Enable High Quality Audio on Linux

Gamunu Balagalla

If you are switched to Linux from Windows you will feel some of the Linux features is much worse than the Windows. One of the best example is sound.

When I compare the sound from Google Play music, YouTube and there is no doubt, music is so much better in the Windows OS because of drivers. In Linux the music sounds like from a can. There is no advanced sound panel to tweak, or is it?

At first, this is what I thought. But actually this is not true, Linux is much more capable and flexible. It happens to be that distribution maintainers of Ubuntu, Arch like distros have configured the default configuration of sound, graphics and fonts to work with all the computers that you can find on the market, or for at least work with most of them.

So, How can you configure the channels, audio depth or Hz, like in windows? It is quite simpler process than you think.

First you have to do little bit of tweaking of the Pulse configuration, to match your preference by creating the ***daemon.conf*** file in your home config directory:

```
$ vim ~/.config/pulse/daemon.conf
```

Or you can directly modify */etc/pulse/daemon.conf*. Although be careful and have a backup of original configuration to restore if you messed it up.

And add or modify below options:

```
default-sample-format = float32le
default-sample-rate = 48000
```

```
alternate-sample-rate = 44100
default-sample-channels = 2
default-channel-map = front-left,front-right
default-fragments = 2
default-fragment-size-msec = 125
resample-method = soxr-vhq
enable-lfe-remixing = no
high-priority = yes
nice-level = -11
realtime-scheduling = yes
realtime-priority = 9
rlimit-rtprio = 9
daemonize = no
```

default-sample-format:

The default sampling format of the sampler. The quality will be different for each sample format. For me, *float32le* appears to be producing the highest quality of sound.

When determining sample format you must select it depending on your CPU architecture's byte order or also called Endianness. You can determine your CPU's byte order by using below command:

```
$ lscpu | grep 'Byte Order'
Byte Order:          Little Endian
```

In my case it's *Little Endian* hence I have selected *float32**le***. If your output returns *Big Endian then* select *float32**be*** sampling format.

Available sample formats:

`u8, s16le, s16be, s24le, s24be, s24-32le, s24-32be, s32le, s32be float32le, float32be, ulaw, alaw`

default-sample-rate and alternate-sample-rate*:*

This determines in ADC or DAC conversion sampling rate and the alternative sample rate. The sound system will determine which to use, either the default or alternative automatically.

In this configuration, I have used 48000Hz. This is more than enough to get higher quality sound. You can learn more about sampling here: Sampling_(signal_processing)

default-sample-channels and default-channel-map:

The number of channels to be sampled and their configuration map. I usually listen to music using my headset so I have selected the number of channels to be 2 and channel map to be front-left and front-right.

In case if you have 2.1 or 5.1 surround sound system, you will have to consult the pulse documentation to figure out how to do tweak the configuration or comment it out, then the system will determine the best settings for you.

default-fragments and default-fragment-size-msec:

Some hardware drivers require the hardware playback buffer to be subdivided into several fragments. These configurations determines the number of fragments and a duration of a single fragment. Defaults are 4 and 25ms so the total buffer will be 100ms long. I have selected 2 and 125ms.

If you have a good sound card you can ignore this configuration since most of newer sound drivers support timer-base scheduling.

resample-method:

The resampling algorithm to use. I have selected soxr-vhq. It is the point sampler, which offers better sound quality than speex-* methods however it is CPU intensive.

Available values:

`src-sinc-best-quality, src-sinc-medium-quality, src-sinc-fastest, src-zero-order-hold, src-linear, trivial, speex-float-N, speex-fixed-N, ffmpeg, soxr-mq, soxr-hq, soxr-vhq.`

enable-lfe-remixing:

This determines the upmixing or downmixing channel behaviour. If disabled output LFE channel is available the signal on the input LFE channel will be ignored.

The Low-frequency effects (LFE) channel is the name of an audio track specifically intended for deep, low-pitched sounds ranging from 3–120 Hz

Scheduling configuration of pulse daemon:

```
high-priority = yes
nice-level = -11
realtime-scheduling = yes
realtime-priority = 9
rlimit-rtprio = 9
daemonize = no
```

high-priority setting makes the pulse daemon is a high priority process. If you enable the realtime-scheduling, it can cause a system lock up. Although this gives you a high priority IO threads of pulse daemon.

Finally you have to configure the alsa to get the best audio output:

```
$ vim /etc/asound.conf
```

The default configuration will be something similar to following:

```
# Use PulseAudio by default
pcm.!default {
  type pulse
  fallback "sysdefault"
  hint {
    show on
    description "Default ALSA Output (currently PulseAudio Sound Server)"
  }
}
```

Change it to this:

```
# Use PulseAudio plugin hw
pcm.!default {
   type plug
   slave.pcm hw
}
```

slave.pcm `**hw**` plugin will communicates directly with ALSA kernel driver. It is a raw communication without any conversion thus giving you a low latency audio output.

Then reboot your computer or restart Pulse and ALSA processes.

That's it. Enjoy your favourite music on Linux hereafter. You will notice that sounds are much better and less noisy than Windows.