Best practices for producing quality digital audio files

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Just as the quality of digital images depends on resolution, color depth, and storage format, the quality of digital audio depends on the sampling rate and bit depth settings, as well as the choice of compressed (lossy or lossless) or uncompressed storage formats.

Most computers' standard configurations support CD and DV quality audio and the more common compressed audio formats. Higher quality (higher sampling rate and/or greater bit depth) audio capture may require additional internal or external hardware for both creation and playback. The recommendations below do not cover hardware, as this area constantly evolves.

General recommendations

Sampling Rate and Bit Depth

The sampling rate determines how many times per second the sound wave is measured; bit depth is the sample size or range of possible numbers used to express the sample. Together, these determine the "resolution" of your audio file, and the size of the data file that contains your audio. Keep in mind that you can't add resolution to an existing digital file by increasing these parameters, and you should consider the quality, recording medium, and ultimate use of any analog source recordings when choosing the quality level for your digital files.

Archival Quality
- Sampling rate = 96 kHz
- Bit depth = 24-bit depth

Acceptable (DVD Quality)
- Sampling rate = 48 kHz
- Bit depth = 16-bit depth

Acceptable (Audio CD Quality)
- Sampling rate = 44.1 kHz
- Bit depth = 16-bit depth

Frequency ranges of different sounds vary, so the requirements for a digital version of your source recordings will also vary. Spoken word recordings are typically acceptable at the Audio CD standard of 44.1 kHz. Field recordings that include natural and/or subtle sounds such as birdsongs or musical recordings should be captured at either Archival or DVD Quality.

The main reason to use one of the higher standards is because you or others plan to do analysis and/or processing of your digital audio files in addition to just listening to them. Remember that you should look to the future: any information that is not captured now may be lost forever. Having the extra samples and/or bits means that any distortion created during processing ends up in the least significant bits, and so disappears when a recording gets transcoded to a more accessible format like audio CD or MP3, which is how most users will listen to them. Information useful in making these decisions is offered below.

File Formats

Ideally you will want to create and submit your audio files to Deep Blue in file formats that are non-proprietary, with a high potential for future readability. You will also want to choose lossless formats for maximum audio fidelity.

Recommended
- AIFF (.aiff, .aif)
- WAV (.wav)

Acceptable
- MP3 (.mp3; see below regarding compression)
- MOV (.mov)

AIFF and WAV are considered Level 1 formats in Deep Blue. They receive our highest level of preservation support, so we make our best effort to maintain all the content, structure and functionality of the work you deposit by taking appropriate preservation actions over time. We can do so because AIFF and WAV are publicly documented, widely used, and uncompressed so they retain all of the original data. MP3 is considered a Level 2/"Limited" format—one for which we will preserve the content of the deposit, but whose appearance and functionality may not be preservable over time. (MOV files are roughly equivalent to WAV files but are technically a proprietary Apple QuickTime format—even though a QuickTime file can contain a standard AIFF file. It is because of this QuickTime "wrapper" that they are only considered acceptable. It is unlikely that MOV files deposited in Deep Blue will become unexpectedly unreadable. If QuickTime formats will become unsupported in the future, there should be ample time to have them transcoded into a more enduring one.)

Compression

Some compression schemes are "lossy," meaning that once an audio file is processed into these forms the original sound at full quality cannot be regenerated from the compressed version. (Moreover, each time lossy compression is applied, more quality is lost.) Other compression schemes are "lossless," meaning that the unmodified original sound at full quality can be faithfully generated from the compressed form. Lossy compression schemes usually result in smaller compressed files since they eliminate details and information many playback mechanisms can't use.

Recommended
- FLAC (Free Lossless Audio Codec; .flac)

Acceptable
- ALAC (Apple Lossless Audio Codec, lossless but proprietary; .alac)
- MP3 (.mp3; lossy but publicly documented)

Not Recommended

AAC/MPEG-2 (Advanced Audio Coding, a proprietary and lossy; .aac)

Real Audio (.ra, .rm, .ram)

Windows Media Audio (.wma)

If you have materials in one of the compressed formats you can still deposit them in Deep Blue, however it is best to submit materials in formats that are not proprietary. So, even though this is still a lossy compression format, MP3 files are preferred over Real Audio or Windows Media files because MP3 is an open standard, supported by multiple vendors. Better, if you have files in proprietary compressed formats such as those just mentioned, convert the files to AIFF or WAV for long-term storage. While some data will already have been lost, converting to a lossless compression scheme will insure that no more information is lost.

Processes like simple 'cuts-only' editing will not benefit from higher resolutions, but if you have a marginal recording that will be digitally processed to remove background noise or tape hiss, equalized, or have its dynamic range significantly altered, a higher resolution may make a difference in your final result. Testing your processing chain with a small sample file will help you make good choices.

To create quality digital audio using direct-to-digital recording

Whatever format your source recordings are in, the first step is to get them into a digital form on a disc so you can process and transfer them to Deep Blue.

If your recordings are already in a disc-based digital form like audio CDs (AIFF files), MP3s or other formats, your job is easier. There are many audio tools available for purchase or as freeware that allow for batch processing of media files from one format to another, so if you need to transcode your audio files all you'll have to do is start the process; you can leave the heavy lifting to your computer.

If your audio recordings are in a digital media-based format like DAT tapes, DV video tapes, MiniDiscs or other formats, then you will have to play your recordings back in real time to capture them to disc. Fortunately, most digital media formats allow for a digital transfer to your computer, eliminating the process of converting from digital to analog and back to digital. It's just like copying a Word document from one computer to another: the copy is literally a clone of the data contained in the original.

To create quality digital audio via conversion from an analog recording

Analog sources require the most effort to get into an appropriate digital form. The hardest part of this process is getting the best possible playback of the source recording. In most cases it is a matter of aligning the playback device for the best playback, but in some cases it can start with trying to find the right device. If you have the recorder that you used to make the original recordings, keep it!

While it is tempting to think you need the absolute best possible digital transfer in terms of sampling rate and bit depth, once you examine what sources you have and what your actual needs are you may find that these standard, common formats suit your needs. You should balance the quality of your source audio recordings, potential future uses, and ease of access. For example, typical cassette recordings of spoken word will not suffer from transfer to a digital form at audio CD quality of 44.1 KHz/16 bits. This can be your master copy, and it offers the added advantage of allowing the direct creation of audio CDs, eliminating the need for transcoding from a higher sample rate or bit depth. This type of transfer is an easy way to digitize your material (devices to do this are available at Groundworks). Subsequent processing to MP3 from an audio CD is also easy.

Finally, digitizing audio from an analog source requires hardware other than the playback device, and this technology changes so fast that we won't discuss it here. We'll be happy to talk to you in person when you are investigating options.

Summary

If you take away nothing else from reading this, remember:

> **Use a supported, non-proprietary format for your files whenever you can.** Available tools for batch processing audio files into a non-proprietary format can make this task much easier.
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> **Balance your efforts in digitizing against the actual use of the recordings.** Is high fidelity reproduction the priority or is access to the content what is really important? Set realistic standards for your content without sacrificing possible future improvements in technology. *This usually means capturing your analog recordings at one level higher than the perceived future need.* Analog recordings originally made at a higher quality or having content that would benefit from better resolution (music, wildlife/nature sounds, etc.) are good candidates for digitizing at higher specifications than standard audio CDs.

Questions?

If you have any questions, please contact us at deepblue@umich.edu and we will be happy to help you. For further information on digital recording and reformatting, please visit the Groundworks at the Digital Media Commons, located in the Duderstadt Center http://www.dc.umich.edu/groundworks/index.htm . They can refer you to experienced digital media professionals that will help you find your best path to success.