The “Bengt workflow” goes like this…

I try to source the best condition vinyl or compact cassette I can find. Never pay much attention to the condition of the cover, the vinyl is more important.

Pretty much the same workflow (digitizing and processing) for vinyl, cassette and reel-to-reel.

DAT is captured using toslink (optical) of the bitstream directly to whatever sample rate is on the tape 44.1 or 48 kHz 16-bit.

I clean the vinyl first from obvious debris, using isopropanol alcohol if needed, then using a Degritter Mark II ultrasonic vinyl cleaning machine with a few drops of ILFOTOL (to ease the water surface tension) and run the Degritter on the “Heavy” cycle. I had a Degritter Mark I machine for many years but the Mark II is definitely an improvement.

For safety’s sake I run a “Furutech Destat III” over the cleaned vinyl and the turntable to remove any possible static electricity buildup.

Then transfer the vinyl using:

- A “Pro-Jext Extension 10 Evolution” turntable with balanced XLR output : https://www.project-audio.com/en/product/xtension-10-evolution/
- With an “Ortofon Quintet Black S” moving coil pickup : https://ortofon.com/products/mc-quintet-black-s
- Connected to a “Pro-Ject Phono Box RS2” RIAA interface, balanced XLR inputs : https://www.project-audio.com/en/product/phono-box-rs2/
- Connected to a “Cosmos ADC” 2 channel Analog to Digital converter, again with balanced XLR inputs. : http://archimago.blogspot.com/2021/09/early-look-e1da-cosmos-adc-affordable.html
- For multichannel reel-to reel tapes I use a MOTU Ultralite Mk5 Analog to digital converter with RCA single end inputs : https://motu.com/en-us/products/gen5/ultralite-mk5/
- Which feeds “Adobe Audition” software via USB for the digitizing which is done at 192 kHz, 32-bit, saved in WAV format : https://www.adobe.com/se/products/audition.html

So now, we have a 192 kHz 32-bit “raw” audio WAV file.

One song in this format is (depending on length) around 500 MB in size. I have tons of diskspace (large NAS) so size isn’t an issue.

This file, if needed, is fed through a program called ClickFix (which isn’t available anymore). A Java application that does the best quality job of removing vinyl clicks and pops automatically.

I then take the “clickfixed” file and trim beginning and end and if needed normalize (not hard limiting) levels to 0 dB, that is, if there already are levels at 0 dB, the material is not affected at all.

If there are other defects that need to be removed, I “cherry-pick” them using “Izotope RX10 Advanced” software : https://www.izotope.com/en/products/rx/features.html. For example if a slow fade in at the beginning of a song needs some background noise removal it is removed in that software.

BTW “Izotope RX10” is prob the industry leader in software for restoring audio, having a plethora of restoring modules for audio in all its forms.

So, by now, the above processed results in a finished “cleaned” file..

Then sometimes I experiment with throwing some “mastering” filters at it, which I normally don’t do. I was just a recent idea since I’ve done that on some digitized 40 year old reel-to-reel tapes. The software used is a suite of modules “Izotope Ozone 11” : https://www.izotope.com/en/products/ozone.html , which is standalone but also can be used as plug-ins (using VST3) in most any DAW (Digital Audio workstation). In my case I use it in Adobe Audition.

I normally use 6 of the modules for the “Mastering” :

- Equalizer - https://www.izotope.com/en/products/ozone/features/equalizer.html
- Stabilizer - https://www.izotope.com/en/products/ozone/features/stabilizer.html
- Impact - https://www.izotope.com/en/products/ozone/features/impact.html
- Imager - https://www.izotope.com/en/products/ozone/features/imager.html
- Dynamic EQ - https://www.izotope.com/en/products/ozone/features/dynamic-eq.html
- Maximizer - https://www.izotope.com/en/products/ozone/features/maximizer.html

I set the parameters to my liking of each of the modules, starting from the default settings or running a section of music through a “wizard” that analyzes it and suggests what it thinks are optimal settings. I take care not to set the “Maximizer” too high, usually only boosting audio between -2 dB and -0.1 dB otherwise the result will be abnormally loud, losing a lot of the dynamics. I never liked the “loudness wars” going on in modern mastering anyway…

Finally (phew!) I downsample from 192 kHz 32-bit WAV to 192 kHz 24 bit WAV (as FLAC can’t handle 32-bit officially as of yet) and finally compress (lossless) to FLAC level 8 for archiving and everyday listening.

The MP3s that I may generate to save space when sending somewhere are downconverted (lossy) to 320 kbps for the sake of being able to e-mail them, not something I normally do.

All downsampling/converting is done using dBpoweramp https://dbpoweramp.com/dmc.htm

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