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LUCRARE DE LICENŢĂ

Detecția și eliminarea distorsiunii din înregistrări audio de pe formate analoage

Conducător ştiinţific

Lect. Dr. Sterca Adrian

Absolvent

**Drimba Alexandru**

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BABEŞ-BOLYAI UNIVERSITY CLUJ-NAPOCA

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SPECIALIZATION COMPUTER SCIENCE - ROMANIAN

DIPLOMA THESIS

Distortion detection and removal on audio recordings from analog formats

Supervisor

Lect. Dr. Sterca Adrian

Author

**Drimba Alexandru**

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Abstract

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Chapter 3

Basic digital signal processing

3.1. Digital audio signal representation

Sound is defined as an oscillation in pressure, stress, particle displacement, particle velocity, etc., propagated in a medium with internal forces (e.g., elastic or viscous), or the superposition of such propagated oscillation [1]. This oscillation can be represented as a continuous function that describes the variation in time of the medium’s pressure, allowing us to “see” sounds. (Fig. 3.1.1)

Sound is transmitted through gases and liquids as longitudinal waves, and through solids both as longitudinal and transverse waves. A transmitting medium is required, so sound cannot travel through vacuum. In a longitudinal wave, the direction of displacement is the same as the direction of propagation, while in a transverse wave, the direction of displacement is perpendicular to the direction of propagation [2]. To better understand the difference between the longitudinal and transversal waves, Fig. 3.1.1 depicts sound waves in air (longitudinal waves), while Fig. 3.1.2 presents transverse waves travelling through a metal wire.

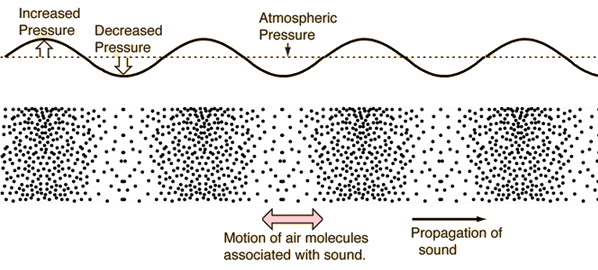
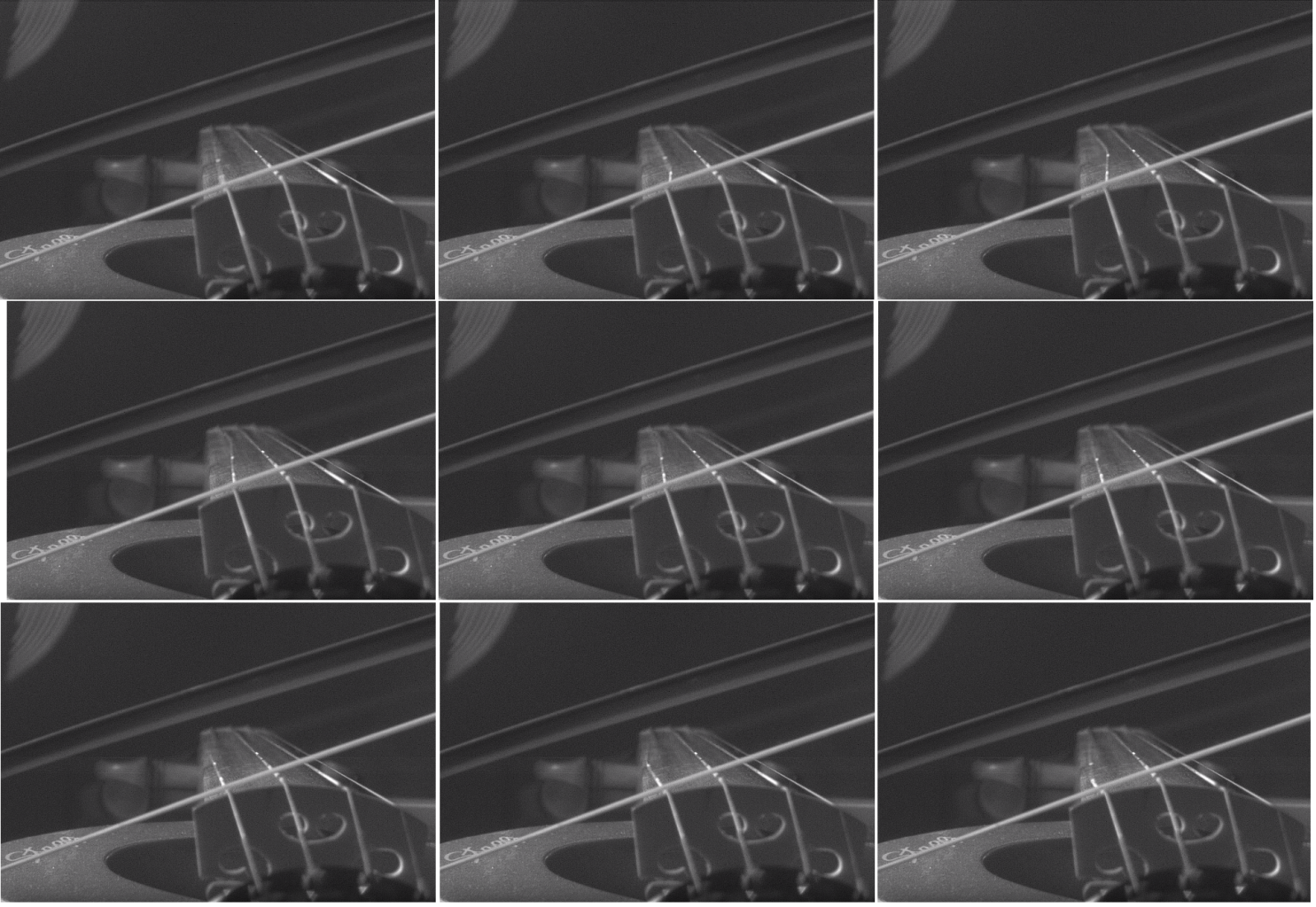


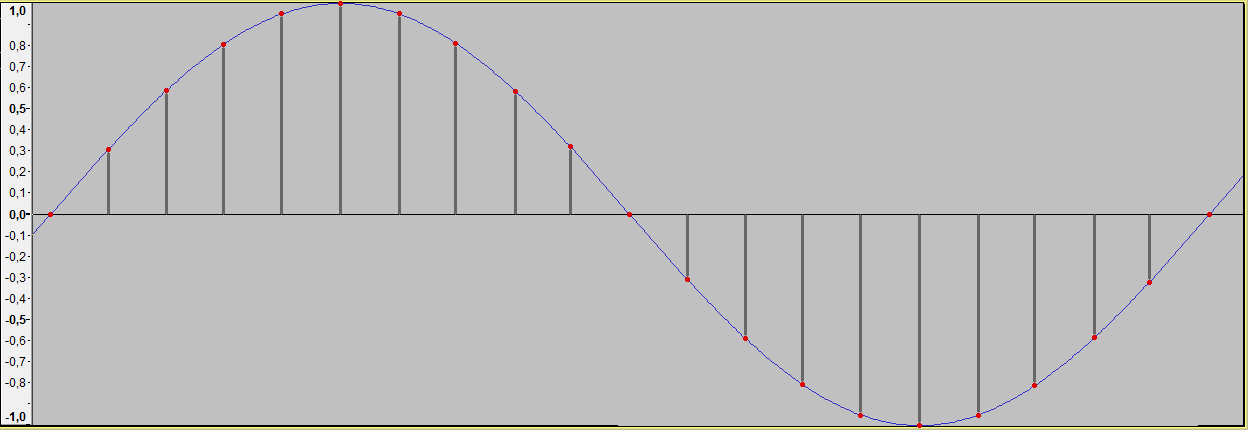
Fig. 3.1.1: Sound Waves in Air. A single-frequency sound wave traveling through air will cause a sinusoidal pressure variation in the air. The air motion which accompanies the passage of the sound wave will be back and forth in the direction of the propagation of the sound, a characteristic of longitudinal waves.[3]

Fig. 3.1.2. Transverse waves seen in a bowed violin chord. [4]. The bow’s movement drags the chord, its displacements being transmitted as transverse waves moving along the chord with the speed of sound of the chord’s material.

Audio signals are representation of sound, typically as an electrical voltage (analog) or as discrete numerical values (digital). Conversion from the analog continuous-time signal to the digital discrete-time signal (also called sampling) is made usually with ADCs (Analog to Digital Converters), and with DACs(Digital to Analog Converters) from digital to analog. A **sample** isa signal’s value at a point in time**.** When converting from analog to digital, some of the information is lost because of factors like:

* 1. discretization – the resulted signal is no longer continuous (precisely defined in every point in time), but discrete: the intensity of the analog signal is recorded at fixed time points. The number of equidistant time points in a second is called **sample rate** (or sampling frequency)**.** The highest frequency that can be carried by the signal, called the **Nyquist frequency**, is given by the following formula:
  2. storage as finite numbers – as opposed to analog values, the precision of the digital values is finite, so only some of the significant digits can be stored.

After getting the sample values from the original analog signal, the samples are then stored as digital numbers in audio files. These sample values can be stored either as uncompressed files, such as the WAV and AU formats, which we’ll be discussing about later, or as compressed files (to decrease file size). Compressed file formats can be lossy (the decompressed data is an approximation of the original), such as MP3, or lossless ( compression preserves the exact original values ), such as FLAC. Audio files typically contain information about the sampling rate, channels (for mono/stereo etc. recordings) and sample encoding (float/integers, signed/unsigned, bit-depth, companding).

Fig. 3.1.3. Conversion from continuous-time to discrete-time. Here, a sine wave is sampled 20 times for each cycle. Each sample is then stored as a 8-bit signed integer ( values in [ -128, 127 ] ). Table below (Table 3.1.1) shows the discretization errors raised at the conversion.

|  |  |  |  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- |
| Sample no. | 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 |
| Actual value | 0,00000 | 0,30902 | 0,58779 | 0,80902 | 0,95106 | 1,00000 | 0,95106 | 0,80902 | 0,58779 | 0,30902 | 0,00000 |
| Sampled value | 0,00000 | 0,30469 | 0,58594 | 0,80469 | 0,94531 | 0,99219 | 0,94531 | 0,80469 | 0,58594 | 0,30469 | 0,00000 |
| Error | 0,00% | 0,43% | 0,19% | 0,43% | 0,57% | 0,78% | 0,57% | 0,43% | 0,19% | 0,43% | 0,00% |

|  |  |  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- |
| Sample no. | 11 | 12 | 13 | 14 | 15 | 16 | 17 | 18 | 19 | 20 |
| Actual value | -0,30902 | -0,58779 | -0,80902 | -0,95106 | -1,00000 | -0,95106 | -0,80902 | -0,58779 | -0,30902 | 0,00000 |
| Sampled value | -0,31250 | -0,59375 | -0,81250 | -0,95313 | -1,00000 | -0,95313 | -0,81250 | -0,59375 | -0,31250 | 0,00000 |
| Error | 0,35% | 0,60% | 0,35% | 0,21% | 0,00% | 0,21% | 0,35% | 0,60% | 0,35% | 0,00% |

Table 3.1.1. Errors at conversion from analog to digital samples. The digital samples are stored as 8-bit signed integers, with values in [ -128, 127 ], rescaled here to [ -1,1 ) to show the error. Analog samples are in the range [ -1,1 ], where -1 is the smallest possible signal value, and 1 is the maximum. We can see the errors are pretty large for this sample encoding.

Bibliography

[1]. American National Standard on Acoustical Terminology, ANSI/ASA S1.1-2013

[2]. Definitions of longitudinal and transverse waves, <http://www.dictionary.com>

[3]. Sound waves in air, <http://hyperphysics.phy-astr.gsu.edu/hbase/Sound/tralon.html>

[4]. Bowed violin string in slow motion, <https://www.youtube.com/watch?v=6JeyiM0YNo4> (11.04.2018)

[]. A tutorial on Burg's method, algorithm and recursion, Cedrick Collomb, 2009:

<http://www.emptyloop.com/technotes/A%20tutorial%20on%20Burg's%20method,%20algorithm%20and%20recursion.pdf>

[]. Direct and Fast Fourier Transforms: <http://www.alwayslearn.com/DFT%20and%20FFT%20Tutorial/DFTandFFT_FFT_Overview.html>

[]. Finite Impulse Response Filter Design:

<https://www.cs.tut.fi/~ts/Mitra_Kaiser.pdf>

[]. Jose Maria Giron-Sierra: *Digital Signal Processing with Matlab Examples, Volume I,* Springer, ISBN 978-981-10-2534-1

[]. Jose Maria Giron-Sierra: *Digital Signal Processing with Matlab Examples, Volume II,* Springer, ISBN 978-981-10-2537-2

[]. Jose Maria Giron-Sierra: *Digital Signal Processing with Matlab Examples, Volume III,* Springer, ISBN 978-981-10-2540-2

[]. FIR Filter Basics: <http://dspguru.com/dsp/faqs/fir/basics/>

[]. FIR Filter Properties: <http://dspguru.com/dsp/faqs/fir/properties/>

[9]. FIR Filter Design: <https://dspguru.com/dsp/faqs/fir/design/>

[10]. IIR Filter Basics: <http://dspguru.com/dsp/faqs/iir/basics/>