Open Source Podcasting Tools

Software to locally record and produce a podcast on Linux

Presenter Background

- Mark Caldwell Walker
- Linux user (Fedora)
- Relevant experience for this topic:
  On-air broadcasting, voice over, book narration,
  Audio production—on Linux
- Podcasts I co-host and edit/engineer:
  CreativeCoasts.org
  FredTechByte.com
- Radio amateur: AC3EW
- Personal introduction website:
  marwalk.net

Topics Covered

- Hardware Components
- Digital Audio Basics
- Audio Components (on Linux)
- Open Source Audio Utilities
- Remote Live Sound (for recording)
- CLI Audio Tools (and Bash scripts to use them)

Hardware Components

- Microphone
- Phantom Power (not always needed)
- Pre-amp
- Cables
- Headphones
- Computer (or portable digital recorder)

Microphones

- **Dynamic**—less sensitive
  - **Diaphragm** is attached to a **coil**, which moves with it through a magnetic field
  - Cannot respond as readily (or as quickly) to subtle low energy sound waves
- **Condenser**—more sensitive
  - **Diaphragm** is used as (or to drive) **a plate** of a specialized **capacitor**
  - Must have **Phantom Power**

Microphone Examples

- **Shure SM7B**
  - Dynamic type
  - Needs pre-amp
  - $399.00

Microphone Examples (cont'd)

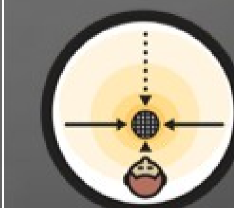
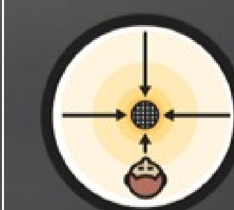
- **eBerry Cobblestone Microphone**
  - Condenser type
  - USB connected and powered
  - $44.99

Microphone Examples (cont'd)

- **Blue Yeti**
  - 3 condenser mics in a directional triangular array
  - USB connected and powered
  - $129.99

Microphone Examples (cont'd)

- **Blue Yeti—Directional Pattern Settings**

PATTERN MODES	PATTERN SETTING SYMBOL	SOUND SOURCE & DIRECTION
CARDIOID MODE Perfect for podcasting, Twitch streaming, music recording, voice overs and instruments. Cardioid mode records sound sources that are directly in front of the microphone, delivering rich, full-bodied sound.		
STEREO MODE Uses both the left and right channels to capture a wide, realistic sound image–ideal for recording acoustic guitar or choir and immersive experiences like ASMR videos.		
OMNIDIRECTIONAL MODE Picks up sound equally from all around the mic. It's best used in situations when you want to capture the ambience of "being there"–like recording a band's live performance, a multi-person podcast or a conference call.		
BIDIRECTIONAL Records from both the front and rear of the microphone–good for recording a duet or a two-person interview.		

Phantom Power

- DC electric power (usually 48v) delivered to a **condenser** microphone
- A **condenser** microphone **<u>will not work</u>** without phantom power
- May (or may not) damage a dynamic mic
- Know what **type of mic** you have, and **read the specs!**

Phantom Power (cont’d)

- Sources of Phantom Power:
  - Pre-amp
  - In-line power insertion unit
  - Mixer board
  - Specialized external power

Cables

- **XLR** (the letters are from legacy history)
  - **X**—Arbitrary inherited type indicator
  - **L**—Locking
  - **R**—Rubber boot on the female version
  - There's no left/right in "LR" as individual mics are monaural
  - The pinouts are basically:
    - hot/positive
    - cold/return
    - ground/shield.

Cables (cont’d)

- Microphone and **audio cables** in general usually carry **only unidirectional** signals.
- **XLR** connector common practices (Generally signal flows from male XLR to female XLR)
  - female XLR to microphone, (to get sound from mic’s male XLR)
  - male XLR to equipment (to provide sound/signal to the next stage in the audio chain)

Cables (converting between types)

- Problem—Sound card 3.5mm (1/8 inch) jacks obviously will not mate with XLR connectors.
- Solution—A cable specially wired with:
  - a female XLR connector at one end for receiving sound output (from a preamp/mixing board, or direct from the microphone)
  - at the other end a stereo mini-plug to go into the mic input jack on the sound card

Mugig Phantom Power Supply

- The **chrome color male XLR** connector carries analog **audio in** from the preamp, and **phantom power out** to the preamp;
- the **black color female XLR** connector takes **audio only out** to a duplicate channel stereo mini-plug **into the sound card** on the DAW.

Pre-amps

- **Cloud Microphones Cl-1 Cloudlifter**

The **black color connector** on the left is a male XLR coming **directly from the mic**

The **chrome color connector** on the right is the **preamp's analog audio output** as well as its **phantom power input**, both going through the same female XLR connector.

Pre-amps (cont’d)

- **CEntrance MicPort Pro USB Mic Preamp**

The **female XLR connection at the top** of the image receives the analog **audio input** from the microphone.

The **USB connection** at the bottom (and **shown in the view to the right**) takes power in from the computer to run the pre-amp, and provides a <u>**digital signal**</u> **out to the computer** through the same USB cable.

This unit **can provide phantom power** to the microphone if needed—selectable through a small toggle button.

Essential Digital Audio Basics

- **Pulse-Code Modulation (PCM)**
  - Linear PCM raw audio, is **just 1s and 0s** in a form that represents **discreet audio levels** for each instantaneous sample saved.
- **Terminology**:
  - **Bit Depth**—the number of bits used per sample, such as 16, 24, and 32 bit float
  - **Sample Rate**—the number of PCM audio samples taken/provided per second, such as 44,100 and 48,000 samples per second

Essential Digital Audio Basics

Each **bit depth** level is **6 dB** of dynamic range:

- **16 bit depth** = **96 dB** of dynamic range = **65,536 levels**
- **24 bit depth** = **144 dB** of dynamic range = **16,777,216 levels**

Actual analog audio (**physical sound waves**) has a **maximum dynamic range of ≈120 dB**, which equates to **20 bit depth**.

The **next binary** related point **is 24** (multiple of 8), the next available audio bit depth choice is **24 bit**.

Essential Digital Audio Basics

- **Nyquist rate**—the **sampling rate** must be at least **twice the highest frequency** in the audio.
  - the highest frequency that can be accurately reproduced at a sample rate of **44,100** samples per second is half that, or 22,050 Hz—that's the standard for **audio CD**s.
  - if you're producing audio for a **DVD**, the standard is **48,000** samples per second.
- **Avoid re-sampling** if at all possible—due to "rounding errors" in the interpolations and other complex processes inside the equipment

Essential Digital Audio Basics

- **Normalization**
  - **Peak—**relative to the loudest sample in the recording, the largest PCM binary value
  - **RMS**—**R**oot, **M**ean, **S**quare (basically <u>average</u>)
- **LUFS** (**L**oudness **U**nits, referenced to **F**ull **S**cale)
  - European Broadcasting Union (EBU) developed **EBU Recommendation R 128**
  - "…uses a sliding rectangular time window of length 0.4 s" (basically in **400 ms increments**)

Software Components

- **ALSA**—Advanced Linux Sound Architecture
  - On most Linux platforms, it's ALSA that provides their audio functionality.
- **PulseAudio**—A way of managing ALSA
  - server/service that sits between the audio applications and the ALSA device kernel modules sending the sound to and from the hardware

Software Components

- **PulseAudio Volume Control**—A GUI tool for volume control on Linux
  - Launch with `pavucontrol` in a CLI shell
  - **PA Terminology**:
    - Source—Sound comes out of sources
      A microphone is an obvious source.
    - Sink—Receives sound from something else
      A sound card or microphone jack is a sink
  - Provides a **real time view** of what sound sources and sinks are active at any instant. **Changes with which devices are active**.

Open Source Audio Utilities

- **SoX**—Sound eXchange
  Swiss Army knife of sound processing programs
  - CLI
  - Processes and converts audio files
- **FFmpeg**—a media file format conversion utility that is very capable
  - CLI
  - Effects processing
  - Convert from .wav to .mp3
  - Normalization (Peak, RMS, and LUFS)

The Levelator® (by the Conversations Network)

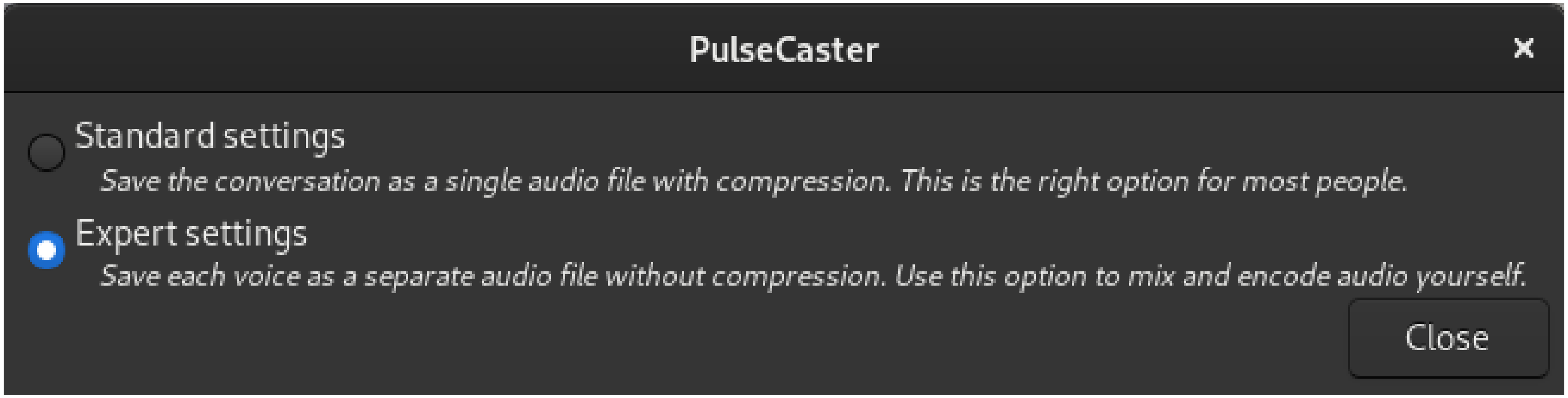
- A problem that neither RMS nor LUFS normalization solves: that of **uneven levels** within an audio file or files.

- Made for Linux, Windows, and Mac (The Windows executable can be run with wine on Linux)

Remote Live Sound

- Podcast episode with **remote** participants
- **PulseCaster** (in your Linux package repositories)
  - Utilizes PulseAudio to split local and remote audio into separate recording files
  - Use with Skype, Zoom, etc.
- **Online Services** for Remote Recording
  - **SquadCast** (subscription based)
  - **Cleanfeed** (more open source oriented)
    - used by professional broadcast stations for remote program transport over the Internet

PulseCaster



Stop Web Apps from Changing audio level

Edit these files (as root):
/usr/share/pulseaudio/alsa-mixer/paths/analog-input-internal-mic.conf
/usr/share/pulseaudio/alsa-mixer/paths_backup/analog-input-internal-mic.conf

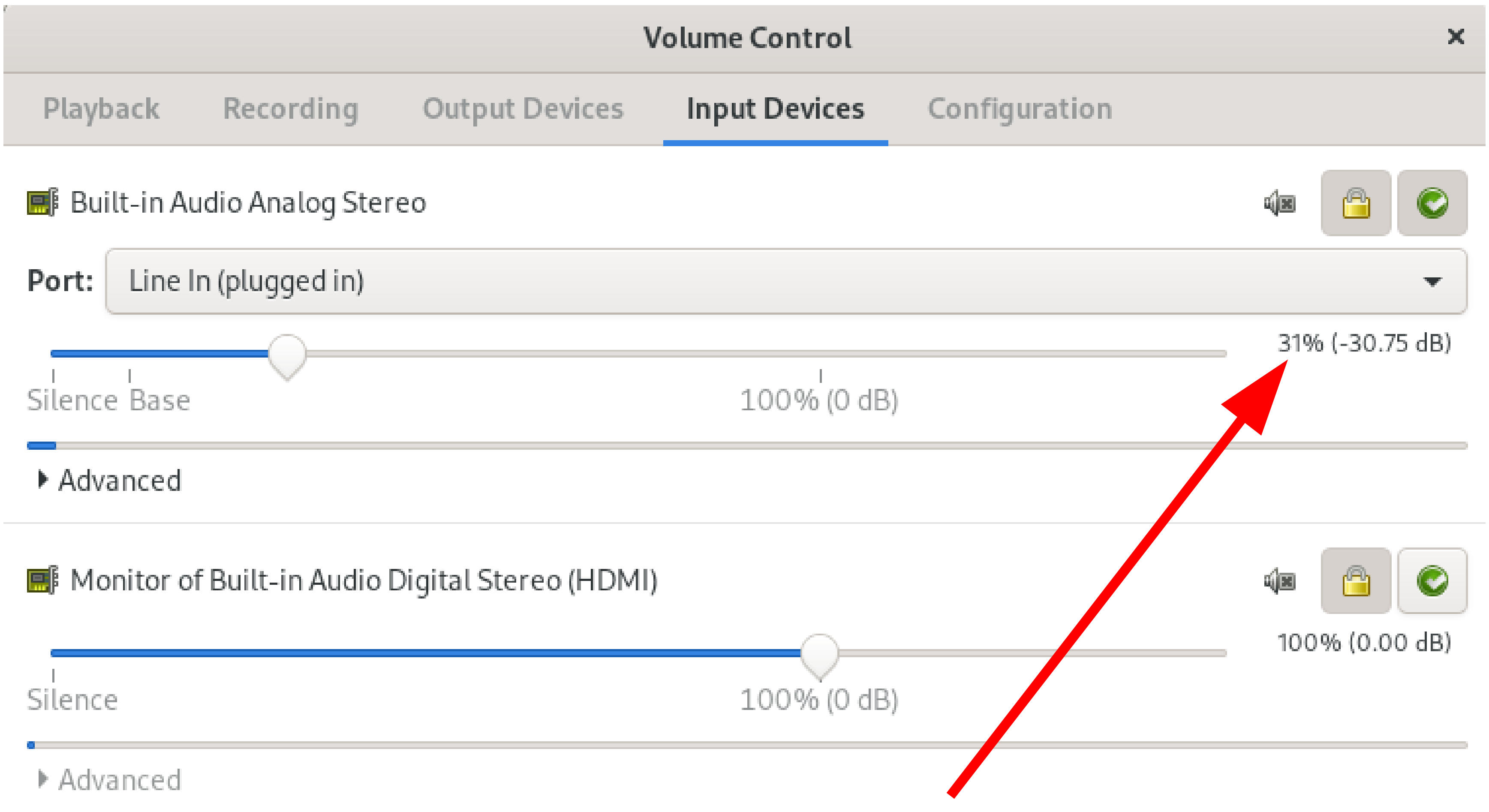
Grep for “volume = merge”
Change to “volume = 30”
(the “30” is for 30% of full level, but you can use any static value)

Example stanza:

```
override-map.2 = all-left,all-right

[Element Internal Mic Boost]
required-any = any
switch = select
volume = 30
```

Stop Web Apps from Changing audio level *PulseAudio Results*



To Recover from Temporary PulseAudio Session Settings

```
[you@localhost ~] $ pulseaudio -k
```

Restarts the PA daemon
And loads the permanently stored configs

Audacity—your go-to editor

- Open Source, and as capable as expensive commercial software
- Extensive feature list (non-exhaustive):
  - Punch and Roll
  - Keyboard shortcuts and macros
  - Effects menu sorted to suit
  - Noise cancellation
  - Labels and Label Tracks
  - Compression, Scrubbing, Tempo adjustment, ...

Audacity—freezes sometimes

Usually in the middle of heavy editing

- Won’t respond to either mouse or keyboard
- Solution: Kill the process with a script

```
#!/bin/bash
#
for KILLAUD in $(ps ax | grep audacity | grep -v grep | awk -F" " '{print $1}')
do
kill ${KILLAUD}
done
#
```

- Then restart Audacity—and recover your work
- This is why CTRL-S should be a muscle reflex

Normalization w/ FFmpeg

```
#! /bin/bash
# To normalize the last .wav file written

ls -1rt *-ed.wav | tail -1 > nfile

origaudio=$(cat nfile)

normlevel="-20" # meaning -20 LUFS (not dBFS)

ffmpeg -i ${origaudio} -af volumedetect -f null /dev/null > orig-RMS.txt 2>&1
```

Continued →

Normalization w/ FFmpeg (cont'd)

```
grep "mean_volume" orig-RMS.txt > orig-RMS-level.txt

grep "max_volume" orig-RMS.txt >> orig-RMS-level.txt

ffmpeg -i ${origaudio} -ac 1 -ar 44100 "1C-${origaudio}"

ffmpeg -i "1C-${origaudio}" -ar 44100 -af loudnorm=I=${normlevel}: \
TP=-3:LRA=7 "LUFS-${origaudio}"

ffmpeg -i "LUFS-${origaudio}" -af volumedetect -f null /dev/null > \
LUFS-RMS.txt 2>&1

grep "mean_volume" LUFS-RMS.txt > LUFS-RMS-level.txt

grep "max_volume" LUFS-RMS.txt >> LUFS-RMS-level.txt
```

Continued →

Normalization w/ FFmpeg (cont'd)

```
Clear

ls -1 ${origaudio}

cat orig-RMS-level.txt | awk -F" " '{print $4 $5}'

printf "\n"

ls -1 "LUFS-${origaudio}"

cat LUFS-RMS-level.txt | awk -F" " '{print $4 $5}'

printf "\n"

rm "1C-${origaudio}"
```

Normalization w/ FFmpeg Example Result

test45-ed.wav
mean_volume:-19.5
max_volume:-2.0

LUFS-test45-ed.wav
mean_volume:-19.4
max_volume:-3.0

RMS measurements in dBFS
of the Original file

RMS measurements in dBFS
of the Resultant file
After Loudness Normalization to
-20 LUFS, -3 dBTP (not dBFS),
Loudness Range of 7 LUFS

Concatenate with SoX Sound eXchange

- File concatenation works with .mp3 files:

  `cat file1.mp3 file2.mp3 > target.mp3`

- File concatenation does not work with .wav files:

  ~~`cat file1.wav file2.wav > target.wav`~~

- Solution: Use SoX to rewrite the .wav header:

  `sox file1.wav file2.wav <... file-n.wav> target.wav`

  - Explanation: The .wav header contains file length information, and the first header isn't changed with just file system `cat` concatenation

Important Audio Specs

- Bit Depth
- Sample Rate
- Normalization Level (RMS/LUFS)
- Max Peak
- Max Noise Floor
- MP3 Bitrate in kb/s (128≈radio; 192≈CD; VBR)

Also:

- No Clipping or Flat-Topping at any level
- No extraneous sound artifacts
- Consistent “room tone”

Practical Examples

- Questions
- Demos
- Experiments

- Audio Production Quick Take Videos youtube.com/channel/UCYhrwwipKrccIc2_xXAD9pQ