Cornfed SIP User Agent

User's Guide

Version 1.1.7

Cornfed Systems, LLC

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1 Introduction

Welcome and thank you for choosing the Cornfed SIP User Agent! The Cornfed SIP User Agent is a Session Initiation Protocol (SIP) based softphone for your IBM-compatible Personal Computer running the Linux operating system. The Cornfed SIP User Agent allows you to make Internet-based telephone calls using an Advanced Linux Sound Architecture (ALSA) or Open Sound System (OSS) sound card with speakers and microphone as your telephone handset.

The Cornfed SIP User Agent is provided as a binary distribution only. The binary executable program is provided free of charge for personal use for users of the Linux operating system. For other terms of use, including any commercial applications, please contact Cornfed Systems.

The Cornfed SIP User Agent includes a web-based monitoring facility. The client logs errors and warnings to the Cornfed Systems web server to assist us in debugging. Every effort is made to insure that no personal information of any kind, including but not limited to, domain names, IP addresses, user information, and passwords, is accessed or transmitted. However, Cornfed Systems cannot be held liable for the transmission of any data deemed inappropriate in any way by you either purposefully or accidentally. Your use of this client gives Cornfed Systems consent to transmit debugging information.

2 Product Overview

This section provides a high-level view of the Cornfed SIP User Agent product. It specifies the available features and the set of software and hardware requirements that a user must meet in order to run the client.

2.1 Features

- Supported RFCs:
    - RFC 2617: HTTP Authentication: Basic and Digest Access Authentication
    - RFC 2778: A Model for Presence and Instant Messaging
    - RFC 2833: RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
    - RFC 3261: The Session Initiation Protocol
    - RFC 3264: An Offer/Answer Model with the Session Description Protocol
    - RFC 3265: SIP-Specific Event Notification
    - RFC 3489bis: Session Traversal Utilities for NAT (STUN)
    - RFC 3550: A Transport Protocol for Real-Time Applications
    - RFC 3863: Presence Information Data Format (PIDF)
    - RFC 3856: A Presence Event Package for SIP
    - RFC 4566: Session Description Protocol
- Supports Digest authentication for registrations and outbound INVITEs
- Support for loose proxy routing using Record-Route and Route headers
- Handles forking of outbound INVITEs by proxies
- Support for G.711 mu-Law and A-Law voice codecs
- Support for G.729a voice codec for commercial use
- Supports SIP compact header forms
- A flexible and scriptable command-line interface client
- A simple GTK based graphical client
- Multi-threaded implementation

2.2 Requirements

- An IBM-compatible (Intel *x86*-based) Personal Computer running the Linux operating system
- An ALSA or OSS compatible sound card. The OSS programming interface is used. An ALSA installation must include the OSS emulation library
- A high-speed Internet connection. This will most likely take the form of a Local-Area Network (LAN) or Cable or Digital Subscriber Line (DSL) broadband connection
- Pthreads library installation

3 Installation

The Cornfed SIP User Agent is distributed in two forms: 1) an RPM binary file and 2) a TAR binary file. For the RPM distribution, a file with name similar to `cornfedsipua-1.1.6-1.i386.rpm` contains the binary executable files and other files including this document. To extract the files, type following command:

```
$ rpm -Uvh cornfedsipua-1.1.6-1.i386.rpm
```

The resulting installation installs a number of files. There are two binaries called `cornfedsip` and `cornfedsip_cli`. The first binary file is the GTK graphical client and the second binary file is Command Line Interface client. Another file is the default ringtone .wav file, `ring.wav`. Another file is called `cornfedsipua.pdf` which contains this documentation file. There may be other files installed as well.

For the TAR distribution, a file with a name similar to `cornfedsipua-1.1.6.tar.gz` contains the binary executables and installation scripts along with other files including this document. To extract the files, type the following command:

```
$ tar xzvf cornfedsipua-1.1.6.tar.gz
```

The result is a directory called `cornfedsipua`. You should cd into this directory and run the install script. Read the `README` file in this directory for specific instructions.

This completes the installation process. To run the client program, type either of the client executable commands at the Linux shell prompt. A menu item entry is also available for the GTK client under the Applications -> Internet menu.

Both clients accept a set of command line arguments. The full specification of the command line arguments is given as:

```
( cornfedsip | cornfedsip_cli )
      [ -intf intf ]
      [ -sport sip_port ]
      [ -rport rtp_port ]
      [ -stun host ]
      [ -soundcard device ]
      [ -ringdevice device ]
      [ -ringfile filename ]
      [ -dns ]
      [ -nodns ]
      [ -config file ]
      [ -noconfig ]
```

The `-intf` argument allows you to set the network interface used for SIP and RTP traffic. The following example illustrates the use of this command line argument:

```
$ sip -intf eth1
```

In this example, the `eth1` interface is used rather than the default `eth0` interface. Note the interface is specified as `eth1`, not `/dev/eth1`.

The `-sport` argument allows you to specify what UDP port the client will use for SIP messaging. The following example illustrates the use of this argument:

```
$ sip -sport 5061
```

In this example, the `5061` UDP port is used rather than the default `5060` port.

The `-rport` argument allows you to specify what UDP port the client will use for RTP media. The following example illustrates the use of this argument:

```
$ sip -rport 5005
```

In this example, the `5005` UDP port is used rather than the default `5004` port.

The `-stun` argument allows you to set the STUN server used for NAT traversal. The *host* argument is the domain name or IP address of the desired STUN server.

The `-soundcard` argument allows you to set the media soundcard device which is used to carry out the voice part of a phone call. The `/dev/dsp` device is assumed if this argument is not present. The following example shows the use of this command line argument:

```
$ sip -soundcard /dev/dsp2
```

Note that this argument uses the full path, `/dev/dsp2`. The device must have an Open Sound System API available for it, either using the Open Sound System or the OSS emulation associated with the Advanced Linux Sound System (ALSA). The device, `/dev/null`, is treated differently. If this device is specified, the soundcard is essentially disabled. `/dev/null` can be specified whether the system you are running the client on has a soundcard or not.

The `-ringdevice` argument allows you to set the ringtone soundcard device which is used to alert the user of an incoming call request. The `/dev/dsp` device is assumed if this argument is not present. The following example shows the use of this command line argument:

```
$ sip -ringdevice /dev/dsp2
```

Note that this argument uses the full path, `/dev/dsp2`. The device must have an Open Sound System API available for it, either using the Open Sound System or the OSS emulation associated with the Advanced Linux Sound System (ALSA).The device, `/dev/null`, is treated a bit differently. If this device is specified, the ringtone is essentially disabled. This ringtone device can be specified whether the system you are running the client on has a soundcard or not.

The `-ringfile` argument allows you to specify a .wav file that will serve as your ringtone. When a call is received, the client will play the ringtone to alert you of the incoming call request.

The -dns and `-nodns` arguments turns DNS lookups on and off, respectively. If DNS lookups are turned on, any field where network addressing is used or modified during client operations can use domain names or IP addresses. If DNS lookups are turned off, fields containing network addressing must use IP addresses only. DNS lookups can manipulated only using the command line arguments. There is no CLI command available to turn DNS lookups on or off.

The `-config` and `-noconfig` arguments are discussed in the section on Config Files.

4 Config Files

The Cornfed SIP User Agent uses a config file to maintain state between client runs. The default location for the config file is the `.cornfed/` directory created and stored in your user home directory. The default config file location can be overridden using the `-config` command line option. The following example illustrates the use of this argument:

```
$ cornfedsip -config ~/myconfig
```

In this example, the specified config file would be read instead of the default. When the client exits, this file would also be updated rather than the default. This means that users will need to have permission to update this file. The `-noconfig` command line option can be used to prevent the client from reading a config file at startup and writing to a config file on exit.

The config file is maintained automatically by the client program. It need not and should not be edited by hand, despite the fact that the file is human readable. When the client starts for the first time, the config file is created with default values. If you change any of these values during the course of the client execution, they will be saved to the config file when you exit the client program.

When the client program is started, the config file is read before any of the command line arguments are parsed. This means the the command line arguments override the config file settings. The two command line arguments to which this does not apply are the `-config` and `-noconfig` arguments.

5 SIP/RTP Basics

The Cornfed SIP User Agent implements a Voice-Over-IP (VoIP) endpoint on your personal computer. This client programs turn your PC into a telephone that can use the Internet to speak to other people that are also running VoIP client software. The Cornfed SIP User Agent is based on Internet Engineering Task Force (IETF) open standards. Specifically, the clients use the Session Initiation Protocol (SIP) for signaling (e.g. to setup and teardown voice calls) and the Real-Time Transport Protocol (RTP) for media (i.e. to carry your voice across the network).

5.1 SIP Addressing

A SIP endpoint is generally labeled by a Uniform Resource Indicator (URI) that looks something like this:

```
sip:613@fwd.pulver.com:5060
```

The `sip:` element is a prefix indicating that standard SIP signaling is to be used. The other elements comprise an endpoint label. The `613` element is called the *user*, the `fwd.pulver.com` element is called the *host*, and the `5060` element is called the *port*. In some circumstances, the user portion of the URI corresponds to a telephone number. The host and port portions correspond to familiar DNS names or IP addresses and UDP port numbers.

In general, the *host* part of the URI can be specified as either an IP address or a Fully Qualified Domain Name (FQDN). Note that if DNS lookups are disabled, you must use an IP address for the host part rather than an FQDN. In the example, the FQDN `fwd.pulver.com` is used. An IP address like `192.246.69.223` could also be used. It is not absolutely necessary to specify the *port* number. If no *port* number is specified in a URI, the Cornfed SIP User Agent assumes the use of the well-known SIP port 5060.

5.2 Network Address Translation

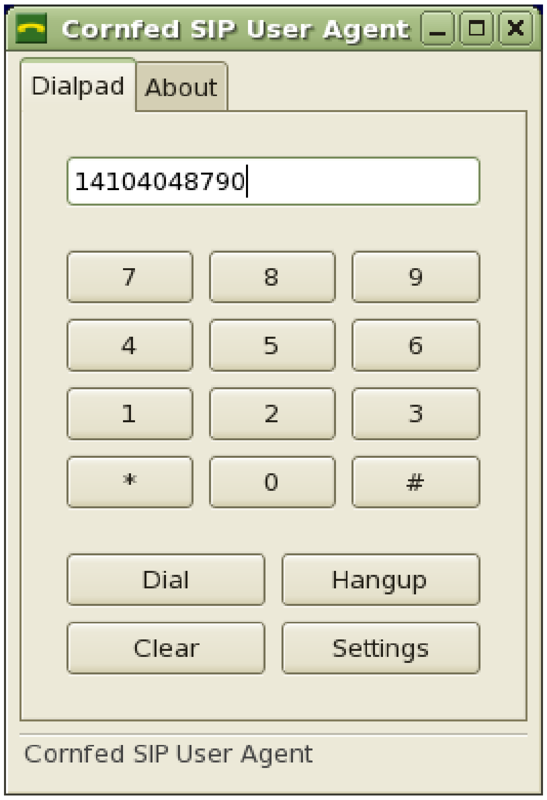
In many cases, the Cornfed SIP User Agent will be used by a computer that is behind a Network Address Translation (NAT) device. For instance, you might be running the client on a corporate network that is firewalled from the Internet and also makes use of NAT. You might also be on a home computer that has some sort of small router that sits between you and your broadband connection.

The Cornfed SIP User Agent uses the STUN protocol to assist with traversal of the NAT. The clients will periodically send STUN messages to a STUN server. The responses returned by the STUN server allow the client to manipulate addressing information in the SIP signaling to allow SIP and RTP traffic to flow between your machine and others on the Internet. If STUN does not work, e.g. if your network blocks STUN response packets, the clients will not be able to traverse the NAT properly.

6 GTK Graphical Client Operations

The first of the Cornfed SIP User Agent implementations is a simple GTK based graphical client that is described in this section. In all the screenshots provided in this documentation, the Fedora Core 6 Clearlooks theme is used, however, the GTK client is theme able. If you switch your basic desktop theme, the GTK client will respond to the change.

Figure 1 introduces the graphical interface. The main window is dominated by a text display, a dialpad, and four control buttons. At the bottom of the window is the Status Bar that provides feedback about the client operations.



14104048790

Cornfed SIP User Agent

Figure 1: The GTK Graphical Client Main Window

The text display shows the dialed digits. When you type in the text display or use the dialpad buttons, the dialed digits are updated. This text is used as the remote party phone number when you make a call.

The digit buttons also provide you the capability to send DTMF when you have a call connected. If you press one of the dialpad buttons during a call, the labeled digit is sent into the call. This capability allows you to navigate voice prompt menus.

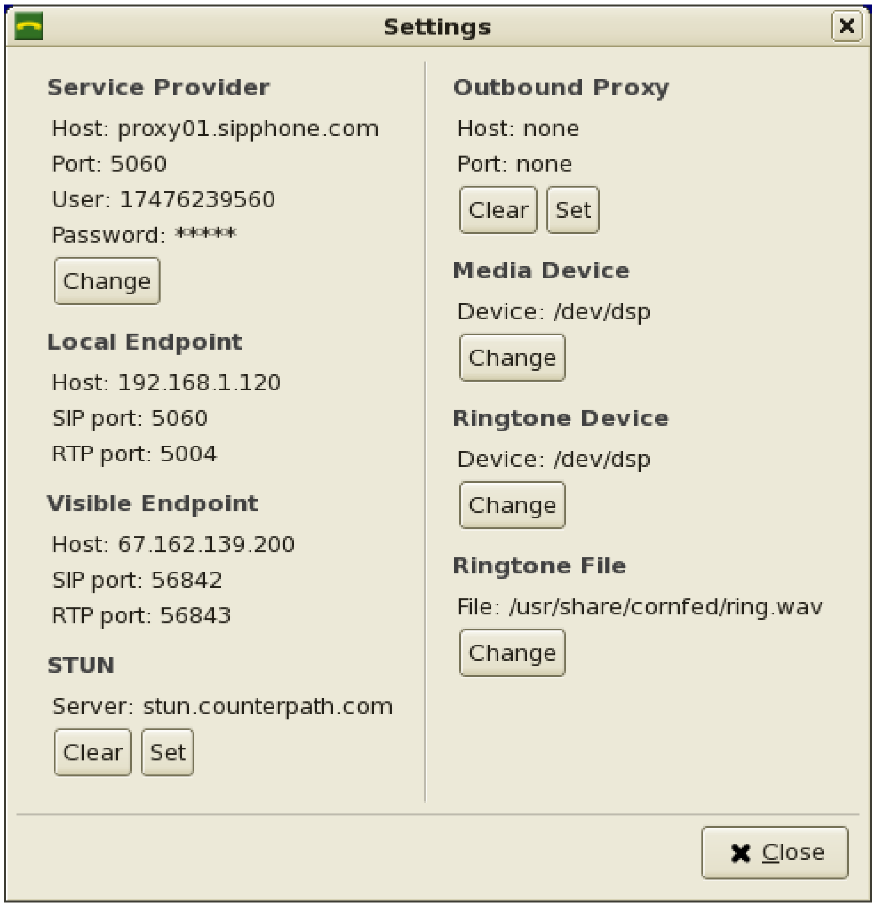
The four control buttons handle setting up and tearing down calls and configuring the client. The Dial button initiates a call and the Hangup button terminates a connected call. The Clear button clears the text display when no call is connected. When the client is not idle, pressing the Clear button will "reset" the client to the idle state. This means clearing any call in whatever state of progress.

The Setting button brings up the client control panel. There are a series of items that can be configured using this control panel. Figure 2 illustrates the elements that can be manipulated.

The first element is the Service Provider. If you press the Change button, you can set your Service Provider information. If you change the Service Provider, the client will try to register with the new Service Provider.

The Local Endpoint and Visible Endpoint elements provide information about your IP addressing. The Visible Endpoint information will be different than the Local Endpoint if your client is behind a NAT.

The STUN Server element allows you to specify the IP addressing of the STUN server. STUN is used to determine the Visible Endpoint addressing information and to keep port bindings alive in a NAT environment.


Settings

Service Provider

Host: proxy01.sipphone.com
Port: 5060
User: 17476239560
Password: *****

Local Endpoint

Host: 192.168.1.120
SIP port: 5060
RTP port: 5004

Visible Endpoint

Host: 67.162.139.200
SIP port: 56842
RTP port: 56843

STUN

Server: stun.counterpath.com

Outbound Proxy

Host: none
Port: none

Media Device

Device: /dev/dsp

Ringtone Device

Device: /dev/dsp

Ringtone File

File: /usr/share/cornfed/ring.wav

Figure 2: GTK Client Control Panel

The Outbound Proxy element allows you to set of clear an Outbound Proxy. If an Outbound Proxy is specified, all SIP messaging is routed through the Outbound Proxy.

The Media Device and Ringtone Device elements allow you to specify which soundcard device is to be used for media and ringtones. These devices can be different. The addressing assumes the OSS device naming.

The Ringtone File element is used to change the file containing your ringtone.

7 Command Line Interface (CLI) Client Basic Operation

The second Cornfed SIP User Agent client is Command-Line Interface (CLI) client. Once the program is started you will see something similar to the following output:

```
$ cornfedsip

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Written by Frank W. Miller

sip>
```

Any time you see the `sip>` prompt, you can enter any of the available commands. If you type the `local` command you will get a list of the local endpoint URI parameters. An example is given below:

```
sip> local

local host   : [192.168.1.120]
local port   : 5060
visible host : [67.162.139.200]
visible port : 3313
```

This command provides you with local and visible network addressing information for your client. The local network address consists of the IP address and UDP port used by your local network adapter on your local machine. The visible network address consists of the IP address and UDP port that are seen by other clients and servers on the Internet if your client is behind a NAT. Other clients and servers will use the visible network address to reach your client if you are behind a NAT device.

By default, the executable program is distributed with debugging turned on. When debugging is enabled, a significant amount of output is dumped to the screen when placing phone calls, primarily dumps of incoming and outgoing SIP messages. Debugging can be turned off by executing the following command:

```
sip> debug off
```

If you want to turn debugging back on, execute this command:

```
sip> debug on
```

If you want to see whether debugging is turned on or off, type this command:

```
sip> debug
Debugging is on
```

7.1 Registering with a Service Provider

Registering with your service provider (e.g. SIPPhone.com) allows others to call you. You can use the `register` command to manage your client's SIP registration. Type the following command to see to whom your registrations are being sent:

```
sip> register

user        : [17476239560]
password    : []
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5060
interval    : 300
buffer      : 10
re-register : yes
expires     : -1
```

You will need to enter a password before you can send this registration. The `user` and `password` usually correspond to the phone number and password issued to you by your Service Provider. To enter a password, execute the following command:

```
sip> register passwd secret
sip> register

user        : [17476239560]
password    : [secret]
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5060
interval    : 300
buffer      : 10
re-register : yes
expires     : -1
```

Note that the `password` field has been updated. You can now register with the service provider by typing the following command:

```
sip> register send
```

If you type the register command again, you'll see some information about your client's registration with the Service Provider:

```
sip> register

user        : [17476239560]
password    : [secret]
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5060
interval    : 300
buffer      : 10
re-register : yes
expires     : 4 minutes 57 seconds
```

Note the `expires` value has changed. The current registration will expire in 4 minutes and 57 seconds. Also note however that the `re-register` value is set to `yes`. This means that the CLI will automatically re-register with the Service Provider when `buffer` seconds are left before the registration expires.

The `user` and `password` are also relevant when making outbound calls. Some Service Providers require a client to be authenticated when they try to make a call. If an outbound call requires authentication, the Cornfed SIP User Agent will use the `user` and `password` associated with the Register Address of Record to perform the authentication.

A registration can also be removed. To remove a previous registration issue the following command:

```
sip> register remove
```

7.2 Initiating a Call

Once you have registered, you can receive incoming calls. To initiate an outbound Internet phone call to another client, you set the remote endpoint with the identification of the endpoint you want to talk to and then dial. Use the `remote` command to see the current remote endpoint settings, type the following command:

```
sip> remote

user        : [14104048790]
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5060
```

The `user` field contains the SIP URI *user* element, the `domain name` field contains the Internet domain name of the host to which you want to make a call, the `host` field contains the IP address to be used as the SIP URI *host* element, and the `port` field contains the SIP URI *port* element. All of these fields can be changed at the command line. To change the `user` field to `13035551212`, type the following commands:

```
sip> remote user 13035551212
sip> remote

user        : [13035551212]
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5060
```

Observe that the `user` field has been changed to `13035551212`. The `port` field can be changed similarly:

```
sip> remote port 5061
sip> remote

user        : [13035551212]
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5061
```

Observe that the `port` field has been changed to `5061`.

The *host* element of the SIP URI can be specified using either an FQDN or an IP address. If an FQDN is specified, the client immediately looks up the associated IP address and places the IP address in the `host` field. If an IP address is specified, the client will clear the `domain name` field.

Once the remote endpoint is set, you simply issue a `dial` command to initiate a phone call.

```
sip> dial
```

This command initiates a call to the specified remote endpoint. If the other side answers, the call is connected and you can begin a conversation with the other party.

7.3 Answering an Incoming Call

It is also possible for another IP phone user to call your client. If they know the network address at which you have registered, they can make calls to you. You can use the `local` command to see the local endpoint settings. Similar to the `remote` command, the following command shows the local endpoint settings:

```
sip> local

local host   : [192.168.1.120]
local port   : 5060
visible host : [67.162.139.200]
visible port : 3515
```

This output shows both a local and a visible network address for your client. If you try to make or receive calls through a Network Address Translation (NAT) box, such as a Residential Gateway, The Cornfed SIP User Agent will use the visible network address information so that other parties on the Internet can reach you.

When another party calls your client, you will see a line similar to the following displayed periodically on your terminal:

```
Incoming call from sip:4104048790@proxy01.sipphone.com
```

In addition, you will hear an audible ringback. When you hear the ringback and/or see this line appear, you have two choices, you can answer the call or refuse the call. If you answer the call, a voice connection is made and you can begin your conversation. If you refuse the call, the other party is told that you are not currently available to take the call. To answer the call use the following command:

```
sip> answer
```

To refuse the call, use the following command:

```
sip> refuse
```

7.4 Ending a Call

To end a call in progress, use the `hangup` command. At the command line, issue the following command:

```
sip> hangup
```

The call is terminated and the conversation in progress is ended, i.e. the speakers go silent and the microphone becomes inactive. If the other party hangs up, the only indication is the conversation being discontinued.

8 CLI Client Details

This chapter provides detailed descriptions of all the Cornfed SIP User Agent CLI commands. The CLI is designed to be *scriptable*. This means that the commands and the responses the CLI provides have been designed to be controlled by automated means. Specifically, the CLI is designed to make scripting using the `expect` scripting language easy.

At the command line, you can type a `?` (or any other invalid command) and a list of all available commands will be displayed:

```
sip> ?

about            : Cornfed SIP User Agent version information
answer           : Answer an incoming call
debug            : Debug settings
dial             : Initiate an outbound call
dialog           : Dialog information
dnd              : Do not disturb
dns              : DNS settings
hangup           : Terminate a connected call
help             : Help information
history          : Call history
local            : Local SIP endpoint settings
log              : Log settings
nat              : NAT settings
outbound         : Outbound proxy settings
play             : Play .wav file
record           : Record .wav file
register         : Register with a SIP server
refuse           : Refuse an incoming call
remote           : Remote SIP endpoint settings
reset            : Reset client
ringtone         : Ringtone settings
rtp              : RTP settings
soundcard        : Soundcard settings
subscribe        : Subscription information
wav              : List of queued .wav files to be played
0123456789ABCD#* : Send DTMF digit during a connected call
```

Several of these commands were introduced previously. Detailed descriptions of all the commands, including hints for how they can be scripted, are provided in this chapter.

An important command to be aware of is the `reset` command. This command simply clears the client state. This is useful when your client and the other endpoint have gotten out of sync. Any retransmissions or waiting that your client is doing will be ended. In addition, it may be necessary to use the `soundcard flush` command, which causes the sound card input buffer to be cleared. These two commands can generally be issued as often as needed in order to return the client to an idle state. Note that a reset command will not clear a completed registration that is counting down towards a re-registration.

8.1 Help

The Cornfed SIP User Agent CLI has an online help system. If you type the `help` command at the CLI prompt, you will get the list of commands shown at the beginning of this chapter. In addition, you can get a summary of the syntax associated with all the commands by typing the following command:

```
sip> help syntax

about             : about
answer            : answer
debug             : debug [ (on | off) ]
dial              : dial
dialog            : dialog
dnd               : dnd [ (on | off) ]
dns               : dns
hangup            : hangup
help              : help [ (syntax | command) ]
history           : history
local             : local
log               : log [ (info | event | connection | warning | error) ]
nat               : nat [ (on | off | stun address) ]
outbound          : outbound [ (host address | port number) ]
play              : play filename
record            : record [ (start | stop | file filename) ]
register           : (register | reg) [ (user name | host address | port
number | passwd password | send | remove | interval seconds | buffer
seconds | rereg | norereg) ]
refuse            : refuse
remote            : remote [ ( user name | host address | port number ) ]
reset             : reset
ringtone          : ringtone [ (file filename | device devicename) ]
rtp               : rtp [ stats ]
soundcard         : soundcard [ (device devicename | flush) ]
subscribe         : (subscription | sub) [ uri ]
wav               : wav [ flush ]
0123456789ABCD#*  : (0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | A | B | C | D
| # | *)+
```

Finally, you can type the `help` *cmd* command where *cmd* is one of the CLI commands. Each command has more detailed description information available when you look at them specifically.

8.2 Answer

When another party calls your client, you will see a line similar to the following displayed periodically on your terminal:

```
Incoming call from sip:4104048790@proxy01.sipphone.com
```

In addition, you will hear an audible ringback. When you hear the ringback and/or see this line appear, you can answer the call and a voice connection is made and you can begin your conversation.

The `answer` command connects an incoming call request. For scripting, the `answer` command outputs several strings. If the current dialog has taken on a bad state, the client will print the output:

```
Bad dialog state
```

If there is no incoming call request, the client will print the output:

```
No incoming call
```

If the call fails to completely connect after the answer command is typed, the client prints:

```
Connection failed
```

Finally, if the call connects successfully, the client prints:

```
Connected
```

An example expect script that handles this output is given as:

```
expect   Incoming call   {
    send   answer\n
    expect {
          Bad dialog state   { ... }
          No incoming call   { ... }
          Connection failed   { ... }
          Connected   { ... }
    }
}
```

8.3 Debug

The `debug` command manages the amount of debugging output that is sent to the user terminal. By default, the executable program is distributed with debugging turned on. When debugging is enabled, a significant amount of output is dumped to the screen when placing phone calls, primarily dumps of incoming and outgoing SIP messages. Debugging can be turned off by executing the following command:

```
sip> debug off
```

If you want to turn debugging back on, execute this command:

```
sip> debug on
```

If you want to see whether debugging is turned on or off, type this command:

```
sip> debug
Debugging is on
```

The amount of debugging information that is sent to the screen when debugging is turned on is controlled by the `log` command.

8.4 Dial

The `dial` command initiates a call. Before issuing the `dial` command, you should make sure the remote endpoint settings are set correctly. When you issue the `dial` command, the client starts a call setup. If the call setup completes, the result is a connected call and you can speak to the called party using your microphone and speakers.

While the dial command is quite simple to issue, the resulting output can be complex due to a number of possible outcomes with the call setup attempt. For scripting, the `dial` command outputs several strings. If the current dialog has taken on a bad state, the client will print the output:

```
Bad dialog state
```

If the client already has a call in progress, the client will print:

```
Not idle
```

If the remote endpoint addressing information is not set correctly, the client will print the output:

```
Remote address not set correctly
```

If the call setup attempt fails, the client prints:

```
Call setup failed
```

If the call setup initiates correctly, the client prints:

```
Calling sip:fwmiller@cornfed.com
```

In this example, the called party is `sip:fwmiller@cornfed.com`. This SIP URI information will vary based on the remote endpoint settings.

At some point, if the call setup is successful, a connection message will be printed by the client:

```
Connected
```

Another possibility is that the call setup attempt times out with no response from the called party. In this case, the client prints:

```
Timed out
```

An example expect script that handles this output is given as:

```
send   dial\n
expect {
      Bad dialog state   { ... }
      Not idle   { ... }
      Remote address not set correctly   { ... }
      Call setup failed   { ... }
      Calling   {
        ...
        expect {
              Call setup failed   { ... }
              Timed out   { ... }
              Connected   { ... }
        }
        ...
    }
}
```

8.5 Dialog

The `dialog` command can be used to show the current state of the SIP dialog between your client and another SIP endpoint. In the following example, output from just after a call that has been initiated by your client and connected is shown. In the example, the `state` field indicates the call is connected. The branch tag value and local and remote tag values are stored as well as the Call-ID. The local and remote CSeq numbers are stored. The local and remote URIs, used in the From and To headers, respectively, are shown. There are various other fields here as well including the route set determined by the Record-Route headers, a set of Via headers, and authorization information.

The `<< Dialog 1 >>` label just prior to the dialog fields shows that this is the first session dialog that is currently being maintained. There are a variety of reasons why you may see multiple dialogs listed when you issue the dialog command. An outbound call may have been forked, a registration may be in progress, or there may be subscription and notification dialogs in progress.

```
sip> dialog

<< Dialog 1 >>
state          : SIPS_CONNECTED
route_set      :
[<sip:198.65.166.131;ftag=de5Fu0tG;lr>]
via_hdrs       :
host [] port -1 branch [z9hG4bKbm04ingm1] transport UDP
call_id        : [vhLYASGotgxm]
local_tag      : [de5Fu0tG]
remote_tag     : [as6dba46a3]
local_seq      : 2
remote_seq     : -1
local_uri      : [sip:192.168.1.120]
remote_uri     : [sip:13039529357@proxy01.sipphone.com]
reg_uri        : [sip:17476239560:tired@proxy01.sipphone.com]
remote_target : [sip:13039529357@130.94.88.86]
authorization  :  [Authorization:  Digest  username="17476239560",
realm="proxy01.sipphone.com",
nonce="4622437f7eaa74c875a435b5b46536832b7c105c",
uri="sip:proxy01.sipphone.com",
response="853e05e043cee72422f8d510fdab3c12"
]
last_resp      :
[]
```

8.6 Do Not Disturb

You can place the Command Line Client into a Do Not Disturb state by typing the following command:

```
sip> dnd on
Do not disturb is on
```

This state causes a 486 Busy Here response to be returned to any incoming INVITE. To turn Do Not Disturb off, type the following command:

```
sip> dnd off
Do not distrub is off
```

If you want to check whether Do Not Disturb is turned on or not, type the command with no argument:

```
sip> dnd
Do not distrub is off
```

8.7 DNS

The `dns` command is used to determine whether DNS lookups are enabled. You can look at whether `dns` lookups are being performed by typing the `dns` command with no arguments:

```
sip> dns
DNS lookups are on
```

8.8 Hangup

To end a call in progress, use the `hangup` command. At the command line, issue the following command:

```
sip> hangup
```

The call is terminated and the conversation in progress is ended, i.e. the speakers go silent and the microphone becomes inactive. If the other party hangs up, the only indication is the conversation being discontinued.

8.9 History

You can display the call activity for your client with the `history` command. Just type `history` at the command line and you will see output that looks something like this:

```
sip> history

Apr 15 09:27:46 Hangup [sip:13039529357@proxy01.sipphone.com]
Apr 15 09:17:43 Connected [sip:13039529357@proxy01.sipphone.com]
Apr 15 09:17:40 Initiated [sip:13039529357@proxy01.sipphone.com]
Apr 13 20:23:01 Hangup [sip:3039529357@38.114.14.210]
Apr 13 20:22:12 Connected [sip:3039529357@38.114.14.210]
Apr 13 20:22:12 Connected [sip:3039529357@38.114.14.210]
Apr 13 20:22:05 Received [sip:3039529357@38.114.14.210]
Apr 13 20:19:39 Connected [sip:13039529357@proxy01.sipphone.com]
Apr 13 20:19:37 Initiated [sip:13039529357@proxy01.sipphone.com]
Apr 13 20:19:07 Initiated [sip:13039529357@sipphone.com]
Apr 13 20:18:12 Initiated [sip:13039529357@sipphone.com]
```

This list of call events is in reverse chronological order. Your client maintains a history file that tracks all of these events. There are four types of historical events, 1) Initiated, 2) Received, 3) Connected, and 4) Hangup. The Initiated and Received events indicated the initiation of a call, for outbound and inbound calls, respectively. The Connected event indicates that a call has been connected and the Hangup event indicates that a call has been terminated.

8.10 Local

The local endpoint settings are used to identify the network addressing information associated with your client. To see the current local endpoint settings, type the following command:

```
sip> local

local host   : [192.168.1.120]
local port   : 5060
visible host : [67.162.139.200]
visible port : 3515
```

This command provides you with local and visible network addressing information for your client. The local network address consists of the IP address and UDP port used by your local network adapter on your local machine. The visible network address consists of the IP address and UDP port that are seen by other clients and servers on the Internet if your client is behind a NAT. Other clients and servers will use the visible network address to reach your client if you are behind a NAT device.

8.11 Log Settings

You can control how much logging information is dumped to the console when the debug setting is on. There are five possible value and they are ordered from most critical to least critical:

`error` (highest, least information)
`warning`
`connection`
`event`
`info` (lowest, most information)

In general, the lower your log setting, the more logging output will be dumped to the console. In the following example, the user sets only error and warning log messages:

```
sip> log warning
Log level warning
```

You can see the current logging level by just typing the `log` command as follows:

```
sip> log
Log level warning
```

8.12 Network Address Translation (NAT)

The Cornfed SIP User Agent has the ability to operate behind a Network Address Translation (NAT) device. NAT detection is automatic and uses the Simple Traversal of UDP over NAT (STUN) protocol to determine visible network address information.

You can see the specified STUN server address with the `nat` command. Just type `nat` at the command line and you should get output similar to the following:

```
sip> nat

domain name : [stun.counterpath.com]
host        : [64.69.76.23]
nat         : on
```

To change the STUN server use the following command:

```
sip> nat stun host
```

This command changes the STUN server to the specified *host* address. The *host* can be either an IP address or an FQDN. When you change this address, the client will begin querying the new server for visible network address information.

By default, the client program is distributed with NAT translation capability turned on. It may be necessary to disable this feature for your environment. NAT translations can be turned off by executing the following command:

```
sip> nat off
```

If you want to turn NAT translations back on, execute this command:

```
sip> nat on
```

8.13 Outbound Proxies

The Cornfed SIP User Agent can be configured to use an outbound SIP proxy. When configured this way, all SIP messages are sent first to the outbound proxy rather than to their configured request or response addresses. To see the currrent outbound proxy setting, type the following command:

```
sip> outbound

outbound proxy host : [168.143.113.40]
outbound proxy port : 5060
```

You can change both the host and port numbers. To configure the outbound proxy host to `fwd.pulver.com`, type the following command:

```
sip> outbound host fwd.pulver.com

domain name : [fwd.pulver.com]
host        : [192.246.69.223]

sip> outbound

outbound proxy host : [192.246.69.223]
outbound proxy port : 5060
```

In this example, a domain name is specified. This domain is immediately converted to its dotted decimal IP address that is stored as the outbound proxy host address. To configure the port number, use the following command:

```
sip> outbound port 5061
sip> outbound

outbound proxy host : [192.246.69.223]
outbound proxy port : 5061
```

Observe the oubound proxy port number has been changed to `5061`. *For the outbound proxy address to be used, both host address and port number must be specified.*

You can use the `outbound host clear` and `outbound port clear` commands to clear the outbound proxy host and port, respectively.

8.14 Playing and Recording .wav Files

The Cornfed SIP User Agent CLI has the ability to play and record .wav files. Playing .wav files will cause a specified .wav file contents to be played out of a connected call. To play a .wav file, execute the following command:

```
sip> play sound.wav
```

The .wav file must be of a certain format. Specifically, it must be a valid .wav file that includes the "`RIFF`", "`WAVE`", "`fmt `", and "`data`" strings in the appropriate .wav header fields. *In addition, the .wav file must be a PCM audio file sampled at 8 Khz using a single channel and 8 bits per sample.*

To record a .wav file, a call must be connected. The *incoming audio only* of the connected call is recorded during the time that recording is enabled. To enable recording, you must first set the name of the file to which the audio is to be recorded using a command like this:

```
sip> record file test.wav
```

You can check which file is currently selected for recording by simply typing `record` with no parameters, as follows:

```
sip> record
Record file [test.wav]
Recording length 0 bytes
```

After connecting a call, you can initiate a recording using the following command:

```
sip> record start
```

After executing this command, the incoming audio stream is recorded to the specified .wav file in addition to being played to your phone speakers. To stop recording, execute the following command:

```
sip> record stop
```

This command causes recording to be stopped. You can then look at the size of the recording file by executing the record command again with no parameters:

```
sip> record
Record file [test.wav]
Recording length 35876 bytes
```

If you begin another recording, the old recording file is deleted. If you want to save the recording you made, either change the recording file name or rename the recording file you just made to another file name. If the recording file does not exist when you start recording, the client will create a new file.

8.15 Register

Registration is the process of notifying a Service Provider (e.g. SIPPhone.com) that your softphone client is available to receive calls. Registrations have an expiration associated with them. Once your client is registered, it watches the expiration timer. The client will automatically re-register with the Service Provider just before the timer expires. This way, your client enjoys uninterrupt incoming call service.

To initiate a registration, you use the `register` command. Type the following command to see to whom your registrations are being sent:

```
sip> register

user        : [17476239560]
password    : []
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5060
interval    : 300
buffer      : 10
re-register : yes
expires     : -1
```

In this case, the Service provider is SIPPhone.com with its host name of `proxy01.sipphone.com`. SIPPhone.com has provided you with a user id which is a phone number, in this case, `17476239560`. The user id can be changed with the `register user` command, the host can be changed with the `register host` command, and the port at which the Service Provider server is can be changed with the `register port` command. To see how to specify the user, host, and port commands, type `help register` at the command line.

You will need to enter a password before you can send this registration. The `user` and `password` usually correspond to the phone number and password issued to you by your Service Provider. To enter a password, execute the following command:

```
sip> register passwd secret
sip> register

user        : [17476239560]
password    : [secret]
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5060
interval    : 300
buffer      : 10
re-register : yes
expires     : -1
```

Note that the `password` field has been updated. You can now register with the service provider by typing the following command:

```
sip> register send
```

If you type the register command again, you'll see some information about your client's registration with the Service Provider:

```
sip> register

user        : [17476239560]
password    : [secret]
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5060
interval    : 300
buffer      : 10
re-register : yes
expires     : 4 minutes 57 seconds
```

Note the `expires` value has changed. The current registration will expire in 4 minutes and 57 seconds. Also note however that the `re-register` value is set to `yes`. This means that the CLI will automatically re-register with the Service Provider when `buffer` seconds are left before the registration expires.

The `user` and `password` are also relevant when making outbound calls. Some Service Providers require a client to be authenticated when they try to make a call. If an outbound call requires authentication, the Cornfed SIP User Agent will use the `user` and `password` associated with the Register Address of Record to perform the authentication.

Control of re-registrations is provided by four commands. `register rereg` turns re-registrations on and `register norereg` turns re-registrations off. If re-registrations are turned off, the client will not try to re-register as the registration expiration expires. The interval and buffer are controlled with the `register interval` and `register buffer` commands. The interval is the suggested amount of time the client sends to the server for registration expirations. The buffer is the duration that remains in the expiration timer when the client attempts to re-register.

A registration can also be removed. To remove a previous registration issue the following command:

```
sip> register remove
```

This command provides basic response strings. When you issue a register send command the client will return a message indicating the registration request has been sent:

```
Registration request sent
```

When a positive response is received that contains the actual expiration, the client indicates that with the following output:

```
Registration expires in 5 minutes
```

If the registration fails, no output is provided by the client. A script should use a timeout to determine failure. An example expect script that handles the positive output is given as:

```
send   reg send\n
expect {
      Registration request sent   {
        ...
        expect {
              Registration expires   { ... }
        }
        ...
    }
}
```

8.16 Refuse

When another party calls your client, you will see a line similar to the following displayed periodically on your terminal:

```
Incoming call from sip:4104048790@proxy01.sipphone.com
```

In addition, you will hear an audible ringback. When you hear the ringback and/or see this line appear, you can refuse the the call by issuing the refuse command.

For scripting, the `refuse` command outputs several strings. If the current dialog has taken on a bad state, the client will print the output:

```
Bad dialog state
```

If there is no incoming call request, the client will print the output:

```
No incoming call
```

If the call is successfully refused, the client prints:

```
Call refused
```

An example expect script that handles this output is given as:

```
expect   Incoming call   {
    send   refuse\n
    expect {
          Bad dialog state   { ... }
          No incoming call   { ... }
          Call refused   { ... }
    }
}
```

8.17 Remote

To initiate an outbound Internet phone call to another client, you set the remote endpoint with the identification of the endpoint you want to talk to and then dial. Use the `remote` command to see the current remote endpoint settings, type the following command:

```
sip> remote

user        : [14104048790]
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5060
```

The `user` field contains the SIP URI *user* element, the `domain name` field contains the Internet domain name of the host to which you want to make a call, the `host` field contains the IP address to be used as the SIP URI *host* element, and the `port` field contains the SIP URI *port* element. All of these fields can be changed at the command line. To change the `user` field to `13035551212`, type the following commands:

```
sip> remote user 13035551212
sip> remote

user        : [13035551212]
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5060
```

Observe that the `user` field has been changed to `13035551212`. The `port` field can be changed similarly:

```
sip> remote port 5061
sip> remote

user        : [13035551212]
domain name : [proxy01.sipphone.com]
host        : [198.65.166.131]
port        : 5061
```

Observe that the `port` field has been changed to `5061`.

The *host* element of the SIP URI can be specified using either an FQDN or an IP address. If an FQDN is specified, the client immediately looks up the associated IP address and places the IP address in the `host` field. If an IP address is specified, the client will clear the `domain name` field.

8.18 Ringtones

A ringtone is played to alert you when an incoming call request is received. A ringtone is implemented by the .wav file that contains the sounds you want to hear when the phone is ringing and the device out which the ringtone is played. You can see the current Ringtone settings by typing:

```
sip> ringtone

Ringtone device [/dev/dsp]
Ringtone .wav file [/usr/share/cornfed/wav/ring.wav]
```

You can change the ringtone device by using the following command:

```
sip> ringtone device /dev/dsp2

Ringtone device [/dev/dsp2]
Ringtone .wav file [/usr/share/cornfed/wav/ring.wav]
```

You can change the ringtone file using the following command:

```
sip> ringtone file ../wav/ring.wav

Ringtone device [/dev/dsp2]
Ringtone .wav file [../wav/ring.wav]
```

The current implementation supports .wav files with a single channel, 8 Khz sample and byte rates, and 8 bits per sample only.

8.19 RTP

The `rtp` command shows information regarding the RTP audio transport associated with a call. The following shows an example output of the `rtp` command:

```
sip> rtp

local host   : [192.168.1.120]
local port   : 5004
visible host : [67.162.139.200]
visible port : 5574
remote host  : [130.94.88.94]
remote port  : 18716
codec        : G.711 mu-law
```

In this example, a call is in progress between `192.168.1.120:5004` (our local host IP address and port) and `130.94.88.94:18716` (the remote host IP address and port). The remote host is actually sending its RTP packets to the visible IP address at `67.162.139.200:5574` since this client is behind a NAT. The codec being used for the call is also listed as G.711 mu-Law in this example.

The Cornfed SIP User Agent uses port 5004 by default for its local RTP port. The remote party specifies the IP address and port of its side of the voice call in the SDP associated with the SIP signaling for the call.

The local RTP port number can be changed as desired. To change the `local port` field to `7078`, type the following commands:

```
sip> rtp port 7078
sip> rtp

local host   : [192.168.1.120]
local port   : 7078
visible host : [67.162.139.200]
visible port : 5575
remote host  : []
remote port  : -1
codec        : -1
```

Observe that the `local port` field has been changed to `7078`. Note also that the `visible port` has changed to `5575`. The NAT device has in this case allocated a new visible UDP port for the different local port.

You can also observe a variety of statistics that are maintained with respect to the media streams. The statistics are broken into two sections, one keeping stats for the life of the client, i.e. across multiple calls, and one for the current (or last) session. The statistics track connect time, packets and bytes in and out and missing packets on the input stream. In addition, a throughput is calculated for each section by taking a simple division of the bytes over the connect duration for both the input and output streams. The following command dumps the current RTP statistics:

```
sip> rtp stats

total connect duration        : 3 min 43 sec
total throughput in           : 20343 bits per sec
total packets in              : 3297
total bytes in                : 567084
total packets missing in      : 1
total throughput out          : 63873 bits per sec
total packets out             : 11128
total bytes out               : 1780480
session connect duration      : 0 min 18 sec
session throughput in         : 69640 bits per sec
session packets in            : 911
session bytes in              : 156692
session packets missing in    : 0
session accumulated jitter in : 0
session throughput out        : 65208 bits per sec
session packets out           : 917
session bytes out             : 146720
```

8.20 Sound Card

In addition to flushing the sound card using the `soundcard flush` command, information about the soundcard status can be obtained. Issue a simple `soundcard` command to generate this data:

```
sip> soundcard
soundcard device [/dev/dsp2]

soundcard input buffer
avail fragments  : 2
total fragments  : 256
fragment size    : 256
avail bytes      : 512

soundcard output buffer
avail fragments : 9
total fragments : 9
fragment size   : 256
avail bytes     : 2304
```

This information describes the parameters associated with the input and output channels of the sound card device.

You can set the Sound Device by using the following command:

```
sip> soundcard device /dev/dsp
```

8.21 .wav File Queue

The client queues any .wav files that you tell it to play out a connected call. To see the contents of the queue, type the `wav` command:

```
sip> wav
Apr 16 22:04:55 <1.1.2c> wav_rec 1 len 160010 frame_size 160 pos
102880
Apr 16 22:04:55 <1.1.2c> wav_rec 2 len 41227 frame_size 160 pos 0
Apr 16 22:04:55 <1.1.2c> wav_rec 3 len 160010 frame_size 160 pos 0
Apr 16 22:04:55 <1.1.2c> wav_rec 4 len 41227 frame_size 160 pos 0
```

The files are listed in the order in which they will be played out. You can clear the queue by typing the `wav flush` command.

8.22 Generating DTMF Tones

The Cornfed SIP User Agent can generate DTMF tones. The feature is useful if you want to need to navigate voicemail system or an auto-attendant. When you have a call connected, simply type a series of DTMF digits at the command prompt. Each of the digits in the list you type will be played out in sequence. There is no delay between the playing of the digits. The following example shows the user sending an `8` DTMF digit to the other party.

```
sip> 8
sip>
```

The following example shows how to quickly navigate a menu system by executing a series of digits in sequence.

```
sip> 113*
sip>
```