HOW TO REFURBISH COMMODORE AMIGA TRACKER AND CHIP TUNES

# Motivation:
Why refurbish / remix and master at all ?

- Original tunes often are poorly mixed or not mixed at all
  - because of missing original author's skills/awareness: E.g. no proper stereo balancing, frequency balancing and leveling. E.g. bass and drums tracks were panned hard to left/right, frequency ranges of instruments overlap, etc..
  - because of technical limitations: most ancient trackers didn't provide EQ's, filters, stereo balancing, compressors, delay, reverb, etc.. E.g. a lot of samples contained DC offset which negatively affects bass frequency range
- Transfer the vibe of the original tunes to different audio gear. E.g. first Amiga Monitors had inbuilt stereo speakers which narrowed the tune's sound, even if left/right balancing was suboptimal. If the same tune's are played back via stereo hifi systems they often sound as ripped apart.
- On the other hand: simply down mixing stereo tunes to mono - many replayers provide such a feature – doesn't make things necessarily better, since an automatic down mixer never can replace proper manual mixing.

Overall Refurbishing Process
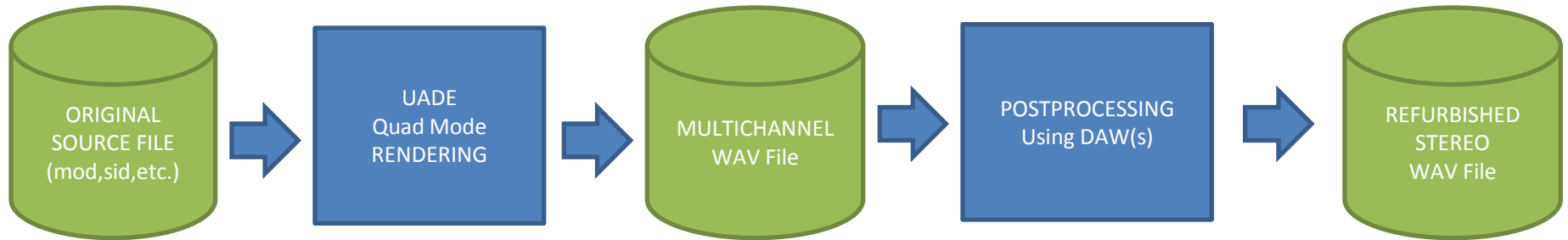
UADE Quad Mode Rendering (WAV file generation)
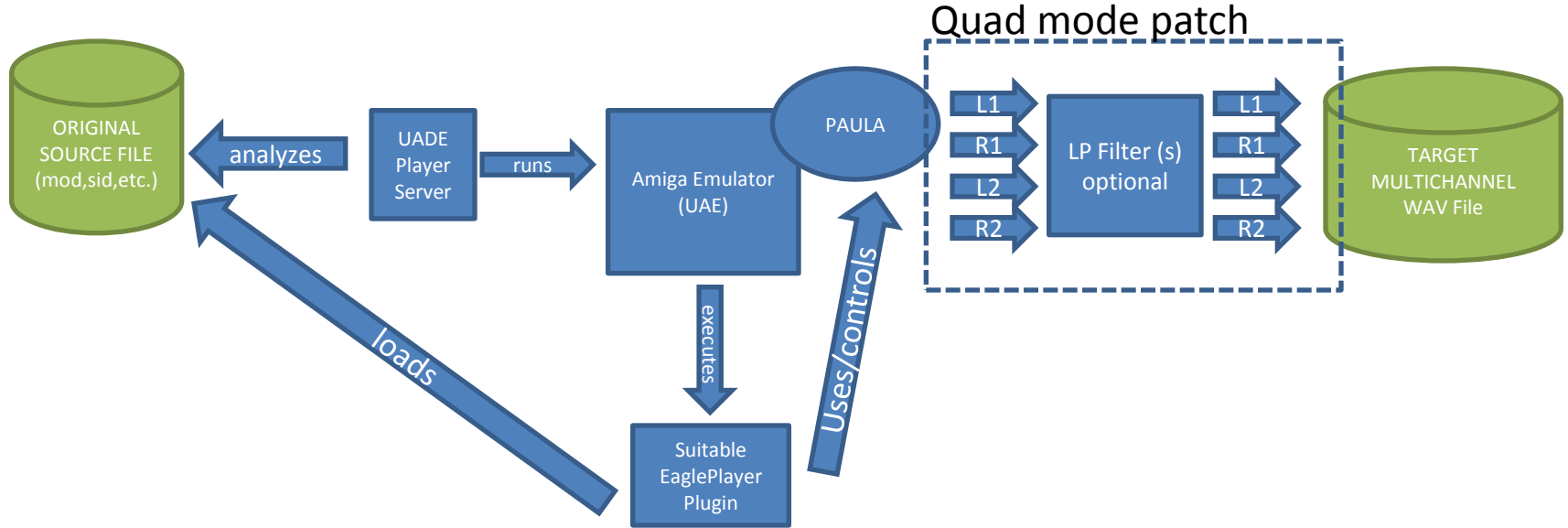
UADE Quad Mode Rendering Installation and execution

- UADE v2.13 player is a Linux freeware which is capable of replaying lots of ancient and modern tracker and chip tune formats. See http://zakalwe.fi/uade/

- UADE v2.13q is a modified version which is capable of exporting the player's output to a multichannel wav file (4 channels exactly). See http://blog.airmann.de/uade-multichannel-audio-support/

- Install UADE v2.13q as explained in the link above

- Execution (Linux bash, export format 44.1 Khz, 16 bit):
uade123 --quadmode --filter=none -f output.wav AHX.Cruisin

UADE Quadmode Rendering suitable export format

- It is recommended to export with disabled Amiga original Lowpass filter(s), because if really necessary suitable lowpass filers(s) can be applied at later mixing stage

- Default sample rate is 44.1 KHz. This is not Amiga standard (about 28,8 KHz). But that makes actually no big difference. Of course it's possible to export with 28,8 KHz and do a high quality SRC to a more suitable format (e.g. 44.1, 88.2, 96 Khz) afterwards (e.g. using SoX etc.).

- Default export dynamic range is 16 bit. This is more than enough, because Amiga PAULA chip sound channels are all 8 bit (0..255). So no 24 bit or 32 bit fp is necessary.
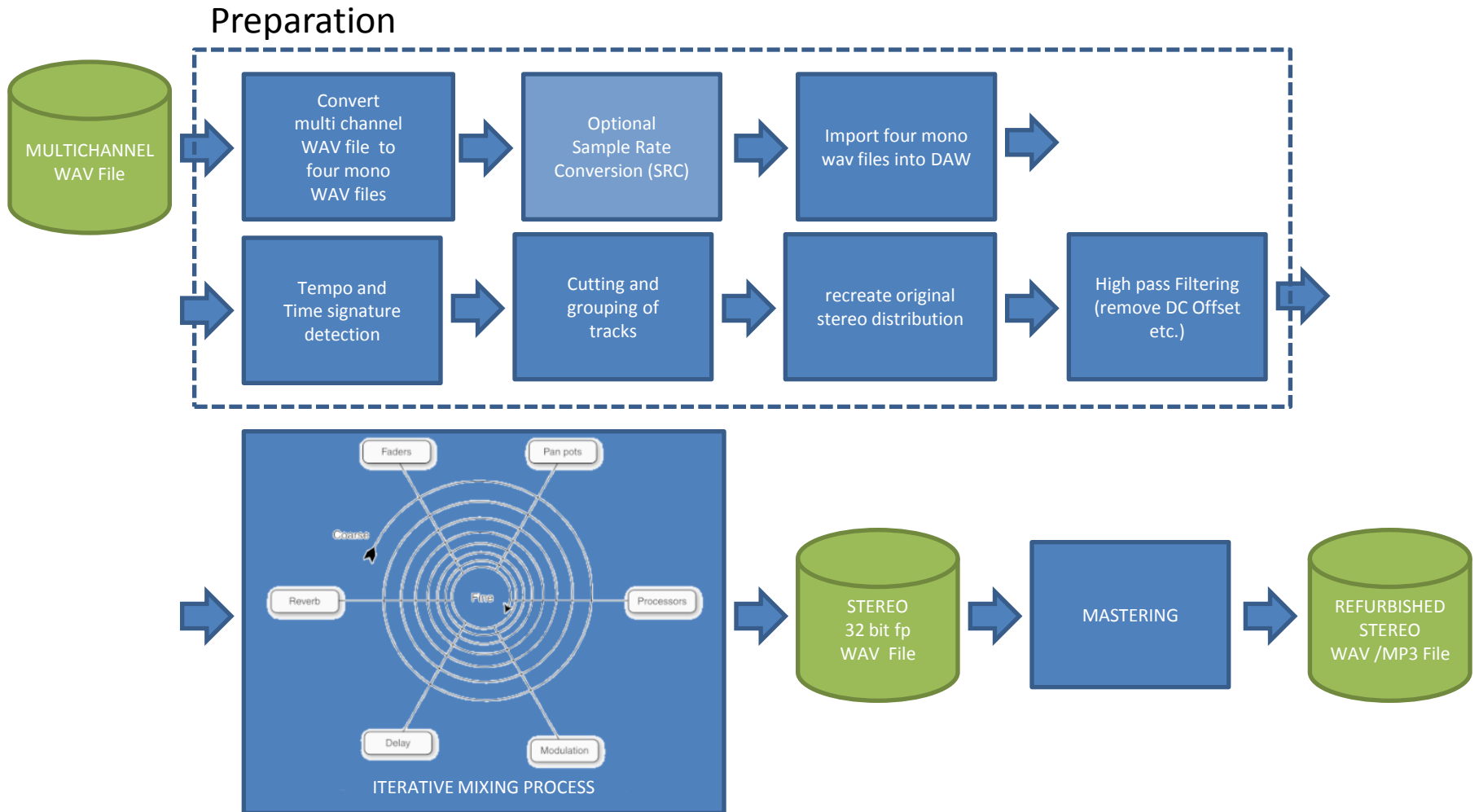
UADE Quad Mode Rendering Internal Data/Signal flow

Preparation and Editing

Post Processing Overview

Preparation

MULTICHANNEL WAV File

| Convert multi channel WAV file to four mono WAV files | Optional Sample Rate Conversion (SRC) | Import four mono wav files into DAW |
| Tempo and Time signature detection | Cutting and grouping of tracks | recreate original stereo distribution | High pass Filtering (remove DC Offset etc.) |

Faders | Pan pots

Coarse

Reverb | Fine | Processors

Delay | Modulation

ITERATIVE MIXING PROCESS

STEREO 32 bit fp WAV File

MASTERING

REFURBISHED STEREO WAV /MP3 File

Post Processing: multichannel import

- Convert multichannel 16 bit wav file to four single Mono 16 bit wav files. Attention: keep track of original left / right channel distribution. Rename the four wav files to L1,L2,R1,R2. HINT: 1=L1, 2=R1,3=L2,4=R2. This reflects the original Amiga stereo output. One also can use the bash script „split_quad.sh", which does the job for you.

- Import the four mono wav files as new separate stereo (!) tracks into a new DAW project and label them according to L/R distribution. Means: original multichannel #1 must be labeled as XXX_L1 and so on:
  - Channel 1 = L1
  - Channel 2 = R1
  - Channel 3 = L2
  - Channel 4 = R2

Post Processing: multichannel import

Post Processing:
tempo and time signature detection

- Why ? Exact tempo and time signature values are important for later track editing – one needs an exact grid for cutting.

- Find out tempo (BPM) and time signature of original tune

  - This can be tricky because some tunes have exotic tempo values like 130,034 BPM. Also in some cases tempo may change at various song positions.

  - For quick analyzation use e.g. BPM analyzer / DJ mixing tool whatever

  - For detailed bpm values use internal functions of your DAW. E.g. Reaper DAW provides a function „Set project tempo from time selection (detect tempo)". E.g. make a sample-precise time selection between two neighbored identical drum patterns. Hint: DAW grid should be enabled and set to 1/16 resolution and the time signature must also be set correctly (normally 4/4)

  - Make sure that tempo value is correct: e.g. check if kick positions are synchronous to grid lines. If tempo value is not absolutely correct, kick positions may be correct at song start but drift away at later song positions.
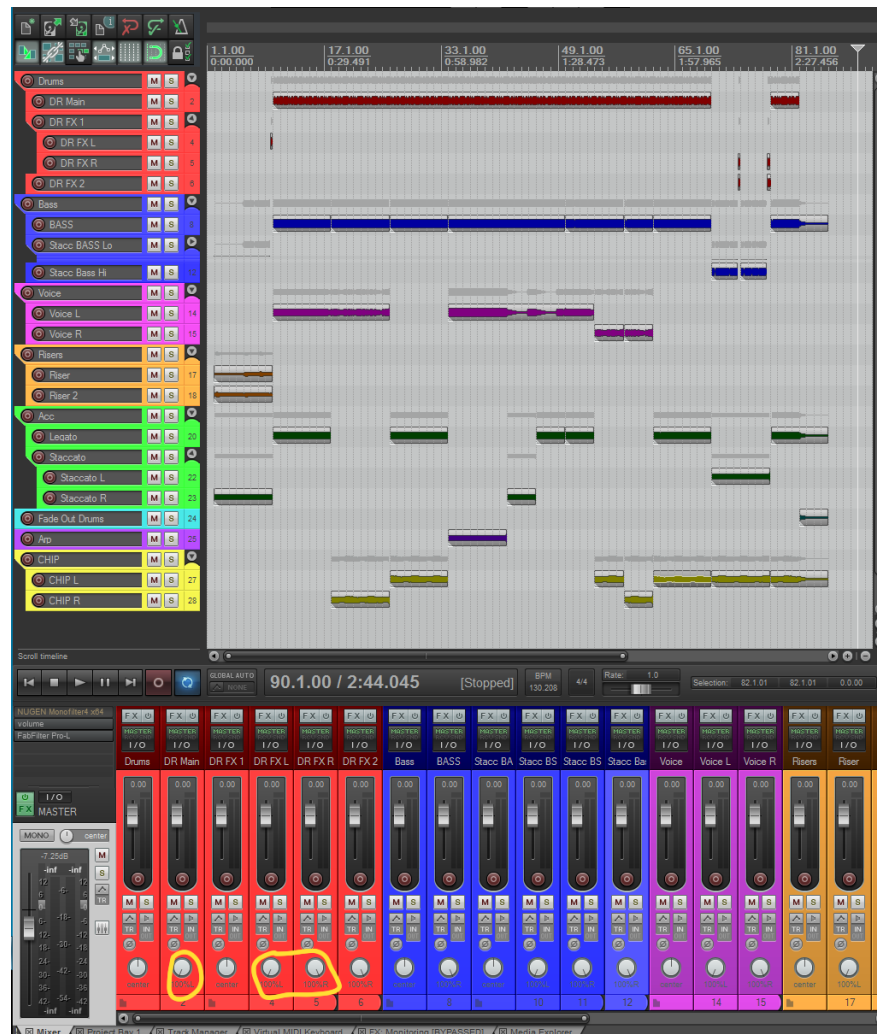
Post Processing:
cutting and grouping

- Why ? Many old tunes mix various instruments on a single track which makes later (re)mixing difficult. Thus, cutting and reorganizing might be necessary.

- Listen to the tracks and cut them according to their content.

- Move the cut pieces to new groups / subgroups e.g.:
  - Bass
  - Drums
    - Snare
    - etc.
  - Voice/Lead
    - Leadsynth A
    - Leadsynth B
    - etc.
  - Pads

Post Processing: recreate original stereo distribution

- Why ? In the first step we want to recreate the original left/right distribution of the tune's parts. During later (re)mixing process we adjust the panning values if necessary

- Since all imported track items are mono, we have to adjust the panning values of the containing stereo tracks

- Check the names of the source WAV file of all items on all tracks. If the name is e.g. XXXX_L1.wav, turn the balance of the containing track to 100% left. If the name is e.g. XXXX_R2.wav, turn the balance to 100% right.

- If there are two items with different L/R settings on the same track, create a new subgroup and split this track into two new tracks: „orig_track"_L and „orig_track"_R.

- For example see screenshot on next page. Have also a look on the paning knobs (yellow marked in the lower part of the screenshot)

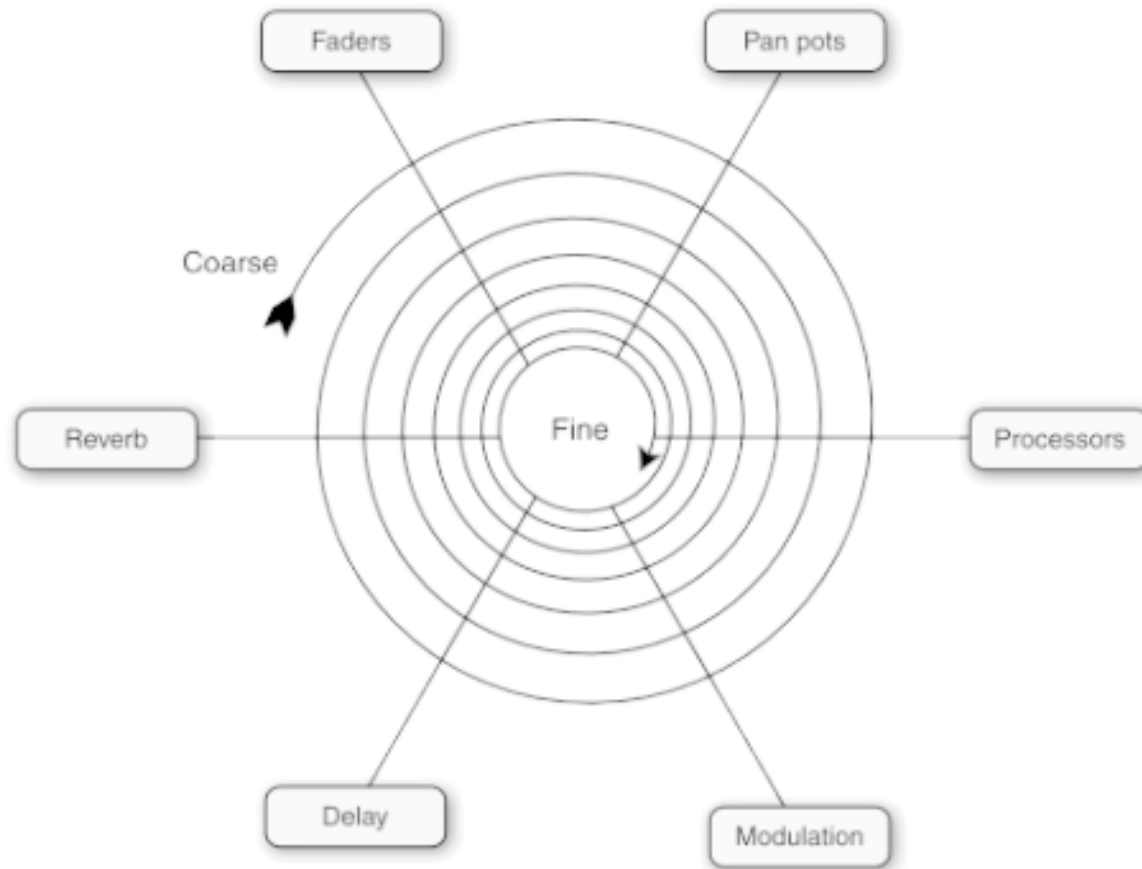# Post Processing:
recreate original stereo distribution

Mixing Process

Post Processing: Mixing Process

- Adjusting track levels and stereo balance. E.g. centering bass and kick drum.
- Adjusting frequency balance (EQing, filtering).
- Adding dsp effects (compression, reverb etc.)
- This process takes most time and is highly subjective. Different persons, different results.
- If the goal is to maintain the original tune's vibe, too much fx should be avoided. E.g. just apply EQ / filtering and compression.
- Proper leveling and stereo balancing is most important.
- Iterative A/B comparisons with original track and optional reference tracks are recommended

Post Processing:
Iterative Mixing Process

Picture © Roey Izhaki „Mixing Audio", Focal Press 2008, page 35

Mastering Process

Post Processing Mastering Process

- 32 bit fp stereo WAV output from mixing process is sent to a mastering engineer
- Example for mastering process (may vary):
  - Import WAV stereo file into mastering DAW
  - DC Offset removal
  - EQing / Compression / Stereo Width adjustments / Saturation / Limiting etc.
  - Loudness adjustment according to various standards (e.g. iTunes, EBU R128, K-System etc.)
  - Iterative A/B comparison with original mix and reference tunes

Post Processing
Mastering Process

- Adding optional fade ins / fade outs
- Final Format conversion
  - Creating of WAV/FLAC Master (44,1 KHz, 16/24 bit stereo): 16/24 bit dithering, SRC (sample rate conversion) if necessary
  - Creation of MP3 / OGG (e.g. 44,1 KHz, 16 bit stereo, 320 Kbit/s)
  - Creation of CD Master (44,1 KHz, 16 bit, stereo): 16 bit dithering, SRC if necessary, creation of red book master

HAPPY REFURBISHING !