Open Source Podcasting Tools

Software to locally record and produce a podcast on Linux

Hardware Components

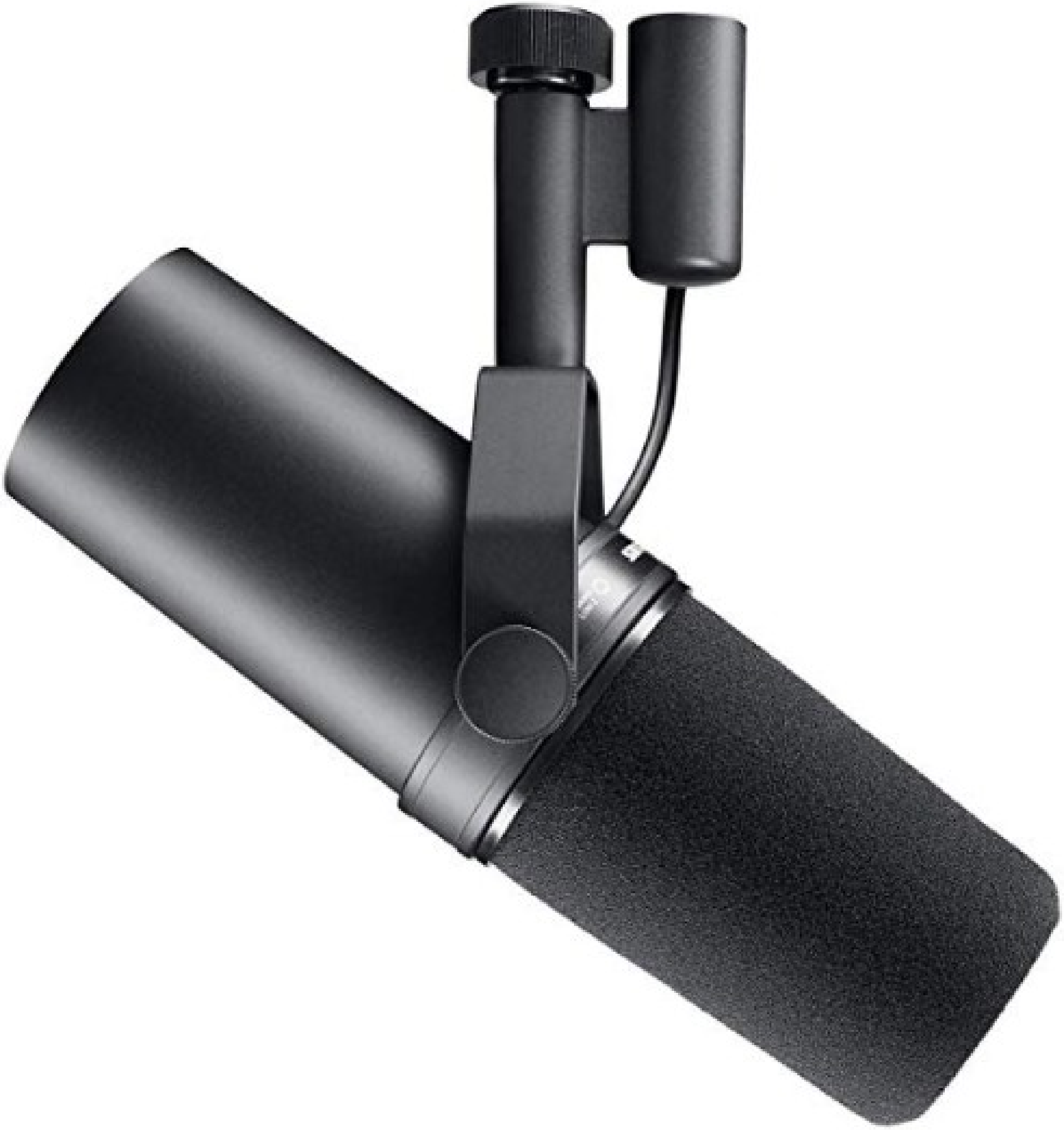
- Microphone
- Phantom Power
- Pre-amp
- Cables
- Headphones
- Computer (or portable digital recorder)

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**

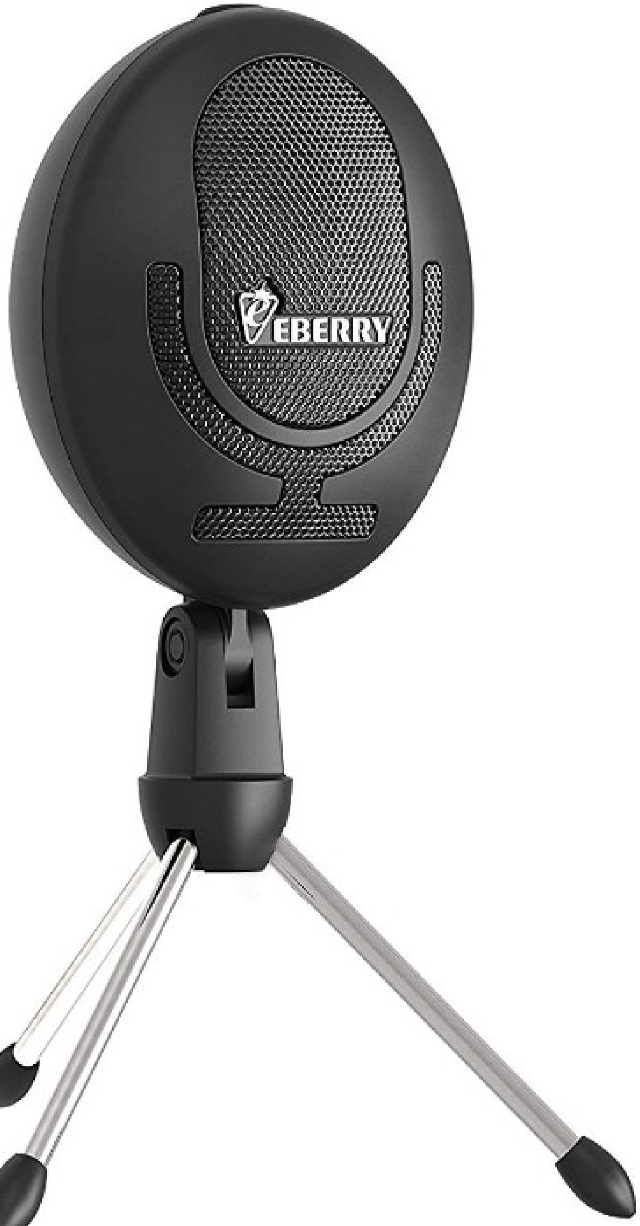
Microphone Examples

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



Microphone Examples

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



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)

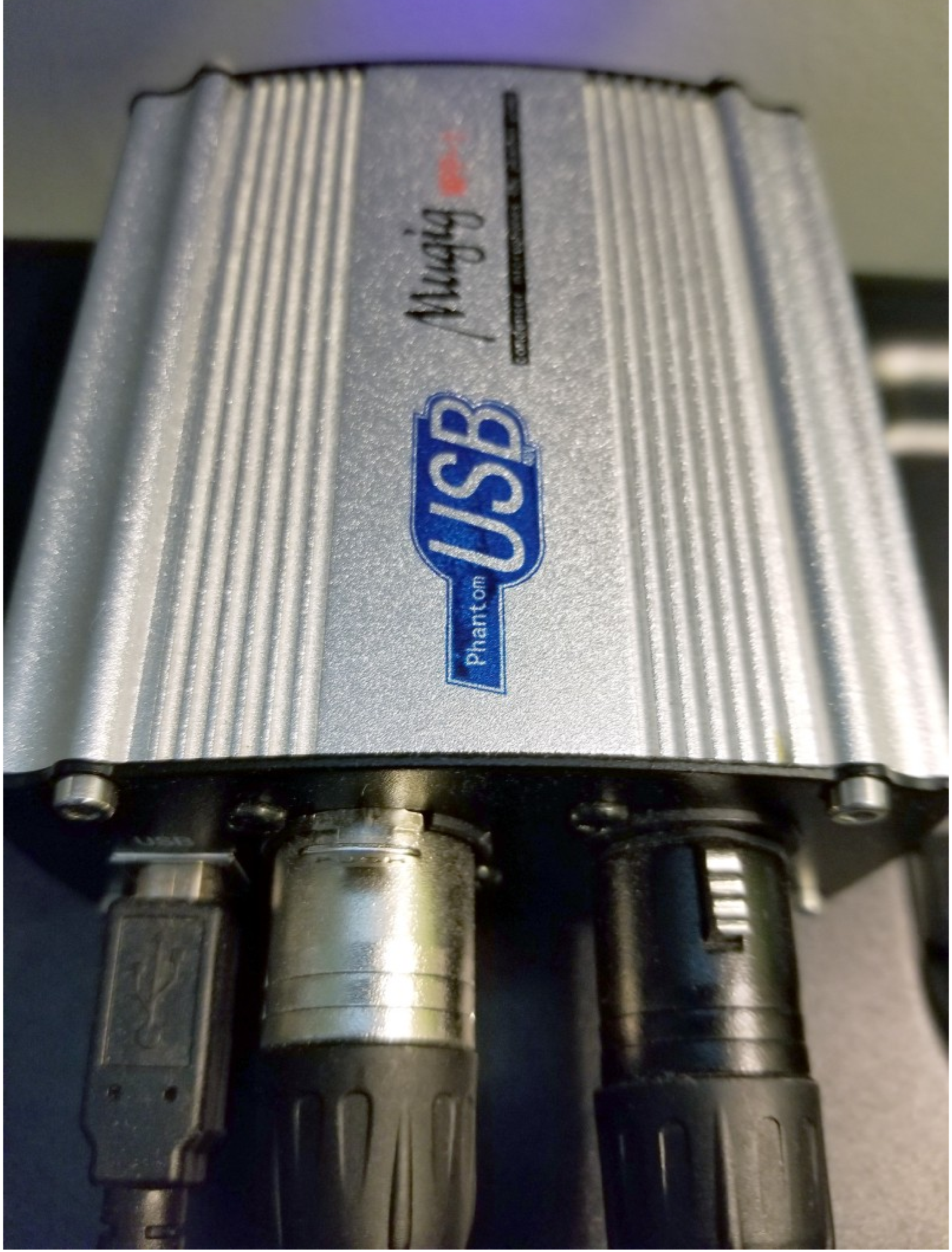
- Microphone and **audio cables** in general usually carry **only unidirectional** signals.
- **XLR** connector common practices (Generally signal flows from male XLR to female XLR)
  - female XLR to microphone, (to get sound from mic's male XLR)
  - male XLR to equipment (to provide sound/signal to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

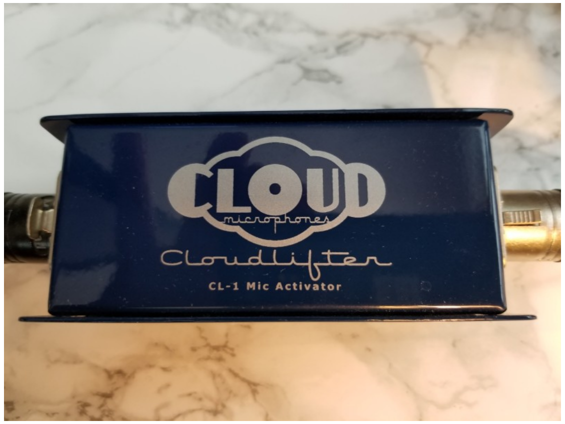
Mugig Phantom Power Supply

- 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**

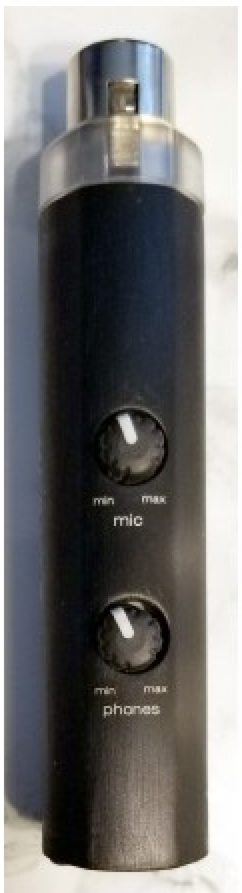


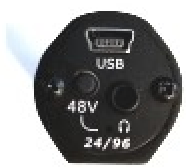
The **black color connector** on the left is a male XLR coming **directly from the mic**

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 <u>**next binary** related point **is 24**</u> (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 average)
- **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 drivers sending the sound to and from the hardware

Software Components

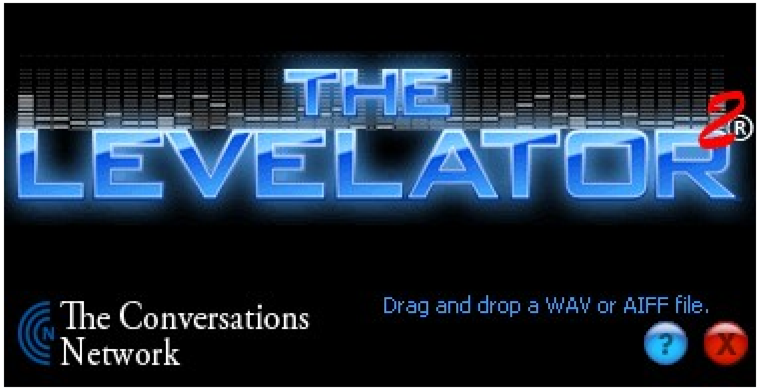
- **PulseAudio Volume Control**—A GUI tool for volume control on Linux
  - Launch with **`pavucontrol`** in a CLI shell
  - **Terminology**:
    - Source—Sound comes out of sources; a microphone is an obvious source.
    - Sink—an input that receives sound from something else—a sound card microphone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

Open Source Audio Apps

- **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

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, …

Audio Specs that Matter

- Bit Depth
- Sample Rate
- Normalization Level (RMS/LUFS)
- Max Peak
- Max Noise Floor
- MP3 Bitrate (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