VoIP client.

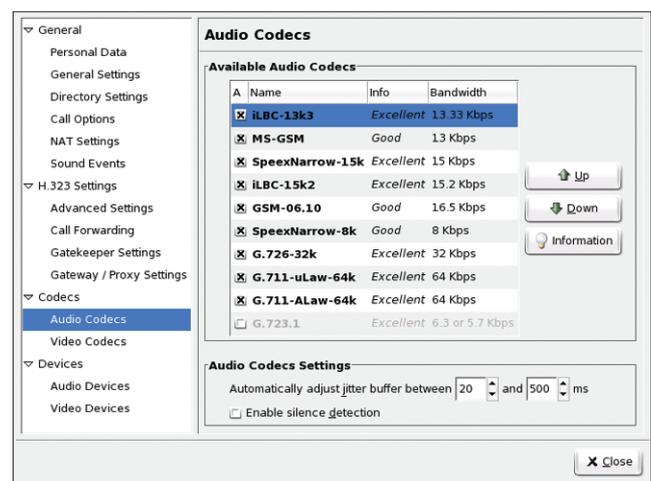
## GnomeMeeting

GnomeMeeting comes with a reputation of being an Internet telephony classic that has dared to make the jump to version 1.0 in the meantime. It was the only VoIP Client in the test field suitable for communication using H.323. SIP support is planned for release 2.0. GnomeMeeting has several outstanding characteristics that make it equally interesting for beginners as well as professionals.

In addition to voice communication via VoIP, the program allows you to hold video conferences and text chats. A comprehensive workshop can be found in Linux User [1]. Those who want to jump right in without tinkering around can launch the built-in configuration wizard.

Installation is very simple, since each of the major distributions includes GnomeMeeting. If you would like to look at the current version 1.0.2, you can download source code or binary (RPM or DEB) from the website. Conveniently, *gnomemeeting.org* also hosts the current source packages for the required libraries, H.323 and Pwlib, which need to be installed first.

This communication genius supports a substantial set of audio codecs that allow users to get the best quality out of their Internet connections. The appropriate configuration window immediately



**Figure 3:** Spoilt for choice: GnomeMeeting has audio codecs to suit any kind of Internet connections, no matter what bandwidth you have.

shows the used bandwidth (see Figure 3). GnomeMeeting uses plug-ins to for audio stream management, and can handle both ALSA and OSS compatible sound cards.

Calling from a local network that uses NAT, is just as easy as with the Cornfed client. GnomeMeeting also has an external service that determines the public IP addresses of the routers and registers them automatically in the configuration. In addition, the client must be registered in the ILS (Internet Locator Service) database of *seconix.com* and the *IP Address Translation* option must be enabled. You need to configure appropriate port forwarding rules at the firewall to support this. Check the FAQ on the GnomeMeeting Web site for more details.

If you use a gatekeeper service, you can also use GnomeMeeting to call normal phones. However, this assumes special hardware like a Quicknet or a Creative VoiceBlaster card. The Gatekeeper serves administrators as a security point on the border between LAN and Internet, where users are required to supply their name and password to authenticate.

It is easy to call other VoIP telephone users. A local address book allows you to sort contacts to be sorted by group, and provides speed dialing. Users need to register with the ILS database to find other GnomeMeeting or MS NetMeeting accounts. They can also be accessed through the address book.

GnomeMeeting is recommended as a mature application that does not leave anything to be desired in terms of H.323 protocol-based VoIP telephony. GnomeMeeting lends itself to intuitive use, and the option to implement a videophone makes the program extremely attractive. When work on implementing SIP is completed, GnomeMeeting might well move up to take the top notch in VoIP softphones.

## IaxComm

The custom-made iaxComm softphone is well-suited to the Asterisk PBX server's proprietary IAX2 protocol. The program lacks a choice of protocols, but it compensates thanks to good functionality and extensive platform independence. IaxComm works equally well on Linux, MacOS X or Win32, as guaranteed by the wxWidgets Framework [2].

The installation on smaller machines can take a while, as the wxWidgets sources need to be recompiled as static libraries and with XRC (XML-based resource system) support. On our lab machine, the install took about an hour. Later, the software run without any further ado, provided we used the binary version.



**Figure 4: The IaxComm GUI is well-organized. There are two rows of buttons for speed dialing, bottom right.**

IaxComm offers all of the facilities one expects from a graphical VoIP phone (see Figure 4). The GUI is full-featured with buttons for dialing, holding calls, and speed dialing. You can use the dial pad to produce DTMG tones, allowing you to connect to a voice mailbox.

The address book is accessed from the options menu, however.

The user selects an account from the drop/down menu. Iaxcomm can connect to multiple servers at the same time and handle up to 100 connections. The GUI displays a list of active lines. For security, IAX2 implements the MD5 challenge/response method MD5 for authentication purposes. Users can require security for internal connections and assign passwords through the program settings.

Automatic gain control, echo cancellation and noise reduction can all be enabled via checkboxes. According to the help file, the GC and echo cancellation should not be run together as this will degrade the output quality significantly.

Unfortunately, iaxComm is limited to the IAX2 protocol, which won't help it to become widely accepted. However, if a company is already using Asterisk as its telephone system, it does get a simple-to-use client free of charge.

## KPhone

The KPhone, SIP telephone is similar GnomeMeeting, being built on the Qt library. It also provides video conferencing client facilities, assuming that a pre-installed VIC package [3]. If you intend to use a distribution version of KPhone, make sure that you have some experience of SIP telephony. You will search in vain for the documentation, and the KPhone Web site is a gaping hole. This is a pity actually, as the program has everything you need to blast off into the world of VoIP.

During our test, two versions came into play: release 3.13 from the Suse Professional 9.1 distribution, and the current source code version 4.02, which is available for download from the website. To compile KPhone 4.02, you need the Qt3 libraries, which should already be on your system, assuming that KDE 3 is your standard desktop. The compilation of the sources ran as expected without problems, although that didn't stop version 4.02 displaying half a screen full of configuration help.

Once started, KPhone ran extremely quickly, offering an easy-to-use graphical interface (see Figure 5).

When the program is first launched, a configuration window opens to prompt the user for critical data like the user and host portion of the SIP URL. In addition, users are prompted to specify whether they have a proxy connection, and whether KPhone should automatically register. After completing these steps, you can fine tune the details in the *Preferences* menu.



**Figure 5: KPhone makes life easy for newbies. You can produce DTMF tones at a push of a button, and use them to query an answering machine.**

The KPhone user is typically busy with two windows: the main window shows the connection status and lets

the user configure whether he or she is on-line, or does not want to be disturbed at the moment, among other things. You can use the right mouse button to access actions such as *Call* or *Send Message*. The call window has a dial pad available for placing calls, or transferring calls to another URI.

There are numerous configuration options. KPhone was the only GPL program in our test that supported a STUN (Simple Traversal of UDP through NAT) server. This allows you to run a SIP Client behind a NAT gateway. On request, the STUN server assigns the client ports on the public IP address for incoming and outgoing communication packets. However, this function is not recommended in connection with symmetrical NAT, since this special kind of port mapping opens up a huge security hole. More information on STUN and the safer successor TURN, among other things, can be found here [4].

In the audio settings, the user can choose between three codecs (G.711u, GSM, iLBC) or change the ring tone. The SIP configuration defines whether KPhone should reject calls from unknown URIs. Besides that, it can reroute incoming calls to a specific URI, which is handy for users who desk hop.

With the exception of the missing documentation, KPhone makes a good impression. Although the option to use STUN is insecure, it is often the only possibility to access phones behind a NAT gateway. And video conferencing support makes KPhone stand out against most other softphones.

### SIPset

SIPset is a simple, but barely usable client. The softphone supports just one codec (PCMU) and is thus very restricted. However, if the user agent is used on both sides in peer-to-peer mode, it is entirely sufficient.

It is too bad that SIPset does not have a bigger choice of codecs, as most other softphones fail to provide a similar range of functions and options. The VoIP client can be launched either in graphic mode (see Figure 6) or in text mode. The program supports both OSS and ALSA audio plug-ins. Even support for IPv6 is integrated.

In order to place calls in a VoIP provider's network, you only need to register your proxy server's IP address along with the appropriate password. The provider proxy registration can be individually re-configured if it expires.

SIPset is also suitable for video telephony. To support this, the program needs the MPEG4IP package, which is available at [5]. Among other things, the Open Source kit contains MPEG-4 and H.261 codecs for video applications, as well as AAC and MP3 codecs for compression/decompression of audio streams.

**Table 1: Softphones - Feature List**

Program	Bonephone	Cornfed	GnomeMeeting	iaxComm	KPhone	SIPset	TkPhone
Version	0.8.9d	0.3.0	1.0.2	20040228	4.0.2	1.5.0	1.0.5
Web	<a href="http://www.iptel.org/products/bonephone/">www.iptel.org/products/bonephone/</a>	<a href="http://www.cornfed.com/products/index.html">www.cornfed.com/products/index.html</a>	<a href="http://www.gnomemeeting.org/">www.gnomemeeting.org/</a>	<a href="http://iaxclient.sourceforge.net/iaxcomm/">http://iaxclient.sourceforge.net/iaxcomm/</a>	<a href="http://www.wirlab.net/kphone/index.html">www.wirlab.net/kphone/index.html</a>	<a href="http://vovida.org/applications/downloads/sipset/">http://vovida.org/applications/downloads/sipset/</a>	<a href="http://www.thekompany.com/products/tkphone/">www.thekompany.com/products/tkphone/</a>
License	GPL	Cornfed	GPL	GPL	GPL	Vovida	-
Prices	free	free	free	free	free	free	\$10US
<b>Functions</b>							
VoIP-Protocol	SIP	SIP	H.323	IAX2	SIP	SIP	SIP
Audio Codecs	PCMU, L16	G.711-uLaw	iLBC, GSM-06.10, MS-GSM, G.711-Alaw, G.711-uLaw, G.726, G.723.1	G.732.1, G.726, GSM, G.711uLaw, G.711Alaw, iLBC, LPC-10, Speex, u.a.	G.711uLaw, GSM, iLBC	PCMU	PCMU, GSM, G.729, Speex
STUN	no	no	no	no	yes	no	yes
SOCKS	no	no	no	no	no	no	yes
ILS Directory	no	no	yes	no	yes	no	yes
Audio Plugins	OSS	ALSA, OSS	ALSA, Quicknet	OSS	OSS, ALSA	OSS, ALSA	OSS, ALSA
Video Plugins	no	no	Video4Linux, (Webcam) AVC (Firewire Cameras)	no	VIC	MPEG4IP	no
IPv6 Support	yes	no	yes	no	yes	yes	no
Bandwidth Profile	33.6 Kbit/s bis 1 Mbit/s	no	no	no	no	no	no
DTMF Tones	yes	yes	yes	yes	yes	no	yes
<b>User Functions</b>							
GUI	yes	yes	yes	yes	yes	yes	yes
CLI	no	yes	no	no	no	yes	no
Addressbook	yes	no	yes	yes	yes	no	yes
Dialpad	yes	yes	yes	yes	yes	no	yes
<b>Multiple</b>							
simultaneous calls	yes	no	yes	yes	yes	no	no
Automatic call	yes	no	yes	yes	yes	yes	yes
Call holding	yes	no	yes	yes	yes	no	no
Video conferencing	no	no	yes	no	yes	yes	no

We had no trouble building the application from the source in our labs. A HTML-formatted manpage provided configuration support, although the page is only available online on the SIPset website. The only known bug to appear was an error in the call-to-URL function. A workaround is available and is likewise to be found on the Sipset website. However, the program crashed now and then, while accepting input into the options window. There were no problems with an existing VoIP connection, however.

Because of the concentration on the PCMU codec, we would hesitate to recommend Sipset, although the program behaved almost perfectly in our tests. Since it already supports video calls, and can handle IPv6, speedy development of this softphone would be welcome.

## TkPhone

The only commercial representative in the test was the TkPhone softphone by TheKompany. At \$10 the program isn't exactly expensive, and it does create a good impression. A free demo version is available for download, although it restricts calls to two minutes.

Before running the *install.sh* script to start the install, the user must decide whether or not to use system font anti-aliasing. Depending on your system, you first need to install one of the three support libraries *noxft*, *xft1*, or *xft2*. If the TkPhone installation script does not find any of these libraries, it stops the setup. The Readme is available as a decision making aid as to which of the three packages should be installed.

Visually, TkPhone can easily hold sway with GnomeMeeting or KPhone (see Figure 7), and can be set up quickly using the configuration menus. The softphone contains support for STUN and the SOCKS service, which was unique in the test group. Thus the program can be



**Figure 6:** SIPset's GUI version does not overload the user: enter an address, click on the Talk button, call.

deployed behind a NAT gateway without having to jump through hoops. With multiple network interfaces, the user selects the appropriate one from a drop-down menu. The registration with a proxy service can occur automatically, if desired.

The choice of codecs is not extensive. However, it still provides a suitable encoder/decoder for each environment. Thus codecs can be selected as needed in the range between 8 KBit/s (G.729) and 64 KBit/s (PCMU). Included in the group is the Speex codec [6], which encodes speech with variable bit rate, as well as iLBC.

TkPhone is easy to use and leaves nothing to be desired. This said, the question arises as to why one should look to a commercial client on Linux as there are enough free clients. Perhaps an account with VoIP gateway provider iConnecthere will convince some users, as they give a special rate to the TkPhone manufacturer. Otherwise, one can completely wipe the program from the disk with the help of the provided uninstallation script.

## So what should I use?

None of the tested programs proved to be completely up to day-to-day use. The connection was always successful, crashes occurred only sporadically with SIPset. GnomeMeeting is recommended for readers who only need to make peer-to-peer calls. The program is very mature and offers extensive documentation, as well as support through a mailing list and the FAQs. Calling normal phone numbers is a little difficult with GnomeMeeting, however, since it currently supports only H.323. And only a few providers, such as Microtelco [7], support H.323 connections.

Things look a little different with SIP, which most VoIP providers support. A SIP client will support both peer-to-peer connections and normal PST connections through a provider's VoIP gateway. KPhone is a good SIP softphone to get started with. Not because it represents the KDE counterpart to GnomeMeeting, but simply because it is available in every distribution and provides the functions one needs. If you can't warm to KPhone, take a look at Cornfed or SIPset. Both come with a GUI and command



**Figure 7:** TkPhone starts up a tidy interface and was the only softphone in the test which also support SOCKS-Servers.

line interface, which has a charm of its own.

The Bonephone Java telephone provides an interesting alternative to the conventional programs. The softphone is slim, fast and offers only the most necessary features. Let's not totally dismiss the commercial product TkPhone, either. Although, you really should avoid spending the \$10 unless you intend get yourself an account with iConnecthere, the TheKompany partner.

For Asterisk users, iaxComm is a mature program that does not leave anything to be desired. Thanks to the wxWidgets toolkit, it runs on any standard platform. However, because it is restricted to the IAX2 protocol, it is (currently) not usable with commercial providers. The really satisfying thing about this test lies in the realization that VoIP softphones are available for Linux for all protocols, requirements and tastes – free of charge and with Open Source code. If VoIP does become a success, it might have a lot to do with the Community's willingness to become involved in development work. ■

## INFO

- [1] GnomeMeeting Workshop: Kilian Krause, Christian Strauf, "Telephonitis": Linux Magazine, Issue 40 March 2004, Page 54
- [2] wxWidgets: <http://www.wxwidgets.org>
- [3] VIC: <http://www.nrg.ee.lbl.gov/vic/>
- [4] STUN/TURN: <http://www.newport-networks.com/whitepapers/fwnatwps3.html>
- [5] MPEG4IP: <http://www.mpeg4ip.net/>
- [6] Speex: <http://www.speex.org/>
- [7] Microtelco: <http://www.linuxjack.com>