SIP Server Usage Instructions

This minimal SIP server can establish calls successfully. To ensure proper operation, make sure to modify the following settings in sip_server.c **before compiling**:

1. **Set the SIP Server's IP Address**: The SIP server uses SIP_SERVER_IP_ADDRESS to generate its Via and Contact headers. It **must** be set to the SIP server's interface address.

#define SIP_SERVER_IP_ADDRESS "192.168.1.128" // Change this to your server's IP address

2. **Configure Location Entries for Softphones/UEs**: Each entry in location_entries represents a softphone/UE with its corresponding phone number, IP address, and SIP port, so that they can be correctly reached by the server.

```
location_entry_t location_entries[] = {
    {"1001", "192.168.1.101", 5060},
    {"1002", "192.168.1.101", 5070},
    {"1003", "192.168.0.1", 5060},
    {"1004", "192.168.0.1", 5070},
    {"1005", "192.168.184.1", 5060},
    {"1006", "192.168.184.1", 5070},
    {"1007", "192.168.192.1", 5060},
    {"1008", "192.168.192.1", 5070},
};  // Update with the actual IP addresses and ports of your softphones/UEs
```

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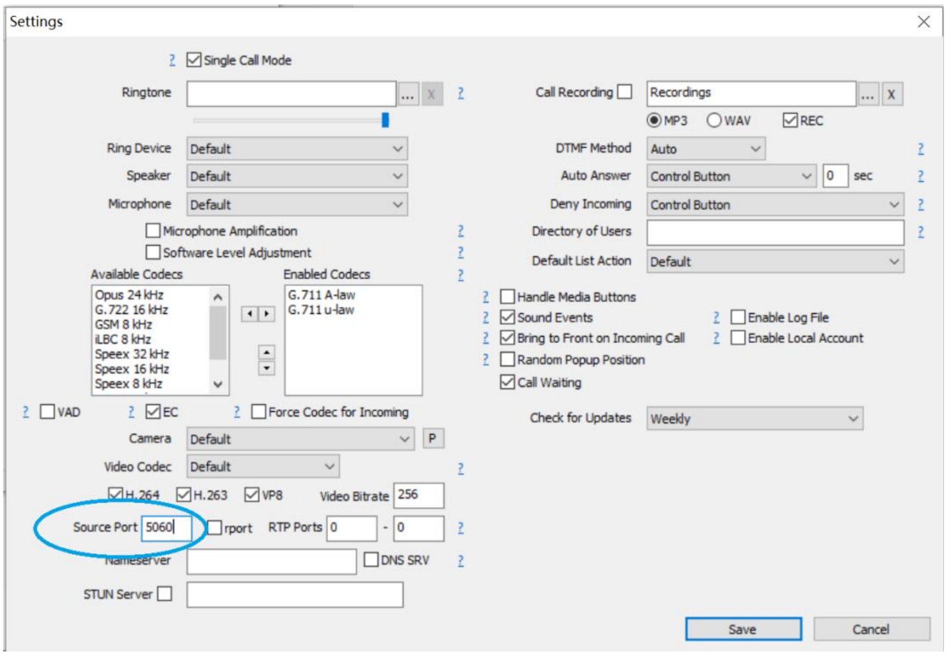
3. **Testing with MicroSIP:**

It is recommended to use **MicroSIP version 3.21.5** for interoperability testing between softphones.

Enter the Settings menu and configure the SIP port as shown in the screenshot, making sure it matches the port specified in the location_entries configuration.

Note: During testing, I found that binding a fixed SIP port on some computers may not work as expected. The softphone may detect a conflict and automatically choose a random port to bind and listen on. Therefore, using a self-developed testing program that allows binding to a fixed SIP port for the callee is a more feasible approach for testing.


Settings
Single Call Mode
Ringtone
Call Recording
Recordings
MP3
WAV
REC
Ring Device
Default
DTMF Method
Auto
Speaker
Default
Auto Answer
Control Button
0
sec
Microphone
Default
Deny Incoming
Control Button
Microphone Amplification
Directory of Users
Software Level Adjustment
Default List Action
Default
Available Codecs
Enabled Codecs
Opus 24 kHz
G.722 16 kHz
GSM 8 kHz
iLBC 8 kHz
Speex 32 kHz
Speex 16 kHz
Speex 8 kHz
G.711 A-law
G.711 u-law
Handle Media Buttons
Sound Events
Enable Log File
Bring to Front on Incoming Call
Enable Local Account
Random Popup Position
Call Waiting
VAD
EC
Force Codec for Incoming
Check for Updates
Weekly
Camera
Default
P
Video Codec
Default
H.264
H.263
VP8
Video Bitrate
256
Source Port
5060
rport
RTP Ports
0
-
0
Nameserver
DNS SRV
STUN Server
Save
Cancel