Embracing Imperfections - Exploring Vintage Sampling Technology & Techniques

Cover Art

Introduction

Technological advancements have significantly improved digital audio quality eliminating imperfections such as noise and aliasing. However, these so-called imperfections of vintage sampling technology can enhance the sonic character of a sample. Through a detailed examination of technical aspects like sample rate, bit depth, aliasing, downsampling, and upsampling, this document offers an insight into sampling techniques that encourages creativity.

Technical Aspects and Digital Signal Processing (DSP)

This chapter specifically refers to a 'sample' in the context of digital signal processing. Refer to the glossary for help.

Sample rate

A sample in the context of DSP is used to describe discrete data points that represents an audio waveform. Sample rate is the number of data points per second that represents an analogue signal digitally, and is measured in hertz [1]. A good way to think of sample rate is that it is the amount of information stored per second. *Figure 1* shows a visual representation of a single sample.

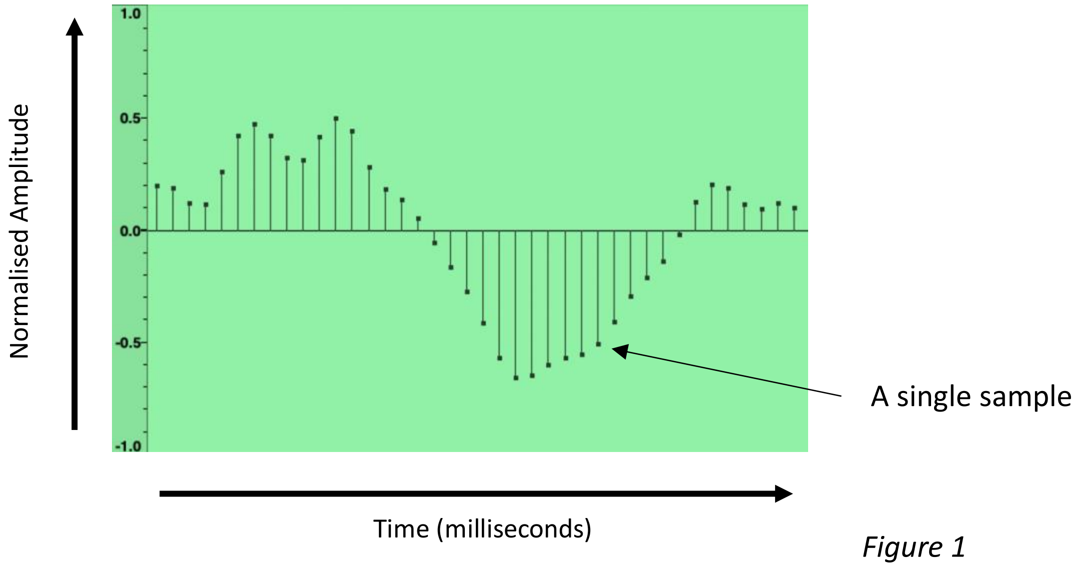

Figure 1

The Nyquist–Shannon sampling theorem states the sample rate must be double the Nyquist frequency to avoid aliasing distortion. Therefore, any frequency below the Nyquist frequency is theoretically free from distortion. The standardised sample rate for music is 44.1 kHz. The following equations represent the relationship between sample rate and the Nyquist frequency [1, 2].

The Nyquist–Shannon sampling theorem:

$$f_n = \frac{1}{2} f_s$$

$$f_s = \frac{N_S}{t}$$

$$f_n = \frac{1}{2}\left(\frac{N_S}{t}\right)$$

Where:

f_n = Nyquist frequency (Hz)
f_s = Sample rate (Hz)
N_S = Number of samples
t = Time/length (s)

From the equations above, a 44.1 kHz sample rate has a Nyquist frequency of 22.05 kHz. Because the upper limit of the human hearing range is 20 kHz, no aliasing should be present in the audible spectrum at this sample rate.

Aliasing

Aliasing occurs when digital audio contains frequencies higher than the Nyquist frequency. Frequency folding, a consequence of aliasing, is the mirroring of frequencies greater than the Nyquist frequency. *Figure 2* shows a visual representation of how a 0 Hz to 8 kHz sine wave sweep undergoes frequency folding when the sample rate is 8 kHz [3]. Note the Nyquist frequency is 4 kHz in this example.

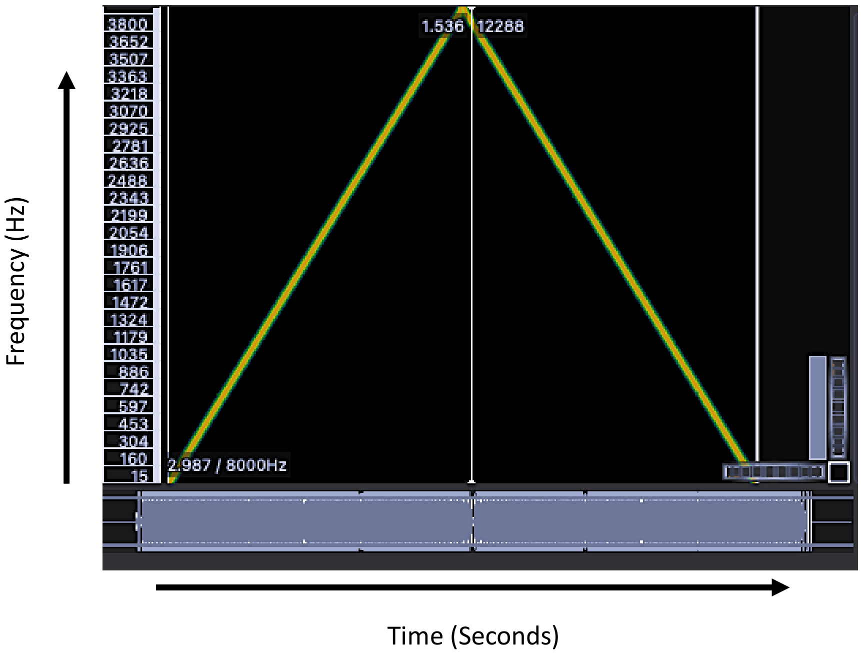

Figure 2

A pure sine wave sweep demonstrates the principles of frequency folding. However, with more complex sounds, frequency folding results in unpredictable frequency interactions. These interactions include, inharmonic distortion, harmonic distortion, intermodulation and phase distortion. While frequency folding doesn't directly create new harmonics, it can introduce artifacts that interact with the original harmonics in a way that are perceived as these effect. Here's how: There is also typically a loss of high frequency information, however this depends on whether anti-aliasing filters and or reconstruction filters are applied to the signal, as well as the sample rate and the spectra of the audio source [1, 10]. Aliasing can cause subtle, significant and unpredictable changes in the perceived tone and quality of a sound. The unpredictable nature of the aliasing adds to its usefulness in a studio context.

Downsampling

Downsampling is the process of intentionally reducing the sample rate of an audio file, introducing aliasing and reducing the file size. A 44100 Hz audio file has double the amount of information compared to an audio file that is 22050 Hz.

Bit depth

Bit depth represents the number of possible amplitude levels each individual sample can contain. A higher bit depth allows for a finer resolution of amplitude and a higher signal to noise ratio as a result of less quantisation noise. The signal to noise ratio is the ratio between the level of the audio signal vs the level of electrical noise - aka noise floor. Therefore, bit depth defines the maximum possible effective dynamic range of a sound. Dynamic range is defined as the difference between the loudest and quietest sound that can be accurately reproduced or recorded. The human ear has a sensitivity range from 0 dB to about 120 dB, equivalent to about 20 bits, and can theoretically only accurately discern volume differences 1 dB apart [2].

Using a low bit depth can introduce quantitation noise to the signal. It can also cause information loss, known as tearing, where the audio level randomly fluctuates between -infinity dB and back to its original value in a distorted way. Listen to Example: 5 (62 OWN FX).wav for a perfect example of both quantisation noise and tearing in effect.

The number of possible amplitude values that an individual sample can hold is represent by the following formulae [1, 2]:

$$r = 2^n$$

Where:

r = Resolution
n = Bit depth

Signal to noise ratio is approximated using the following formulae:

$$SNR = 20 \log r$$

$$SNR = 20 \log 2^n$$

Where:

SNR = Signal to noise ratio
n = Bit depth

Bit Depth	Resolution	Signal-noise ratio (dB)
8	256	48
9	512	54
10	1024	60
11	2048	66
12	4096	72
13	8192	78
14	16384	84
15	32768	90
16	65536	96
24	16777216	144
32	4294967296	192

Figure 3

From the table above, *figure 3*, 24-bit audio has a larger signal to noise ratio than sensitivity range of human hearing meaning that it should technically be impossible to hear the noise floor of 'clean' audio at this resolution [1]. Note that one of the benefits of using 32 bit audio is that it has a significantly higher headroom meaning that clipped signals or extremely quiet recordings can be scaled without any additional noise or distortion [5].

Resampling & Sampling

According to the Ableton Live manual, the Redux device's downsampling introduces "inharmonic frequency content into the spectrum". However, it fails to mention that downsampling can introduce frequency content resulting in a sound that is more appealing than the original. Although low sample rates and bit depth are generally viewed as undesirable intentionally using these characteristics can improve the sonic character of a sound.

Instead of using a saturation effect on a channel I suggest trying a bit crusher, downsampler, or resampler such as TAL-DAC, TAL-Sampler, PotenzaDSP Amigo, Ableton Redux, Logic Bitcrusher, Beatskillz Sample X, RX950, waveTracing SP950, Airwindows Pockey2, DeRez2, BitGlitter, ASTRID Ami-Sampler, BitDOS, or my own Max for Live sampling device PAULA. Another great technique is loading an audio file into Audacity and resampling/changing the project sample rate. Then export the audio file as an unsigned 8-bit .wav file. You can also save audio as .IFF 8-bit file (Amiga IFF, SVX8, SV16) that works on original Amiga hardware and can be opened in ProTracker 2 clone or Amigo on a modern computer. Buying an Amiga or an old sampler such as the AKAI S950 or a E-mu 4XT Ultra is also an option. You can even upgrade these devices, replacing the floppy disk drive with a USB – floppy disk emulator.

A good method of sampling is to loopback the stereo audio output of your computer to a virtual stereo input channel via your audio interface, allowing you to sample any sound from your computers output. Alternatively, you can reamp your own interface using physical patch cables to connect the output into the input, or through an amp and back into the interface. Be cautious of feedback using these methods.

In the 80s and 90s, artists were constrained by sample rate and bit depth limitations, profoundly shaping the production methods and sonic character of electronic music. Modern advancements in sampling technology have relegated downsampling and bit crushing to a bygone era, preserving their distinctive influence as a unique hallmark of its time when digital became affordable and accessible to the masses.

ProTracker 2 Sampling

ProTracker 2 Overview

ProTracker, a music creation program for the Amiga computer, allows artist to sequence short digital samples and manipulate them with effects. ProTracker was unique in how it managed samples and only had 4 channels of 8-bit audio. A free open-source clone of ProTracker 2.3D which runs on modern computers is available here.

ProTracker 2.3D workflow and limitations

ProTracker has only 3 octaves of notes from C1 to B3 with each note assigned its own sample rate. The ProTracker notes frequencies table shows the sample rate assigned to each note in *figure 4*. The lower the note, the lower the sample rate. The internal clock of the Amiga is synced by its video output which is why the table is noted PAL. Sample play back is mono only [4].

ProTracker is unique in how it manages sampling. When importing a sample into ProTracker, a note from the table must be specified [6]. The audio is then resampled at this rate and fixed in pitch at the specified key. A3 +4 finetune (28604 Hz) is the highest note at which resampling is possible [8].

ProTracker 2.3D was initially limited to a maximum sample size of 64 kB, which had to fit within the available RAM on the Amiga. Based on the 64 kB sample length limitation, a sample fixed at E3 (20864 Hz) fits a maximum of 3.14 seconds of audio. Note that later revisions of ProTracker doubled this limit to 128 kB [9].

It was common to sample drums, cymbals, and other high frequency sounds at F3 or above to retain the high frequency information [8]. For other sounds and instruments, it would be best to experiment with different notes to resample to. In the ProTracker 2.3D clone manual refers to E3 as “Lo-quality” [9]. Choosing a note to sample to requires experimentation, for example, an 808 kick doesn’t contain high frequency information therefore it may be fine to sample at C2 or even C1 based on the Nyquist frequency.

ProTracker note frequencies (PAL)[7]

Note	Sample Rate (Hz)
C-1	4144
C#1	4390
D-1	4655
D#1	4926
E-1	5231
F-1	5542
F#1	5872
G-1	6223
G#1	6593
A-1	6982
A#1	7389
B-1	7830
C-2	8287
C#2	8779
D-2	9309
D#2	9852
E-2	10463
F-2	11084
F#2	11745
G-2	12445
G#2	13185
A-2	13964
A#2	14779
B-2	15694
C-3	16574
C#3	17559
D-3	18668
D#3	19705
E-3	20864
F-3	22168
F#3	23489
G-3	24803
G#3	26273
A-3	27928
A#3	29557
B-3	31388

Figure 4

One technique of saving memory was done by transposing a sample up by an octave and resampling at the same sample rate, halving the sample length and therefore halving the file size [4]. This technique is known as upsampling. *Figure 5* shows the difference in storage size of an 808 kick / sub bass demonstrating the effect of this technique.

Name	Size
1 (14 OWN BASS).wav	750 KB
2 (14 OWN BASS).wav	33 KB
3 (14 OWN BASS).wav	16 KB

Figure 5

Where:

1 (14 OWN BASS).wav = Original (48 kHz, 32 bit)
2 (14 OWN BASS).wav = Downsampled (8.29 kHz, 8 bit)
3 (14 OWN BASS).wav = Upsampled (8.29 kHz, 8 bit)

Pitching the sound up an octave shifts the frequency information up the spectrum by multiple of 2. Note that transposing the sample up an octave at the same sample rate (8.29 kHz) doesn't drastically affect the sound because the 808 doesn't contain high frequency information and the Nyquist frequency is 4.15 kHz. This was done to save space in the RAM and also on storage such as 1.44 MB floppy disk.

Sample CD

Sample Material

File names are organised into ‘Own’, ‘Resampled’ and ‘Vinyl’. I decided to split the samples into different sections; Breaks, Bass, Leads, Pads, FX, and Vox as I felt that this best described the material in general.

The ‘Own’ samples are sounds which have been individually made by myself using hardware, software and recordings. Below is a list of all equipment used.

- Hardware: Roland JV-1080, Roland JX-8p, Roland SH-9, Moog Theremin, Casio CZ-5000, KORG TR-Rack, KORG Minilogue, KORG Wavestation, Arturia Microfreak, AKAI S1000, Mackie 1604, Yamaha AX-496, Yamaha 02R96 and a Toshiba D-VR17 VHS Recorder.
- Soft synths: IK Multimedia DCO-X, IK Multimedia 99, Roland JD-800, Roland D50, Roland SH-101, Ableton Analog, TAL-Sampler, FB-7999, KORG Triton, KORG M1, Arturia Emulator II V, Arturia CMI V and Granulator II.
- Microphones / recorders: Zoom H4n, Zoom H1n, AKG C414, Sennheiser MKH 418-S, and a Shure SM58.

The ‘Resampled’ samples are sounds which have been sampled from any digital source such as YouTube, .wav, .mp3, etc.

The ‘Vinyl’ samples were sampled by myself using a Technics SL-1200 MK3 turntable. I went through about a thousand records in DJ Don’t Fuck About’s collection of jazz, funk, soul, dub and reggae.

The CD version includes extra content.

PAULA 1.93

Based on the Amiga 1200 and ProTracker 2.3D, I decided to create my own resampler using Max for Live. The main reason I created PAULA was to overcome the sample length limitation, emulate the resampling behaviour to a specific note like ProTracker but in real-time and to support stereo sampling. I was able to match the audio spectrum of the downsampling to ProTracker 2.3D clone [6]. I also decided to include additional features such as being able to adjust the LED filter cutoff, variable bit rate and ADSR envelopes. The device is dependent on TAL-DAC VST3 due to its real time variable reconstruction filter [10]. I recommend sampling the output of PAULA. *Figure 6* shows screenshot of PAULA 1.93.

Figure 6

Conclusion

Modern recording capabilities have perfected digital audio quality, rendering it virtually flawless by eliminating undesirable artifacts such as aliasing and noise. However, exploring and emulating the characteristics of vintage sampling technologies by manipulating sample rate and bit depth can yield desirable sonic qualities. By embracing the imperfections inherent in older technologies, artists can not only enrich their sound but also gain a deeper understanding of the history and evolution of sound design. As we continue to innovate in digital audio, it is valuable to remember and integrate this knowledge.

Glossary

ADSR Envelope: A parameter used in synthesizers and samplers to shape the sound's amplitude over time. ADSR stands for Attack, Decay, Sustain, and Release, which are the four stages of the envelope that control how the sound evolves.

Aliasing: The overlapping of frequency information resulting from a sample rate below the Nyquist frequency. This overlap, also known as frequency folding, can result in inharmonic distortion, harmonic distortion, intermodulation and phase distortion

Anti-Aliasing: A technique used to minimize aliasing that typically involves filtering out frequencies near the Nyquist frequency.

Amiga 1200: A home computer introduced by Commodore in 1992, widely used for gaming and music production.

Bit Depth: Represents the number of possible amplitude levels each individual sample can contain. A higher bit depth allows for a finer resolution of amplitude and a higher signal-to-noise ratio as a result of less quantization noise.

DAC/ADC: Digital to Analogue Converter / Analog to Digital Converter.

Downsampling: The process of reducing the sample rate of an audio signal.

Digital Signal Processing: The manipulation of digital data representing a signal to alter its properties using algorithms and mathematical techniques.

Dynamic Range: The difference between the loudest and quietest sound that can be accurately reproduced or recorded, often measured in decibels (dB).

Decibels (dB): A unit of measurement for the intensity of sound. It’s a logarithmic scale used to quantify the ratio of a particular sound level compared to a reference.

Discrete: Mathematically discrete means distinct and countable, like separate steps on a ladder; whereas continuous refers to smooth, unbroken ranges of values, like a ramp.

Frequency folding: The mirroring of frequencies greater than the Nyquist frequency as a result of aliasing.

Floppy disk: A magnetic storage medium released in 1971 for storing and transferring digital data. Common sizes included 3.5-inch and 5.25-inch disks, with the former holding up to 1.44 MB of data.

Hertz (Hz): The unit of frequency, defined as one cycle per second. Used to measure periodic phenomena. 1 kHz = 1000 Hz.

Harmonic distortion: A type of distortion that occurs when harmonics (multiples of the original frequency) are added to the signal.

Intermodulation distortion: Distortion that occurs when different frequencies mix together and produce new frequencies that are not harmonically related to the original signal.

Loopback: A process where the output of a system is fed back into its input.

Max for Live: A platform that integrates the Max programming language with Ableton Live, allowing users to create custom instruments, effects, and tools for audio and visuals.

Nyquist Frequency: The maximum frequency a signal can contain to be accurately reconstructed.

Nyquist–Shannon Sampling Theorem: A fundamental principle in digital signal processing that states that the sample rate must be double the Nyquist frequency to avoid aliasing distortion.

Noise floor: The background noise in a recording or audio signal, such as electrical interference or quantisation noise.

ProTracker 2.3D: A music tracker for the Amiga computer.

Quantization Noise: Distortion introduced due to the quantisation of the discrete data. It arises from the rounding of values to discrete digital levels.

RAM: Random access memory, a type of computer memory used for temporarily storing data that is being actively used or processed by the computer.

Reconstruction Filter: A filter used in digital to analogue conversion (DAC) to smooth out aliasing and reconstruct a continuous analogue waveform. An anti-aliasing technique.

Resampling: The process of recording a previously digitised signal.

Sampling: The process of converting a continuous analogue signal into a discrete digital signal by taking regular measurements of the amplitude of the analogue signal.

Sample: Refers to a single discrete data point that forms part of a waveform in terms of DSP. Or an entire sound recording, in terms of music production.

Signal-to-Noise Ratio (SNR): The ratio between the level of the audio signal vs the level of electrical noise (noise floor). A higher SNR indicates a 'cleaner' signal with less noise.

Spectrum/Spectra: The representation of a signal's frequency content. Most spectra used in the context of sound engineering such a parametric EQ are graphs of frequency (Hz) vs gain (dB).

Tearing: Distortion that occurs when the audio level randomly fluctuates between -infinity dB and its original value due to low bit depth as a result of quantisation error.

Upsampling: The process of increasing the sample rate of an audio signal.

VHS: VHS (Video Home System) is an analogue audiovisual recording format using magnetic tape cassettes introduced in 1976. It became the dominant home video format in the 1980s and 1990s.

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"Do I spend too much time thinking and not enough doing?
Did I try my hardest at any of my dreams?
Did I purposely let others discourage me when I knew I could?
Will I die, never knowing what I could have been or could have done?
Do not let these doubts restrain or trouble you
Just point yourself in the direction of your dreams
Find your strength in the sound
And make your transition
Make your transition
Make your transition"

Galaxy 2 Galaxy – Transition
Underground Resistance
Christa Robinson, Cornelius Harris, DJ Dex

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