Lab 5 - QEMU-based Audio

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This lab involves modifying a WSPR-decoder to use pulseaudio on a QEMU emulated ARM processor. Once this is done, the WSPR-decoder can not only access local (emulated) hardware peripherals within QEMU, it can also use network-based audio souces from a local area network or the internet. After recompilation, the program that you write should also work on the Beaglebone.

After completing this lab, you should have a working WSPR decoder, interfaced to pulseaudio which you can test within QEMU.

1. Laboratory Experiment

Part 1 - Pulseaudio (40%)

Update your copy of the lab questions to get the lab5-audio files using the `git pull` command. Follow the instructions below to install pulseaudio on aarch64 ARM under `qemu` Debian Linux.

Start by booting QEMU.

```
$ qemu-system-aarch64 -M virt -cpu cortex-a53 -m 1G -initrd initrd.img-4.19.0-16-arm64 \
    -kernel vmlinuz-4.19.0-16-arm64 -append "root=/dev/vda2 console=ttyAMA0" \
    -drive if=virtio,file=debian-3607-aarch64.qcow2,format=qcow2,id=hd \
    -net user,hostfwd=tcp::10022-:22 -net nic -nographic -device intel-hda -device hda-duplex
&
```

Rather than login directly, I prefer to ssh to the QEMU machine so I can run Xwindows.

```
$ ssh -Y elec3607@localhost -p 10022
```

You can also copy files to it.

```
scp -P 10022 -r lab5-audio elec3607@localhost:
```

Install the alsa and pulseaudio.

```
$ sudo apt install libasound2 libasound2-plugins libasound2-doc alsa-utils pulseaudio pavucon
trol paprefs libpulse-dev libcanberra-gtk-dev
```

```
$ sudo usermod -aG audio,pulse,pulse-access elec3607
```

Then you have to log out and log in again.

```
$ aplay -l
**** List of PLAYBACK Hardware Devices ****
card 0: Intel [HDA Intel], device 0: Generic Analog [Generic Analog]
  Subdevices: 1/1
  Subdevice #0: subdevice #0
$ pulseaudio --start
$ pactl info
Server String: /run/user/1000/pulse/native
Library Protocol Version: 32
Server Protocol Version: 32
Is Local: yes
Client Index: 2
Tile Size: 65472
User Name: elec3607
Host Name: debian
Server Name: pulseaudio
Server Version: 12.2
Default Sample Specification: s16le 2ch 44100Hz
Default Channel Map: front-left,front-right
Default Sink: alsa_output.platform-4010000000.pcie-pci-0000_00_02.0.analog-stereo
Default Source: alsa_input.platform-4010000000.pcie-pci-0000_00_02.0.analog-stereo
Cookie: 1295:3f1d
```

Then you can start it up via systemctl.

```
$ systemctl --user enable pulseaudio
$ systemctl --user start pulseaudio
Job for pulseaudio.service failed because the control process exited with error code.
See "systemctl --user status pulseaudio.service" and "journalctl --user -xe" for details.
$ sudo shutdown -r now
```

Wait for everything to come back up and login again via ssh

```
$ systemctl --user status pulseaudio
● pulseaudio.service - Sound Service
   Loaded: loaded (/usr/lib/systemd/user/pulseaudio.service; enabled; vendor pre
   Active: active (running) since Sat 2021-04-10 15:59:48 AEST; 4s ago
 Main PID: 336 (pulseaudio)
   CGroup: /user.slice/user-1000.slice/user@1000.service/pulseaudio.service
           ├─336 /usr/bin/pulseaudio --daemonize=no
           └─357 /usr/lib/aarch64-linux-gnu/pulse/gsettings-helper
```

Now edit `/etc/pulse/default.pa` and append the line:

```
load-module module-null-sink sink_name=MySink format=s16le channels=1 rate=12000
```

We can restart the daemon and configure with.

```
$ systemctl --user restart pulseaudio
$ pacmd list-sinks
```

```
2 sink(s) available.
  * index: 0
        name: <alsa_output.platform-4010000000.pcie-pci-0000_00_02.0.analog-stereo>
        driver: <module-alsa-card.c>
        flags: HARDWARE HW_MUTE_CTRL HW_VOLUME_CTRL DECIBEL_VOLUME LATENCY FLAT_VOLUME DYNAMI
C_LATENCY
        state: IDLE
        suspend cause: (none)
        priority: 9039
        volume: front-left: 30419 /  46% / -20.00 dB,   front-right: 30419 /  46% / -20.00 dB
                balance 0.00
        base volume: 65536 / 100% / 0.00 dB
        volume steps: 65537
        muted: no
        current latency: 1283.09 ms
        max request: 344 KiB
        max rewind: 344 KiB
        monitor source: 0
        sample spec: s16le 2ch 44100Hz
        channel map: front-left,front-right
                     Stereo
        used by: 0
        linked by: 0
        configured latency: 2000.00 ms; range is 0.50 .. 2000.00 ms
        card: 0 <alsa_card.platform-4010000000.pcie-pci-0000_00_02.0>
        module: 6
        properties:
                alsa.resolution_bits = "16"
                device.api = "alsa"
                device.class = "sound"
                alsa.class = "generic"
                alsa.subclass = "generic-mix"
                alsa.name = "Generic Analog"
                alsa.id = "Generic Analog"
                alsa.subdevice = "0"
                alsa.subdevice_name = "subdevice #0"
                alsa.device = "0"
                alsa.card = "0"
                alsa.card_name = "HDA Intel"
                alsa.long_card_name = "HDA Intel at 0x10040000 irq 50"
                alsa.driver_name = "snd_hda_intel"
                device.bus_path = "platform-4010000000.pcie-pci-0000:00:02.0"
                sysfs.path = "/devices/platform/4010000000.pcie/pci0000:00/0000:00:02.0/soun
d/card0"
                device.bus = "pci"
                device.vendor.id = "8086"
                device.vendor.name = "Intel Corporation"
                device.product.id = "2668"
                device.product.name = "82801FB/FBM/FR/FW/FRW (ICH6 Family) High Definition Au
dio Controller (QEMU Virtual Machine)"
                device.form_factor = "internal"
                device.string = "front:0"
                device.buffering.buffer_size = "352800"
                device.buffering.fragment_size = "176400"
                device.access_mode = "mmap+timer"
                device.profile.name = "analog-stereo"
                device.profile.description = "Analog Stereo"
                device.description = "Built-in Audio Analog Stereo"
```

```
                alsa.mixer_name = "QEMU Generic"
                alsa.components = "HDA:1af40022,1af40022,00100101"
                module-udev-detect.discovered = "1"
                device.icon_name = "audio-card-pci"
        ports:
                analog-output-lineout: Line Out (priority 9900, latency offset 0 usec, availa
ble: unknown)
                        properties:

        active port: <analog-output-lineout>
    index: 1
        name: <MySink>
        driver: <module-null-sink.c>
        flags: DECIBEL_VOLUME LATENCY FLAT_VOLUME DYNAMIC_LATENCY
        state: IDLE
        suspend cause: (none)
        priority: 1000
        volume: mono: 65536 / 100% / 0.00 dB
                balance 0.00
        base volume: 65536 / 100% / 0.00 dB
        volume steps: 65537
        muted: no
        current latency: 1576.69 ms
        max request: 46 KiB
        max rewind: 46 KiB
        monitor source: 2
        sample spec: s16le 1ch 12000Hz
        channel map: mono
                     Mono
        used by: 0
        linked by: 0
        configured latency: 2000.00 ms; range is 0.50 .. 2000.00 ms
        module: 20
        properties:
                device.description = "Null Output"
                device.class = "abstract"
                device.icon_name = "audio-card"
$ pacmd list-sources
3 source(s) available.
    index: 0
        name: <alsa_output.platform-4010000000.pcie-pci-0000_00_02.0.analog-stereo.monitor>
        driver: <module-alsa-card.c>
        flags: DECIBEL_VOLUME LATENCY DYNAMIC_LATENCY
        state: SUSPENDED
        suspend cause: IDLE
        priority: 1030
        volume: front-left: 65536 / 100% / 0.00 dB,   front-right: 65536 / 100% / 0.00 dB
                balance 0.00
        base volume: 65536 / 100% / 0.00 dB
        volume steps: 65537
        muted: no
        current latency: 0.00 ms
        max rewind: 0 KiB
        sample spec: s16le 2ch 44100Hz
        channel map: front-left,front-right
                     Stereo
        used by: 0
        linked by: 0
```

```
        configured latency: 0.00 ms; range is 0.50 .. 2000.00 ms
        monitor_of: 0
        card: 0 <alsa_card.platform-4010000000.pcie-pci-0000_00_02.0>
        module: 6
        properties:
                device.description = "Monitor of Built-in Audio Analog Stereo"
                device.class = "monitor"
                alsa.card = "0"
                alsa.card_name = "HDA Intel"
                alsa.long_card_name = "HDA Intel at 0x10040000 irq 50"
                alsa.driver_name = "snd_hda_intel"
                device.bus_path = "platform-4010000000.pcie-pci-0000:00:02.0"
                sysfs.path = "/devices/platform/4010000000.pcie/pci0000:00/0000:00:02.0/soun
d/card0"
                device.bus = "pci"
                device.vendor.id = "8086"
                device.vendor.name = "Intel Corporation"
                device.product.id = "2668"
                device.product.name = "82801FB/FBM/FR/FW/FRW (ICH6 Family) High Definition Au
dio Controller (QEMU Virtual Machine)"
                device.form_factor = "internal"
                device.string = "0"
                module-udev-detect.discovered = "1"
                device.icon_name = "audio-card-pci"
  * index: 1
        name: <alsa_input.platform-4010000000.pcie-pci-0000_00_02.0.analog-stereo>
        driver: <module-alsa-card.c>
        flags: HARDWARE HW_MUTE_CTRL HW_VOLUME_CTRL DECIBEL_VOLUME LATENCY DYNAMIC_LATENCY
        state: SUSPENDED
        suspend cause: IDLE
        priority: 9039
        volume: front-left: 65536 / 100% / 0.00 dB,   front-right: 65536 / 100% / 0.00 dB
                balance 0.00
        base volume: 65536 / 100% / 0.00 dB
        volume steps: 65537
        muted: no
        current latency: 0.00 ms
        max rewind: 0 KiB
        sample spec: s16le 2ch 44100Hz
        channel map: front-left,front-right
                     Stereo
        used by: 0
        linked by: 0
        configured latency: 0.00 ms; range is 0.50 .. 2000.00 ms
        card: 0 <alsa_card.platform-4010000000.pcie-pci-0000_00_02.0>
        module: 6
        properties:
                alsa.resolution_bits = "16"
                device.api = "alsa"
                device.class = "sound"
                alsa.class = "generic"
                alsa.subclass = "generic-mix"
                alsa.name = "Generic Analog"
                alsa.id = "Generic Analog"
                alsa.subdevice = "0"
                alsa.subdevice_name = "subdevice #0"
                alsa.device = "0"
                alsa.card = "0"
```

```
                alsa.card_name = "HDA Intel"
                alsa.long_card_name = "HDA Intel at 0x10040000 irq 50"
                alsa.driver_name = "snd_hda_intel"
                device.bus_path = "platform-4010000000.pcie-pci-0000:00:02.0"
                sysfs.path = "/devices/platform/4010000000.pcie/pci0000:00/0000:00:02.0/soun
d/card0"
                device.bus = "pci"
                device.vendor.id = "8086"
                device.vendor.name = "Intel Corporation"
                device.product.id = "2668"
                device.product.name = "82801FB/FBM/FR/FW/FRW (ICH6 Family) High Definition Au
dio Controller (QEMU Virtual Machine)"
                device.form_factor = "internal"
                device.string = "front:0"
                device.buffering.buffer_size = "352800"
                device.buffering.fragment_size = "176400"
                device.access_mode = "mmap+timer"
                device.profile.name = "analog-stereo"
                device.profile.description = "Analog Stereo"
                device.description = "Built-in Audio Analog Stereo"
                alsa.mixer_name = "QEMU Generic"
                alsa.components = "HDA:1af40022,1af40022,00100101"
                module-udev-detect.discovered = "1"
                device.icon_name = "audio-card-pci"
        ports:
                analog-input-linein: Line In (priority 8100, latency offset 0 usec, availabl
e: unknown)
                        properties:

        active port: <analog-input-linein>
    index: 2
        name: <MySink.monitor>
        driver: <module-null-sink.c>
        flags: DECIBEL_VOLUME LATENCY DYNAMIC_LATENCY
        state: SUSPENDED
        suspend cause: IDLE
        priority: 1000
        volume: mono: 65536 / 100% / 0.00 dB
                balance 0.00
        base volume: 65536 / 100% / 0.00 dB
        volume steps: 65537
        muted: no
        current latency: 0.00 ms
        max rewind: 46 KiB
        sample spec: s16le 1ch 12000Hz
        channel map: mono
                     Mono
        used by: 0
        linked by: 0
        configured latency: 0.00 ms; range is 0.50 .. 2000.00 ms
        monitor_of: 1
        module: 20
        properties:
                device.description = "Monitor of Null Output"
                device.class = "monitor"
                device.icon_name = "audio-input-microphone"
```

The Volume Control program allows you to monitor what sources and sinks are available. Play the sample data file and run `pavucontrol`.

```
$ paplay data/iq-16b.wav&
$ pavucontrol&
```

- In the Configuration tab, select Off as the Built-in Audio profile
- If you look at Input Devices and Show: All Input Devices, you should only see Monitor of Null Output and the meter should indicate something is playing.
- If you look at Output Devices and Show: All Output Devices, you should only see Null Output and the meter should indicate something is playing. `

Part 2 - Recording Program (50%)

In the parecfile directory, `parecfile.c` is a program that records some data via pulseaudio, and then writes it to stdout. As its name suggests, the pulseaudio simple interface is very simple and its documentation is available **[here (https://www.freedesktop.org/wiki/Software/PulseAudio/Documentation/)](https://www.freedesktop.org/wiki/Software/PulseAudio/Documentation/)**. Using the `parecfile/parecfile.c` code as an example, modify `wsprcan/wsprd.c` so that instead of reading its input from a wav file, it reads it from pulseaudio.

Demonstrate that your program works by playing a file in the background, and decoding it with your modified program.

```
$ paplay data/iq-16b.wav &
$ wsprcan/k9an-wsprd
Writing data/wf-1618037425.wav
mode  -1 -3.1   0.001437 -3  VK2RG QF56 30
mode -19 -2.9   0.001455 -2  VK3GOD QF23 23
mode -21 -2.9   0.001478 -2  VK4YEH QG62 37
<DecodeFinished,data/wf-1618037425.wav,3>
[3]+  Done                    paplay data/iq-16b.wav
```

Part 3 - wsprwait (10%)

Study the bash script wsprwait. Explain to the tutor what it does and why it might be useful in this project.