SIP phone

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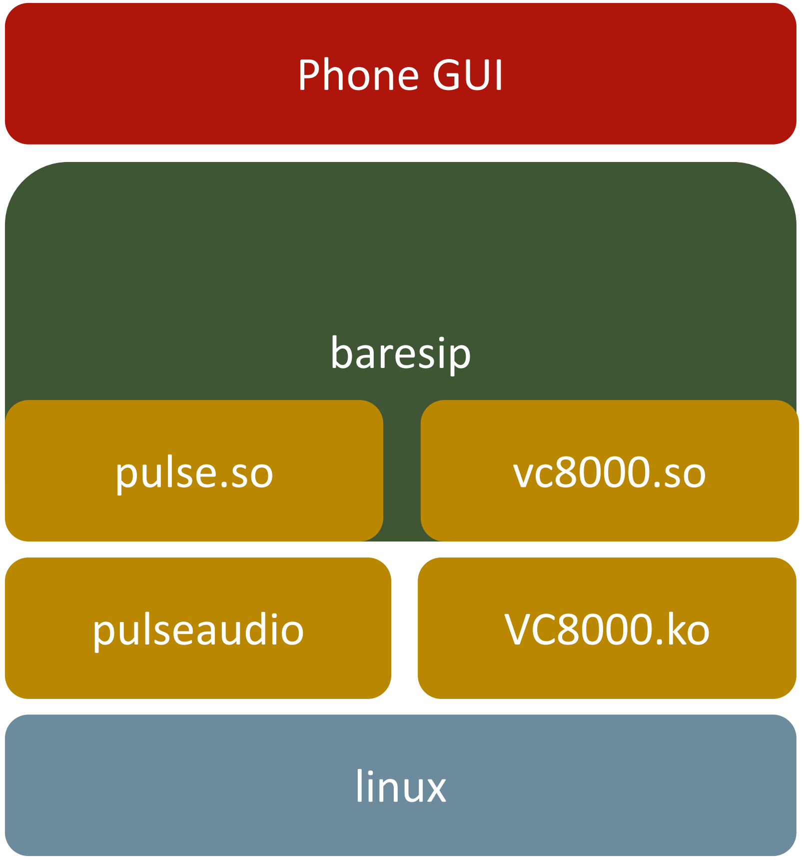
Outline

- baresip
- pulseaudio with WebRTC AEC3
- Phone GUI

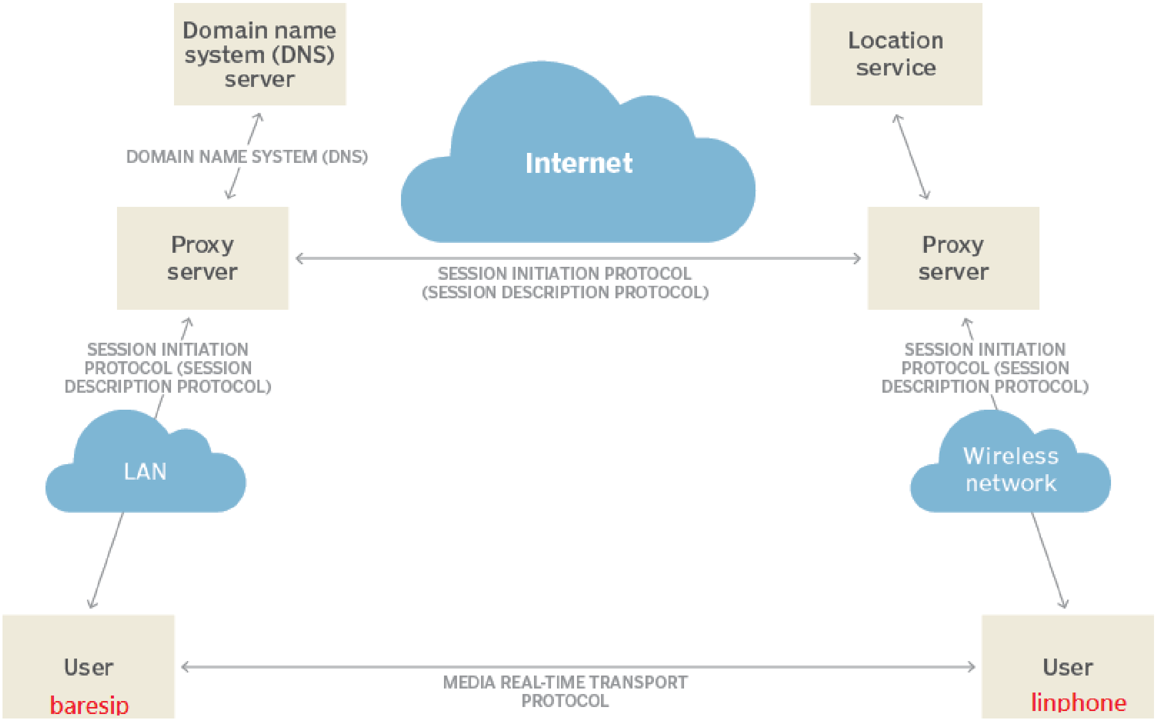
SIP phone

- Architecture



baresip

- A portable and modular SIP User-Agent with audio and video support
- Source: https://github.com/baresip/baresip



baresip

- On MA35D1
  - VC8000 h264 decoder
    - 1080P: 30fps
    - 720P: 60fps
  - WebRTC AEC3 support (software AEC)
  - Multi calls support
- Configuration path
  - /home/root/.baresip/

baresip

- Configuration (basic)
  - Audio device

```
# Audio
audio_path          /usr/share/baresip
audio_player        alsa,default
audio_source        alsa,default
audio_alert         alsa,default
ausrc_srate         16000
auplay_srate        16000
ausrc_channels      2
auplay_channels     2
```

  - Video device

```
# Video
#video_source       fakevideo
video_display       fb
video_size          1024x600
```

baresip

- Audio driver

  # Audio driver Modules

module	alsa.so
#module	pulse.so
#module	jack.so

- Audio codec

  # Audio codec Modules (in order)

#module	mpa.so
#module	codec2.so
#module	ilbc.so
module	speex.so

baresip

- Video display

```
# Video display modules
module          fb.so
#module         x11.so
#module         sdl.so
```

- Video codec

```
# Video codec Modules (in order)
#module         avcodec.so
#module         vp8.so
#module         vp9.so
module          vc8000.so
```

baresip

- VC8000

#vc8000	
vc8000_viddst_auto_size	yes
vc8000_viddst_size	1024x600
vc8000_viddst_offset_x	0
vc8000_viddst_offset_y	0
vc8000_panel_size	1024x600
vc8000_pp_enable	yes

- Call (Option for multi call)

# Call	
call_local_timeout	120
call_max_calls	4
call_hold_other_calls	no

pulseaudio with WebRTC AEC3

- A sound server system
  - Middleware between applications and hardware devices, either using ALSA or OSS.
  - Per-application volume controls
  - Support for multiple audio sources and sinks
  - Built- in sample conversion and resampling
- Good for multi calls application
  - Mixer and echo cancel module support
- Download from OpenNuvoton

  # git clone https://github.com/OpenNuvoton/pulseaudio_webrtc.git
- Configuration path
  - /home/root/.pulse/

pulseaudio with WebRTC AEC3

- pulseaudio support on baresip
  - Modify the configuration of baresip
    - Audio device

```
# Audio
audio_path        /usr/share/baresip
audio_player      pulse
audio_source      pulse
audio_alert       pulse
ausrc_srate       16000
auplay_srate      16000
ausrc_channels    2
auplay_channels   2
```

    - Audio driver

```
# Audio driver Modules
#module           alsa.so
module            pulse.so
#module           jack.so
```

pulseaudio with WebRTC AEC3

- Enable pulseaudio daemon

  root@ma35d1-evb:~#pulseaudio –D

- Set WebRTC AEC3 to default audio source/sink device

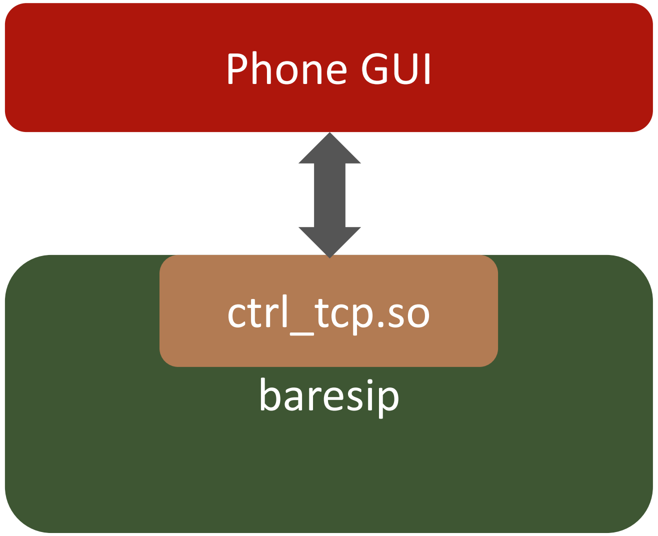
  root@ma35d1-evb:~#pactl load-module module-echo-cancel use_master_format=1 aec_method=webrtc source_name=echoCancel_source sink_name=echoCancel_sink

  root@ma35d1-evb:~#pactl set-default-source echoCancel_source

  root@ma35d1-evb:~#pactl set-default-sink echoCancel_sink

Phone GUI

- A simple GUI demo for baresip front-end. Base on QT.
- Using TCP socket (localhost:4444) to communicate with baresip
- Message exchange by JSON data format



Phone GUI

- Message format
  - len:{JSON body},
  - Command example

```
{
  "command" : "dial",
  "params"  : "sip:alice@atlanta.com",
  "token"   : "qwerasdf“        ←Optional. Included in the response if present
 }
```

Phone GUI

- Response example


```
{
 "response" : true,
 "ok"       : true,
 "data"     : "",
 "token"    : "qwerasdf"
}
```


- Event example


```
{
 "event"      : "true",
 "class"      : "call",
 "type"       : "CALL_CLOSED",
 "param"      : "Connection reset by peer",
 "accountaor" : "sip:alice@atlanta.com",
 "direction"  : "incoming",
 "peeruri"    : "sip:bob@biloxy.com",
 "id"         : "73a12546589651f8"
}
```

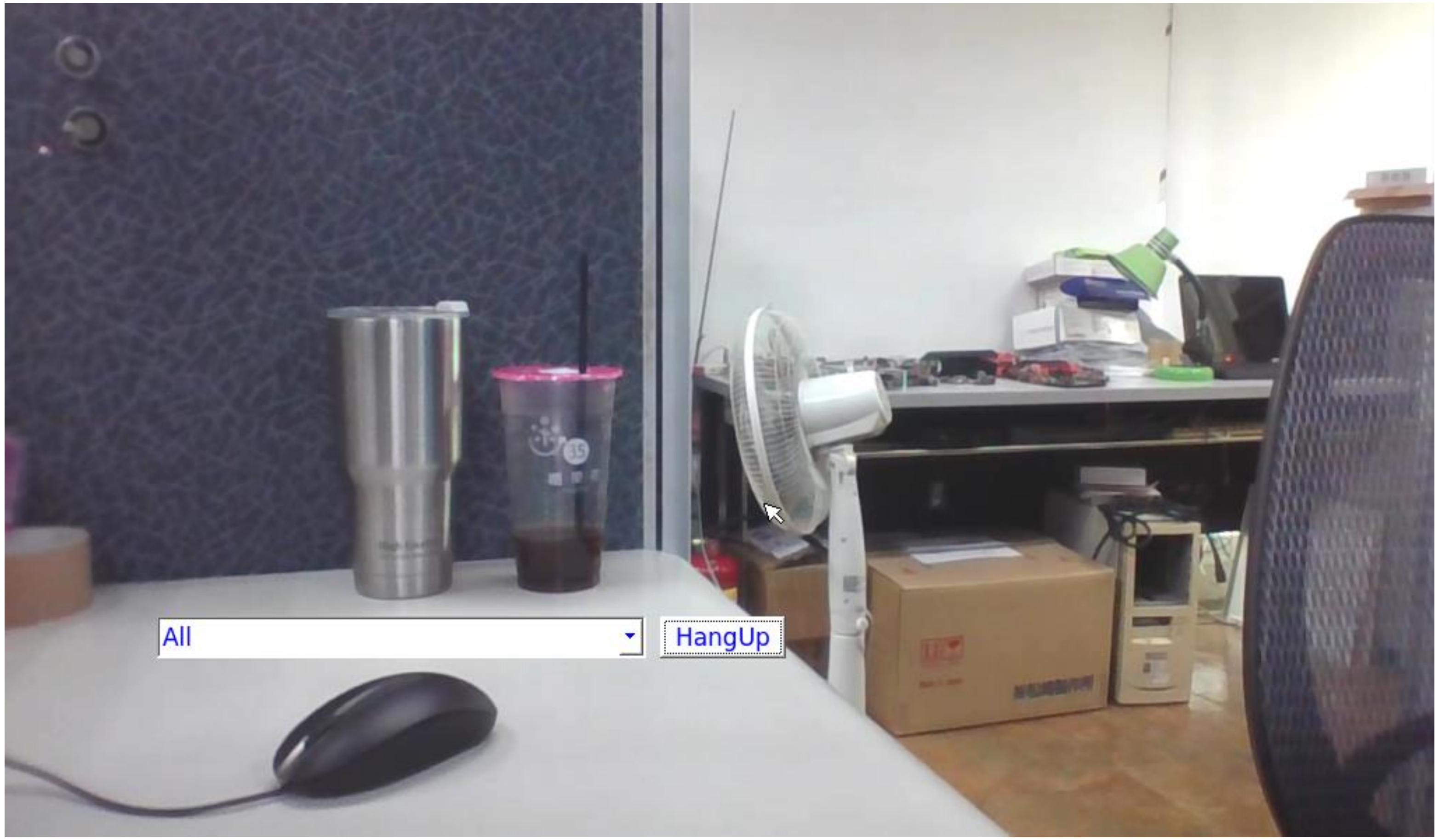
Phone GUI

- Phone GUI support on baresip
  - Modify the configuration of baresip

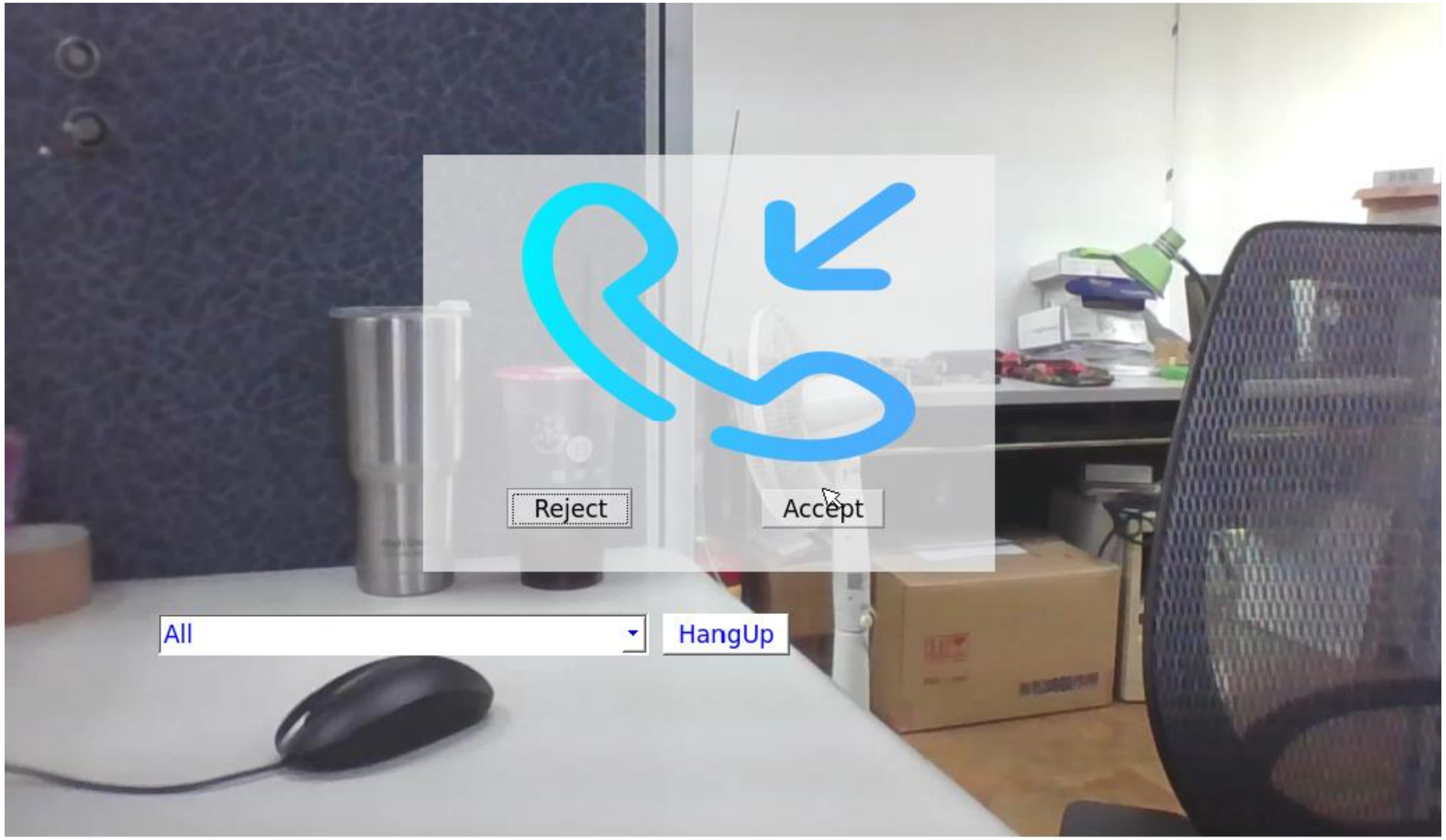
```
# Application Modules
#module_app          syslog.so
#module_app          mqtt.so
module_app           ctrl_tcp.so
module_app           vidloop.so
module_app           ctrl_dbus.so
#module_app          httpreq.so
#module_app          multicast.so
```

Demo - Standby

Demo - Talking



Demo – Incoming call during talking



Demo – Two calls

All | HangUp

All
sip:user@192.168.31.226
sip:chchen59.phone@sip.linphone.org

Joy of innovation

NUVOTON

Thank You

Danke

Merci

ありがとう

Gracias

Kiitos

감사합니다

धन्यबाद

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