Kernelversion:	2.6.35-32-generic-pae
Alsaversion:	1.0.23 - no pulseaudio is beeing used
MPDVersion:	0.17 alpha
Audiochip:	Realtek ALC 888
Mainboard:	Jetway JNC96FL-525-LF
CPU:	Intel Atom D525+NM10
Loaded driver files:	snd_hda_codec_realtek; snd_hda_intel; snd_had_codec; snd_hwdep; snd_pcm; snd_timer

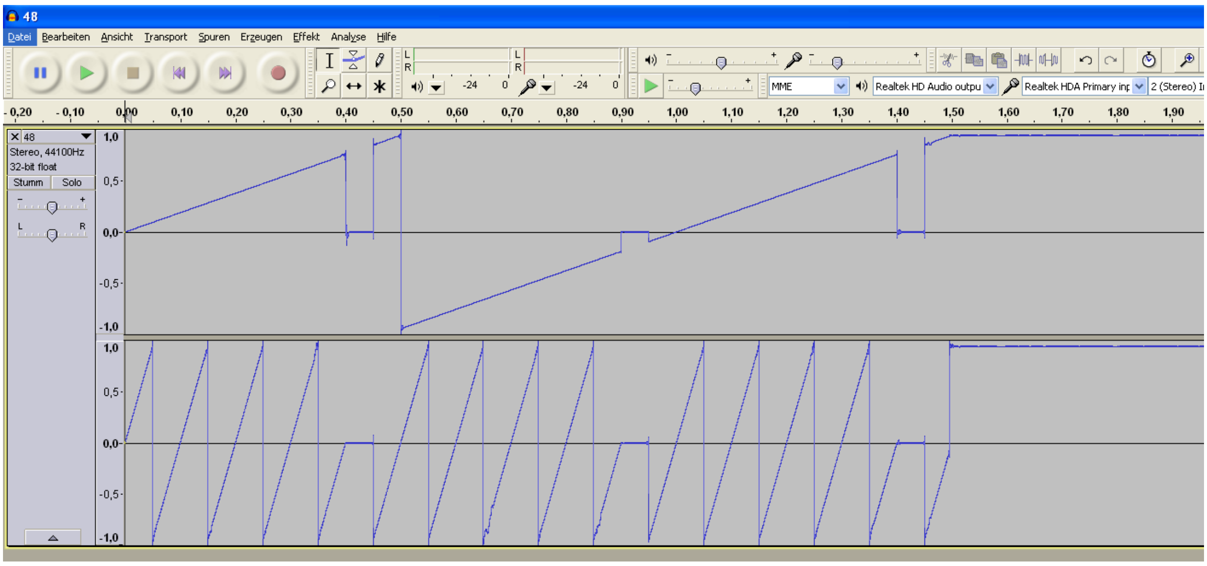
Problem description:

We have some strange behavior in our audio application using the hardware and software components named above. Basically the main problem seems to be a buffer or memory management issue. Once a playback is stopped the very last sample is repeated forever, independently from sampling rate or sound file format. This sample leads to a constant DC voltage after the DA converters.

Problem Constant DC Sample:

As mentioned above, the last sample that has been played (prior to pause or stop current playback) will be repeated forever. Hence we get a constant digital “DC” from the S/Pdif output.
If we mute the SPDIF Output using the ALSAMIXER, there is no S/Pdif signal, hence no DC. By demuting again, the same sample appears again.
The annoying thing is, that this sample always stays for a while if we play another sound file. If the sample has been at a high value, this leads to a loud noise in the speaker. The following pictures illustrate this problem.

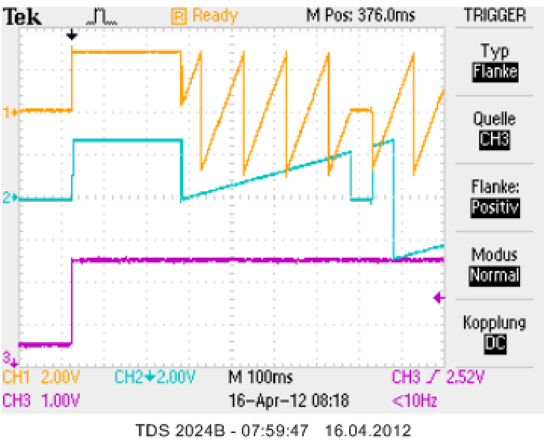
First picture shows the original soundfile display in audacity – this picture is for reference only. It shows, how the signal should be read on the oscilloscope.

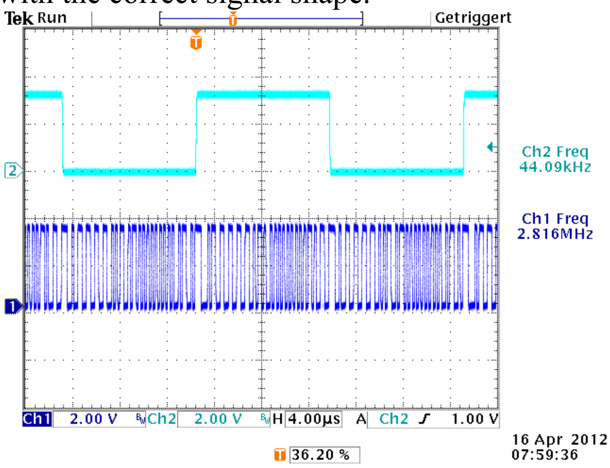


The DC offset at the end of the file is to demonstrate that the last sample stays forever, even if the file is stopped.

Fileformat 44.1kHz FLAC 16Bit

The file shown above is played as 44,1kHz FLAC 16Bit; after hitting stop in the player application, we get a constant sample at –6dBFS, until we play again. This sample is then repeated for about 300ms before the file starts with the correct signal shape.

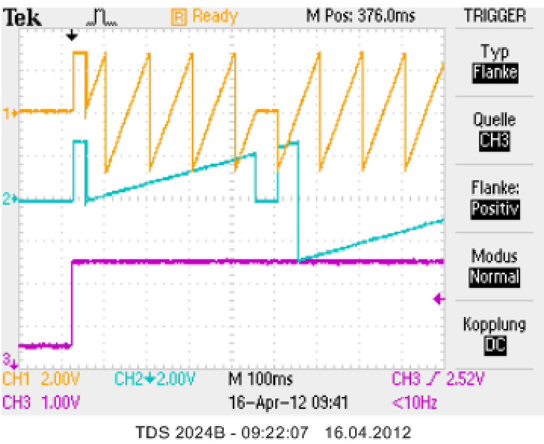


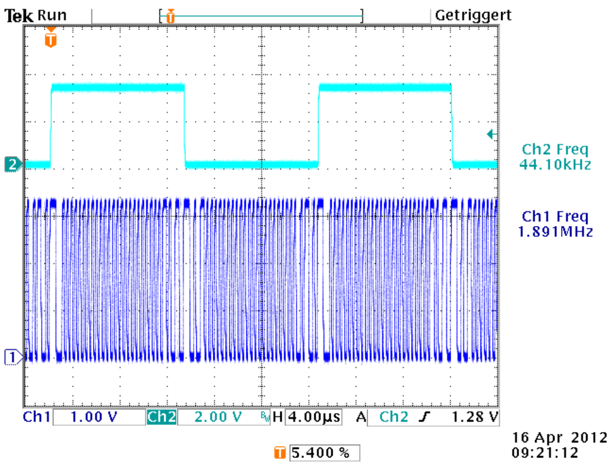


The magenta curve shows when the play button is pressed. The constant value for about 300ms at the beginning of the signal shape is the faulty condition.

The right picture shows the constant sample in the S/P-DIF stream, although the player is in stop mode, hence no sample should be read except digital zero.

Fileformat 44.1kHz FLAC 24Bit

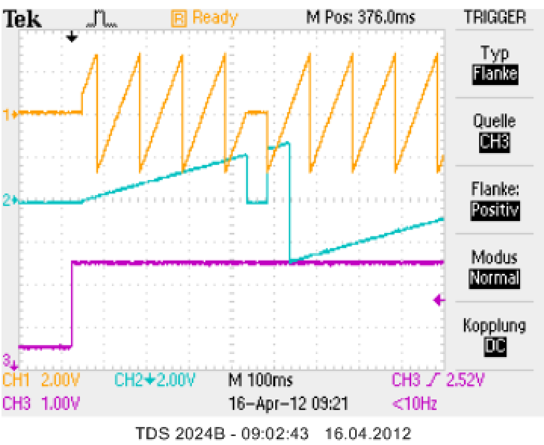


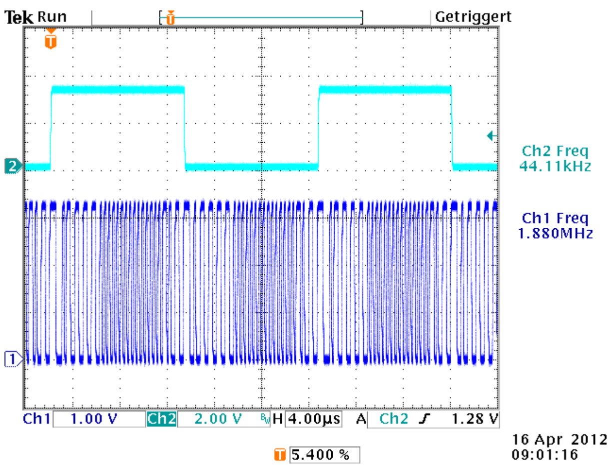


The magenta curve shows when the play button is pressed. There is still a constant sample but the time period is shorter then at 16Bit.

When in stop mode, there is still a constant sample in the S/P-DIF stream of –6dBFS.

Fileformat 44.1kHz WAV 16Bit



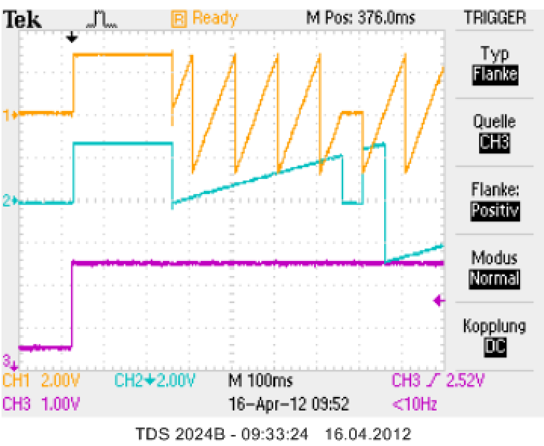


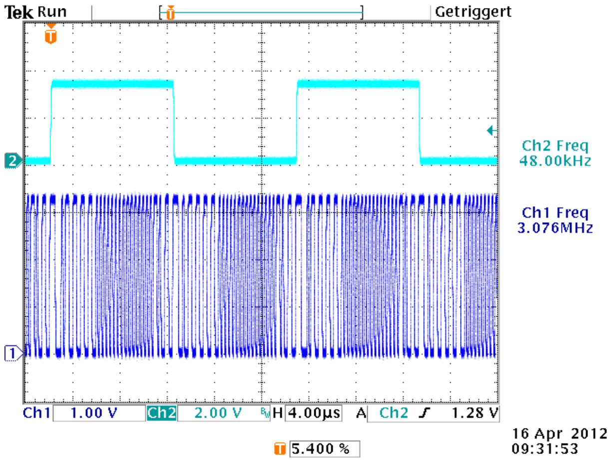
The magenta curve shows when the play button is pressed. There is no constant sample but the first few samples are missing.
When in stop mode, there is still a constant sample in the S/P-DIF stream of –6dBFS. The curios thing is, that this samples is not repeated when we play again the file from the beginning.

Other Files

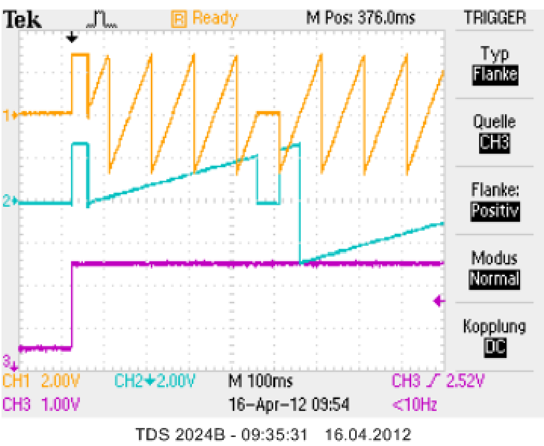
The next figures show the same scenario at different sample rates as well as different file formats. See the corresponding headline for the details.

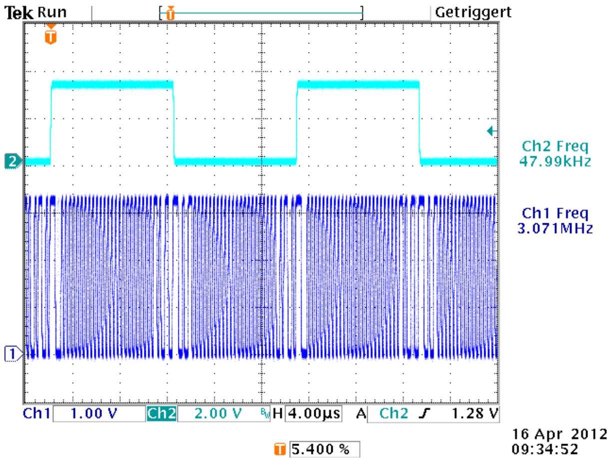
Fileformat 48kHz FLAC 16Bit



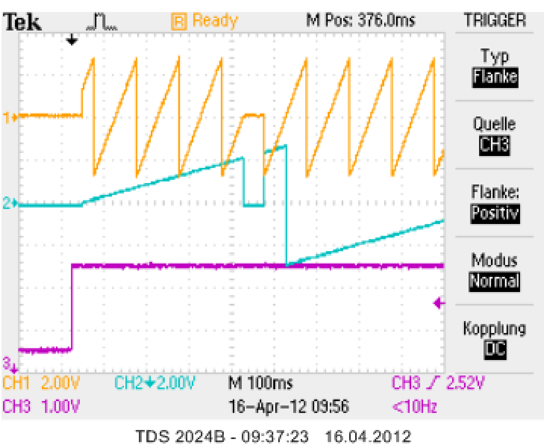


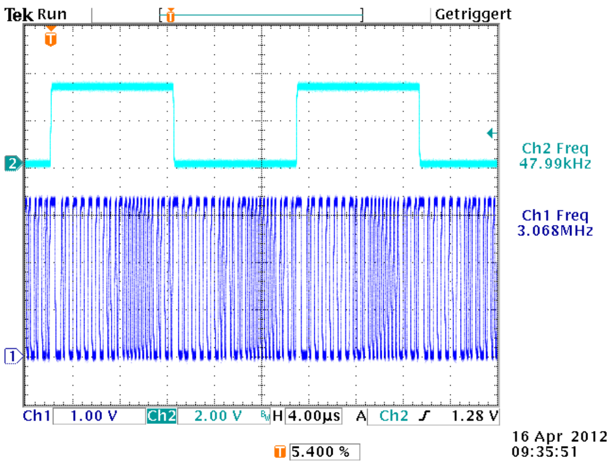
Fileformat 48kHz FLAC 24Bit



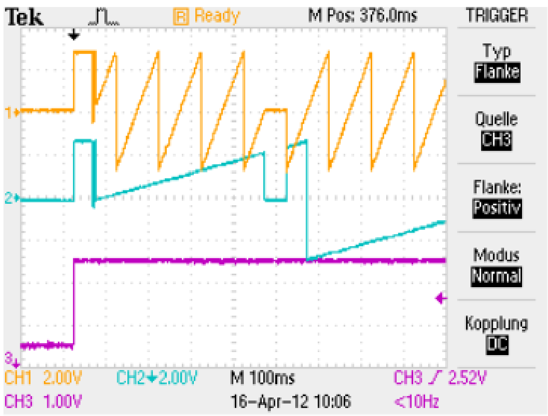


Fileformat 48kHz WAV 16Bit

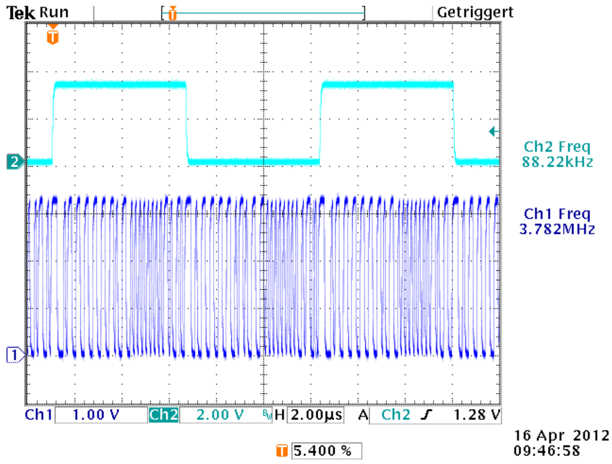




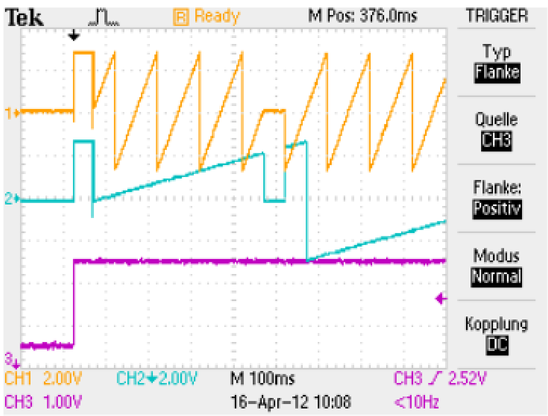
Fileformat 88.2kHz FLAC 16Bit



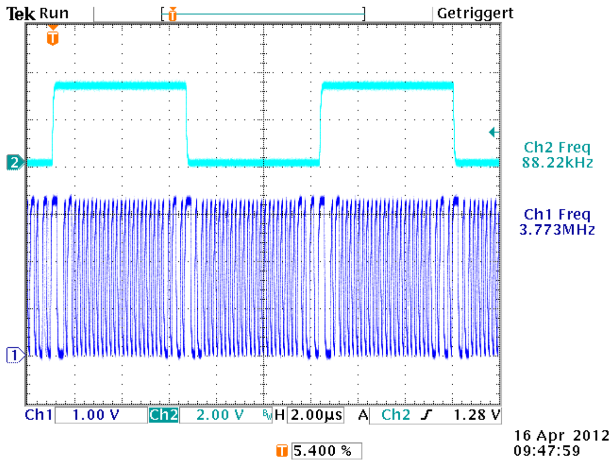
TDS 2024B - 09:47:39 16.04.2012



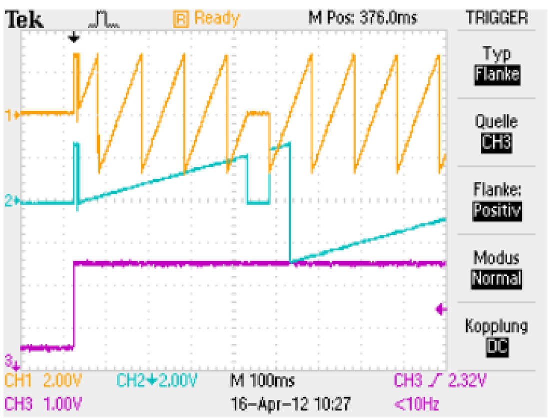
Fileformat 88.2kHz FLAC 24Bit



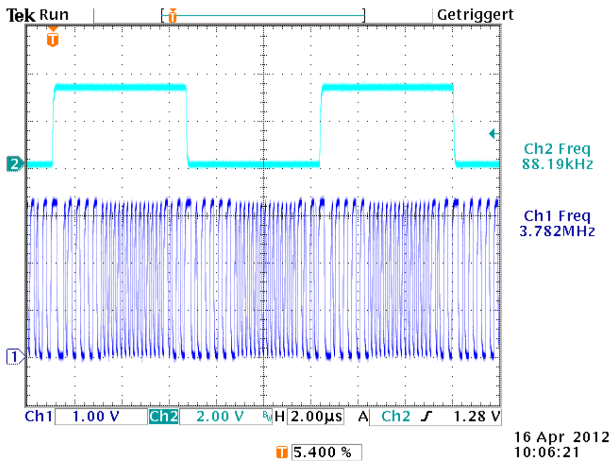
TDS 2024B - 09:49:23 16.04.2012



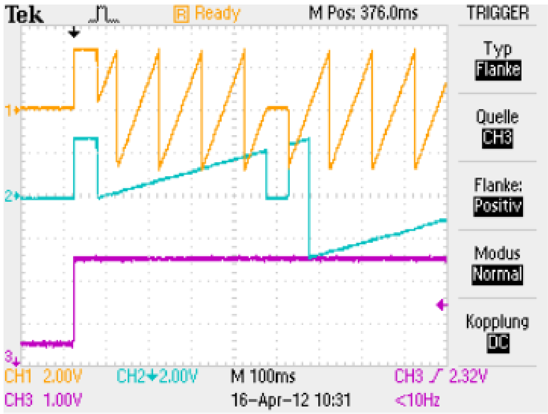
Fileformat 88.2kHz WAV 16Bit



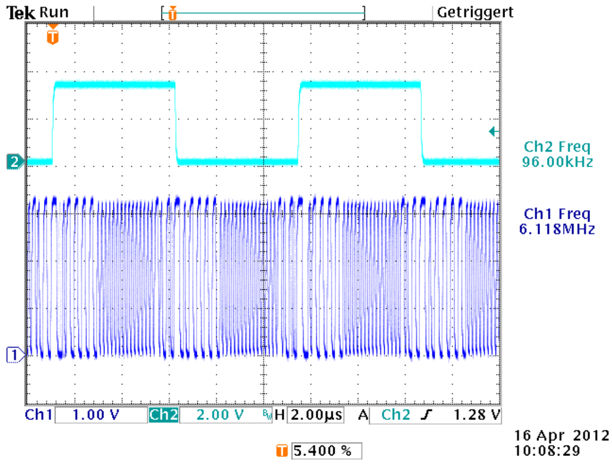
TDS 2024B - 10:08:11 16.04.2012



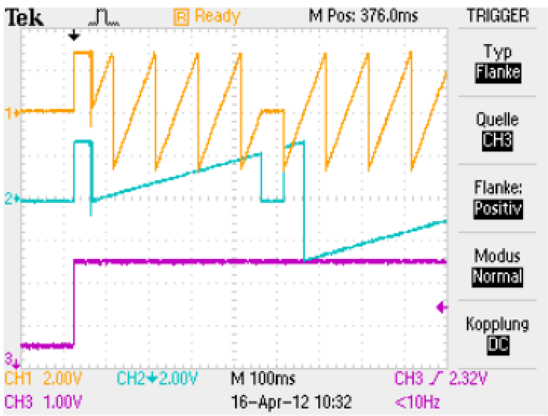
Fileformat 96kHz FLAC 16Bit



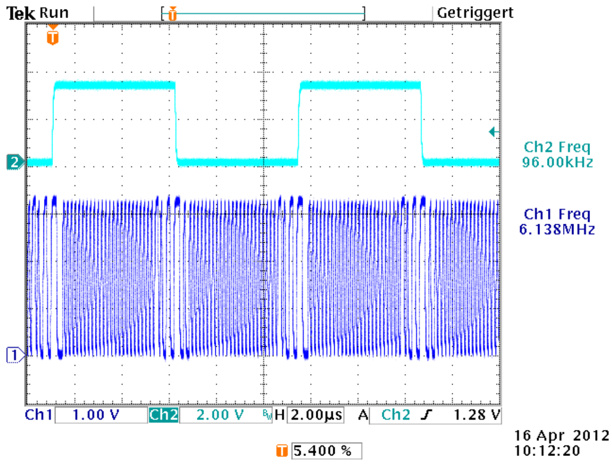
TDS 2024B - 10:12:24 16.04.2012



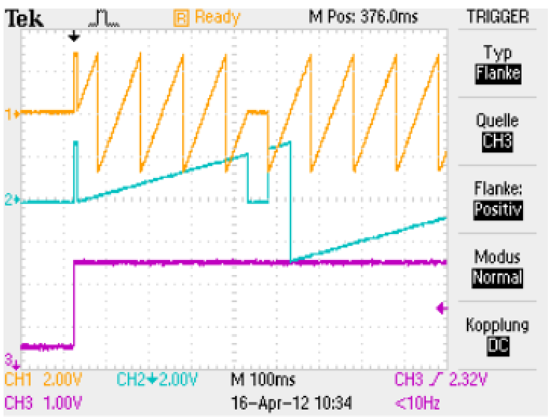
Fileformat 96kHz FLAC 24Bit



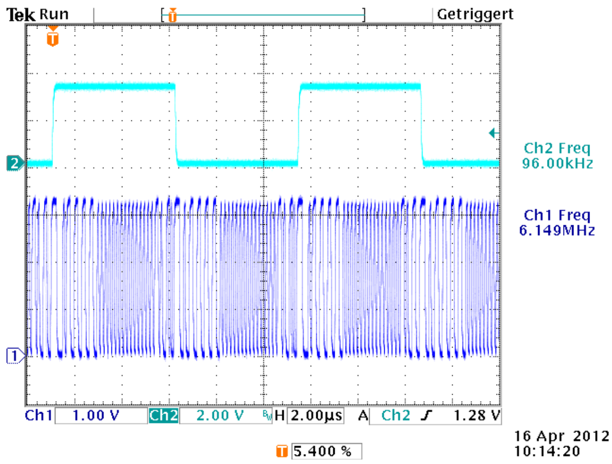
TDS 2024B - 10:13:11 16.04.2012



Fileformat 96kHz WAV 16Bit

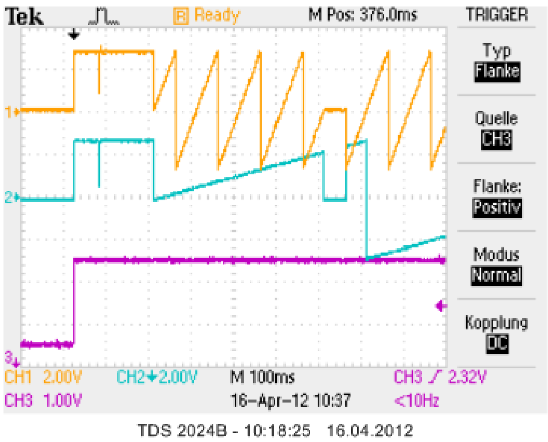


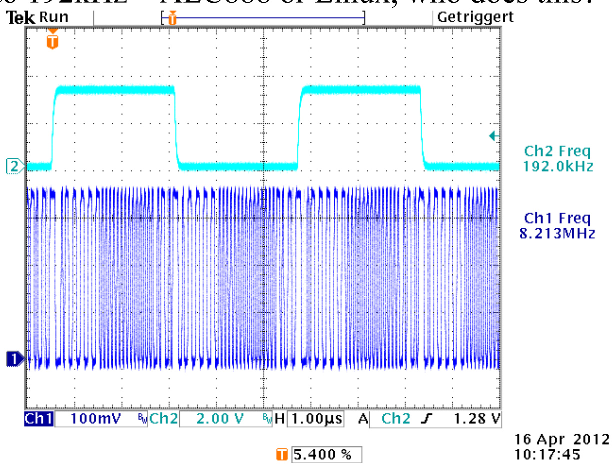
TDS 2024B - 10:15:29 16.04.2012



Fileformat 176.4kHz FLAC 16Bit

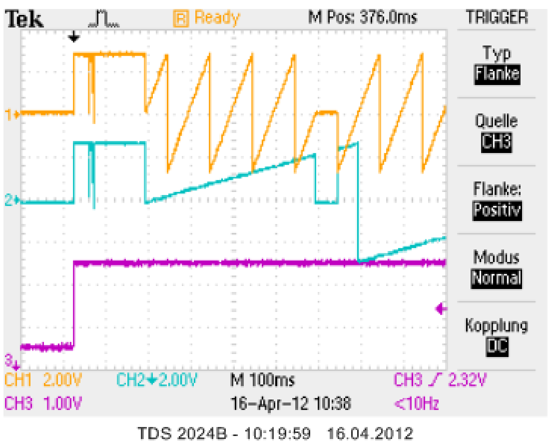
This sampling rate is automatically converted to 192kHz – ALC888 or Linux, who does this?

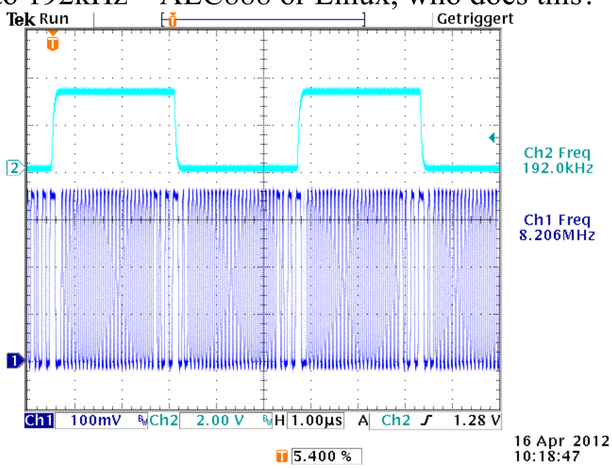




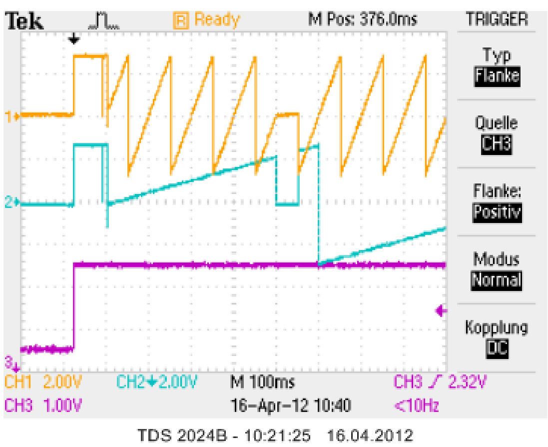
Fileformat 176.4kHz FLAC 24Bit

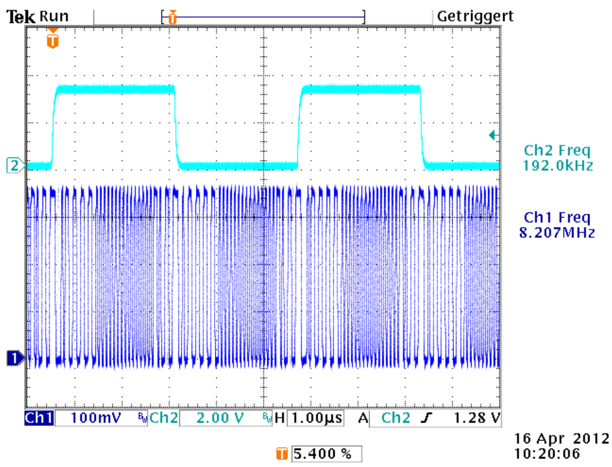
This sampling rate is automatically converted to 192kHz – ALC888 or Linux, who does this?



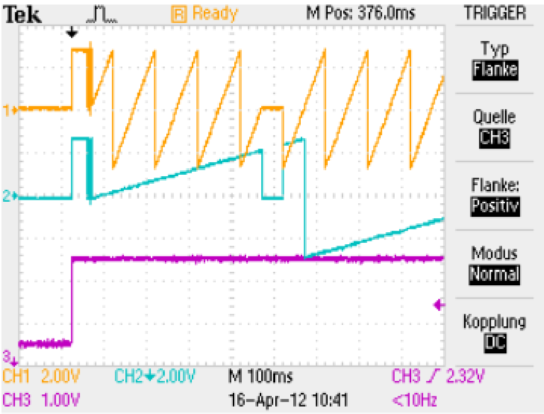


Fileformat 192kHz FLAC 16Bit

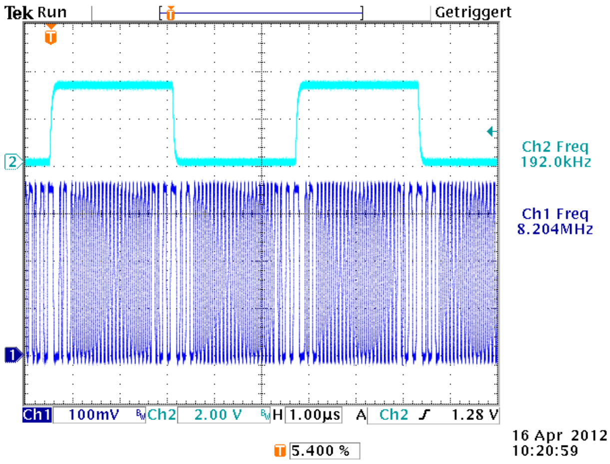




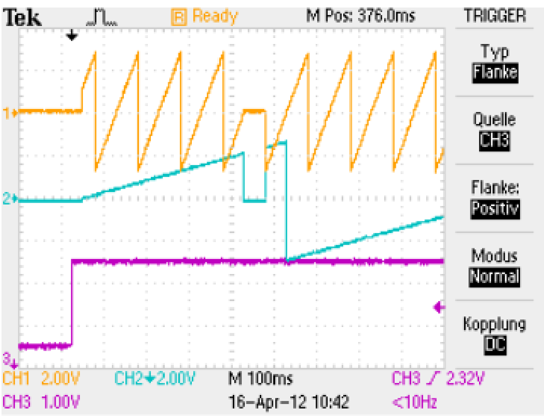
Fileformat 192kHz FLAC 24Bit



TDS 2024B - 10:22:10 16.04.2012



Fileformat 192kHz WAV 16Bit



TDS 2024B - 10:23:16 16.04.2012