Subject: Re: Digital Music Formats
Posted by Adveser on Mon, 02 Aug 2010 21:41:37 GMT
View Forum Message <> Reply to Message

This is what I know:

for some reason, people seem to hate WMA and AAC, the proprietary formats of MS and Apple. So I would avoid those.

There is Ogg Vorbis, which is better at lows and highs than MP3.

But MP3 is far better at mids, which is where the vast majority of the information is. Your hearing is logarithmic, so it seems like there are more low and mid frequencies than there are compared to high, if that makes sense.

This is what I use in LAME with Razorlame's front end: -b 320 -m s -h --resample 48 -q 0 -k

I'll explain those switches:

I use 320kbps constant, even though there is no additional loss using Variable bitrates.

I recently saw the voice prints of some "busy" sections of songs encoded at different rates. 320 was virtually identical to the wav file and things got really poor below 192kbps. But that is nothing new. Never ever go below 128kbps.

I only use Stereo. I don't trust any of the schemes to save more space by encoding the channels differently. I have heard claims that J-Stereo is superior because there is more room to do less compression.

the -h means high quality. It's an encoder setting telling it to "take your time and do it right because this guy doesn't care how slow the program runs"

I resample to 48Khz. Why not? It is only adding bandwidth, which comes in handy when using DSPs anything else that modifies sound. That is my opinion and it is not generally shared by the masses. Some are in the camp that what sounds best is the best and some are in the camp that modifying anything is a reduction in quality. I seem to, or used to when soundcards sucked, get a better sound from the higher sample rate. At the very least this was preventing the K-Mixer from doing it for me. These days I don't really worry about it since I don't get my files from the CD anymore and don't have to encode my own files often enough. I still resample when I encode, if only because it does no harm that I can hear. Sorry for the length of the section. Go with 44.1 if you want accuracy, go with 48khz if you are modifying the sound digitally.

-q 0 is another setting for encoder that basically means don't ever take a shortcut with the math. It tells it to work slowly and do as many calculations as necessary to get the best result. This is the "real" quality setting. The lower the better.

- k tells the encoder not to use any filters such as passband filters. This switch may be obsolete. I don't know anymore. If someone does please tell me the new command line switch.

That's my take on the settings I use.