Command Line Control of an Asterisk Confbridge

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Contents

0.1	Introduction		
	0.1.1	Software and Hardware Used	2
0.2	Prepar	ration and Set-up	3
	0.2.1	Overview of system	3
	0.2.2	Asterisk Server	3
	0.2.3	Important CLI Commands	3
	0.2.4	Configuration File Descriptions	4
	0.2.5	SIP Clients	6
0.3	C++	Confbridge Recorder	7

0.1 Introduction

This document details the method used to configure an asterisk server and describes a C++ program used to record calls made in a confbridge.

The aim is to create a platform where node operators can group call through an Asterisk confbridge.

0.1.1 Software and Hardware Used

- Ubuntu 15.04
- Asterisk 13.1.0
- Code::Blocks 13.12
- SFLphone 1.4.1

0.2 Preparation and Set-up

0.2.1 Overview of system

Each node is assigned a SIP account and an extension number. This number can be used for one-on-one calls. Furthermore, a confbridge is created. This is essentially a conference call with a specific extension. Recording of the confbridge is achieved using a C++ program.

0.2.2 Asterisk Server

The server is required to have Asterisk installed:

```
sudo apt-get install asterisk
```

Once installed, the command line interface can be accessed as follows:

```
sudo asterisk -r
```

Figure 1: Asterisk CLI

The Asterisk server is configured by editing three .conf files:

- sip.conf manage SIP accounts and set the server IP.
- extensions.conf control what happens when extensions are dialled.
- confbridge.conf configure a confbridge.

These files are located at:

```
/etc/asterisk
```

These files must be replaced with the provided ones, in order to allow recording. It is recommended to make a copy of the original contents before replacing.

0.2.3 Important CLI Commands

- Whenever a .conf file is altered, the reload command must be used to refresh the server.
- To restart the server, use *core restart now*.
- View users present in a confbridge, confbridge list.
- Commands can be made from terminal, without entering the CLI:

```
sudo asterisk -rx "command"
```

```
For example: sudo asterisk -rx "confbridge list"
```

0.2.4 Configuration File Descriptions

sip.conf

```
[general]
bindaddr=0.0.0.0:5060
                                  ; listen on IPv4 wildcard, UDP default port
localnet=127.0.0.1/255.255.255.0 ; server IP, must be changed accordingly
[111]
                                   ;account username/extension number
type=friend
                                   ; account can make and recieve calls
host=dynamic
                                   ; dynamic IP address
secret=123
                                   ;account password
[222]
                                   ; create as many accounts as needed
type=friend
host=dynamic
secret=234
[333]
type=friend
host=dynamic
secret=345
[444]
type=friend
host=dynamic
secret=456
```

extensions.conf

```
[default]

exten => 100,1,Answer() ; if 100 is dialed, server answers
; ask node opperator for name and announce arrival to others
exten => 100,2,Set(CONFBRIDGE(user,announce_join_leave)=yes)
exten => 100,3,ConfBridge(100,NeXtRad) ; link opperator to confbridge

;if other extension called, dial that number (one-on-one calls).
exten => _XXX,1,Dial(SIP/${EXTEN})
```

${\bf confbridge.conf}$

```
[general]
; --- ConfBridge User Profile Options ---
[default_user]
type=user
music_on_hold_when_empty=yes
; --- ConfBridge Bridge Profile Options ---
[default_bridge]
type=bridge
[NeXtRad]
                                                 ; confbridge name
type=bridge
record_file=/var/spool/asterisk/NeXtRAD.wav
                                                 ;save location
record_conference=no
                                                  ;no record from start
; --- ConfBridge Menu Options ---
[sample_user_menu]
type=menu
*=playback_and_continue(conf-usermenu)
*1=toggle_mute
```

1=toggle_mute

*4=decrease_listening_volume

4=decrease_listening_volume

*6=increase_listening_volume

6=increase_listening_volume

*7=decrease_talking_volume

7=decrease_talking_volume

*8=leave_conference

8=leave_conference

*9=increase_talking_volume

9=increase_talking_volume

0.2.5 SIP Clients

SFLphone 1.4.1

This client was used during testing of the asterisk server. Note that the status must be registered in order to work.



Figure 2: Account Settings

Figures 3 and 4 display the account settings of a functioning account.

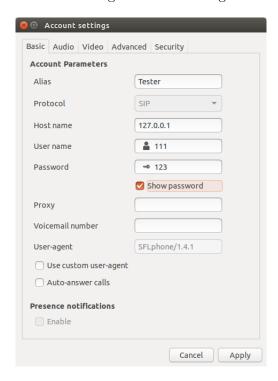


Figure 3: Basic Account Settings

NB Note that the port number is different to the one specified in the *sip.conf* file.

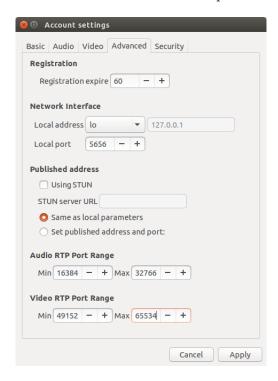


Figure 4: Advanced Account Settings

To test the server confbridge, dial 100 using SFLphone. The user is prompted to speak his/her name and then press the hash key. The user is announced to any other users present in the confbridge and the conference call commences.

Linphone 3.6.1

Linphone is a cross-platform SIP client (iOS, Android, Windows, OS X and Linux). It has been tested to work perfectly with this system on iOS and Linux. The set-up is extremely similar to the method described above. Friendly reminder: port number is 5656. (NOT 5060)

0.3 C++ Confbridge Recorder

Upon compiling the code in Code::Blocks, the following console application will appear:

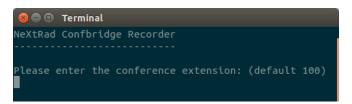


Figure 5: Welcome Screen

The user is then prompted to enter his/her password. If the program is successful in initiating recording, the output shown in figure 6 will appear. During development, the only location which Asterisk had permission to store recorded files was: /var/spool/asterisk

```
② ■ Terminal

~$ asterisk -rx "confbridge record start 100"

[sudo] password for darryn:
Recording started

Type 'stop' to end the recording
```

Figure 6: Recording Success

NB Note that a confbridge only exists once a user is present. Thus, a user must be in the confbridge before recording can take place. The following error occurs if no users are present:

```
Terminal

-$ asterisk -rx "confbridge record start 100"

[sudo] password for darryn:

Conference not found.

Command 'confbridge record start 100' failed.

Type 'stop' to end the recording
```

Figure 7: Confbridge Error

Upon typing stop,

Figure 8: Final Options

Appendix A: Code Listing

```
//includes
#include <iostream>
#include <string>
#include <stdlib.h>
//namespaces
using namespace std;
//global variables
string conf Number;
string command;
string option;
//functions
void welcome();
void stop();
void start();
void start()
          system("clear\n"); //clear console
          cout << "~\$ asterisk -rx \"confbridge record start " + confNumber + "\""
                    << endl << endl; //print command to terminal</pre>
          \verb|command| = "sudo asterisk -rx \ \ | confbridge record start " + confNumber + | confNumber + 
                   "\""; //create\ string\ of\ command
          system(command.c_str()); //run command in terminal
          cout << "\nType 'stop' to end the recording" << endl;</pre>
         while(true)
                    cin >> option;
                    if (option == "stop")
                   {stop();}
void stop()
          system("clear\n"); //clear console
          cout << "~$ asterisk -rx 'confbridge record stop " + confNumber + "'" <<</pre>
                     endl << endl; //print command to terminal</pre>
          command = "sudo asterisk -rx \"confbridge record stop " + confNumber + "
                  \""; //create string of command
          \verb|system(command.c_str()); // run command in terminal|\\
          cout << "\nOptions: " << endl;</pre>
          cout << "- 'open' to open the location of the recording" << endl;</pre>
          cout << "- 'start' to start a new recording" << endl;</pre>
          cout << "- 'exit' to close the program" << endl;</pre>
          while(true)
                    cin >> option;
                   if (option == "start")
                   {start();} else
                   if (option == "exit")
                   {exit(0);} else
                   if (option == "open")
                    {system("cd /var/spool/asterisk");} //unable to gain access
         }
```