SIP phone

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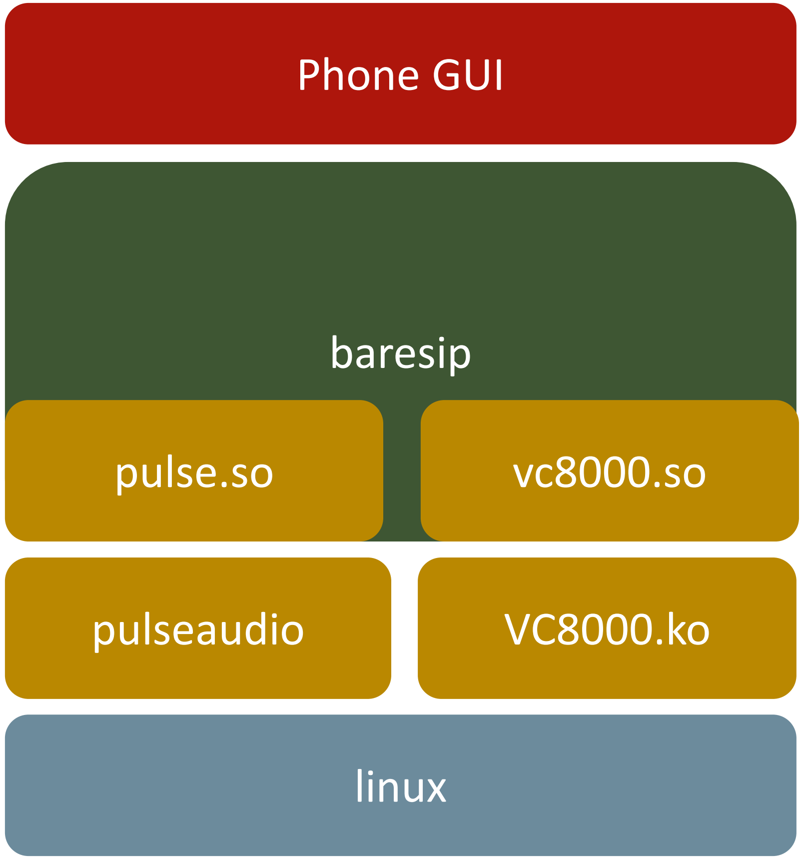
Outline

- baresip
- pulseaudio with WebRTC AEC3
- Phone GUI
- VC8000

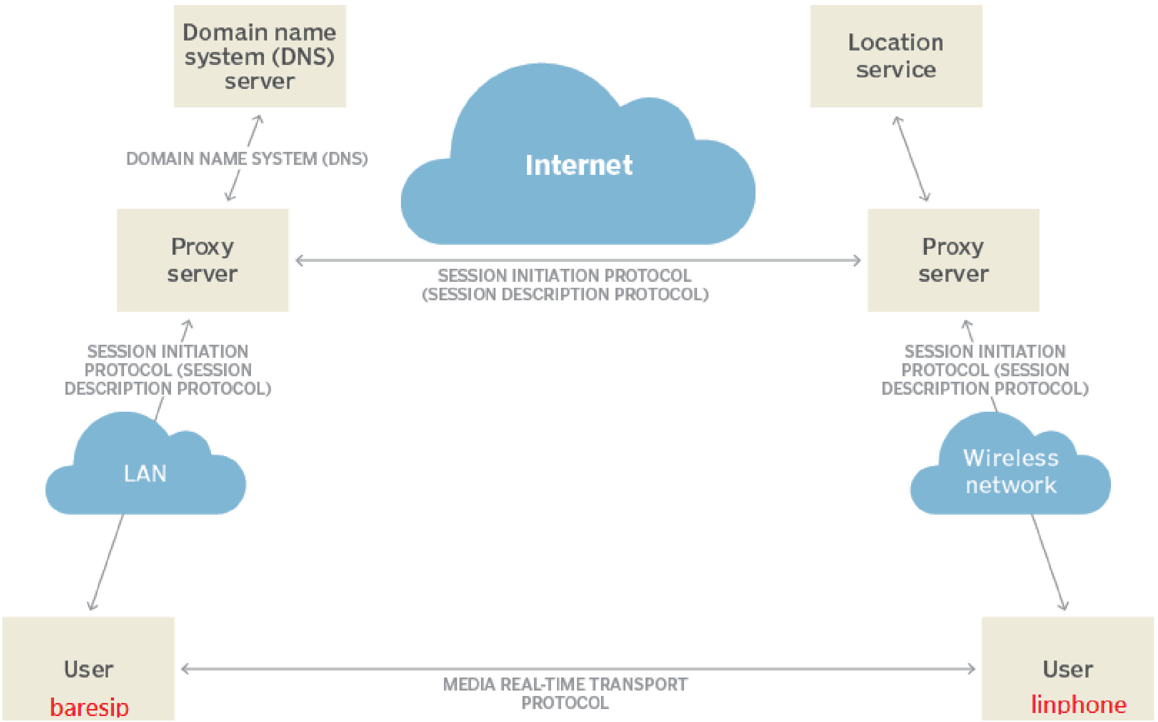
SIP phone

- 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

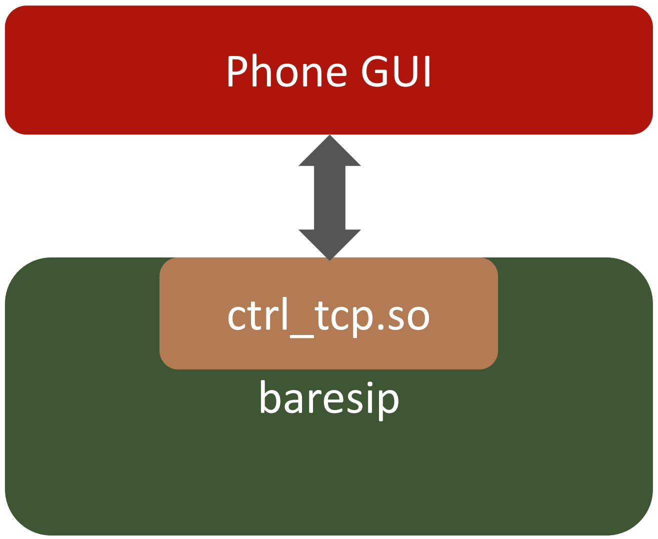
  root@ma35d1-evb:~#pactl load-module module-echo-cancel use_master_format=1 aec_method=webrtc source_name=echoCancel_source sink_name=echoCancel_sink

  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
```

VC8000

- MA35D1 H264/JPEG hardware decoder
- Multi decode instance support
- Post processing (scaling, rotation) support.
- Decode image to frame buffer 0(/dev/fb0) directly.

```
root@ma35d1-evb:/lib/modules/5.4.110#
root@ma35d1-evb:/lib/modules/5.4.110# modinfo ma35d1-vc8000.ko
filename:       /lib/modules/5.4.110/ma35d1-vc8000.ko
description:    MA35D1 VC8000 H264/JPEG driver (V4L2 M2M)
version:        0.4.1
author:         Nuvoton Technology Corporation
license:        GPL v2
srcversion:     B821B2D6F370AAB906675F9
alias:          of:N*T*Cnuvoton,ma35d1-vc8kC*
alias:          of:N*T*Cnuvoton,ma35d1-vc8k
depends:
intree:         Y
name:           ma35d1_vc8000
vermagic:       5.4.110 SMP mod_unload aarch64
parm:           debug:Debug level - higher value produces more verbose messages (int)
root@ma35d1-evb:/lib/modules/5.4.110# _
```

VC8000

- VC8000 kernel configuration
  - https://github.com/OpenNuvoton/MA35D1_linux-5.4.y/blob/master/arch/arm64/boot/dts/nuvoton/ma35d1.dtsi

```
reserved-memory {
        #address-cells = <2>;
        #size-cells = <2>;
        ranges;
        display_buf: display_buf@0 {
                reg = <0x0 0x8C800000 0x0 0x3000000>; /* 48MiB */
                no-map;
        };
        vc8k_buf: vc8k_buf@0 {
                reg = <0x0 0x8B000000 0x0 0x1800000>; /* 24MiB, used by H264 reference frame */
                no-map;
        };
};
```

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VC8000

```
/* VC8000 engine */
vc8k:vc8k@40290000 {
        compatible = "nuvoton,ma35d1-vc8k";
        reg = <0x0 0x40290000 0x0 0x1000>;
        clocks = <&clk vc8k_gate>;
        nuvoton,sys = <&sys>;
        interrupts = <GIC_SPI 28 IRQ_TYPE_LEVEL_HIGH>;
        memory-region = <&vc8k_buf>;
        pp_out_enable = "yes";  /* enable post processing */
        fb_width = <1024>;      /* frame buffer width */
        fb_height = <600>;      /* frame buffer height */
        fb_fmt = "RGB888";      /* RGB888, RGB565, or YUV420P */
        vid_max_width = <1920>; /* video source maximum width */
        vid_max_height = <1080>;    /* video source maximum height */
        vid_max_instance = <1>; /* video decode maximum instance */
        vid_refbuf_cnt = <6>;   /* video P/B frame maximum reference number */
        status = "okay";
};
```

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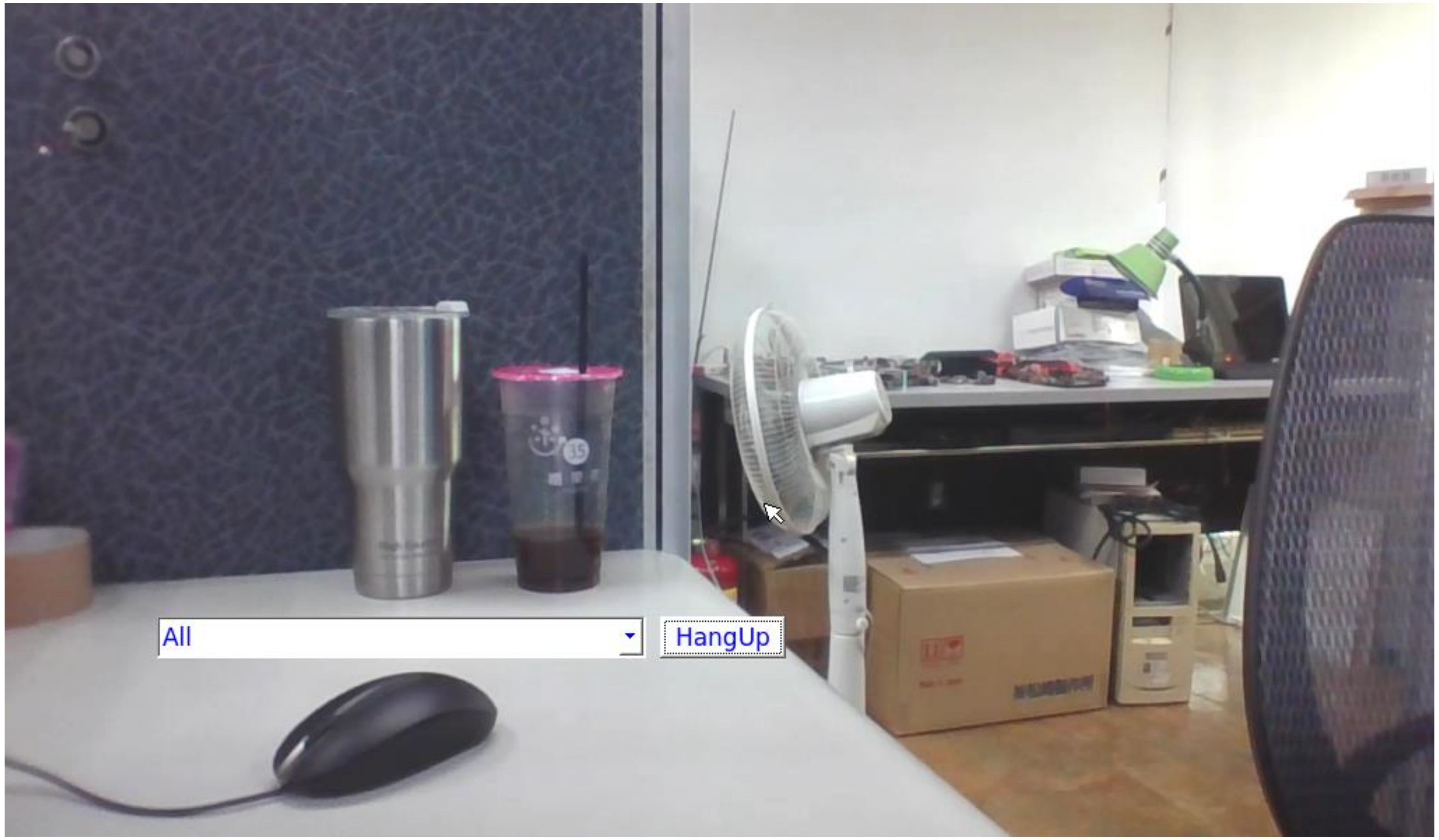
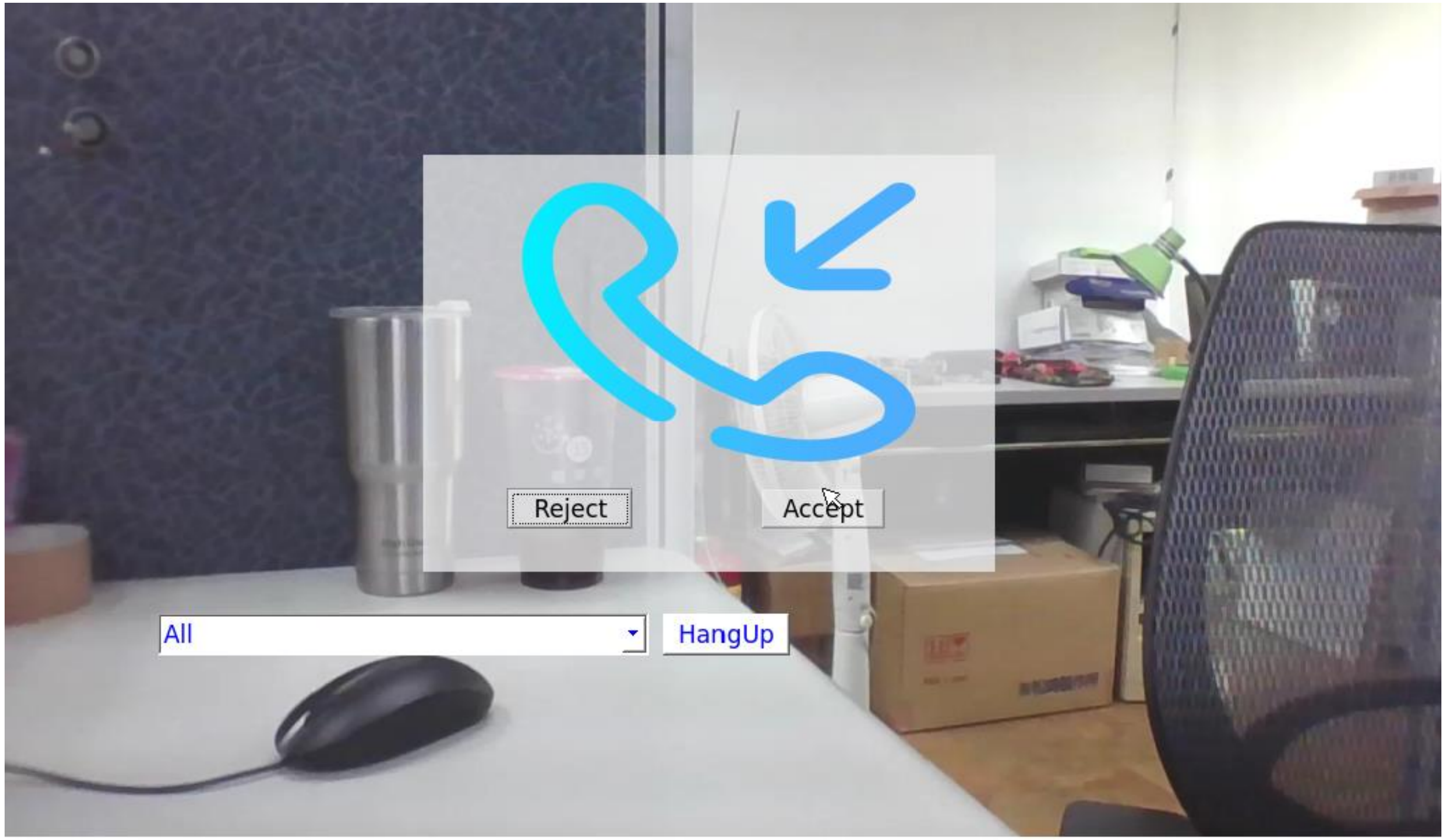
Demo - Standby

Demo - Talking



Demo – Incoming call during talking



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Demo – Multi calls

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

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Thank You

Danke

Merci

ありがとう

Gracias

Kiitos

감사합니다

धन्यबाद

شكرا لك

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