

# Psychophysics of musical elements in the discrete-time representation of sound

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Notes, ornaments and intervals are examples of basic musical elements. Their representation as discrete-time digital audio plays a central role in software for music creation and design. Nevertheless, there is no systematic relation, in analytical terms, of these musical elements to the sonic samples. Such a compendium enables scientific experiments in precise and trustful ways, among educational and artistic uses. This paper presents a comprehensive description of music in digital audio, within an unified approach. Musical elements, like pitch, duration and timbre are expressed by equations on sample level. This quantitatively relates characteristics of the discrete-time signal to musical qualities. Internal variations, e.g. tremolos, vibratos and spectral fluctuations, are also considered to operate within a note. Moreover, the generation of musical structures such as rhythmic meter, pitch intervals and cycles, are attained canonically with notes. The availability of these resources in scripts is provided in public domain within the MASSA toolbox - Music and Audio in Sequences and Samples. Authors observe that the implementation of sample-domain analytical results as open source can encourage concise research. As further illustrated in the paper, MASSA has already been employed by users for diverse purposes, including acoustics experimentation, art and education. The efficacy of these physical descriptions was confirmed by the synthesis of small musical pieces. As shown, it is possible to synthesize whole albums through collage of scripts and parametrization.

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## I. INTRODUCTION

Music is commonly defined as the art made by sounds and silences. Sound is the longitudinal wave of mechanical pressure. The frequency bandwidth between  $20Hz$  and  $20kHz$  is appreciated by the human hearing system with boundaries dependent on the person, climate conditions and sonic characteristics themselves<sup>1</sup>. Considered the speed of sound as  $\approx 343.2m/s$ , these limits imply wavelengths of  $\frac{343.2}{20} = 17.16m$  and  $\frac{343.2}{20000} = 17.16mm$ .

Human perception of sound involves captivation by bones, stomach, ears, transfer functions of head and torso, and processing by the nervous system<sup>1</sup>. Besides that, the ear is a dedicated organ for the appreciation of these waves. Its mechanism decomposes sound into its sinusoidal spectrum and delivers to the nervous system. The sinusoidal components are crucial to musical phenomena, as one can perceive in the composition of sounds of musical interest (such as harmonic sounds and noises, exposed in sections II and III), and sonic structures of musical interest (such as tunings and scales, in section IV).

The representation of sound is commonly called au-

dio (although these terms are often used without distinction). Audio expresses waves prominent from the capture of microphones or from direct synthesis, although these sources entangle as captured sounds are processed to generate new sonorities. Digital audio specified by protocols that eases file transferring and storage often imply a quality loss. Standard representation of digital audio, on the other hand, assures perfect reconstruction of the analog wave, within any convenient precision. This paradigm consists of equally spaced samples, by  $\lambda_s$  durations, each specified by a fixed number of bits. This is the Pulse Code Modulation (PCM) representation of sound. A sound in PCM audio is characterized by a sampling frequency  $f_s = \frac{1}{\lambda_s}$  (also called the sampling rate), which is the number of samples used for representing a second of sound; and by a bit depth, which is the number of bits used for representing the amplitude of each sample. Figure 1 shows 25 samples of a PCM audio with 4 bits each. The fixed  $2^4 = 16$  values for the amplitude of each sample, with the regular spacing  $\lambda_s$ , yields the quantization error or noise. This noise diminishes as bit depth increases.

The Nyquist theorem states that the sampling frequency is twice the maximum frequency of the represented signal. Thus, for general musical purposes, it is necessary to have samples in a rate at least twice the highest frequency heard by humans, that is,  $f_s \geq 2 \times 20kHz = 40kHz$ . This is the fundamental reason for the adoption of sampling frequencies like  $44.1kHz$  and  $48kHz$ , standards in Compact Disks (CD) and broadcast systems (radio and television), respectively.

Within this framework, musical notes can be charac-

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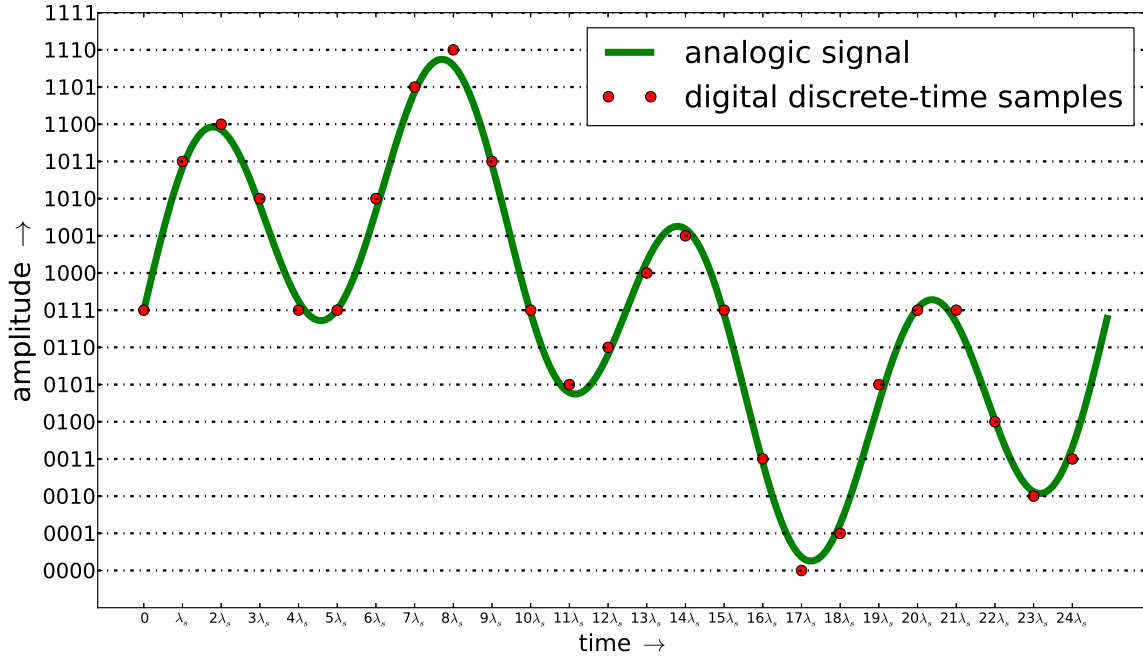


FIG. 1. Pulse Code Modulation (PCM) audio: an analogical signal is represented by 25 samples with 4 bits each.

terized. The note still stands paradigmatic as the 'fundamental unit' of musical structures and, in practice, it can unfold into sounds that uphold other approaches<sup>2</sup>. Notes are also convenient for another reason: the average listener – and considerable part of the specialists – presupposes rhythmic and pitch organization (made explicit in section IV) as fundamental musical properties, and these are developed in traditional musical theory in terms of notes.

#### A. Contributions and paper organization

This article aims at representing musical structures and artifices by their discrete-time sonic characteristics. Results include mathematical relations, usually in terms of samples, and their direct computer program implementations. Despite the general interests involved, there are few books and computer implementations that tackle the subject. These mainly focus on computer implementations and ways to mimic traditional instruments, with scattered mathematical formalisms. A compilation of the works and their contributions is in the bibliography<sup>3</sup>. To the best of the author's knowledge, there is a lack of articles on the topic. Moreover, although current computer implementations use the analytical descriptions presented in this study in an implicit manner, it seems that there has been no concise and mathematical description of the processes implemented.

In order to address this concise description of musical

elements and structures, in terms of the digitized sound, the objectives of this paper are:

1. Present a concise set of relations among musical basic elements and sequences of PCM audio samples.
2. Introduce a framework of sound synthesis with control at sample level and with potential uses in psycho-acoustical experiments and high-fidelity synthesis.
3. Provide the accessibility of the developed framework. The analytic relations presented in this article are implemented as public domain scripts, i.e. small computer programs using accessible technologies for better distribution and validation. This constitute the MASSA toolbox, available in an open Git repository<sup>4</sup>. These scripts are written in Python and make use of external libraries, such as Numpy that performs calls to Fortran routines for better computational efficiency. Part of the scripts have been transcribed to JavaScript and native Python with readiness, what favors their use in Web browsers such as Firefox and Chromium<sup>5–8</sup>. Furthermore, these are all open technologies, that is, published using licenses that allow copy, distribution and use of any part for research and derivatives. This way, the work presented here is embedded in recommended practices for availability and eases co-authorship processes<sup>9,10</sup>.

4. To provide a didactic presentation of the content to favor its apprehension and usage. It is worthwhile to mention that this subject comprises diverse topics on signal processing, music, psycho-acoustics and programming.

The remaining parts of this work are organized as follows: section II characterizes the basic musical note; section III further develops internals of the musical note; section IV tackles the organization of musical notes in higher levels of musical structure<sup>11?–14</sup>. As these descriptions embody topics such as psycho-acoustics, cultural traditions, formalisms and protocols, the text points to external complements as needed<sup>15–17</sup>.

The next section is a minimum text in which musical elements are presented side-by-side with the discrete-time samples they result. In order to account for validation and sharing, implementations on computer code of each one of these relations are gathered in the MASSA toolbox together with little musical montage resulting from them.

## II. CHARACTERIZATION OF THE DISCRETE-TIME MUSICAL NOTE

In diverse artistic and theoretical contexts, music is conceived as constituted by fundamental units called notes, “atoms” that constitute music itself<sup>11,12,18</sup>. Nowadays, these notes are understood as central elements of certain musical paradigms. In a cognitive perspective, the notes are seen as discretized elements that easy and enrich the flow of information through music<sup>1,14</sup>. Canonically, the principal properties of a musical note are duration, volume, pitch and timbre<sup>14</sup>. These qualities can be managed quantitatively, dictated by the evenly time spaced sonic samples.

All the relations described in this section are implemented at the file `eqs2.1.py` of the MASSA toolbox. Musical pieces *5 sonic portraits* and *reduced-fi* are also available online in order to corroborate the concepts.

### A. Duration

The sample frequency  $f_s$  is defined as the number of samples in each second of the discrete-time signal. Let  $T_i = \{t_i\}$  be an ordered set of real samples separated by  $\delta_s = 1/f_s$  seconds. A musical note of duration  $\Delta$  seconds is presented as a sequence  $T_i^\Delta$  of  $\lfloor \Delta \cdot f_s \rfloor$  samples. That is, the integer part of the multiplication is considered, and an error of at most  $\delta_s$  missing seconds is admitted, which is usually fine for musical purposes. As an example, if  $f_s = 44.1kHz \Rightarrow \lambda_s = \frac{1}{44100} \approx 23$  microseconds. Thus:

$$T_i^\Delta = \{t_i\}_{i=0}^{\lfloor \Delta \cdot f_s \rfloor - 1} \quad (1)$$

With  $\Lambda = \lfloor \Delta \cdot f_s \rfloor$ , the number of samples in the sequence, the more condensed notation is  $T_i^\Delta = \{t_i\}_0^{\Lambda-1}$ .

### B. Volume

The sensation of sound volume depends on reverberation and harmonic distribution, among other characteristics described on section III. One can get volume variations through the potency of the wave<sup>19</sup>:

$$pot(T_i) = \frac{\sum_{i=0}^{\Lambda-1} t_i^2}{\Lambda} \quad (2)$$

The final volume is dependent on the speakers amplification of the signal. Thus, what matters is the relative potency of a note in relation to the other ones around it, or the potency of a music section in relation to the rest. Differences in volume are measured in decibels, calculated directly from the amplitudes through energy or potency:

$$V_{dB} = 10 \log_{10} \frac{pot(T'_i)}{pot(T_i)} \quad (3)$$

The quantity  $V_{dB}$  has the decibel unit ( $dB$ ). For each  $10dB$  it is associated a “doubled volume”. A handy reference is  $10dB$  for each step in the intensity scale: *pianissimo*, *piano*, *mezzoforte*, *forte* e *fortissimo*. Other useful references are  $dB$  values related to double amplitude or potency:

$$\begin{aligned} t'_i &= 2 \cdot t_i \Rightarrow pot(T'_i) = 4 \cdot pot(T_i) \Rightarrow \\ &\Rightarrow V'_{dB} = 10 \log_{10} 4 \approx 6dB \end{aligned} \quad (4)$$

$$\begin{aligned} pot(T'_i) &= 2 \cdot pot(T_i) \Rightarrow \\ &\Rightarrow V'_{dB} = 10 \log_{10} 2 \approx 3dB \end{aligned} \quad (5)$$

and the amplitude gain for a sequence whose volume has been doubled ( $10dB$ ):

$$\begin{aligned} 10 \log_{10} \frac{pot(T'_i)}{pot(T_i)} &= 10 \Rightarrow \\ \Rightarrow \sum_{i=0}^{\lfloor \Delta \cdot f_s \rfloor - 1} t_i'^2 &= 10 \sum_{i=0}^{\Lambda-1} t_i^2 = \sum_{i=0}^{\Lambda-1} (\sqrt{10} \cdot t_i)^2 \quad (6) \\ \therefore t'_i &= \sqrt{10} t_i \Rightarrow t'_i \approx 3.16 t_i \end{aligned}$$

As shown, it is necessary an amplitude increase by a bit more than a factor of 3 for a doubled volume. These values are guides for increasing or decreasing the absolute values on the sample sequences with musical purposes. The conversion from decibels to amplitude gain (or attenuation) is straightforward:

$$A = 10^{\frac{V_{dB}}{20}} \quad (7)$$

where  $A$  is the multiplicative factor that relates the amplitudes before and after the amplification.

### C. Pitch

As exposed in the previous subsections, to a note corresponds a sequence  $T_i$  in which duration and volume are directly related to the size of the sequence and the amplitude of its samples, respectively. The pitch is specified by the fundamental frequency  $f_0$  whose cycle has duration  $\delta_{f_0} = 1/f_0$ . This duration, multiplied by the sampling frequency  $f_s$  results on the number of samples per cycle:  $\lambda_{f_0} = f_s \cdot \delta_{f_0} = f_s / f_0$ .

For didactic reasons, be  $f_0$  such that it divides  $f_s$  and  $\lambda_{f_0}$  results integer. If  $T_i^f$  is a sonic sequence of fundamental frequency  $f$ , then:

$$T_i^f = \{t_i^f\} = \{t_{i+\lambda_f}^f\} = \left\{t_{i+\frac{\lambda_f}{f}}^f\right\} \quad (8)$$

In the next section, frequencies  $f$  that does not divide  $f_s$  will be considered. This restriction does not imply in lost of generality of this section's content.

### D. Timbre

In a sound with harmonic spectrum, the (wave) period corresponds to the duration of the cycle, given by the inverse of the fundamental frequency. The trajectory of the wave inside the period - called the waveform - defines a harmonic spectrum and, thus, a timbre<sup>20</sup>. Sonic spectra with minimum differences can result in timbres with crucial expressive differences and, consequently, different timbres can be produced using different spectra<sup>1</sup>.

The simplest case is the spectrum that consists of only of the fundamental frequency  $f$ . This defines the sinusoid: a frequency in pure oscillatory movement called 'simple harmonic movement'. Be  $S_i^f$  a sequence whose samples  $s_i^f$  describes a sinusoid of frequency  $f$ :

$$S_i^f = \{s_i^f\} = \left\{\sin\left(2\pi \frac{i}{\lambda_f}\right)\right\} = \left\{\sin\left(2\pi f \frac{i}{f_s}\right)\right\} \quad (9)$$

where  $\lambda_f = \frac{f_s}{f} = \frac{\delta_f}{\lambda_s}$  is the number of samples in the period.

In a similar fashion, other waveforms are applied in music for their spectral qualities and simplicity. While the sinusoid is an isolated point in the spectrum, these waves present a succession of harmonic components. These waveforms are specified in equations 9, 10, 11 and 12, illustrated in Figure 2. These artificial waveforms are traditionally used in music for synthesis and oscillatory control of variables. They are also useful outside musical contexts<sup>21</sup>.

The sawtooth presents all components of the harmonic series with decreasing energy of  $-6dB/octave$ . The sequence of temporal samples can be described as:

$$D_i^f = \{d_i^f\} = \left\{2 \frac{i \% \lambda_f}{\lambda_f} - 1\right\} \quad (10)$$

The triangular waveform present only odd harmonics falling with  $-12dB/octave$ :

$$T_i^f = \{t_i^f\} = \left\{1 - \left|2 - 4 \frac{i \% \lambda_f}{\lambda_f}\right|\right\} \quad (11)$$

The square wave preserves only odd harmonics falling at  $-6dB/octave$ :

$$Q_i^f = \{q_i^f\} = \begin{cases} 1 & \text{for } (i \% \lambda_f) < \lambda_f/2 \\ -1 & \text{otherwise} \end{cases} \quad (12)$$

The square wave can be used in a subtractive synthesis with the purposes of mimicking a clarinet. This instrument has only the odd harmonic components and the square wave is convenient with its abundant energy in high frequencies. The sawtooth is a common starting point for a subtractive synthesis, because it has both odd and even harmonics with high energy. In general, these waveforms are appreciated as excessively rich in sharp harmonics, and attenuator filtering on treble and middle parts of the spectrum is specially useful for reaching a more natural and pleasant sound. The relatively attenuated harmonics of the triangle wave makes it the more functional - among the listed ones - to be used in the synthesis of musical notes without any treatment. The sinusoid is often a nice choice, but a problematic one. While pleasant if not loud in a very high pitch (above 500Hz, it requires careful dosage), the pitch is not accurately detected, specially in low frequencies. It requires a big amplitude to give sinusoid volume, if compared to other waveforms. Both particularities are seen as a consequence of the non-existence of pure sinusoidal sounds in nature<sup>1</sup>.

Figure 2 presents the waveforms described in equations 9, 10, 11 and 12 for  $\lambda_f = 100$  (period of 100 samples). If  $f_s = 44.1kHz$ , the PCM standard in Compact Disks, the wave has fundamental frequency  $f = \frac{f_s}{\lambda_f} = \frac{44100}{100} = 441 \text{ Herz}$ , around A4, just above the central "C", whatever the waveform is.

The spectrum of each basic waveform is in Figure 3. The isolated and exactly harmonic components of the spectrum is a consequence of the fixed period usage. The sinusoid consists of an unique node in the spectrum, pure frequency. The figure exhibit the spectra described: the sawtooth is the only waveform with a complete harmonic series (odd and even components); triangular and square waves has the same components (odd harmonics), decaying at  $-12dB/octave$  and  $-6dB/octave$ , respectively.

The harmonic spectrum is composed of frequencies  $f_n$  that are multiples of the fundamental frequency:  $f_n = (n+1)f_0$ . As the human linear perception of pitch follows a geometric progression of frequencies, the spectrum

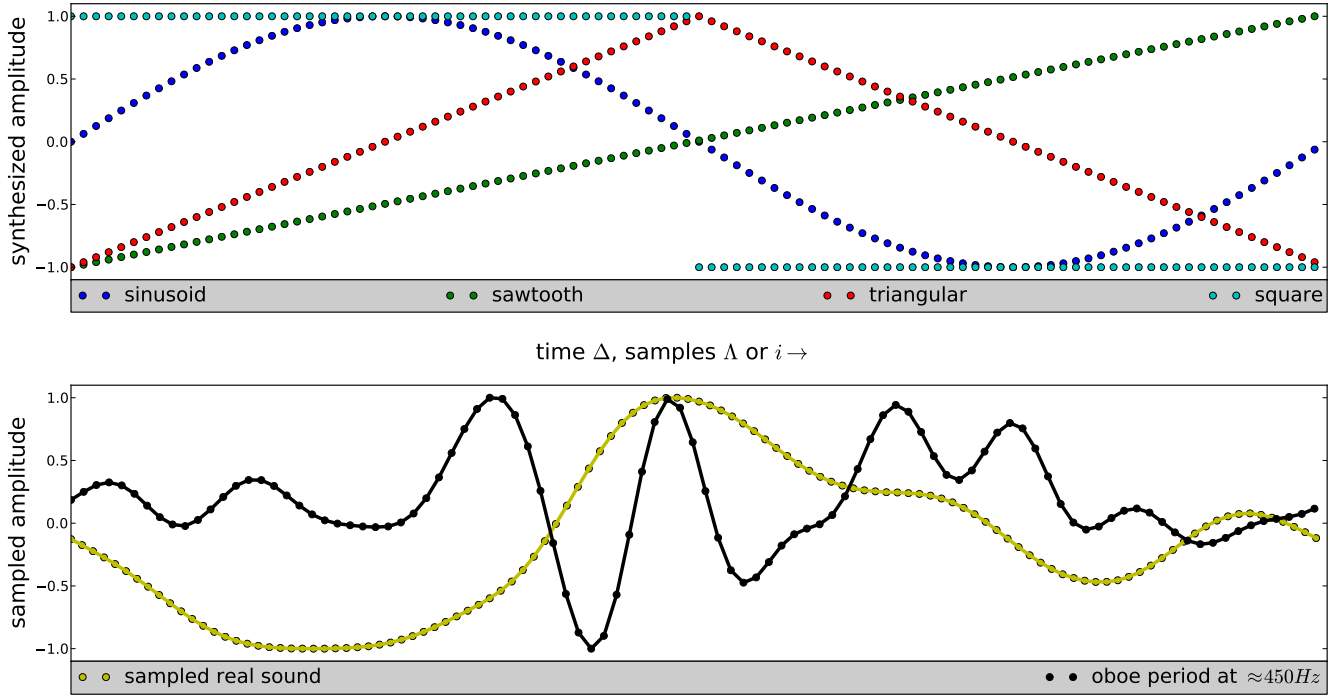


FIG. 2. Basic musical waveforms: (a) the synthetic waveforms; (b) the real waveforms.

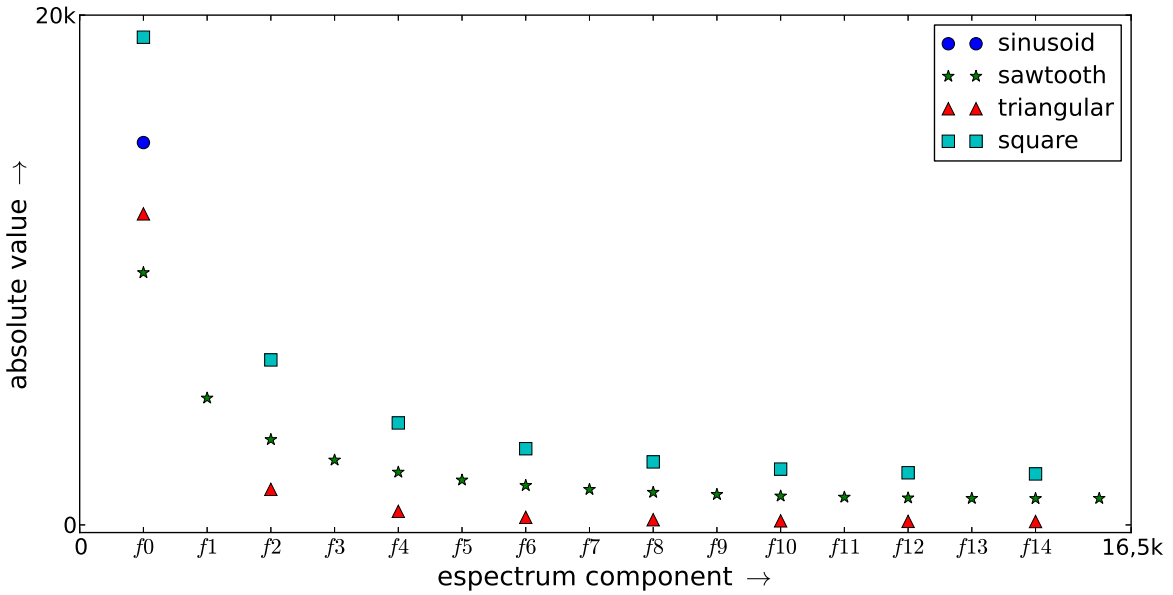


FIG. 3. Spectrum of basic artificial musical waveforms.

has notes different from the fundamental frequency (see equation 84). Additionally, the number of harmonics will be limited by the Nyquist frequency  $f_s/2$ .

From a musical perspective, it is critical to internalize that energy in a component of frequency  $f_n$  means an oscillation in the constitution of the sound, purely

harmonic and in that frequency  $f_n$ . This energy, specifically concentrated on the frequency  $f_n$ , is separated by the ear for further cognitive processes (this separation is done in many species with mechanisms similar to the human cochlea<sup>1</sup>).

The sinusoidal components are usually the main re-

sponsible for timbre qualities. If they are not presented in harmonic proportions (small number relations), the sound is perceived as noisy or dissonant, in opposition to sonorities with an unequivocally established fundamental. Furthermore, the notion of absolute pitch relies on the similarity of the spectrum to the harmonic series<sup>1</sup>.

In the case of a fixed length period and waveform, the spectrum is perfectly harmonic and static, and each waveform is compound of specific proportions of harmonic components. High curvatures are sign of high harmonics in the wave. Figure 2 depicts a wave, labeled as “sampled real sound”, with a period of  $\Lambda_f = 114$  samples, extracted from a relatively well behaved recorded sound. The oboe wave was also sampled in  $44.1kHz$ . The chosen period for sampling was relatively short, with 98 samples, and corresponds to the frequency  $\frac{44100}{98} = 450Hz$ , which corresponds to a slightly out-of-tune A4 pitch. One can notice from the curvatures: the oboe’s rich spectrum in high frequencies and the lower spectrum of the real sound.

The sequence  $R_i = \{r_i\}_0^{\Lambda_f-1}$  of samples in the real sound of Figure 2 can be taken as basis for a sound  $T_i^f$  in the following way:

$$T_i^f = \{t_i^f\} = \left\{r_{(i \% \Lambda_f)}\right\} \quad (13)$$

The resulting sound has the momentary spectrum of the original waveform. As a consequence of its repetition in an identical form, the spectrum is perfectly harmonic, without noise and with variations typical of the natural phenomenon. This can be observed in Figure 4, that shows the spectrum of the original oboe note and a note with same duration, whose samples consists of the repetition of cycle of Figure 2. Summing up, the natural spectrum exhibits variations in the frequencies of the harmonics, in their intensities and some noise, while the note made from the sampled period has a perfectly harmonic spectrum.

### E. Spectrum at sampled sound

The presence and behavior of these sinusoidal components in the discretized sound have some particularities. Considering a signal  $T_i$  and its corresponding Fourier decomposition  $\mathcal{F}\langle T_i \rangle = C_i = \{c_k\}_0^{\Lambda-1}$ , the recomposition is the sum of the frequency components as time samples<sup>22</sup>:

$$\begin{aligned} t_i &= \frac{1}{\Lambda} \sum_{k=0}^{\Lambda-1} c_k e^{j \frac{2\pi k}{\Lambda} i} \\ &= \frac{1}{\Lambda} \sum_{k=0}^{\Lambda-1} (a_k + j.b_k) [\cos(w_k i) + j.\sin(w_k i)] \end{aligned} \quad (14)$$

where  $c_k = a_k + j.b_k$  defines the amplitude and phase of each frequency:  $w_k = \frac{2\pi k}{\Lambda}$  in radians or  $f_k = w_k \frac{\Lambda}{2\pi} =$

$\frac{f_a}{\Lambda} k$  in Hertz, taking into account the respective limits in  $\pi$  and in  $\frac{f_a}{2}$  given by the Nyquist Theorem. As can be inferred,  $j$  is the complex number, with  $j^2 = -1$ .

For a sound signal, samples  $t_i$  are real and are given by the real part of equation 14:

$$\begin{aligned} t_i &= \frac{1}{\Lambda} \sum_{k=0}^{\Lambda-1} [a_k \cos(w_k i) - b_k \sin(w_k i)] \\ &= \frac{1}{\Lambda} \sum_{k=0}^{\Lambda-1} \sqrt{a_k^2 + b_k^2} \cos \left[ w_k i - \tan^{-1} \left( \frac{b_k}{a_k} \right) \right] \end{aligned} \quad (15)$$

Equation 15 shows how the imaginary term of  $c_k$  adds a phase to the real sinusoid: the terms  $b_k$  enables the phase sweep  $[-\frac{\pi}{2}, +\frac{\pi}{2}]$  given by  $\tan^{-1} \left( \frac{b_k}{a_k} \right)$  which has this image. The terms  $a_k$  specifies the right or left side of the trigonometric circle, which completes the phase domain:  $[-\frac{\pi}{2}, +\frac{\pi}{2}] \cup [\frac{\pi}{2}, \frac{3\pi}{2}] \equiv [2\pi]$ .

Figure 5 shows two samples and its spectral components. In this case, the Fourier decomposition has one unique pair of coefficients  $\{c_k = a_k - j.b_k\}_0^{\Lambda-1=1}$  relative to frequencies  $\{f_k\}_0^1 = \left\{w_k \frac{\Lambda}{2\pi}\right\}_0^1 = \left\{k \frac{f_a}{\Lambda=2}\right\}_0^1 = \left\{0, \frac{f_a}{2} = f_{\max}\right\}$  with energies  $e_k = \frac{(c_k)^2}{\Lambda=2}$ . The role of amplitudes  $a_k$  is clearly observed with  $\frac{a_0}{2}$ , the fixed offset<sup>23</sup> and  $\frac{a_1}{2}$ , oscillation amplitude with frequency given by the relation  $f_k = k \frac{f_a}{\Lambda=2}$ . This case have special relevance. It is necessary at least 2 samples to represent an oscillation and it yields the Nyquist frequency  $f_{\max} = \frac{f_a}{2}$ , which is the maximum frequency in a sound sampled with  $f_a$  samples per second<sup>24</sup>.

All fixed sequences  $T_i$  of only 3 samples also have just 1 frequency, since their first harmonic has 1, 5 samples and exceeds the bottom limit of 2 samples, i.e. the frequency of the harmonic would exceed the Nyquist frequency:  $\frac{2.f_a}{3} > \frac{f_a}{2}$ . The coefficients  $\{c_k\}_0^{\Lambda-1=2}$  are present in 3 frequency components. One is relative to zero frequency ( $c_0$ ), and the other two ( $c_1$  and  $c_2$ ) have the same role for the reconstruction of sinusoid with  $f = f_a/3$ .

$\Lambda$  real samples  $t_i$  result in  $\Lambda$  complex coefficients  $c_k = a_k + j.b_k$ . The coefficients  $c_k$  are equivalent two by two, corresponding to the same frequencies and with same contribution to its reconstruction. They are complex conjugates:  $a_{k1} = a_{k2}$  and  $b_{k1} = -b_{k2}$  and, as a consequence, the modules are equal and phases have opposite signs. Remembering that  $f_k = k \frac{f_a}{\Lambda}$ ,  $k \in \{0, \dots, \lfloor \frac{\Lambda}{2} \rfloor\}$ . When  $k > \frac{\Lambda}{2}$ , the frequency  $f_k$  is mirrored by  $\frac{f_a}{2}$  in this way:  $f_k = \frac{f_a}{2} - (f_k - \frac{f_a}{2}) = f_a - f_k = f_a - k \frac{f_a}{\Lambda} = (\Lambda - k) \frac{f_a}{\Lambda} \Rightarrow f_k \equiv f_{\Lambda-k}, \forall k < \Lambda$ .

The same can be observed with  $w_k = f_k \cdot \frac{2\pi}{f_a}$  and the periodicity  $2\pi$ : it follows that  $w_k = -w_{\Lambda-k}$ ,  $\forall k < \Lambda$ . Given the cosine (an even function) and the inverse tangent (an odd function), the components in  $w_k$  and  $w_{\Lambda-k}$  contribute with coefficients  $c_k$  and  $c_{\Lambda-k}$  in the reconstruction of the real samples.

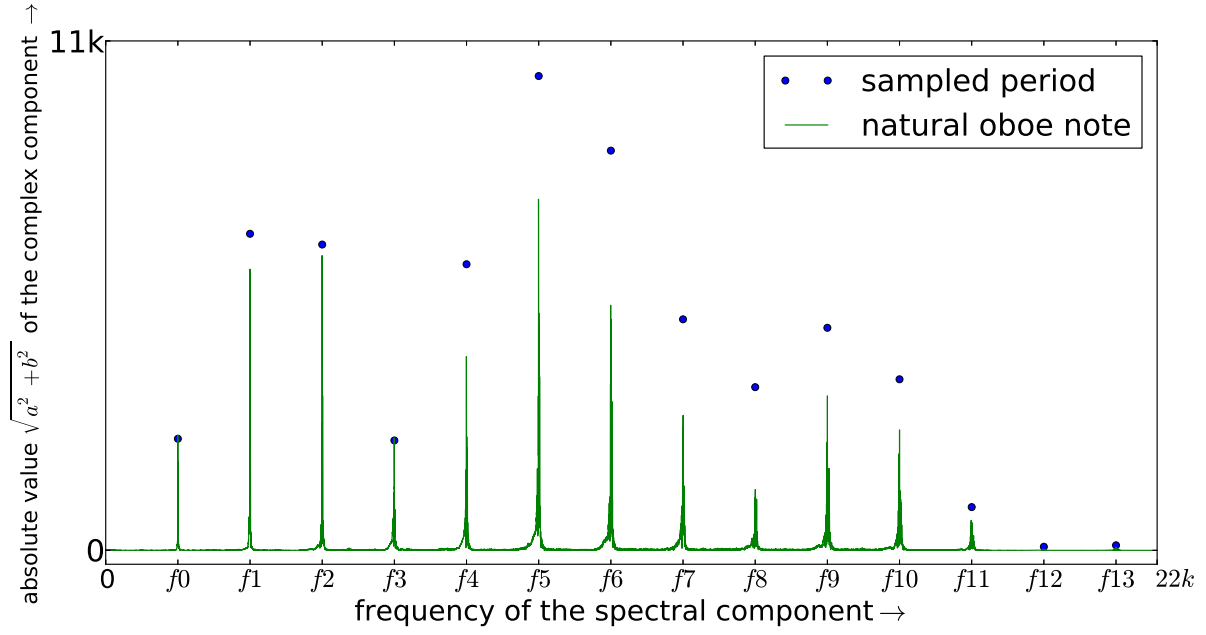


FIG. 4. Spectrum of the sonic waves of a natural oboe note and from a sampled period. The natural sound has fluctuations in the harmonics and in its noise, while the sampled period note has a perfectly harmonic spectrum.

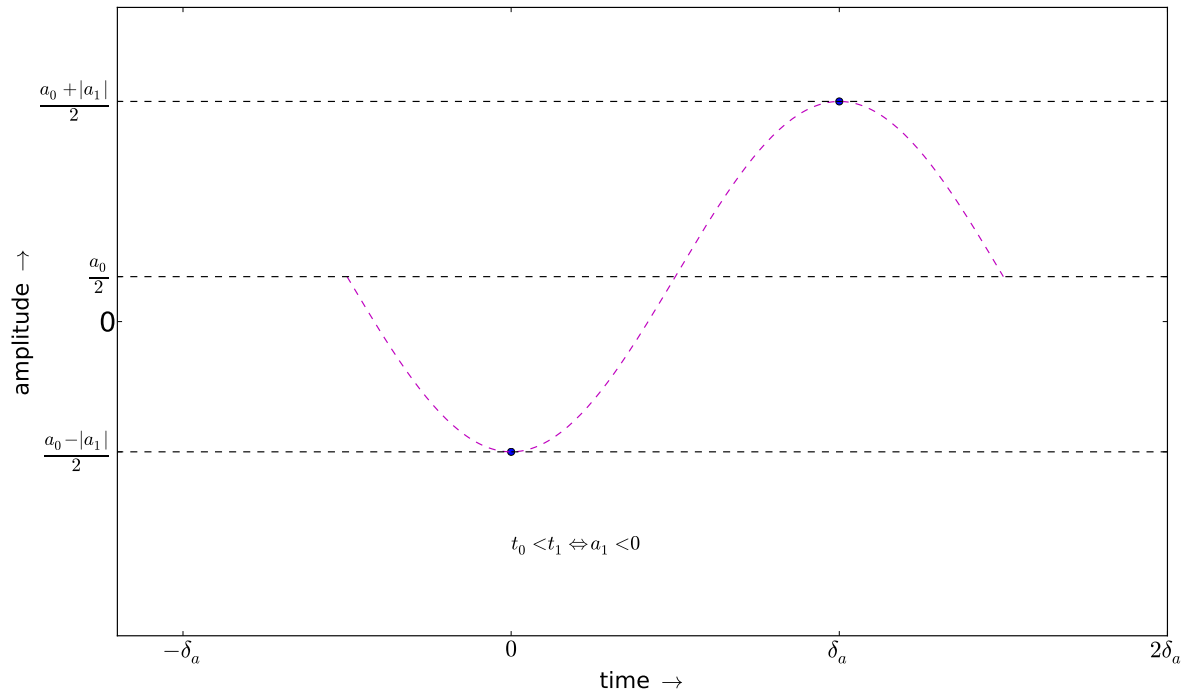


FIG. 5. Oscillation of 2 samples (maximum frequency for any  $f_a$ ). The first coefficient reflects a detachment (*offset* or *bias*) and the second coefficient specifies the oscillation amplitude.

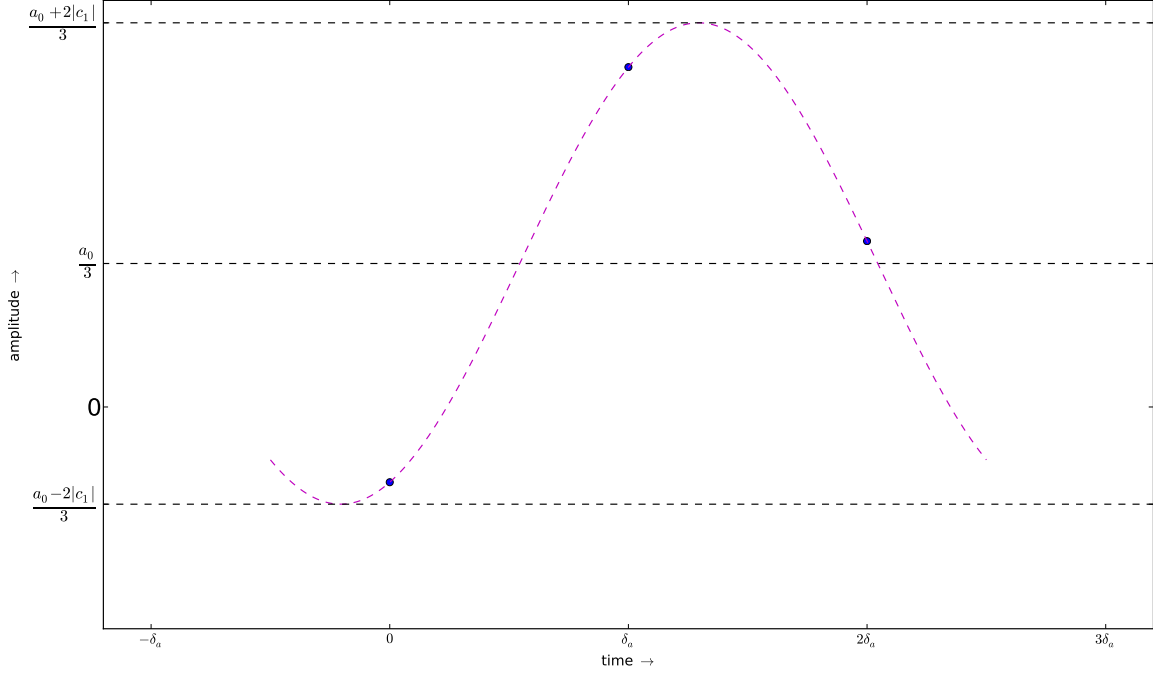


FIG. 6. Three fixed samples presents only one non-null frequency.  $c_1 = c_2^*$  and  $w_1 \equiv w_2$ .

In other words, in a decomposition of  $\Lambda$  samples, the  $\Lambda$  frequency components  $\{c_i\}_0^{\Lambda-1}$  are equivalents in pairs, except for  $f_0$ , and, when  $\Lambda$  is even, for  $f_{\Lambda/2} = f_{\max} = \frac{f_a}{2}$ . Both components are isolated, i.e. there is one and only component in frequency  $f_0$  or  $f_{\Lambda/2}$  (if even  $\Lambda$ ). One can verify this assertion considering  $k = 0$  and  $k = \Lambda/2$ . Uncoiled:  $f_{\Lambda/2} = f_{(\Lambda-\Lambda/2)=\Lambda/2}$  and  $f_0 = f_{(\Lambda-0)=\Lambda} = f_0$ . Furthermore, these two frequencies (zero and Nyquist frequency) does not have phase variation, being their coefficients strictly real. In this way, it is possible to conclude the number  $\tau$  of equivalent coefficient pairs:

$$\tau = \frac{\Lambda - \Lambda \% 2}{2} + \Lambda \% 2 - 1 \quad (16)$$

This discussion makes clear the equivalences 17, 18 and 19:

$$f_k \equiv f_{\Lambda-k}, \quad w_k \equiv -w_{\Lambda-k}, \quad \forall \quad 1 \leq k \leq \tau \quad (17)$$

$T_i \Rightarrow a_k = a_{\Lambda-k}$  and  $b_k = -b_{\Lambda-k}$ , and thus:

$$\sqrt{a_k^2 + b_k^2} = \sqrt{a_{\Lambda-k}^2 + b_{\Lambda-k}^2}, \quad \forall \quad 1 \leq k \leq \tau \quad (18)$$

$$tg^{-1}\left(\frac{b_k}{a_k}\right) = -tg^{-1}\left(\frac{b_{\Lambda-k}}{a_{\Lambda-k}}\right), \quad \forall \quad 1 \leq k \leq \tau \quad (19)$$

with  $k \in \mathbb{N}$ .

To expose the general case for components combination in each sample  $t_i$ , one can gather relations in equation 15 for the real signal reconstruction, relations of modules 18 and phases equivalences 19, the number of paired coefficients 16, and equivalence of paired frequencies 17:

$$t_i = \frac{a_0}{\Lambda} + \frac{2}{\Lambda} \sum_{k=1}^{\tau} \sqrt{a_k^2 + b_k^2} \cos \left[ w_k i - tg^{-1} \left( \frac{b_k}{a_k} \right) \right] + \frac{a_{\Lambda/2}}{\Lambda} \cdot (1 - \Lambda \% 2) \quad (20)$$

with  $a_{\Lambda/2} = 0$  if  $\Lambda$  odd.

With 4 samples it is possible to represent 1 or 2 frequencies in any proportions (i.e. with independence). Figure 7 depicts the basic waveforms with 4 samples and its two (possible) components. The individual contributions sum to the original waveform and a brief inspection reveals the major curvatures resulting from the higher frequency, while the fixed offset is captured in the zero frequency component.

Figure 8 shows the harmonics for the basic waveforms of equations 9, 10, 11 and 12 for the case of 4 samples. There is only 1 sinusoid for each waveform, with the exception of the sawtooth, which has even harmonics.

Figure 9 presents the sinusoidal decomposition for 6 samples and figure 10 presents the decomposition of the



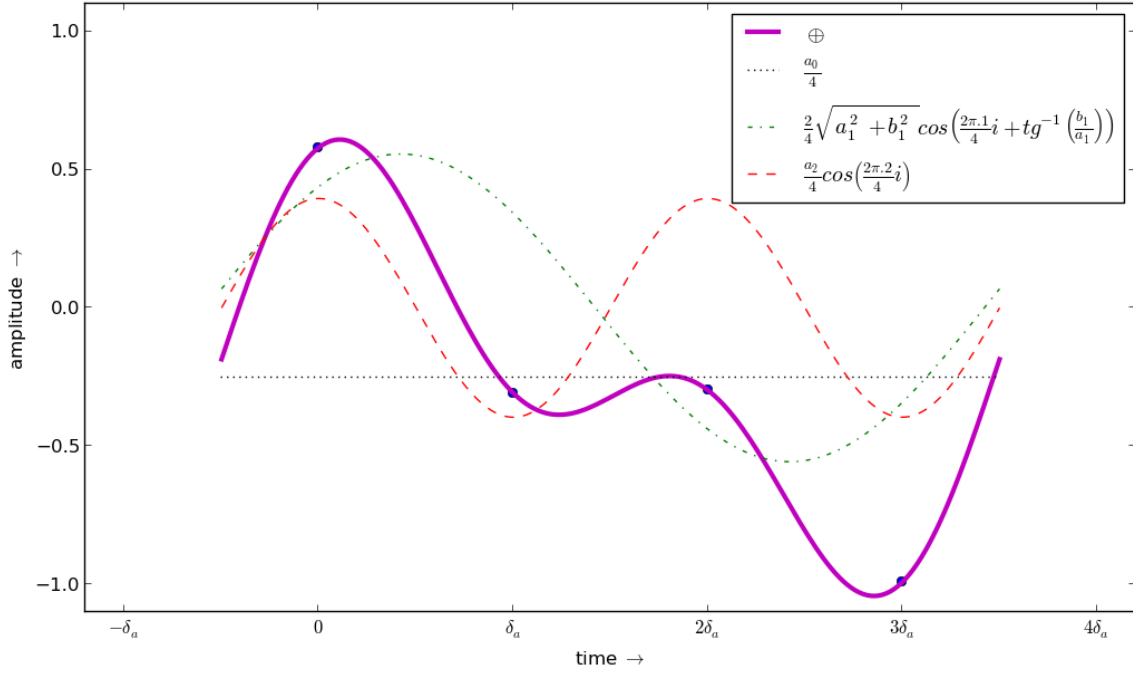


FIG. 7. Frequency components for 4 samples.

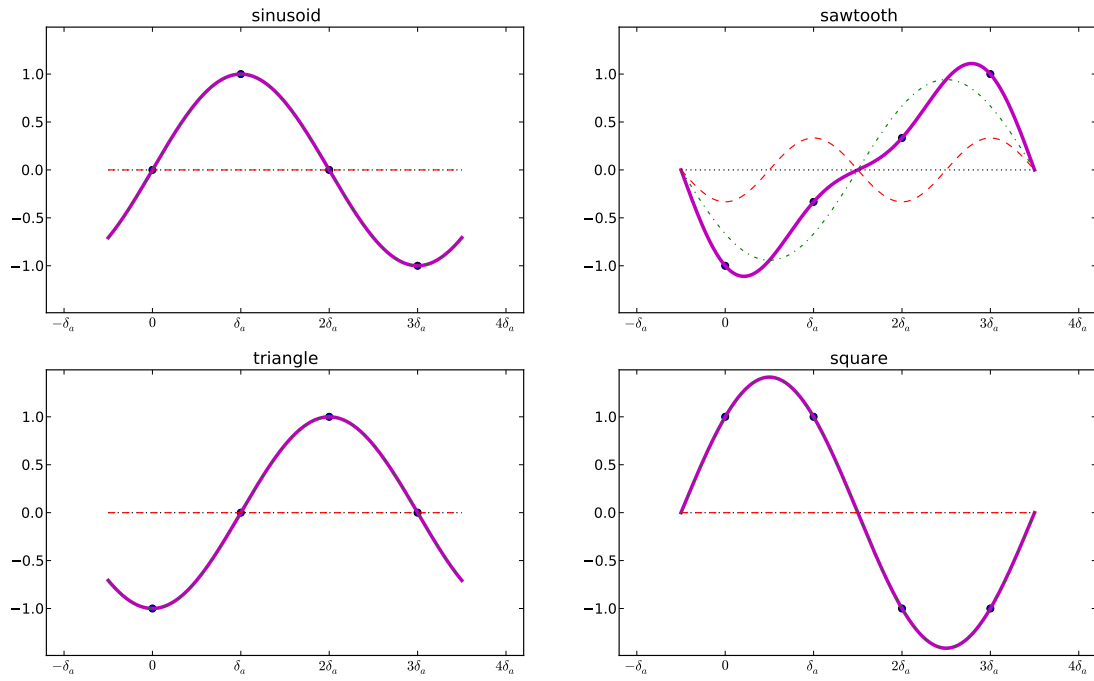


FIG. 8. Basic wave forms with 4 samples.

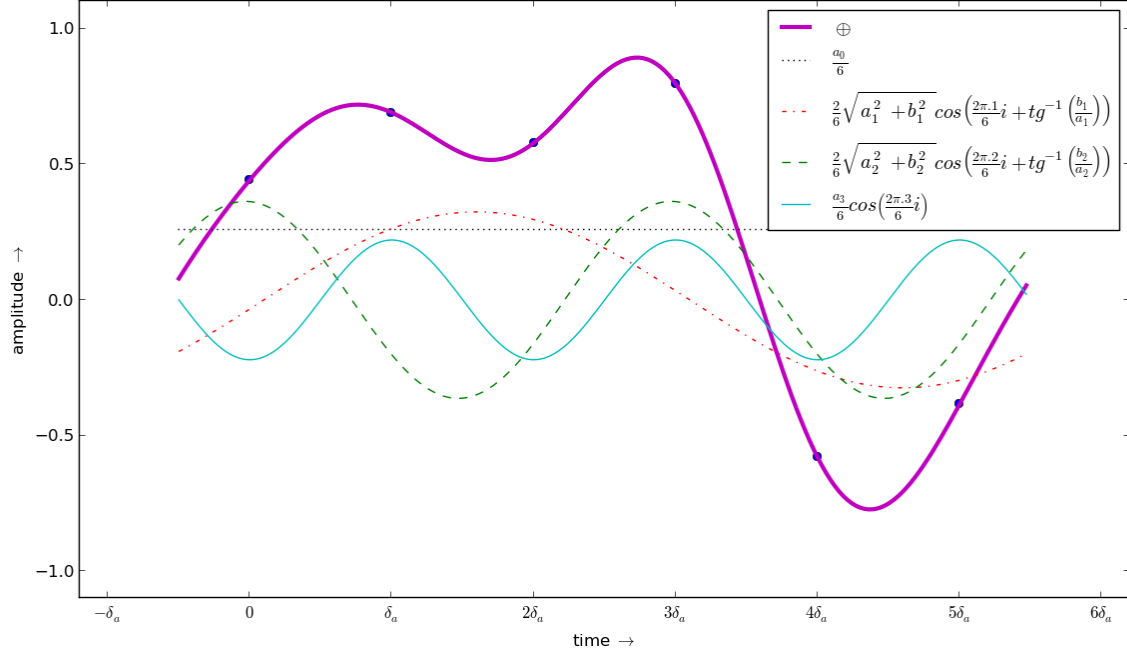


FIG. 9. Frequency components for 6 samples: 4 sinusoids, one of them is the *bias* with zero frequency.

basic wave forms. In this case, the waveforms have spectra with fundamental differences: square and triangular have the same components but with different proportions, while the sawtooth have an extra component.

#### F. The basic note

Be  $f$  such that it divides  $f_a$ <sup>25</sup>. A sequence  $T_i$  of sonic samples separated by  $\delta_a = 1/f_a$  describes a musical note with a frequency of  $f$  Hertz and  $\Delta$  seconds of duration if, and only if, it has the periodicity  $\lambda_f = f_a/f$  and size  $\Lambda = \lfloor f_a \cdot \Delta \rfloor$ :

$$T_i^{f, \Delta} = \{t_i \%_{\lambda_f}\}_{0}^{\Lambda-1} = \left\{t_i^f \%_{\left(\frac{f_a}{f}\right)}\right\}_0^{\Lambda-1} \quad (21)$$

The note by itself does not specify a timbre. Nevertheless, it is necessary to choose a waveform for the samples  $t_i$  to have a value. An unique period from the basic waveforms can be used to specify the note, where  $\lambda_f = \frac{f_a}{f}$  is the number of samples at the period. Here,  $L_i^{f, \delta_f}$  is the sequence that describes a period of the waveform  $L_i^f \in \{S_i^f, Q_i^f, T_i^f, D_i^f, R_i^f\}$  with duration  $\delta_f = 1/f$  (as given by equations 9, 10, 11 and 12) and  $R_i^f$  is a sampled real waveform:

$$L_i^{f, \delta_f} = \left\{l_i^f\right\}_0^{\delta_f \cdot f_a - 1} = \left\{l_i^f\right\}_0^{\lambda_f - 1} \quad (22)$$

Therefore, the sequence  $T_i$  will consist in a note of duration  $\Delta$  and frequency  $f$  if:

$$T_i^{f, \Delta} = \left\{t_i^f\right\}_0^{\lfloor f_a \cdot \Delta \rfloor - 1} = \left\{l_i^f \%_{\left(\frac{f_a}{f}\right)}\right\}_0^{\Lambda-1} \quad (23)$$

#### G. Spatial localization and spatialization

Although it is not one of its four basic properties, a musical note always has a spatial localization: the note source position at the ordinary physical space. The reverberation the environment in which sound occurs is the matter of *spatialization*. Both fields, spatialization and spatial localization, are widely valued by audiophiles and music industry<sup>26</sup>.

##### 1. Spatial localization

It is understood that the perception of sound localization occurs in our nervous system by three informations: the delay of incoming sound between both ears, the difference of sound intensity at each ear and the filtering realized by the human body, including its chest, head and ears<sup>1,27,28</sup>.

Considering only the direct incidences in each ear, the equations are quite simple. An object placed at  $(x, y)$ , as in figure 11, is distant of each ear by:

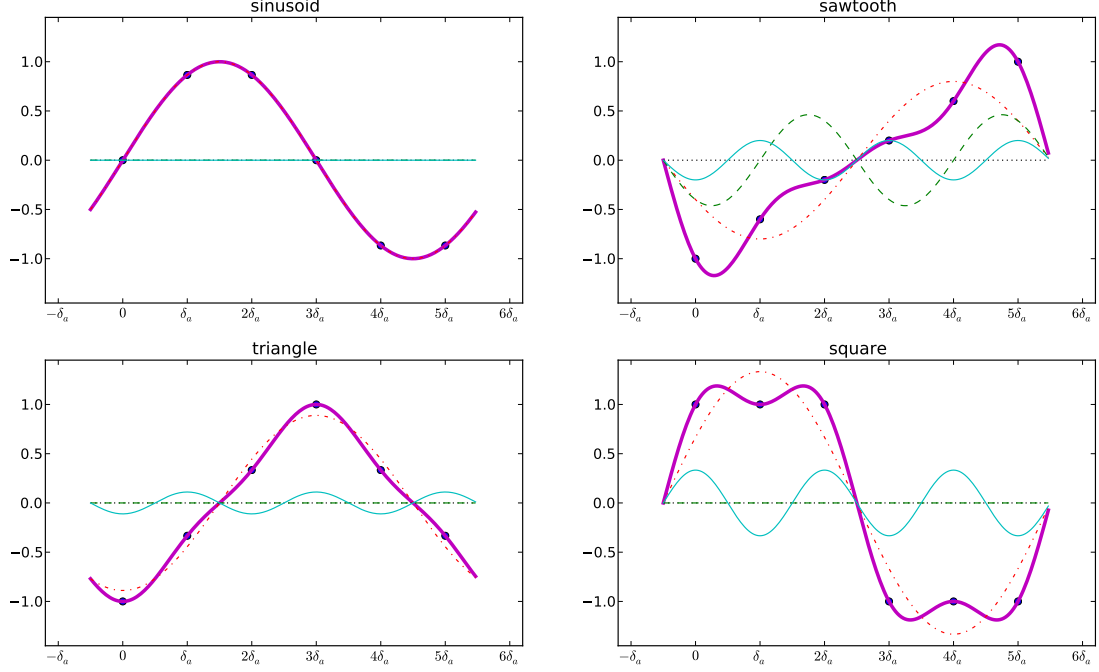


FIG. 10. Basic waveforms with 6 samples: triangular and square waveforms have odd harmonics, with different proportions and phases; the sawtooth has even harmonics.

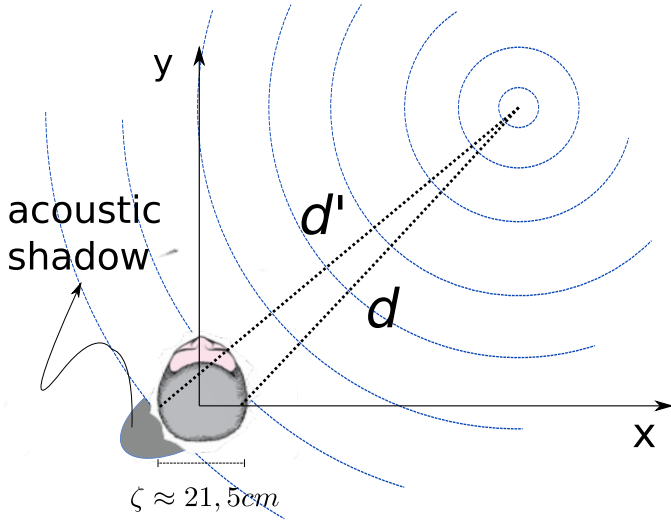


FIG. 11. Detection of sound source localization: schema used to calculate Interaural Time Difference (ITD) and Interaural Intensity Difference (IID).

$$d = \sqrt{\left(x - \frac{\zeta}{2}\right)^2 + y^2} \quad (24)$$

$$d' = \sqrt{\left(x + \frac{\zeta}{2}\right)^2 + y^2}$$

Where  $\zeta$  is the distance between ears  $\zeta$ , known to be  $\zeta \approx 21,5cm$  in an adult human. Immediate calculations result in the ITD:

$$ITD = \frac{d' - d}{v_{sound \text{ at air}} \approx 343.2} \text{ seconds} \quad (25)$$

and in the Interaural Intensity Difference:

$$IID = 20 \log_{10} \left( \frac{d}{d'} \right) \text{ decibels} \quad (26)$$

which, converted to amplitude, yields  $IID_a = \frac{d}{d'}$ . The  $IID_a$  can be used as a multiplicative constant to the right channel of a stereo sound signal:  $\{t'_i\}_0^{\Lambda-1} = \{IID_a \cdot t_i\}_0^{\Lambda-1}$ , where  $\{t_i\}$  are samples of the wave incident in the left ear. It is possible to use IID together with ITD as a time advance for the right channel. It is a

crucial vestige to localization perception of bass sounds and percussive sonorities<sup>28</sup>. With  $\Lambda_{ITD} = \lfloor ITD \cdot f_a \rfloor$ :

$$\begin{aligned} \Lambda_{ITD} &= \left\lfloor \frac{d' - d}{343, 2} f_a \right\rfloor \\ IID_a &= \frac{d}{d'} \\ \left\{ t'_{(i+\Lambda_{ITD})} \right\}_{\Lambda_{ITD}}^{\Lambda+\Lambda_{ITD}-1} &= \{ IID_a \cdot t_i \}_0^{\Lambda-1} \\ \{ t'_i \}_0^{\Lambda_{ITD}-1} &= 0 \end{aligned} \quad (27)$$

with  $t_i$  as the right channel and  $t'_i$  the left channel. If  $\Lambda_{ITD} < 0$ , it is only needed to change  $t_i$  by  $t'_i$  and to use  $\Lambda'_{ITD} = |\Lambda_{ITD}|$  and  $IID'_a = 1/IID_a$ .

Although simple until here, spatial localization depends considerably of other cues. By using only ITD and IID it is possible to specify solely the horizontal angle (azimuthal)  $\theta$  given by:

$$\theta = \tan^{-1} \left( \frac{y}{x} \right) \quad (28)$$

with  $x, y$  as presented in Figure 11. Henceforth, there are problems when  $\theta$  fall upon the so called "cone of confusion": the same pair of ITD and IID results in a large number of points inside the cone. On those points the inference of the azimuthal angle depends specially of the attenuative filtering for high frequencies, since the head interferes much more in the treble than in bass waves<sup>27,28</sup>. Also relevant to the hearing of lateral sources is that low frequencies diffracts and the wave arrive to the opposite ear with a delay of  $\approx 0,7ms$ .<sup>26</sup>

Figure 11 depicts the acoustic shadow of the cranium, an important phenomenon to perception of source azimuthal angle in the cone of confusion. The cone itself is not shown in figure 11 because it is not exactly a cone and its precise dimensions were not encountered in the literature. Given the filtering and diffraction dependent of the sound spectrum, it is hard, if not impossible, to correctly draw the confusion cone. Even so, the cone of confusion can be understood as a cone with its top placed in the middle of head and growing out in the direction of each ear<sup>27</sup>.

On the other hand, the complete localization, including height and distance of sound source, is given by the Head Related Transfer Function (HRTF)<sup>27</sup>. There are well known open databases of HRTF, such as CIPIC, and it is possible to apply those transfer functions in a sonic signal by convolution (see equation 42)<sup>29</sup>. Each human body has its filtering and there are techniques to generate HRTFs to be universally used<sup>30</sup>.

## 2. Spatialization

Spatialization results from sound reflections and absorptions by room/environment surface where the note

is played. The sound propagates through the air with a speed of  $\approx 343,2m/s$  and can be emitted from a source with any directionality pattern. When a sound pulse encounters a surface there is reflection. In this reflection occurs: 1) the inversion of propagation speed component that is normal to the surface; 2) energy absorption, specially in trebles. The sonic waves propagate until they reaches inaudible levels (and even further). As a sonic front reaches the human ear, it can be described as the original sound, with the last reflection point as the source, and the absorption filters of each surface it has reached. It is possible to simulate reverberations that are impossible in real systems. For example, it is possible to use asymmetric reflections with relation to the axis perpendicular to the surface, or to increase specific frequency bands (known as 'resonances'), both characteristics are not found in real systems.

There are some reverberation models that are less related to each independent reflection, exploring valuable information to the auditory system. In fact, reverberation can be modeled with a set of 2 temporal and spectral sections:

- First period: 'first reflections' are more intense and scattered.
- Second period: 'late reverberation' is practically a dense succession of indistinct delays with exponential decay and statistical occurrences.
- First band: the bass has some resonance bandwidths relatively spaced.
- Second band: mid and treble have progressive decay and smooth statistical fluctuations.

Smith III points that reasonable concert rooms have total reverberation time of  $\approx 1.9$  seconds, and that the period of first reflections is around 0.1 seconds. These values suggest that, in the given conditions, there are perceived wave pulses which propagates for 652.08 meters long ( $83.79k$  samples in  $f_a = 44.1kHz$ ) before reaching the ear. In addition, sound reflections made after propagation for 34.32 meters long ( $4.41k$  samples in  $f_a = 44.1kHz$ ) have incidences less distinct by hearing. These first reflections are particularly important to spacial sensation. The first incidence is the direct sound, described by ITD and IID of equations 25 and 26. Admitting that each one of the first reflections, before reaching the ear, will propagate at least  $3-30m$ , dependent of room dimension, the separation between the first reflections is  $8-90ms$  ( $\approx 350-4000$  samples in  $f_a = 44.1kHz$ ). It is possible to experimentally verify that the number of reflections increases with squared proportion  $\approx k \cdot n^2$ . A discussion about the use of convolutions and filtering to favor implementation of these phenomena is provided in subsection III F, specially at the paragraphs about reverberation.

## H. Musical use

Once defined the basic note, it is convenient to build musical structures with sequences based on these particles. The sum of elements with same index of  $N$  sequences  $T_{k,i} = \{t_{k,i}\}_{k=0}^{N-1}$  with same size  $\Lambda$  results in the overlapped spectral contents of each sequence, in a process called mixing:

$$\{t_i\}_0^{\Lambda-1} = \left\{ \sum_{k=0}^{N-1} t_{k,i} \right\}_0^{\Lambda-1} \quad (29)$$

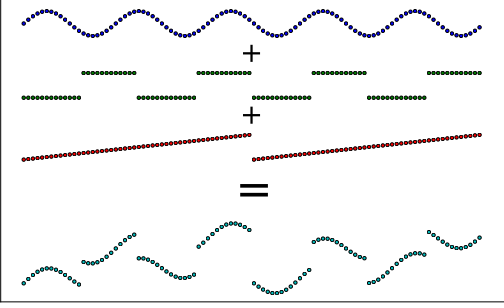


FIG. 12. Mixing of three sound sequences. The amplitudes are directly overlapped.

Figure 12 illustrates this overlapping process of discretized sound waves, each with 100 samples. It is possible to conclude that, if  $f_a = 44.1kHz$ , the frequencies of the sawtooth, square and sine wave are, respectively:  $\frac{f_a}{100/2} = 882Hz$ ,  $\frac{f_a}{100/4} = 1764Hz$  and  $\frac{f_a}{100/5} = 2205Hz$ . The samples duration are very short  $\frac{f_a=44.1kHz}{100} \approx 2$  milliseconds. One can complete the sequence with zeros to sum (mix) sequences with different sizes.

The mixed notes are generally separated by the ear according to the physical laws of resonance and by the nervous system<sup>1</sup>. This musical notes mixing process results in musical harmony whose intervals between frequencies and chords of simultaneous notes guide subjective and abstract aspects of music and its appreciation<sup>31</sup>, which is tackled in section IV.

Sequences can be concatenated in time. If the sequences  $\{t_{k,i}\}_0^{\Lambda_k-1}$  of size  $\Lambda_k$  represent  $k$  musical notes, their concatenation in a unique sequence  $T_i$  is a simple melodic sequence, or melody of its own:

$$\{t_i\}_0^{\sum \Delta_k-1} = \{t_{l,i}\}_0^{\sum \Delta_k-1}, \quad (30)$$

$l \text{ smallest integer} : \Lambda_l > i - \sum_{j=0}^{l-1} \Lambda_j$

This mechanism is demonstrated in Figure 13 with same sequences of Figure 12. Although the sequences are short for the usual sample rates, it is easy to observe the concatenation of sound sequences. Besides that, each note has a duration larger than 100ms if  $f_a < 1kHz$ .

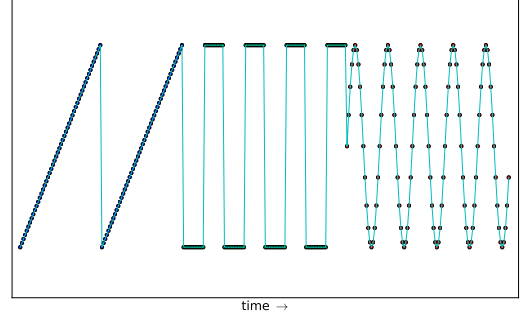


FIG. 13. Concatenation of three sound sequences by temporal overlap of its samples.

The musical piece *reduced-fi* explores the temporal juxtaposition of notes, resulting in a homophobic piece. The vertical principle is demonstrated at the *sounic pictures*, static sounds with peculiar spectrum. Both pieces were written in Python and are available as part of the *MASSA toolbox*.<sup>32</sup>

With the basic musical digital note carefully described, the next section develops the temporal evolution of its contents as in *glissandi* and volume envelopes. Filtering of spectral components and noise generation complements the musical note as a self contained unity. Section IV is dedicated to the organization of these notes as music by using metrics and trajectories, with regards to traditional music theory.

## III. VARIATION IN THE BASIC MUSICAL NOTE

The basic digital music note defined in section II has following parameters: duration, pitch, intensity (volume) and timbre. This is a useful and paradigmatic model, but does not exhaust all the aspects of a musical note.

First of all, characteristics of the note modifie along the note itself<sup>19</sup>. For example, a 3 seconds piano note has intensity with abrupt rise at beginning and progressive decay, has spectrum variations with harmonics decaying some others that emerge along time. These variations are not mandatory, but sound synthesis guidance for musical uses, as it reflects how sounds appear in nature. This is so considered to be true, that there is a rule of thumb: to make a sound that incites interest by itself, do internal variations on it<sup>1</sup>. To explore all the ways in which variations occur within a note is out of the scope of any work, given the sensibility of the human ear and the complexity of human sound cognition. In the following, primary resources are presented to produce varia-

tions in the basic note. It is worthwhile to recall that all the relations in this and other sections are implemented in Python and published as the public domain MASSA toolbox. The musical pieces *Transita para metro*, *Vibra e treme*, *Tremolos*, *vibratos e a frecuencia*, *Trenzinho de caipiras impulsivos*, *Ruidosa faixa*, *Bela rugosi*, *Chorus infantil*, *ADa e SaRa* were made to validate and illustrate concepts of this section. The code that synthesizes these pieces are also part of the toolbox<sup>32</sup>.

### A. Lookup Table

The *Lookup Table* (or simply LUT), is an array for indexed operations which substitutes continuous and repetitive calculation. It is used to reduce computational complexity and to use functions without direct calculation, as data sampled from nature. In music its usage transcends these applications, as it simplifies many operations and makes possible the use a single wave period to synthesize sounds in the whole audible spectrum, with any waveform.

Be  $\tilde{\Lambda}$  the wave period in samples and  $\tilde{L}_i = \{\tilde{l}_i\}_0^{\tilde{\Lambda}-1}$  the samples  $\tilde{l}_i$  (refer to Equation 22), a sequence  $T_i^{f,\Delta}$  with samples of a sound with frequency  $f$  and duration  $\Delta$  can be obtained by means of  $\tilde{L}_i$ :

$$T_i^{f,\Delta} = \left\{ t_i^f \right\}_0^{\lfloor f_a \cdot \Delta \rfloor - 1} = \left\{ \tilde{l}_{\gamma_i \% \tilde{\Lambda}} \right\}_0^{\Delta-1}, \quad (31)$$

where  $\gamma_i = \left\lfloor i \cdot f \frac{\tilde{\Lambda}}{f_a} \right\rfloor$

In other words, with the right LUT indexes ( $\gamma_i \% \tilde{\Lambda}$ ) it is possible to synthesize sounds at any frequency. Figure 14 illustrates the calculation of  $\{t_i\}$  sample from  $\{\tilde{l}_i\}$  for  $f = 200\text{Hz}$ ,  $\tilde{\Lambda} = 128$  and adopting the sample rate of  $f_s = 44.1\text{kHz}$ . Though this is not a practical configuration (as assigned below), it makes possible a graphical visualization of the procedure.

The calculation of the integer  $\gamma_i$  introduces noise which decreases as  $\tilde{\Lambda}$  increases. In order to use this calculation in sound synthesis, with  $f_s = 44.1\text{kHz}$ , the standard guidelines suggests the use of  $\tilde{\Lambda} = 1024$  samples, since it does not produce relevant noise on the audible spectrum. The rounding or interpolation method are not decisive in this process<sup>33</sup>.

The expression defining the variable  $\gamma_i$  can be understood as  $f_s$  being added to  $i$  at each second. If  $i$  is divided by the sample frequency,  $\frac{i}{f_a}$  is incremented in 1 at each second. Multiplied by the period, it results in  $i \frac{\tilde{\Lambda}}{f_a}$ , which covers the period in each second. Finally, with frequency  $f$  it results in  $i \cdot f \frac{\tilde{\Lambda}}{f_a}$  which completes  $f$  periods  $\tilde{\Lambda}$  in 1 second, i.e. the resulting sequence presents the fundamental frequency  $f$ .

There are important considerations here: it is possible to use practically any frequency  $f$ . Limits exist only in low frequencies when the size of the table  $\tilde{\Lambda}$  is not sufficient for the sample rate  $f_a$ . The lookup procedure is virtually costless and replaces calculations by simple indexed searches (what is generally understood as an optimization process). Unless otherwise stated, this procedure will be used along all the text for every applicable case. LUTs are broadly used in computational implementations for music. A classical usage of LUTs is known as *Wavetable Synthesis*, which consists of many LUTs used together to generate a quasi-periodic music note<sup>13,34</sup>.

### B. Incremental Variations of Frequency and Intensity

As stated by the Weber and Fechner<sup>35</sup> law, the human perception has a logarithmic relation with the stimulus. In other words, the exponential progression of a stimulus is perceived as linear. For didactic reasons, and given its use in AM and FM synthesis (subsection III E), linear variation is discussed first.

Consider a note with duration  $\Delta = \frac{\Lambda}{f_a}$ , in which the frequency  $f = f_i$  varies linearly from  $f_0$  to  $f_{\Lambda-1}$ . Thus:

$$F_i = \{f_i\}_0^{\Lambda-1} = \left\{ f_0 + (f_{\Lambda-1} - f_0) \frac{i}{\Lambda - 1} \right\}_0^{\Lambda-1} \quad (32)$$

$$\Delta_{\gamma_i} = f_i \frac{\tilde{\Lambda}}{f_a} \Rightarrow \gamma_i = \left\lfloor \sum_{j=0}^i f_j \frac{\tilde{\Lambda}}{f_a} \right\rfloor \quad (33)$$

$$\gamma_i = \left\lfloor \sum_{j=0}^i \frac{\tilde{\Lambda}}{f_a} \left[ f_0 + (f_{\Lambda-1} - f_0) \frac{j}{\Lambda - 1} \right] \right\rfloor$$

$$\left\{ t_i^{\overline{f_0, f_{\Lambda-1}}} \right\}_0^{\Lambda-1} = \left\{ \tilde{l}_{\gamma_i \% \tilde{\Lambda}} \right\}_0^{\Lambda-1} \quad (34)$$

where  $\Delta_{\gamma_i} = f_i \frac{\tilde{\Lambda}}{f_a}$  is the LUT increment between two samples given the sound frequency of the first sample. Therefore, it is handy to calculate the elements  $t_i^{\overline{f_0, f_{\Lambda-1}}}$  by means of the period  $\left\{ \tilde{l}_i \right\}_0^{\Lambda-1}$ .

The equations 32, 33 and 34 are related with the linear progression of the frequency. As stated above, the frequency progression *perceived* as linear follows an exponential progression, i.e. a geometric progression of frequency is perceived as an arithmetic progression of pitch. It is possible to write:  $f_i = f_0 \cdot 2^{\frac{i}{\Lambda-1} n_8}$  where  $n_8 = \log_2 \frac{f_{\Lambda-1}}{f_0}$  is the number of octaves between  $f_0$  and  $f_{\Lambda-1}$ .

Therefore,  $f_i = f_0 \cdot 2^{\frac{i}{\Lambda-1} \log_2 \frac{f_{\Lambda-1}}{f_0}} = f_0 \cdot 2^{\log_2 \left( \frac{f_{\Lambda-1}}{f_0} \right) \frac{i}{\Lambda-1}} = f_0 \left( \frac{f_{\Lambda-1}}{f_0} \right)^{\frac{i}{\Lambda-1}}$ . Accordingly, the equations for linear pitch transition are:

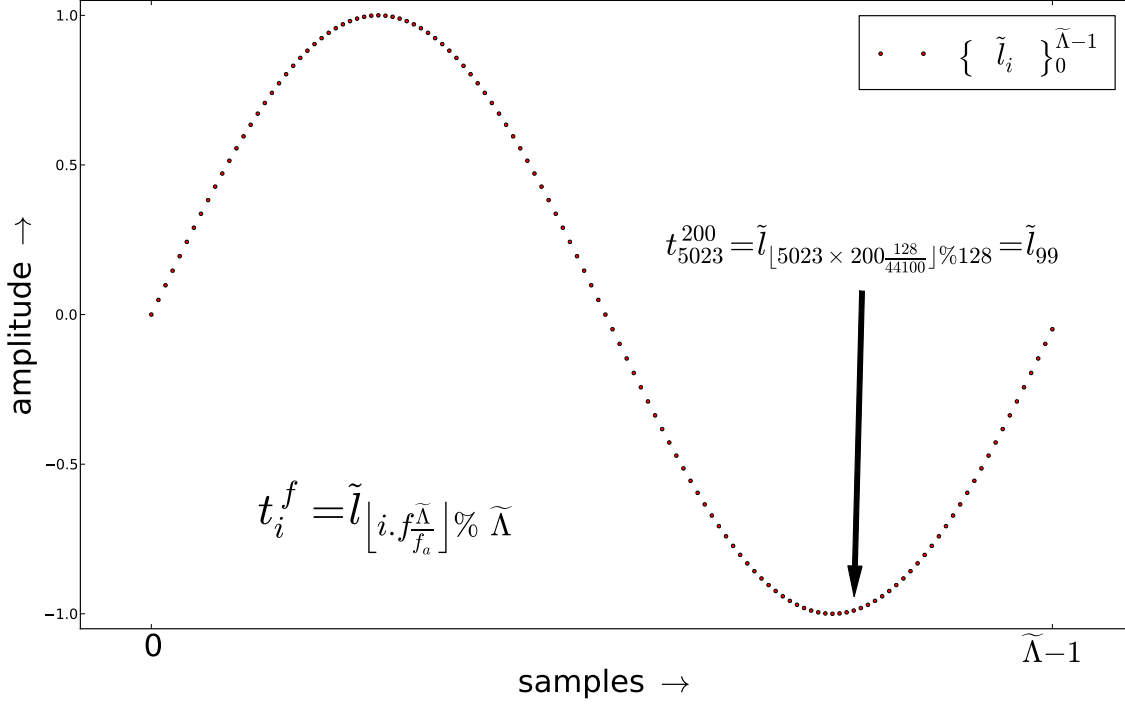


FIG. 14. Search in the *lookup table* to synthesize sounds in different frequencies using an unique waveform with high resolution.

$$F_i = \{f_i\}_0^{\Lambda-1} = \left\{ f_0 \left( \frac{f_{\Lambda-1}}{f_0} \right)^{\frac{i}{\Lambda-1}} \right\}_0^{\Lambda-1} \quad (35)$$

$$\Delta_{\gamma_i} = f_i \frac{\tilde{\Lambda}}{f_a} \Rightarrow \gamma_i = \left\lfloor \sum_{j=0}^i f_j \frac{\tilde{\Lambda}}{f_a} \right\rfloor \quad (36)$$

$$\gamma_i = \left\lfloor \sum_{j=0}^i f_0 \frac{\tilde{\Lambda}}{f_a} \left( \frac{f_{\Lambda-1}}{f_0} \right)^{\frac{j}{\Lambda-1}} \right\rfloor$$

$$\left\{ \overline{t_{i, f_0, f_{\Lambda-1}}} \right\}_0^{\Lambda-1} = \left\{ \tilde{l}_{\gamma_i \% \tilde{\Lambda}} \right\}_0^{\Lambda-1} \quad (37)$$

The term  $\frac{i}{\Lambda-1}$  converts the interval  $[0, 1]$  and it is possible to power it to a value in such a way that the beginning of the transition will be smoother or steeper. This procedure is useful for energy variations with the purpose of changing the volume<sup>36</sup>. It is sufficient to multiply the original sequence by the sequence  $a_{\Lambda-1}^{\left(\frac{i}{\Lambda-1}\right)^\alpha}$ , where  $\alpha$  is the given coefficient and  $a_{\Lambda-1}$  is a fraction of the original amplitude, which is the value to be reached at the end of transition.

Thus, for amplitude variations:

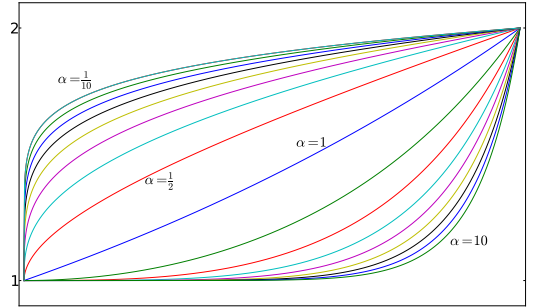


FIG. 15. Intensity transitions to different values of  $\alpha$  (see equations 38 and 39).

$$\{a_i\}_0^{\Lambda-1} = \left\{ a_0 \left( \frac{a_{\Lambda-1}}{a_0} \right)^{\left(\frac{i}{\Lambda-1}\right)^\alpha} \right\}_0^{\Lambda-1} = \left\{ (a_{\Lambda-1})^{\left(\frac{i}{\Lambda-1}\right)^\alpha} \right\}_0^{\Lambda-1} \quad (\text{with } a_0 = 1) \quad (38)$$

and

$$T'_i = T_i \odot A_i = \{t_i \cdot a_i\}_0^{\Lambda-1} = \left\{ t_i \cdot (a_{\Lambda-1})^{\left(\frac{i}{\Lambda-1}\right)^\alpha} \right\}_0^{\Lambda-1} \quad (39)$$

It is often convenient to have  $a_0 = 1$  in order to start a new sequence with the original amplitude and then progressively change it. If  $\alpha = 1$ , the amplitude variation follows the geometric progression whose defines the linear variation of volume. Figure 15 depicts transitions between values 1 and 2 and for different values of  $\alpha$ , a gain of  $\approx 6dB$  as given by equation 4.

Special attention should be dedicated while considering  $a = 0$ . In equation 38,  $a_0 = 0$  results in a division by zero and if  $a_{\Lambda-1} = 0$ , there will be multiplication by zero. Both cases make the procedure useless, once no number different from zero can be represented as a ratio in relation of zero. It is possible to solve this dilemma choosing a number that is small enough like  $-80dB \Rightarrow a = 10^{\frac{-80}{20}} = 10^{-4}$  as the minimum volume for a *fade in* ( $a_0 = 10^{-4}$ ) or for a *fade out* ( $a_{\Lambda-1} = 10^{-4}$ ). A linear fade can be used to reach zero amplitude. Another common solution is the use of the quartic polynomial term  $x^4$ , as it reaches zero without these difficulties and gets reasonably close to the curve with  $\alpha = 1$  as it withdraws from zero<sup>13</sup>.

For linear amplification – but not linear perception – it is sufficient to use an appropriate sequence  $\{a_i\}$ :

$$a_i = a_0 + (a_{\Lambda-1} - a_0) \frac{i}{\Lambda - 1} \quad (40)$$

Here the conversion between decibels and amplitude is convenient. In this way, the equations 7 and 39 specify a transition of  $V_{dB}$  decibels:

$$T'_i = \left\{ t_i 10^{\frac{V_{dB}}{20} \left( \frac{i}{\Lambda-1} \right)^\alpha} \right\}_0^{\Lambda-1} \quad (41)$$

for the general case of amplitude variations following a geometric progression. The greater the value of  $\alpha$ , the smoother the sound introduction and more intense its end.  $\alpha > 1$  results in volume transitions commonly called *slow fade*, while  $\alpha < 1$  results in *fast fade*<sup>37</sup>.

The linear transitions will be used for AM and FM synthesis, while logarithmic transitions are proper tremolos and vibratos, as developed in subsection III E. A non-oscillatory exploration of these variations is in the music piece *Transita para metro*, which code is online as part of the MASSA toolbox<sup>32</sup>.

### C. Application of Digital Filters

This subsection is limited to a description of sequences processing, by convolution and differential equations, and immediate applications, as its complexity escapes the scope of this study<sup>38</sup>. Filter applications can be part of the synthesis process or made subsequently as part of processes commonly called “sound treatment”.

### 1. Convolution and finite impulse response (FIR) filters

Filters applied by means of convolution are known by the acronym FIR (Finite Impulse Response) and are characterized by having a finite sample representation. This sample representation is called ‘impulse response’  $\{h_i\}$ . FIR filters are applied in the time domain of digital sound by means of convolution with the respective impulse response of the filter<sup>39</sup>. For the purposes of this work, the convolution is defined as:

$$\begin{aligned} \{t'_i\}_0^{\Lambda_t + \Lambda_h - 2} &= \{ (T_j * H_j)_i \}_0^{\Lambda_t' - 1} \\ &= \left\{ \sum_{j=0}^{\min(\Lambda_h - 1, i)} h_j t_{i-j} \right\}_0^{\Lambda_t' - 1} \\ &= \left\{ \sum_{j=\max(i+1-\Lambda_h, 0)}^i t_j h_{i-j} \right\}_0^{\Lambda_t' - 1} \end{aligned} \quad (42)$$

where  $t_i = 0$  for the samples not given. In other words, the sound  $\{t'_i\}$ , resulting from the convolution of  $\{t_i\}$ , with the impulse response  $\{h_i\}$ , has each  $i$ -th sample  $t_i$  overwritten by the sum of its last  $\Lambda_h$  samples  $\{t_{i-j}\}_{j=0}^{\Lambda_h-1}$  multiplied one-by-one by samples of the impulse response  $\{h_i\}_0^{\Lambda_h-1}$ . This procedure is illustrated in Figure 16, where the impulse response  $\{h_i\}$  is in its retrograde form, and  $t'_{12}$  and  $t'_{32}$  are two calculated samples using the convolution given by  $(T_j * H_j)_i = t'_i$ . The final signal always has the length of  $\Lambda_t + \Lambda_h - 1 = \Lambda_t'$ .

With this procedure it is possible to apply reverberators, equalizers, *delays*, to name a few of a variety of other filters for sound processing and to obtain musical/artistic effects.

The impulse response can be provided by physical measures or by pure synthesis. An impulse response for a reverberation application, for example, can be obtained by sound recording of the environment when someone triggers a snap which resembles an impulse, or obtained by a sinusoidal sweep whose Fourier transform approximates its frequency response. Both are impulse responses which, properly convoluted with the sound sequence, result in the own sound with a reverberation that resembles the original environment where the measure was made<sup>13</sup>.

The inverse Fourier transform of an even and real envelope is an impulse response of a FIR filter. Convoluted with a sound, it performs the frequency filtering specified by the envelope. The greater the number of samples, the higher the envelope resolution and the computational complexity, which should often be weighted, for the convolution is expensive.

An important property is the time shift caused by convolution with a shifted impulse. Despite computationally expensive, it is possible to create *delay lines* by means of sound convolution with an impulse response that has an impulse for each reincidence of the sound. In figure 17



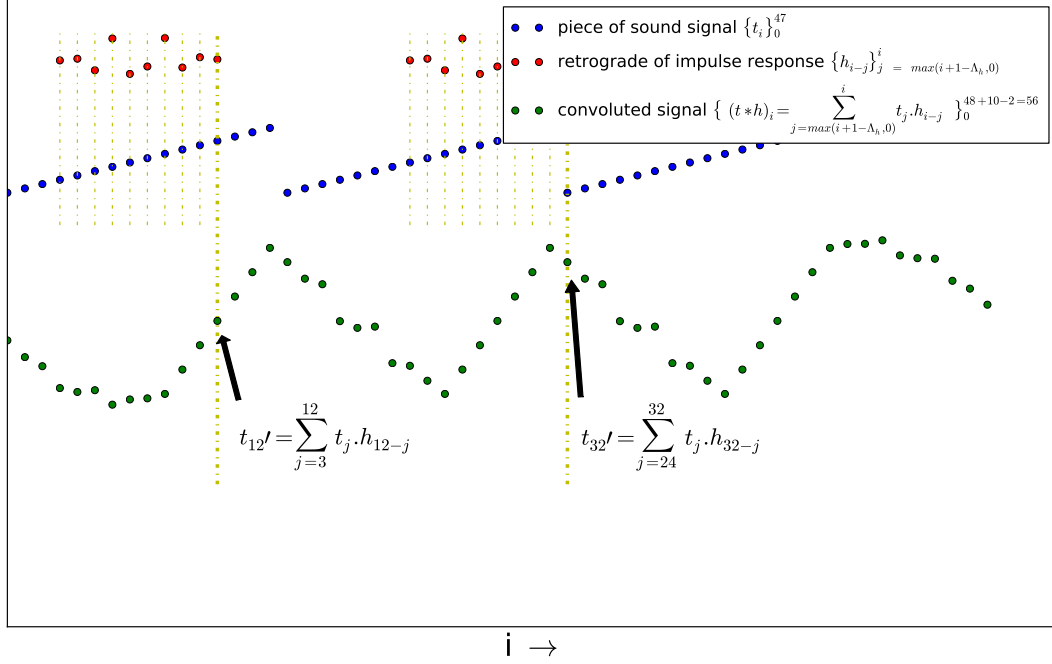


FIG. 16. Graphical interpretation of convolution. Each resulting sample is the sum of the previous samples of a signal, with each one multiplied by the retrograde of the other sequence.

it is possible to observe the shift caused by convolution with an impulse. Depending on the intensity of the impulses, the result is perceived as rhythm (from an impulse for each couple of seconds to about 20 impulses per second) or as pitch (from about 20 impulses per second and higher frequencies). In the latter case, the process resembles granular synthesis, reverbs and equalization.

## 2. Infinite impulse response (IIR) filters

This class of filters, known by the acronym IIR, are characterized by having an infinite time representation, i.e. the impulse response does not converge to zero. Its application is usually made by the following equation:

$$t'_i = \frac{1}{b_0} \left( \sum_{j=0}^J a_j \cdot t_{i-j} + \sum_{k=1}^K b_k \cdot t'_{i-k} \right) \quad (43)$$

In the most cases it is possible to normalize the variables:  $a'_j = \frac{a_j}{b_0}$  and  $b'_k = \frac{b_k}{b_0} \Rightarrow b'_0 = 1$ . Equation 43 is called 'difference equation' because the resulting samples  $\{t'_i\}$  are given by differences original samples  $\{t_i\}$  and previous resulting ones  $\{t'_{i-k}\}$ .

There are many methods and tools to obtain IIR filters. The text below lists a selection of them for didactic

purposes and as a reference. They are well behaved filters whose aspects are described in figure 18.

For filters of simple order, the cutoff frequency  $f_c$  is where the filter performs an attenuation of  $-3dB \approx 0.707$  of the original amplitude. For band-pass and band-reject (or 'notch') filters, this attenuation has two specifications:  $f_c$  (in this case, the 'center frequency') and bandwidth  $bw$ . In both frequencies  $f_c \pm bw$  there is an attenuation of  $\approx 0.707$  of the original amplitude. There is sound amplification in band-pass and band-reject filters when the cutoff frequency is low and the band width is large enough. In trebles, those filters present only a deviation of the expected profile, expanding the envelope to the bass.

For filters with other frequency responses, it is possible to apply them successively. Another possibility is to use a biquad 'filter receipt'<sup>40</sup> or the calculation of Chebichev filter coefficients<sup>41</sup>. Both alternatives are explored by referenced studies, in special<sup>42,43</sup>, and by the collection of filters maintained by the *Music-DSP* community of the Columbia University<sup>21,44</sup>.

1. Low-pass with a simple pole, module of the frequency response in the upper left corner of figure 18. The general equation has the cutoff frequency  $f_c \in (0, \frac{1}{2})$ , fraction of the sample frequency  $f_s$  in which occurs an attenuation of  $3dB$ . The coefficients  $a_0$  and  $b_1$  of the IIR filter are given by

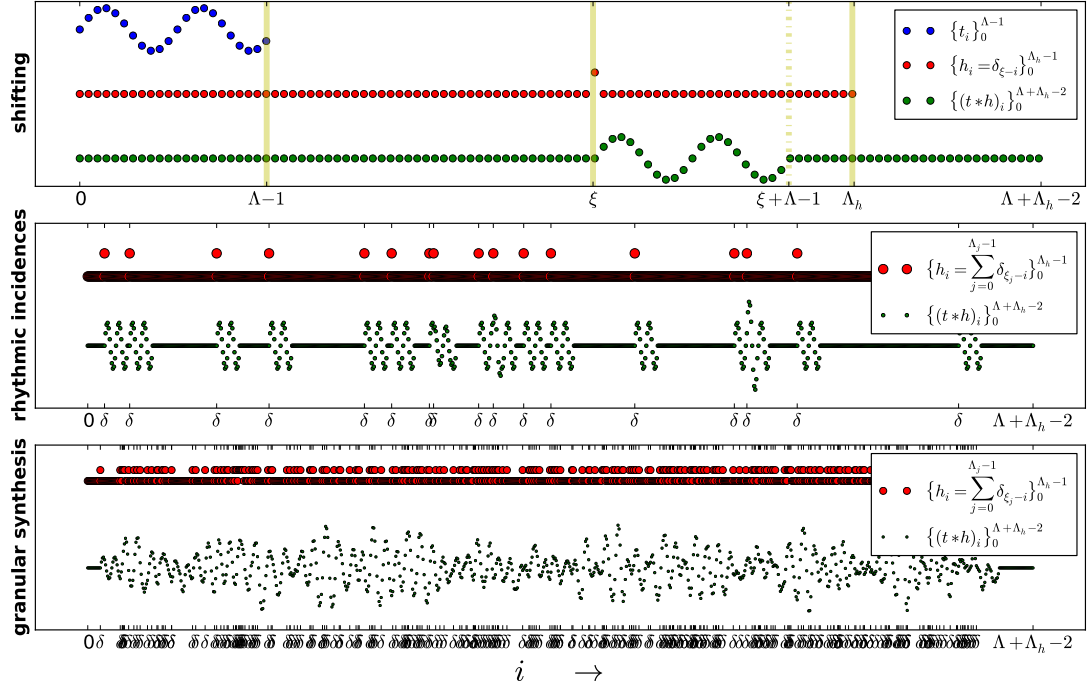


FIG. 17. Convolution with the impulse: shifting (a), delay lines (b) and granular synthesis (c). Disposed in increasing order of its pulse density.

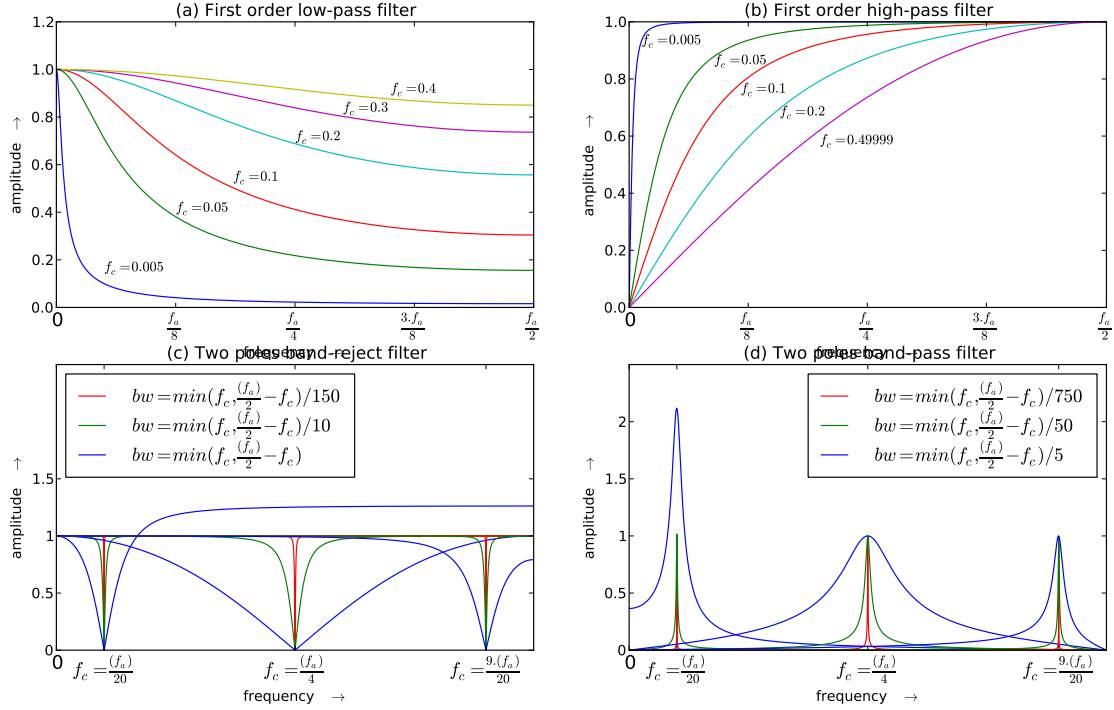


FIG. 18. Modules for the frequency response (a), (b), (c) and (d) for IIR filters of equations 44, 45, 47 and 48 respectively, considering different cutoff frequencies, center frequencies and band width.

means of the intermediate variable  $x \in [e^{-\pi}, 1]$ :

$$\begin{aligned} x &= e^{-2\pi f_c} \\ a_0 &= 1 - x \\ b_1 &= x \end{aligned} \quad (44)$$

2. High-pass filter with a simple pole, module of its frequency responses at the upper right corner of figure 18. The general equation with cutoff frequency  $f_c \in (0, \frac{1}{2})$  is calculated by means of the intermediate variable  $x \in [e^{-\pi}, 1]$ :

$$\begin{aligned} x &= e^{-2\pi f_c} \\ a_0 &= \frac{x+1}{2} \\ a_1 &= -\frac{x+1}{2} \\ b_1 &= x \end{aligned} \quad (45)$$

3. Notch filter. This filter is parametrized by a center frequency  $f_c$  and bandwidth  $bw$ , both given as fraction of  $f_s$ , therefore  $f, bw \in (0, 0.5)$ . Both frequencies  $f_c \pm bw$  has  $\approx 0.707$  of the amplitude, i.e. attenuation of  $3dB$ . Auxiliary variables  $K$  and  $R$  are defined as:

$$\begin{aligned} R &= 1 - 3bw \\ K &= \frac{1 - 2R \cos(2\pi f_c) + R^2}{2 - 2 \cos(2\pi f_c)} \end{aligned} \quad (46)$$

The band-pass filter in the lower left corner of figure 18 has the following coefficients:

$$\begin{aligned} a_0 &= 1 - K \\ a_1 &= 2(K - R) \cos(2\pi f_c) \\ a_2 &= R^2 - K \\ b_1 &= 2R \cos(2\pi f_c) \\ b_2 &= -R^2 \end{aligned} \quad (47)$$

The coefficients of band-reject filter, depicted in the lower right of figure 18, are:

$$\begin{aligned} a_0 &= K \\ a_1 &= -2K \cos(2\pi f_c) \\ a_2 &= K \\ b_1 &= 2R \cos(2\pi f_c) \\ b_2 &= -R^2 \end{aligned} \quad (48)$$

## D. Noise

Generally, sounds without an explicit pitch are called noise<sup>14</sup>. They are important musical sounds, as noise present in piano notes, violin, etc. Besides that, the majority of percussion instruments does not exhibit an unequivocal pitch and their sounds are generally regarded as noise<sup>1</sup>. In electronic music, including electro-acoustic and dance genres, noise has diverse uses and frequently characterizes the music style<sup>13</sup>.

The absence of a definite pitch is due to the lack of a perceptible harmonic organization in the sinusoidal components of the sound. Hence, there are many ways to generate noise. The use of random values to generate the sound sequence  $T_i$  is an attractive method but not outstandingly useful because it tends to produce white noise with little or no variations<sup>13</sup>.

Another way is to generate noise is by using the desired spectrum, from which it is possible to perform the inverse Fourier transform. The spectral distribution should be done with care: if phases of components present proeminent correlation, the synthesized sound will concentrate energy in some periods of its duration.

Some noise with static spectrum are listed below. They are called *colored noise* since they are associated with colors. Figure 19 shows the spectrum profile and the corresponding sonic sequence side-by-side. All five noises were generated with the same phase for each component, making it possible to observe the contributions of different parts of the spectrum.

- The white noise has its name because its energy is distributed equally among all frequencies. It is possible to obtain white noise with the inverse transform of the following coefficients:

$$\begin{aligned} c_0 &= 0, \quad \text{to avoid bias} \\ c_i &= e^{j \cdot x}, \quad x \text{ random} \in [0, 2\pi], \quad i \in \left[1, \frac{\Lambda}{2} - 1\right] \\ c_{\Lambda/2} &= 1, \quad \text{if } \Lambda \text{ even} \\ c_i &= c_{\Lambda-i}^*, \quad \text{for } i > \frac{\Lambda}{2} \end{aligned} \quad (49)$$

The exponential  $e^{j \cdot x}$ , is an artifice to obtain unitary module and random phase for the value of  $c_i$ . Besides that,  $c_{\Lambda/2}$  is always real (as discussed in the previous section).

- The pink noise is characterized by a decrease of  $3dB$  per octave. This noise is useful for testing electronic devices, and it has prominent presence in nature<sup>1</sup>.

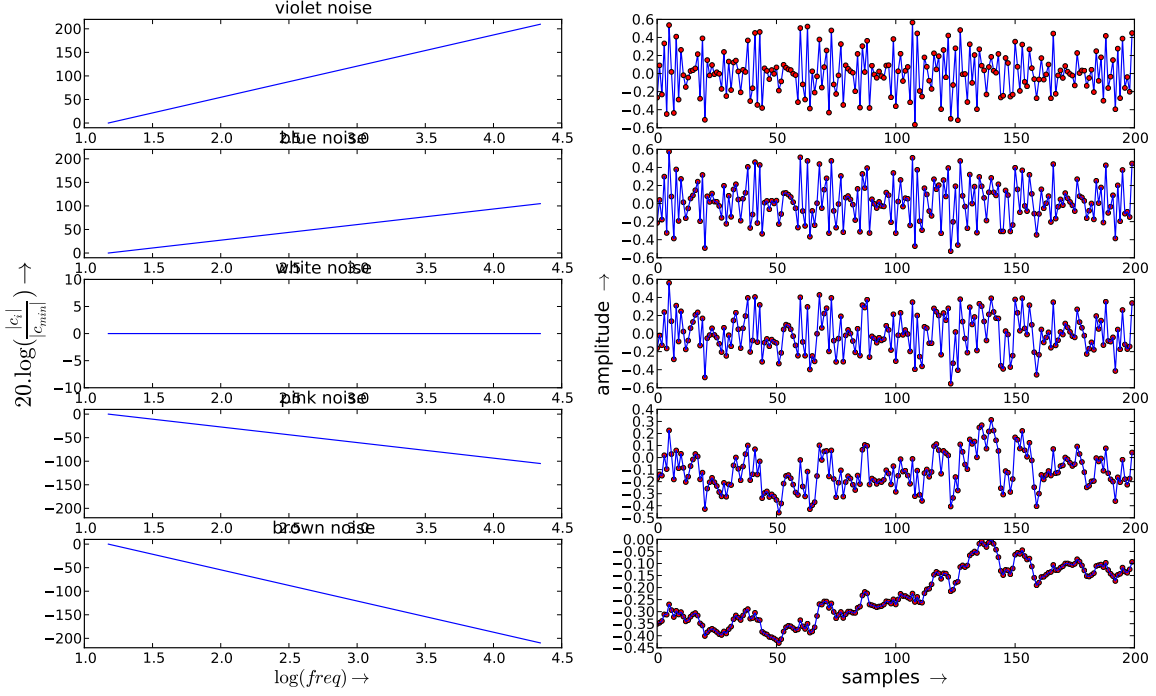


FIG. 19. Colors of noise generated by equations 49, 50, 51, 52 and 53: spectrum and waveforms.

$$\begin{aligned}
 f_{\min} &\approx 15Hz \\
 f_i &= i \frac{f_a}{\Lambda}, \quad i \leq \frac{\Lambda}{2}, \quad i \in \mathbb{N} \\
 \alpha_i &= \left(10^{-\frac{3}{20}}\right)^{\log_2\left(\frac{f_i}{f_{\min}}\right)} \\
 c_i &= 0, \quad \forall i : f_i < f_{\min} \\
 c_i &= e^{j \cdot x} \cdot \alpha_i, \quad x \text{ random} \in [0, 2\pi], \quad \forall i : f_{\min} \leq f_i < f_{\lceil \Lambda/2 \rceil} \\
 c_{\Lambda/2} &= \alpha_{\Lambda/2}, \quad \text{if } \Lambda \text{ even} \\
 c_i &= c_{\Lambda-i}^*, \quad \text{for } i > \Lambda/2
 \end{aligned} \tag{50}$$

The minimum frequency  $f_{\min}$  is chosen with regard to the human hearing, since a sound component with frequency below  $\approx 20Hz$  is virtually inaudible.

Other noises can be made by similar procedures. Simple modifications are needed, specially in the equation that defines  $\alpha_i$ .

- The brown noise received this name after Robert Brown, who described the Brownian movement<sup>45</sup>

What characterizes this noise is the decrease of  $6dB$  per octave. In this way,  $\alpha_i$  in the equation set 50 is:

$$\alpha_i = \left(10^{-\frac{6}{20}}\right)^{\log_2\left(\frac{f_i}{f_{\min}}\right)} \tag{51}$$

- In the blue noise there is a gain of  $3dB$  per octave in a band limited by the minimum frequency  $f_{\min}$  and the maximum frequency  $f_{\max}$ . Therefore, the corresponding equation is also based on the equations set 50:

$$\begin{aligned}
 \alpha_i &= \left(10^{\frac{3}{20}}\right)^{\log_2\left(\frac{f_i}{f_{\min}}\right)} \\
 c_i &= 0, \quad \forall i : f_i < f_{\min} \text{ or } f_i > f_{\max}
 \end{aligned} \tag{52}$$

- The violet noise is similar to the blue noise, but its gain is  $6dB$  per octave:

$$\alpha_i = \left(10^{\frac{6}{20}}\right)^{\log_2\left(\frac{f_i}{f_{\min}}\right)} \tag{53}$$

- The black noise has higher losses than  $6dB$  for octave:

$$\alpha_i = \left(10^{-\frac{\beta}{20}}\right)^{\log_2\left(\frac{f_i}{f_{\min}}\right)}, \quad \beta > 6 \tag{54}$$

- The gray noise is defined as a white noise subject to one of the ISO-audible curves. Those curves are obtained by experiments and are imperative to obtain  $\alpha_i$ . An implementation of ISO 226, which is the last revision of these curves, is in the MASSA toolbox<sup>32</sup>.

This subsection exposed only noise with static spectrum. There are also characterizations for noise with dynamic spectrum during the time, and noises which are fundamentally transient, like clicks and chirps. The former are easily modeled by an impulse relatively isolated, while chirps are not in fact a noises, but a fast scan of some given frequency band<sup>13</sup>.

The noise from equations 49, 50, 51, 52 and 53 are presented in figure 19. As stated above, the spectra were built with the same phase and frequency for each coefficient, making it straightforward to observe the contribution of treble harmonics and bass frequencies.

### E. Tremolo and vibrato, AM and FM

Vibrato is a periodic variation in pitch (frequency) and tremolo is a period variation in volume (intensity)<sup>46</sup> For the general case, the vibrato is described as follows:

$$\gamma'_i = \left\lfloor i f' \frac{\tilde{\Lambda}_M}{f_s} \right\rfloor \quad (55)$$

$$t'_i = \tilde{m}_{\gamma'_i \% \tilde{\Lambda}_M} \quad (56)$$

$$f_i = f \left( \frac{f + \mu}{f} \right)^{t'_i} = f \cdot 2^{t'_i \frac{\nu}{12}} \quad (57)$$

$$\begin{aligned} \Delta_{\gamma_i} = f_i \frac{\tilde{\Lambda}}{f_s} &\Rightarrow \gamma_i = \left\lfloor \sum_{j=0}^i f_j \frac{\tilde{\Lambda}}{f_s} \right\rfloor = \\ &= \left\lfloor \sum_{j=0}^i \frac{\tilde{\Lambda}}{f_s} f \left( \frac{f + \mu}{f} \right)^{t'_j} \right\rfloor = \left\lfloor \sum_{j=0}^i \frac{\tilde{\Lambda}}{f_s} f \cdot 2^{t'_j \frac{\nu}{12}} \right\rfloor \end{aligned} \quad (58)$$

$$T_i^{f, vbr(f', \nu)} = \left\{ t_i^{f, vbr(f', \nu)} \right\}_0^{\Lambda-1} = \left\{ \tilde{t}_{\gamma_i \% \tilde{\Lambda}} \right\}_0^{\Lambda-1} \quad (59)$$

For the proper realization of the vibrato, it is important to pay attention to both tables and sequences. Table  $\tilde{M}_i$  with length  $\tilde{\Lambda}_M$  and the sequence with indices  $\gamma'_i$  make the sequence  $t'_i$  which is the oscillation pattern in the frequency while table  $\tilde{L}_i$  with length  $\tilde{\Lambda}$  and the sequence with indices  $\gamma_i$  make  $t_i$  which is the sound itself. Variables  $\mu$  and  $\nu$  quantify the vibrato intensity:

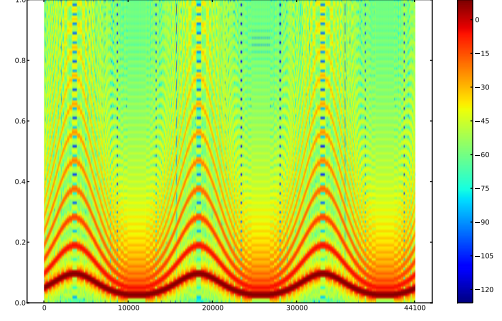


FIG. 20. Spectrogram of a sound with a sinusoidal vibrato of 3Hz and one octave of depth in a 1000Hz sawtooth wave, considered  $f_s = 44.1kHz$ .

- $\mu$  is a direct measure of how many Hertz are involved in the upper limit of the oscillation, while
- $\nu$  is the direct measure of how many semitones (or half steps) are involved in the oscillation ( $2\nu$  is the number of semitones between the upper and lower peaks of frequency oscillations of the sound  $\{t_i\}$  caused by the vibrato).

It is convenient to use  $\nu = \log_2 \frac{f+\mu}{f}$  in this case because the maximum frequency increase is not equivalent to the maximum frequency diminish, but the semitone variation remains.

Figure 20 is the spectrogram of an artificial vibrato for a note with 1000Hz (between a B and a C), in which pitch deviation reaches one octave above and one below. Practically any waveform can be used to generate a sound and the vibrato oscillation pattern, with virtually any oscillation frequency and pitch deviation<sup>47</sup>. Those oscillations with precise waveforms and arbitrary amplitudes are not possible in traditional music instruments and, in this way, it introduces novelty in the artistic possibilities.

Tremolo is similar:  $f'$ ,  $\gamma'_i$  and  $t'_i$  remains the same. The amplitude sequence to be multiplied by the original sequence  $t_i$  turns in:

$$a_i = 10^{\frac{V_{dB}}{20} t'_i} = a_{\max}^{t'_i} \quad (60)$$

and finally:

$$\begin{aligned} T_i^{tr(f')} &= \left\{ t_i^{tr(f')} \right\}_0^{\Lambda-1} = \{ t_i \cdot a_i \}_0^{\Lambda-1} = \\ &= \left\{ t_i \cdot 10^{\frac{V_{dB}}{20} t'_i} \right\}_0^{\Lambda-1} = \left\{ t_i \cdot a_{\max}^{t'_i} \right\}_0^{\Lambda-1} \end{aligned} \quad (61)$$

where  $V_{dB}$  is the oscillation depth in decibels of tremolo and  $a_{\max} = 10^{\frac{V_{dB}}{20}}$  is the maximum amplitude gain. The measurement in decibels is pertinent because the maximum increase in amplitude is not equivalent to the related maximum decrease, while the difference in decibels remains.

Figure 21 shows the amplitude of sequences  $\{a_i\}_0^{\Lambda-1}$  and  $\{t'_i\}_0^{\Lambda-1}$  for three oscillations of a tremolo with a sawtooth waveform. The curvature is due to the logarithmic progression of the intensity. The tremolo frequency is  $1,5Hz$  because  $f_a = 44,1kHz \Rightarrow \text{duration} = \frac{t_{\max}=82000}{f_a} = 2s \Rightarrow \frac{3\text{oscillations}}{2s} = 1,5 \text{ oscillations per second (Hz)}$ .

The music piece *Vibra e trema* explores these possibilities given by tremolos and vibratos, both used in conjunction and independently, with frequencies  $f'$ , different depths ( $\nu$  and  $V_{dB}$ ), and progressive parameters variations (tremolos and vibratos occur many times together in a traditional music instrument and voices). Aiming a qualitative appreciation, the piece also develops a comparison between vibratos and tremolos in logarithmic and linear scales. Its source code is available online as part of the MASSA toolbox.

The proximity of  $f'$  to  $20Hz$  generates rugosity in both tremolos and vibratos. This rugosity is largely appreciated both in traditional classical music and current electronic music, specially in the *Dubstep* genre. Rugosity is also generated by spectral content that produce beating<sup>48,49</sup>. The sequence *Bela Rugosi* explores this rugosity threshold with concomitant tremolos and vibratos at the same voice, with different intensity and waveforms. The respective code is available online in the MASSA toolbox.

As frequency increases further, these oscillations no longer remains noticeable individually. In this case, the oscillations are audible as pitch. Then,  $f'$ ,  $\mu$  and the waveform together, changes the spectrum of original sound  $T_i$  in different ways for both tremolos and vibratos. They are called AM (*Amplitude Modulation*) and FM (*Frequency Modulation*) synthesis, respectively. These techniques are well known, with applications in synthesizers like *Yamaha DX7*, and even with applications outside music, like in telecommunications, for data transfer by means of electromagnetic waves (e.g. AM and FM radios).

For musical goals, it is possible to understand FM based on the case of sines and, in occurrences of greater complexity, to decompose the signals into their respective Fourier components (i.e. sines as well). In this way, the FM synthesis performed with a sinusoidal vibrato with frequency  $f'$  and depth  $\mu$  in a sinusoidal sound  $T_i$  with frequency  $f$  generates bands centered in  $f$  and far from each other with a distance of  $f'$ :

$$\begin{aligned} \{t'_i\} &= \left\{ \cos \left[ f \cdot 2\pi \frac{i}{f_s - 1} + \mu \cdot \sin \left( f' \cdot 2\pi \frac{i}{f_s - 1} \right) \right] \right\} = \\ &= \left\{ \sum_{k=-\infty}^{+\infty} J_k(\mu) \cos \left[ f \cdot 2\pi \frac{i}{f_s - 1} + k \cdot f' \cdot 2\pi \frac{i}{f_s - 1} \right] \right\} = \\ &= \left\{ \sum_{k=-\infty}^{+\infty} J_k(\mu) \cos \left[ (f + k \cdot f') \cdot 2\pi \frac{i}{f_s - 1} \right] \right\} \end{aligned} \quad (62)$$

where

$$\begin{aligned} J_k(\mu) &= \\ &= \frac{2}{\pi} \int_0^{\frac{\pi}{2}} \left[ \cos \left( \bar{k} \frac{\pi}{2} + \mu \cdot \sin w \right) \cdot \cos \left( \bar{k} \frac{\pi}{2} + k \cdot w \right) \right] dw \\ \bar{k} &= k \% 2, \quad k \in \mathbb{N} \end{aligned} \quad (63)$$

is the Bessel function<sup>42,50</sup> which specifies the amplitude of each component in FM synthesis.

On these equations, the frequency variation introduced by  $\{t'_i\}$  does not respect the geometric progression that yield linear pitch variation, but reflects equation 32. The use of equations 57 for FM are described in<sup>3</sup>, where the spectral content of the FM synthesis is calculated for oscillations in the logarithmic scale. In fact, the simple and attractive FM behavior are observed only with linear variations, such as in 62).

For the amplitude modulation (AM):

The resulting sound is the original one together with the reproduction of its spectral content below and above the original frequency, with the distance  $f'$  from  $f$ . Again, this is obtained by variations in the linear scale of the amplitude. An exposition of the spectrum of an AM performed with oscillations in the logarithmic amplitude scale is again in<sup>3</sup>, and it again loses the simple and attractive behavior.

The sequence  $T_i$ , with frequency  $f$ , called 'carrier', is modulated by  $f'$ , called 'modulator'. In FM and AM jargon,  $\mu$  and  $\alpha = 10^{\frac{V_{dB}}{20}}$  are 'modulation indexes'. The following equations are defined for the vibration pattern of the modulator sequence  $\{t'_i\}$ :

$$\gamma'_i = \left\lfloor i f' \frac{\tilde{\Lambda}_M}{f_s} \right\rfloor \quad (64)$$

$$t'_i = \tilde{m}_{\gamma'_i \% \tilde{\Lambda}_M} \quad (65)$$

The modulator  $\{t'_i\}$  into the carrier  $\{t_i\}$ , for FM, is applied as:

$$f_i = f + \mu \cdot t'_i \quad (66)$$

$$\Delta_{\gamma_i} = f_i \frac{\tilde{\Lambda}}{f_s} \Rightarrow \gamma_i = \left\lfloor \sum_{j=0}^i f_j \frac{\tilde{\Lambda}}{f_s} \right\rfloor = \left\lfloor \sum_{j=0}^i \frac{\tilde{\Lambda}}{f_s} (f + \mu \cdot t'_j) \right\rfloor \quad (67)$$

$$T_i^{f, FM(f', \mu)} = \left\{ t_i^{f, FM(f', \mu)} \right\}_0^{\Lambda-1} = \left\{ \tilde{t}_{\gamma_i \% \tilde{\Lambda}} \right\}_0^{\Lambda-1} \quad (68)$$

where  $\tilde{t}$  is the waveform period with a length of  $\tilde{\Lambda}$  samples, used for the carrier signal.

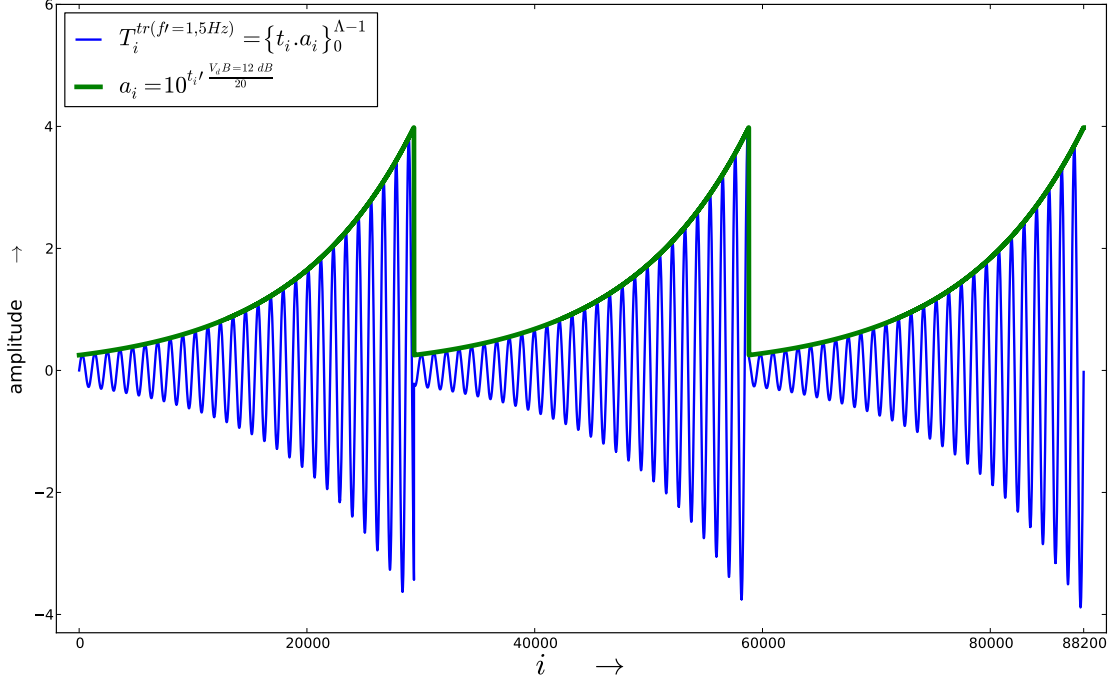


FIG. 21. Tremolo with a depth of  $V_{dB} = 12dB$ , with a sawtooth waveform as its oscillatory pattern, with  $f' = 1.5Hz$  in a sine of  $f = 40Hz$  (considered a sample frequency of  $f_s = 44.1kHz$ ).

To perform AM, it is needed to modulate the signal  $\{t_i\}$  with  $\{t'_i\}$  using the following equations:

$$a_i = 1 + \alpha \cdot t'_i \quad (69)$$

$$T_i^{f, AM(f', \alpha)} = \left\{ t_i^{f, AM(f', \alpha)} \right\}_0^{\Lambda-1} = \{t_i \cdot a_i\}_0^{\Lambda-1} = \{t_i \cdot (1 + \alpha \cdot t'_i)\}_0^{\Lambda-1} \quad (70)$$

## F. Musical usages

At this point, the musical possibilities exploded. All characteristics, like pitch (given by frequency), timbre (given by the waveform, filters and noise), volume (manipulated by intensity) and duration (given by the number of samples), can be considered in an absolute form or treated during duration of sound (clearly, with the exception of duration itself).

The following musical usages comprehend a collection of possibilities with the purpose of exemplifying types of sound manipulation that result in musical material. Some of them are deeply discussed in the next section.

### 1. Relations between characteristics

An interesting possibility is to use relations between parameters of a tremolo or vibrato, and some parameters of the basic note like frequency. In this way, it is possible to have a vibrato frequency proportional to note pitch, or a tremolo depth inversely proportional to pitch. Therefore, with equations 55, 57 and 60, it is perfectly possible to arbitrate:

$$\begin{aligned} f^{vbr} &= f^{tr} = func_a(f) \\ \nu &= func_b(f) \\ V_{dB} &= func_c(f) \end{aligned} \quad (71)$$

with  $f^{vbr}$  and  $f^{tr}$  as  $f'$  in the referenced equations. They can also be associated with vibrato and tremolo oscillation frequency of equation 55.  $\nu$  and  $V_{dB}$  are the respective depth values of vibrato and tremolo. Functions  $func_a$ ,  $func_b$  and  $func_c$  are arbitrary and dependent on musical intentions. The music piece *Tremolos, vibratos e a frequencia* explores such bonds and exhibit variations in the oscillation waveform with the purpose of building a *musical language* (details in the next section). The corresponding code is on is also available online as part of the MASSA toolbox.

## 2. Convolution for rhythm and meter

It is possible to use a musical pulse – such as specified by a BPM tempo – and to place an impulse at the start of each beat, with the purpose of establishing metric and rhythms: the convolution with an impulse shifts the sound to impulse position. For example, two impulses equally spaced builds a binary division into the pulse. Two signals, one with 2 impulses and the other with 3 impulses, both equally spaced in the pulse duration, yields a pulse maintenance with a rhythmic in which eases both binary or ternary divisions of the pulse, and this is found in many ethnic and traditional musical styles<sup>51</sup>. Absolute values of the impulses implies in proportions among the amplitudes of the sonic reincidences. The use of convolution with impulses in this context is explored in the music piece *Trenzinho de caipiras impulsivos*. These features embrace the creation of ‘sound amalgams’ based on granular synthesis, for now refer specially to the figure 24. This piece is a link to next section contents. The source code of the music piece is online, as part of the MASSA toolbox<sup>32</sup>.

## 3. Moving source and receptor, Doppler effect

Retaken the exposition in subsection II G, when an audio source or receptor is moving, its characteristics are ideally updated at each sample of the digital signal. Speed components should be found for each ear. In this way, given the audio source speed (or velocity)  $v_s$ , with positive values if the source moves away from receptor, and receptor speed  $v_r$ , positive when it gets closer to audio source, the frequency is given by the well known Doppler shift:

$$f = \left( \frac{v_{sound} + v_r}{v_{sound} + v_s} \right) f_0 \quad (72)$$

With both frequencies  $f$  and  $f_0$ , and the IID from the new source position, it is possible to create the Doppler effect. There is an addendum to improve the fidelity of the physical phenomena: to increase the received potency. It is possible to understand this potency gain as being proportional to the wave shrinking or expansion:  $\Delta P = P_0 \left( \frac{v_r - v_s}{343.2} \right)$ , where  $P_0$  is signal potency and  $P$  the potency at receptor.

In this way, it is possible to obtain both amplitude and frequency of a moving audio source. Being this audio source in front of receptor with  $y_0$  meters of horizontal distance and  $z_0$  meters of height, the distance is given by  $D_i = \left\{ d_i = \sqrt{y_i^2 + z_0^2} \right\}_0^{\Lambda-1}$ , where  $y_i = y_0 + (v_s - v_r) \frac{i}{f_s}$  with  $v_s$  and  $v_r$  both horizontal, having null non- $y$  components. Amplitude changes with the distance and with the potency factor cited above (see subsection II B for potency to amplitude conversion).

$$A_i = \left\{ \frac{z_0}{d_i} A_{\Delta P} \right\}_0^{\Lambda-1} = \left\{ \frac{z_0}{\sqrt{y_i^2 + z_0^2}} \sqrt{\frac{v_r - v_s}{343.2} + 1} \right\}_0^{\Lambda-1} \quad (73)$$

The amplitude change caused by the distance is even, while the change caused by the potency variation is antisymmetric in relation to the crossing of source with receptor. The frequency has a symmetric progression in relation to pitch. In other words, the same semitones (or fractions) added during the approach are decreased during the departure. Besides that, the transition is abrupt if source and receptor intersect with zero distance, otherwise, there is a monotonic progression. In the given case, where there is a static height  $z_0$ , observed the speed component in the direction given by observer and source positions, the frequencies  $F_i$  at the observer is given by:

$$F_i = \{f_i\}_0^{\Lambda-1} = \left\{ \frac{v_{sound} + v_r \frac{y_i}{\sqrt{z_0^2 + y_i^2}}}{v_{sound} + v_s \frac{y_i}{\sqrt{z_0^2 + y_i^2}}} f_0 \right\}_0^{\Lambda-1} \quad (74)$$

There is a Python implementation of the Doppler effect as describe above, and also considering the intersection between audio source and receptor, at the MASSA toolbox<sup>32</sup>.

## 4. Filters and noises (subsections III D and III C)

With the use of filters, the possibilities are even wider. Convolve a signal to have a reverberated version of it, to remove its noise, to distort or to treat the audio aesthetically in other ways. For example, it is possible to simulate sounds as if originated in an old television or telephone, by the use of a band-pass filter, allowing only frequencies between  $1kHz$  and  $3kHz$ . By rejecting the frequency of electric oscillation (usually  $50Hz$  or  $60Hz$ ) and harmonics, one can remove noises caused by audio devices connected to a usual power supply. A more musical application is to perform filtering in specific bands and to use those bands as an additional parameter to the notes.

Inspired by traditional music instruments, it is possible to apply a time-dependent filter<sup>1</sup>. Chains of these filters can perform complex and more accurate filtering routines. The music piece *Ruidosa faixa* explores filters and many kinds and noise synthesis. The source code is available online as part of MASSA toolbox.

A sound can be altered through different filtering and then mixed to create an effect known as *chorus*. Based on what happens in a choir of singers, the sound is performed using small and potentially arbitrary modifications of parameters like center frequency, presence (or absence) of vibrato or tremolo and its characteristics, equalization, volume, etc. As a final result, those versions of the original sound are mixed together (see equation 29). The music piece *Chorus infantil* implements a chorus in many



ways with different sounds and its source code is also available in the MASSA toolbox.

## 5. Reverberation

Using the same terms of subsection II G, the late reverberation can be achieved by a convolution with a section of pink, brown or black noise, with exponential decay of amplitude along time. Delay lines can be added as a prefix to the noise with the decay, and this contemplates both time parts of the reverberation: the first reflections and the late reverberation. It is possible to improve the quality by calculating the geometric trajectory and (mainly low-pass, subsection III C) filtering by each surface where wavefront reflected before reaching the ear in the first 100–200 milliseconds. The colored noise can be gradually introduced with a *fade-in*: the initial moment given by direct incidence of sound (i.e. without any reflection and given by ITD and IID), reaching its maximum at the beginning of the 'late reverberation', when the geometric incidences loses their relevance to the statistical properties of noise decay.

As an example, consider  $\Delta_1$  as the duration of the first section and  $\Delta_R$  as the duration of total reverberation ( $\Lambda_1 = \Delta_1 f_s$ ,  $\Lambda_R = \Delta_R f_s$ ). Be  $p_i$  the probability of a sound to repeat in the  $i$ -th sample. Amplitude falls with exponential decay. Following subsection II G, the reverberation  $R_i^1$  of the first period can be described as:

$$R_i^1 = \{r_i^1\}_{\Lambda_1-1}^{\Lambda_1-1} : r_i^1 = \begin{cases} 10^{\frac{V_{dB}}{20} \frac{i}{\Lambda_R-1}} & \text{with probability } p_i = \left(\frac{i}{\Lambda_1}\right)^2 \\ 0 & \text{with probability } 1 - p_i \end{cases} \quad (75)$$

where  $V_{dB}$  is the total decay in decibels, typically  $-80dB$  or  $-120dB$ . Reverberation  $R_i^2$  of the second period can be emulated by a brown noise  $R_i^m$  (or by a pink noise  $R_i^r$ ) with exponential amplitude decay of the waveform:

$$R_i^2 = \{r_i^2\}_{\Lambda_1}^{\Lambda_R-1} = \left\{ 10^{\frac{V_{dB}}{20} \frac{i}{\Lambda_R-1}} \cdot r_i^m \right\}_{\Lambda_1}^{\Lambda_R-1} \quad (76)$$

Considered:

$$R_i = \{r_i\}_{\Lambda_1}^{\Lambda_R-1} : r_i = \begin{cases} 1 & \text{if } i = 0 \\ r_i^1 & \text{if } 1 \leq i < \Lambda_1 - 1 \\ r_i^2 & \text{se } \Lambda_1 \leq i < \Lambda_R - 1 \end{cases} \quad (77)$$

, a reverberated auralization of a sound can be achieved by simple convolution of  $R_i$  (called reverberation 'impulse response') with the sound sequence  $T_i$  as described in subsection III C.

Reverberation is well known by causing great interest in the listeners and to provide more enjoyable sonorities. Besides that, modifications in the reverberation

space consists on a common resource (almost a *cliché*) to surprise and attract the listener. An implementation of the reverb recipe described is on the MASSA toolbox<sup>32</sup>

## 6. ADSR Envelopes

The volume variation along the duration of sound is crucial to our timbre perception. The volume envelope known as ADSR (*Attack-Decay-Sustain-Release*), has many implementations in both hardware and software synthesizers. A pioneer implementation can be found in the Hammond Novachord synthesizer of 1938 and some variants are cited below<sup>52</sup>.

The scholastic ADSR envelope is characterized by 4 parameters: attack duration (time in which sound reaches its maximum amplitude), decay duration (follows the attack immediately), level of sustain volume (in which the volume remains stable after the decay) and release duration (after sustain section, this is the duration needed for amplitude to reach zero or final value). Note that the sustain duration is not specified because it is the difference between the duration itself and the durations of attack, decay and sustain.

The ADSR envelope with durations  $\Delta_A$ ,  $\Delta_D$  and  $\Delta_R$ , with total duration  $\Delta$  and sustain level  $a_S$ , given by the fraction of the maximum amplitude, to be aplyed to any sound sequence  $T_i = \{t_i\}$ , can be presented as:

$$\begin{aligned} \{a_i\}_0^{\Lambda_A-1} &= \left\{ \xi \left( \frac{1}{\xi} \right)^{\frac{i}{\Lambda_A-1}} \right\}_0^{\Lambda_A-1} \quad \text{or} \\ &= \left\{ \frac{i}{\Lambda_A-1} \right\}_0^{\Lambda_A-1} \\ \{a_i\}_{\Lambda_A}^{\Lambda_A+\Lambda_D-1} &= \left\{ a_S^{\frac{i-\Lambda_A}{\Lambda_D-1}} \right\}_{\Lambda_A}^{\Lambda_A+\Lambda_D-1} \quad \text{or} \\ &= \left\{ 1 - (1 - a_S) \frac{i - \Lambda_A}{\Lambda_D - 1} \right\}_{\Lambda_A}^{\Lambda_A+\Lambda_D-1} \\ \{a_i\}_{\Lambda_A+\Lambda_D}^{\Lambda_A+\Lambda_D+\Lambda_R-1} &= \{a_S\}_{\Lambda_A+\Lambda_D}^{\Lambda_A+\Lambda_D+\Lambda_R-1} \\ \{a_i\}_{\Lambda_A+\Lambda_D}^{\Lambda_A+\Lambda_D+\Lambda_R-1} &= \left\{ a_S \left( \frac{\xi}{a_S} \right)^{\frac{i-(\Lambda_A+\Lambda_D)}{\Lambda_R-1}} \right\}_{\Lambda_A+\Lambda_D}^{\Lambda_A+\Lambda_D+\Lambda_R-1} \quad \text{or} \\ &= \left\{ a_S - a_S \frac{i + \Lambda_R - \Lambda}{\Lambda_R - 1} \right\}_{\Lambda_A+\Lambda_D}^{\Lambda_A+\Lambda_D+\Lambda_R-1} \end{aligned} \quad (78)$$

with  $\Lambda_X = \lfloor \Delta \cdot f_s \rfloor \quad \forall \quad X \in (A, D, R, )$  and being  $\xi$  a small value that provides a satisfactory *fade in* and *fade out*, e.g.  $\xi = 10^{\frac{-80}{20}} = 10^{-4}$  or  $\xi = 10^{\frac{-40}{20}} = 10^{-2}$ . The lower the  $\xi$ , the slower the *fade*, like the  $\alpha$  illustrated in Figure 15. The stated second in equations 78 attend both introduction and ending of sound from zero intensity because they are linear. Schematically, figure 22 shows the ADSR envelope in a classical implementation

that supports many variations. For example, between attack and decay it is possible to add an extra partition where the maximum amplitude remains for more than a peak. Another common example is the use of more elaborated tracings of attack or decay. The music piece *ADa e SaRa*, available in MASSA, explores many configurations of the ADSL envelope.

$$\{t_i^{ADSR}\}_0^{\Lambda-1} = \{t_i \cdot a_i\}_0^{\Lambda-1} \quad (79)$$

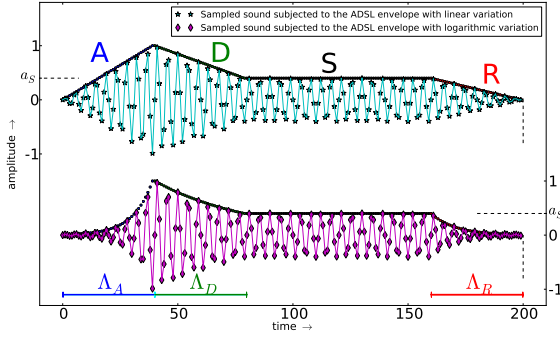


FIG. 22. An ADSR envelope (Attack, Decay, Sustain, Release) applied to an arbitrary sound sequence. The linear variation of the amplitude is above, in blue. Below the amplitude variation is exponential.

#### IV. ORGANIZATION OF NOTES IN MUSIC

Consider  $S_j = \left\{ s_j = T_i^j = \{t_i^j\}_{i=0}^{\Lambda_j-1} \right\}_{j=0}^{H-1}$ , the sequence  $S_j$  of  $H$  musical events  $s_j$ . Be  $S_j$  a ‘musical structure’, composed by events  $s_j$  which are musical structures themselves, e.g. notes. This section is dedicated to techniques that make  $S_j$  interesting and enjoyable for audition.

The elements of  $S_j$  can be overlapped by mixing them together, as in equation 29 and figure 12, for building intervals and chords. This reflects the ‘vertical thought’ in music. On the other hand, the concatenation of the events in  $S_j$ , as in equation 30 and in figure 13, builds melodic sequences and rhythms, which are associated with the musical ‘horizontal thought’. The fundamental frequency  $f$  and the starting moment (attack) are generally the most important characteristics of the elements  $s_j$  in  $S_j$ . This makes possible the creation of music made by pitches (both harmony and melody) and by the presence of temporal metrics and rhythms.

#### A. Tuning, intervals, scales and chords

##### 1. Tuning

Doubling the frequency is equivalent to ascending one octave ( $f = 2f_0$ ). The octave division in twelve steps is the canon of classical western music. Its usage has also been observed outside western tradition, like ceremonial/religious and ethnic context<sup>11</sup>. In this system, ideally, the factor given by  $\varepsilon = 2^{\frac{1}{12}}$  defines a semitone. A note grid along the hearing spectrum in which, given the frequency  $f$ , all the possible fundamental frequencies are separated by intervals, multiples of  $\varepsilon$ . Twelve semitones (or half steps), equidistant to the human ear, reaches an octave, equivalent to human year. Therefore, if  $f = 2^{\frac{1}{12}} f_0$ , there is a semitone between  $f_0$  and  $f$ .

This absolute accuracy is common in computational implementations. Real music instruments, however, exhibit deviations in order to make compatible deviations from the harmonics of notes. Besides that, the fixed reference  $\varepsilon = 2^{\frac{1}{12}}$  characterizes an equally tempered tuning. There are tunings in which intervals are ratios of low-order integers, based on observations of physical phenomena. These tunings were first formalized around two thousand years before the arrival of the equal temperament<sup>1</sup>. ). Two emblematic tunings are:

- The **just intonation**, defined by association of diatonic scale notes with ratios of low-order integers, as found in the harmonic series. Considered the ionian mode, the white piano keys from C to C, as seen below in subsection IV A 3, i.e. intervals 1, 2M, 3M, 4J, 5J, 6M, 7M, 8, are considered ratios of frequency: 1, 9/8, 5/4, 4/3, 3/2, 5/3, 15/8, 2/1. The following intervals are also considered: the semitone 16/15, the ‘minor tone’ 10/9, and the ‘major tone’ 9/8. There are many ways to perform the division of the 12 notes in the just intonation.
- The **Pythagorean tuning**, based on the interval 3/2 (perfect fifth). The Ionian mode ratios becomes: 1, 9/8, 81/64, 4/3, 3/2, 27/16, 243/128, 2/1. These intervals are also considered to yield notes in a scale. Besides the intervals in the mode, also used are the minor second 256/243, the minor third 32/27, the augmented fourth 729/512, the diminished fifth 1024/729, the minor sixth 128/81 and the minor seventh 16/9.

In order to attend for micro-tonality<sup>53</sup>. It is possible to use non-integer real values as factors of  $\varepsilon = 2^{\frac{1}{12}}$  between frequencies, or to maintain integer values and change  $\varepsilon$ . For example, a tuning really near of the harmonic series itself is proposed with the equal division of the octave in 53 notes:  $\varepsilon_2 = 2^{\frac{1}{53}}$ .<sup>54</sup> In this division, notes are related by means of integers, factors of  $\varepsilon_2$  relating frequencies by a valid interval. Note that if  $S_i$  is a pitch sequence related by means of  $\varepsilon_1$ , a direct mapping for notes related by  $\varepsilon_2$

is a new sequence  $S'_i = \{s'_i\} = \left\{s_i \frac{\varepsilon_1}{\varepsilon_2}\right\}$ . The music piece *Micro tom* explores microtonal features and its code is part of MASSA toolbox.

## 2. Intervals

Using the ratio  $\varepsilon = 2^{\frac{1}{12}}$  between note frequencies (i.e. one semitone) the intervals in the twelve note system can be represented by integers. Table I summarizes each interval: its traditional nomenclature, consonance and dissonance properties, and number of semitones.

The nomenclature, based on conveniences for tonal music and practical aspects of note manipulation, can be specified as follows<sup>1,11</sup>:

- Intervals are seen first by the as number of steps between notes: first (unison), second, third, forth, fifth, sixth, seventh and eighth (octave). The ninth, tenth, eleventh, etc. are compound intervals made by one or more octaves plus a “simple” interval inside the octave, which characterizes the compound interval. The intervals are represented by numeric digits, e.g. 1, 3, 5 are a unison, a third and a fifth, respectively.
- Qualities of each interval: perfect consonances – i.e. unison, forth, fifth and octave – are ‘perfect’. The imperfect consonances – i.e. thirds and sixths – and dissonances – i.e. seconds and sevenths – can be major and minor. Exception for the tritone.
- The perfect fourth can be a perfect consonance or a dissonance according to the context and theoretical background. As a general rule, it can be considered a consonance except when it resolves into a fifth or a third with movimentation of just one note .
- Tritone is a dissonance in Western music because it characterizes the “dominant” chord in the tonal system (see subsection IV B) and represents the instability. Some cultures have the interval as a consonance, using it in melodies and chants as a stable interval.
- A major interval decreased by one semitone results in a minor interval. A minor interval increased by one semitone results in a major interval.
- A perfect interval (unison, perfect forth, perfect fifth, perfect octave), or a major interval (major second M2, major third M3, major sixth M6 or major seventh M7), increased by one semitone results in an augmented interval (e.g. augmented third aug3 with five semitones). The augmented forth is also called tritone (aug4 tri TT).
- A perfect interval or a minor interval (minor second m2, minor third m3, minor sixth m6 or minor seventh m7), decreased by one semitone results in a diminished interval. The diminished fifth is also called tritone (dim5 tri TT).
- An augmented interval increased by one semitone results in a ‘doubly augmented’ interval and a diminished interval decreased by one semitone results in a ‘doubly diminished’ interval.
- Notes played simultaneously configure a harmonic interval.
- Notes played as a sequence in time configure a melodic interval. The order of the notes: first the lowest note or the highest note, results in an ascending or descending interval, respectively.
- If the lower pitch is raised one octave, or if the upper pitch is lowered one octave, the interval is inverted. The sum of an interval and its inversion is 9 (e.g. m7 is inverted to M2:  $m7 + M2 = 9$ ). An inverted major interval results in a minor interval and vice-versa. An inverted augmented interval results in a diminished interval and vice-versa (inverting a doublyi-augmented results in a doubly-diminished and vice-verse, etc.). An inverted perfect interval results in a perfect interval as well.
- An interval higher than an octave is called a ‘compound interval’ and is classified like the interval between the same notes but in the same octave. They are also reached by adding a 7 to the interval: P11 is an octave plus a forth ( $7 + P4 = P11$ ), M9 is an octave plus a major second ( $7 + M2 = M9$ ).

The augmented/diminished intervals and the doubly augmented/doubly diminished intervals are consequences of the tonal system. Scale degrees (subsection IV A 3) are in fact different pitches, with specific uses and functions. Henceforth, in a *C flat* major scale, the tonic – first degree – is *C flat*, not *B*, and the leading tone – seventh degree – is *B flat*, not *A sharp* or *C double flat*. In a similar way, the second degree of a scale can be one semitone from first degree, like the leading tone (seventh degree at one ascending semitone from the first degree), where there is a diminished third between the two semitones of the seventh and second scale degrees<sup>14</sup>.

The descriptions presented summarizes the traditional theory of the music intervals<sup>14</sup>. The music piece *Intervalos entre alturas* explores these intervals in both independent and related manners. The source code is available online in the MASSA toolbox<sup>32</sup>.

## 3. Scales

A scale is an ordered set of pitches. Usually, scales repeat at each octave. The ascending sequence with all notes from the octave division in 12 equal intervals, separated by the ratio  $\varepsilon = 2^{\frac{1}{12}}$ , is known as the chromatic scale with equal temperament. There are 5 perfectly

TABLE I. Music intervals together with their traditional notation, basic classification for dissonance and consonance, and number of semitones. Unison, fifth and octave are the perfect (P) consonances. Major (M) and minor (m) thirds and sixths are the imperfect consonances. Minor seconds and major sevenths are the harsh dissonances. Major seconds and minor sevenths are the mild dissonances. Perfect fourth is a special case, being a perfect consonance when considered as an inversion of the perfect fifth and a dissonance or a imperfect consonance otherwise. The second special case is the tritone (A4 or aug4, d5 or dim5, tri, TT). This interval is consonant in some cultures. For tonal music, the tritone indicates dominant and seeks urgent resolution in a third or sixth. Due to this instability it is considered a dissonant interval.

| consonances                      |                      |                     |
|----------------------------------|----------------------|---------------------|
| perfects:                        | traditional notation | number of semitones |
| imperfects:                      | P1, P5, P8           | 0, 7, 12            |
|                                  | m3, M3, m6, M6       | 3, 4, 8, 9          |
| dissonances                      |                      |                     |
| fortes:                          | traditional notation | number of semitones |
| brandas:                         | m2, M7               | 1, 11               |
|                                  | M2, m7               | 2, 10               |
| special cases                    |                      |                     |
| consonance or dissonance:        | traditional notation | number of semitones |
| dissonance in Western tradition: | P4                   | 5                   |
|                                  | tritone, aug4, dim5  | 6                   |

symmetric divisions of the octave within the chromatic scale. These divisions are considered scales by easy and peculiar uses they provide. Considering integers  $e_i$  used as powers of the factor  $\varepsilon = 2^{\frac{1}{12}}$  to multiply lower note frequency ( $f = \varepsilon^{e_i} f_0$ ), the scales are as follows:

$$\begin{aligned}
\text{chromatic} &= E_i^c = \{e_i^c\}_0^{11} = \{0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11\} = \{i\}_0^{11} \\
\text{whole tones} &= E_i^t = \{e_i^t\}_0^5 = \{0, 2, 4, 6, 8, 10\} = \{2.i\}_0^5 \\
\text{minor thirds} &= E_i^{tm} = \{e_i^{tm}\}_0^3 = \{0, 3, 6, 9\} = \{3.i\}_0^3 \\
\text{major thirds} &= E_i^{tM} = \{e_i^{tM}\}_0^2 = \{0, 4, 8\} = \{4.i\}_0^2 \\
\text{tritones} &= E_i^{tt} = \{e_i^{tt}\}_0^1 = \{0, 6\} = \{6.i\}_0^1
\end{aligned} \tag{80}$$

Therefore, the third note of the whole tone scale with  $f_0 = 200\text{Hz}$  is  $f_3 = \varepsilon^{e_3} f_0 = 2^{\frac{6}{12}} \cdot 200 \approx 282.843\text{Hz}$ . These ‘scales’, or patterns, generate stable structures by their intern symmetries and can be repeated in an efficient and sustained way. Section IV G specially discusses symmetries. The music piece *Cristais* exposes each one of these scales, in both melodic and harmonic counterpart and their source code is part of MASSA toolbox.

Diatonic modes are:

aeolian = natural minor scale =

$$= E_i^m = \{e_i^m\}_0^6 = \{0, 2, 3, 5, 7, 8, 10\}$$

$$\text{locrian} = E_i^{mlo} = \{e_i^{mlo}\}_0^6 = \{0, 1, 3, 5, 6, 8, 10\}$$

ionian = major scale =

$$= E_i^M = \{e_i^M\}_0^6 = \{0, 2, 4, 5, 7, 9, 11\}$$

$$\text{dorian} = E_i^{md} = \{e_i^{md}\}_0^6 = \{0, 2, 3, 5, 7, 9, 10\}$$

$$\text{phrygian} = E_i^{mf} = \{e_i^{mf}\}_0^6 = \{0, 1, 3, 5, 7, 8, 10\}$$

$$\text{lydian} = E_i^{ml} = \{e_i^{ml}\}_0^6 = \{0, 2, 4, 6, 7, 9, 11\}$$

$$\text{mixolydian} = E_i^{mmi} = \{e_i^{mmi}\}_0^6 = \{0, 2, 4, 5, 7, 9, 10\} \tag{81}$$

They have only major, minor and perfect intervals. The unique exception is the tritone that is presented as an augmented fourth or a diminished fifth.

Diatonic scales follow a circular pattern of successive intervals *tone, tone, semitone, tone, tone, tone, semitone*. Thus, it is possible to write:

$$\begin{aligned}
\{d_i\} &= \{2, 2, 1, 2, 2, 2, 1\} \\
e_0 &= 0 \\
e_i &= d_{(i+\kappa)\%7} + e_{i-1} \quad \text{for } i > 0
\end{aligned} \tag{82}$$

with  $\kappa \in \mathbb{N}$ . For each mode there is only one value for  $\kappa \in [0, 6]$  by which  $\{e_i\}$  matches. For example, a brief inspection reveals that  $e_i^{ml} = d_{(i+2)\%7} + e_{i-1}^{ml}$ . Then,  $\kappa = 2$  for the lydian mode.

The minor scales has two additional forms, named melodic and harmonic:

$$\begin{aligned}
\text{natural minor} &= E_i^m = \{e_i^m\}_0^6 = \\
&= \{0, 2, 3, 5, 7, 8, 10\} \\
\text{harmonic minor} &= E_i^{mh} = \{e_i^{mh}\}_0^6 = \\
&= \{0, 2, 3, 5, 7, 8, 11\} \\
\text{melodic minor} &= E_i^{mm} = \{e_i^{mm}\}_0^{14} = \\
&= \{0, 2, 3, 5, 7, 9, 11, 12, 10, 8, 7, 5, 3, 2, 0\}
\end{aligned} \tag{83}$$

The different ascending and descending contour of the melodic minor scale is required by the leading tone (seventh and last degree, separated by one semitone from the octave, enhances tonic polarization) in the ascending trajectory, which is not necessary in the descending mode, and therefore it recovers the normal form. The harmonic scale presents the leading tone but does not avoid the augmented second between the sixth and seventh degrees, it does not consider the melodic trajectory<sup>31</sup>. Other scales can be represented in a similar form, like the pentatonic and the modes of limited transposition by Messiaen<sup>55</sup>.

Although not a scale, the harmonic series is often used as such. Besides, this point is a convenient moment to put it plain simple:

$$\begin{aligned}
H_i &= \{h_i\}_0^{19} = \\
&\{0, 12, 19 + 0.02, 24, 28 - 0.14, 31 + 0.2, 34 - 0.31, \\
&36, 38 + 0.04, 40 - 0.14, 42 - 0.49, 43 + 0.02, \\
&44 + 0.41, 46 - 0.31, 47 - 0.12, \\
&48, 49 + 0.05, 50 + 0.04, 51 - 0.02, 52 - 0.14\}
\end{aligned} \tag{84}$$

In this scale, the frequency of  $i$ -th note  $h_i$  the frequency of  $i$ -th harmonic  $f = \varepsilon^{h_i} f_0$  from the spectra generated by  $f_0$ . Nature exhibits this frequencies within sounds and with all kinds of distortions.

#### 4. Chords

The simultaneous occurrence of three or more notes is observed by means of chords. Chords are based on triads, specially in tonal music. Triads are built by two successive thirds within 3 notes: root, third and fifth. If the lower note of a chord is different from the root, this chord is an inverted chord as oposed to the chord in its root position. A closed position is any in which no chord note fits between two consecutive notes<sup>14</sup>, any non-closed position is an open position. In closed and fundamental position, and with fundamental in 0, triads are described as:

$$\begin{aligned}
\text{major triad} &= A_i^M = \{a_i^M\}_0^2 = \{0, 4, 7\} \\
\text{minor triad} &= A_i^m = \{a_i^m\}_0^2 = \{0, 3, 7\} \\
\text{diminished triad} &= A_i^d = \{a_i^d\}_0^2 = \{0, 3, 6\} \\
\text{augmented triad} &= A_i^a = \{a_i^a\}_0^2 = \{0, 4, 8\}
\end{aligned} \tag{85}$$

To have another third superimposed to the fifth, is sufficient to add 10 as the highest note in order to form a tetrad with minor seventh, or add 11 in order to form a tetrad with major seventh. Inversions and open positions can be obtained with the addition of  $\pm 12$  to selected component.

Incomplete triadic chords, with extra notes ('dirty' chords), and non-triadic are also common.

For general guidance:

- A fifth confirms the root (fundamental).
- Major or minor thirds, suggests major or minor chord qualities.
- Every tritone, specially if built between a major third and a minor seventh, tends to resolve in a third or a sixth.
- Note duplication is avoided. If duplication is needed, the preferred order is: the root, fifth, third and seventh.
- Note omission is avoided in the triad. If needed, the fifth is first considered for omission, than third and, if needed, the fundamental.
- It is possible to build chords with notes different from triads, specially if they respect a recurrent logic or musical concatenation that justifies these different notes.
- Chords built by successive intervals different of thirds – like fourths and seconds – are recurrent in compositions of advanced tonalism or experimental character.
- The repetition of chord successions (or their characteristics) fixes a trajectory and makes possible the introduction of exotic formations without incoherence.

#### B. Atonal and tonal harmonies, harmonic expansion and modulation

Omission of basic tonal chaining is the key to obtain modal and atonal harmonies. In absence of these minimal tonal structures, the harmony is considered modal if the notes match with some diatonic scale (see equations 81) or if notes are presented in a small number. If basic tonal chainings are absent and notes does not match any diatonic scale and are diverse and dissonant (by relation

with each other) enough to avoid reduction by polarization, the harmony is atonal. In this classification, the modal harmony is not tonal or atonal and is reduced to the incidence of notes within the diatonic scale (or simplifications) and to the absence of tonal structures. It is possible to note, following this concept, that atonal harmony is hard to be realized and, indeed, no matter how dissonant and diverse is a set of notes, tonal harmonies arise if not avoided<sup>56</sup>.

## 1. Atonal harmony

In fact, the techniques around atonal music aim at structures for avoiding direct relation with modes and tonality. Manifesting such structures are of such difficulty that the dodecafonism arouse. The purpose of dodecafonism is to use a set of notes (ideally 12 notes), and to perform each note, one by one, in the same order. In this context, the tonic becomes difficult to be established. Nevertheless, the Western hearing searches for tonal traces in music and easily finds them by unexpected and tortuous paths.

The use of dissonant intervals (specially tritones) without resolution, reinforces the absence of tonality. Considering this context, while creating a music piece, it is possible to:

- Repeat notes. By considering immediate repetition as an extension of previous incidence, the use of same note in sequence does not add relevant information.
- To play adjacent notes at the same time, making harmonic intervals and chords.
- Present durations and pauses with liberty, respecting notes order.
- Vary by extension, transposition, translation, inversion, retrograde and retrograde inversion. See subsections IVE and IVI for more details.
- Account for the presence of note structures, it is possible to use variations in orchestration, articulation, spatialization, among others.

The atonal harmony can be observed, paradigmatically, within these presented conditions (a simple dodecaphonic model). Most of what was written by great dodecaphonic composers, like Alban Berg and even Schoenberg, had the purpose of mixing tonal and atonal techniques.

## 2. Tonal harmony

In the XX century, rhythmic music and with emphasis on sonorities/timbres, extended the concepts of tonality and harmony. Hence, tonal harmony has strong influence

in artistic movements and commercial venues. In addition, the dodecafonism itself is considered of tonal nature because it denies tonal characteristics of polarization.

In tonal or modal music, chords – like the ones listed in equations 85– built on the root note of each degree of a scale – such as listed in equations 81 – form the pillars of the harmonic field. Music harmony aims at studying the incidence of chord progressions and chaining rules. Even a monophonic melody generates harmonic fields, making it possible to observe the suggested chords at each passage.

In the traditional tonal music, a scale has its tonic (first degree) on any note, and can be major (with the same notes of Ionian mode) or minor (same notes as Eolian mode, 'natural minor', which has both harmonic and melodic versions, see equations 83). The scale is the base for triads, each with its root in a degree:  $\hat{1}, \hat{2}, \hat{3}, \hat{4}, \hat{5}, \hat{6}, \hat{7}$ . To build triads, the third and the fifth notes above the root is considered together with the root (or fundamental).  $\hat{1}, \hat{3}, \hat{5}$  is the first degree chord, built on top of scale's first degree and central for tonal music. The chords of the fifth degree  $\hat{5}, \hat{7}, \hat{2}$  ( $\hat{7}$  sharp when a minor scale) and of the fourth degree  $\hat{4}, \hat{6}, \hat{1}$  are secondary. After these, other degrees are considered. The 'traditional harmony' comprises conventions and stylistic techniques to create chainings with such chords<sup>31</sup>.

The 'functional harmony' ascribes functions to these three central chords and tries to understand their use by means of these functions. The chord built on top of first degree is the **tonic** chord ( $T$  or  $t$  for a major or minor tonic, respectively) and its function (role) consists of maintaining a center, usually referred as a "ground" for the music. The chord built on the fifth degree is the **dominant** ( $D$ , the dominant is always major) and its function is to lean for the tonic (the dominant chord asks for a conclusion and this conclusion is the tonic). Thus, the dominant chord guides the music to the tonic. The triad built under the fourth degree is the **subdominant** ( $S$  or  $s$  for a major or minor subdominant, respectively) and its function is to deviate the music from the tonic. The process aims at confirming the tonic using tonic-dominant-tonic chains which are expanded by using other chords in different ways.

The remaining triads are associated to these three most important chords. In the major scale, the associated relative (relative tonic  $Tr$ , relative subdominant  $Sr$  and relative dominant  $Dr$ ) is the triad built a third below, and the associated counter-relative (counter-relative tonic  $Ta$ , counter-relative subdominant  $Sa$  and the counter-relative dominant  $Da$ ) is the triad built in a third above. In the minor scale the same happens, but the triad a third below is called counter-relative ( $tA$ ,  $sA$ ) and the triad a third above is called relative ( $tR$ ,  $sR$ ). The precise functions and musical effects of these chords are controversial. Table II shows relations between the triads built in each degree of the major scale.

The dominant counter-relative should form a minor chord. It explains the change in the fourth degree by a

TABLE II. Summary of tonal harmonic functions of the major scale. Tonic is the music center, the dominant goes to the tonic and the subdominant moves the music away from the tonic. The three chords can, at first, be freely replaced by their respective relative and counter-relatives.

| relative                    | main chord of the function               | counter-relative                |
|-----------------------------|--|---------------------------------|
| $\hat{6}, \hat{1}, \hat{3}$ | tonic: $\hat{1}, \hat{3}, \hat{5}$       | $\hat{3}, \hat{5}, \hat{7}$     |
| $\hat{3}, \hat{5}, \hat{7}$ | dominant: $\hat{5}, \hat{7}, \hat{2}$    | $[\hat{7}, \hat{2}, \hat{4}\#]$ |
| $\hat{2}, \hat{4}, \hat{6}$ | subdominant: $\hat{4}, \hat{6}, \hat{1}$ | $\hat{6}, \hat{1}, \hat{3}$     |

semitone above  $\hat{7}\#$ . The diminished chord  $\hat{7}, \hat{2}, \hat{4}$ , is generally considered a ‘dominant seventh chord with omitted root’<sup>57</sup>. In minor mode, there is a change in  $\hat{7}$  by an ascending semitone to make possible the existence of a unique semitone separating  $\hat{7}$  and  $\hat{1}$ , and making the dominant possible (that should be major and goes to the tonic). In this way, the dominant is always major, for both major and minor scale and, because of that, even in a minor tone, the relative dominant remains a third below and in the counter-relative, a third above.

### 3. Tonal expansion: individual functions and chromatic mediant

Each one of these chords can be confirmed and developed by performing their individual dominant or subdominant, which are the triads based a fifth above or a fifth below, respectively. These individual dominants and subdominants, in the same way, have also subdominants and individual dominants of their own. Given a tonality, any chord can occur, no matter how distant it is from the harmonic field and from the notes within the scale. The unique condition is that the occurrence presents a coherent trajectory of dominants and subdominants to the original tonality.

Mediants, or ‘chromatic mediant’, are two for each chord: the upper mediant, formed by the root at the third of original chord; and the lower mediant, formed by the fifth at the third of original chord. Both chords also are formed by a third, but with a chromatic alteration regarding the original chord. If two chromatic alterations exist, i.e. two notes altered by one semitone each regarding the original chord, it is a ‘doubly chromatic mediant’. Again, there are two forms for each chord: the upper form, with a third in the fifth of the original triad; and the lower form, with a third in the root of the original triad. It is interesting to observe that a major chord has major chromatic and doubly chromatic mediant. A minor chord has minor chromatic and doubly chromatic mediant. (Recall that relatives and counter-relatives have opposite major-minor quality.) This relation between chords is considered advanced tonalism, sometimes even considered expansion and dissolution of tonalism, with strong and impressing effects although perfectly consonant triads. Chromatic mediant are used since the end of Romanticism by Wagner, Litz, Richard

Strauss, among others, and are quite simple to be realized<sup>31,58</sup>.

## 4. Modulation

Modulation is the change of key (tonic, or tonal center) in a music. It is characterized by observance of start and end keys, and transition artifacts. Keys are always conceived as related by fifths and their relative and counter-relatives. Ways to perform modulation:

- Transposing the discourse to a new key, without any preparation. It is a common Baroque procedure although incident at other periods as well. Sometimes it is called phrasal modulation or unprepared modulation.
- Careful use of an individual dominant, and perhaps also the individual subdominant, to confirm change in key and harmonic field.
- Use of chromatic alterations to reach a chord in the new key starting from a chord in the previous key. Called chromatic modulation.
- Featuring an unique note, possibly repeated or suspended with no accompaniment, common to start and end keys, constitute a peculiar way to introduce the new harmonic field.
- Changing the function, without changing the notes, of a chord to contemplate a new key. This procedure is called enharmony.
- Maintaining the tonal center and changing the key quality from major to minor (or vice-verse) comprehends a parallel modulation. Keys with same tonic and different quality is known as homonyms.

The dominant has great importance and is a natural pivot into modulations, which leads to the circle of fifths<sup>31,57–59</sup>. Other inventive ways to modulate are possible, to point but one common example, the minor thirds tetrad ( $E_i^{tm}$  in equations 80) can be sustained to bridge to other tonalities, with the ease of its both tritones. The music piece *Acorde cedo* explores these chord relations. Its code is online as part of MASSA toolbox<sup>32</sup>.

## C. Counterpoint

Counterpoint is defined as the conduction of simultaneous melodic lines, or “voices”. The bibliography covers systematic ways to conduct voices, leading to scholastic genres like canons, inventions and fugues. It is possible to summarize counterpoint rules, and it is known that Beethoven – among others – also outlined such a briefing of counterpoint.

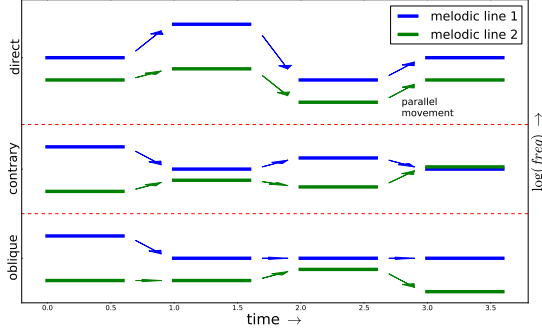


FIG. 23. Different motions of counterpoint aiming to preserve independence between voices. There are 3 types of motion: direct, contrary and oblique. The parallel motion is a type of direct motion.

The purpose of counterpoint is to conduct voices in a way that they sound independent. In order to do that, the relative motion of voices (in pairs) is crucial and categorized as: direct, oblique and contrary (see Figure 23). The parallel motion is an direct motion. The gold rule here is to take care of the direct motions, avoiding them when leading to a perfect consonance. The parallel motion should occur only between imperfect consonances and no more than three consecutive times. Dissonances can be unadmitted or used when followed and preceded by consonances of joint degrees, i.e. neighbor notes in a scale. The motions that lead to a neighbor note in the scale sound coherent. When having 3 or more voices, the melodic importance lies on the higher and lower voices, in this order<sup>60–62</sup>.

These rules were used in the music piece *Conta ponto*, the source code is available online in MASSA toolbox.

#### D. Rhythm

Rhythmic notion is dependent on events separated by durations<sup>14</sup>. These events can be heard individually if spaced by at least 50–63ms. For the temporal separation between them to be perceived as duration, the period should even a bit larger, around 100ms<sup>17</sup>. It is possible to summarize duration heard as rhythm or pitch, and its transition, as in table III<sup>17,63</sup>.

The duration transition span is minimized because the limits are not well defined. In fact, the duration where someone begins to perceive a fundamental frequency or a separation between occurrences, depends on the listener and sonic characteristics<sup>1,17</sup>.

The rhythmic metric is commonly based on a basic duration called pulse. The pulse typically comprehends durations between 0.25 – 1.5s (240 and 40BPM, respectively). In music education and cognitive studies, it is common to associate these range of frequencies to the durations of the heart beat, movements of inspiration and

expiration and steps of a walking or running person<sup>1,14</sup>.

The pulse is subdivided into equal parts and is also repeated in sequence. These relations (division and concatenation) usually follow relations of small order integers. The occurrences, in written and ethnic music, of musical pulse division (and their sequential grouping at time), are, in ascendent disposition: 2, 4 and 8, after that 3, 6 (two groups of 3 or 3 groups of 2) and 9 and 12 (three and 4 groups of 3). At last, the prime numbers 5 and 7, complementing 1-9 and 12. Other metrics are less common, like division and grouping in 13, 17, etc, and are mainly used in contexts of experimental music and classical music of XX and XXI. No matter how complex they seem, metrics are common compositions and decompositions of 1-9 equal parts<sup>1,51</sup>. A schematic exposition is shown in figure 24.

Dual relations (simple measures and binary divisions) are frequent in dance rhythms and celebrations, and are called “imperfect”. Ternary relations are common in ritualistic music and is related to the sacred, and are called “perfect”.

Strong units (accents) fall in the ‘head’ of bars, the downbeats, of which the first in every bar is the greatest. The head of a unit is the first part of the subdivision. In binary divisions (2, 4 and 8), strong units alternate with weak units (e.g. division in 4 is: strong, weak, average strong, weak). In ternary divisions (3, 6 and 9) two weak units succeeds the downbeat (e.g. division in 3 is: strong, weak, weak). Division in 6 is considered compound but can also occur as a binary division. Binary division units which suffers a ternary division yields two units divided in tree units each: strong (subdivided in strong, weak, weak) and weak (subdivided in strong, weak, weak). Another way to perform the division in 6 is ternary division units that divides as binary, resulting in: a strong unit (subdivided in strong and weak) and two weak units (subdivided in strong and weak for each).

The accent in the weak beat is the backbeat, whereas a note starting in a weak beat persisting across the strong beat is the syncope.

Notes can occur inside and outside of these ‘musical metric’ divisions. In most well behaved cases, notes occur exactly on these divisions, with greater incidence strong beat attacks. In extreme cases, time metric cannot be perceived<sup>1</sup>. Small variations along the grid are crucial for musical interpretation and different styles<sup>13</sup>.

Be the pulse the grouping level  $j = 0$ , level  $j = -1$  is the first pulse subdivision, level  $j = 1$  is the first pulse agglomeration and so on. In this way,  $P_i^j$  is the  $i$ -th unit at grouping level  $j$ :  $P_{10}^0$  is the tenth pulse,  $P_3^1$  is the third grouped unit (and, possibly, the third measure),  $P_2^{-1}$  is the second part of pulse subdivision.

The limits of  $j$  are of special interest: pulse divisions are durations perceivable as rhythm; furthermore, the pulses sum, at its maximum, a music or a cohesive set of musics. In other words, a duration given by  $P_i^{\min(j)}$ ,  $\forall i$ , is greater than 50ms and the durations summed together  $\sum_{\forall i} P_i^{\max(j)}$  are less than a few minutes or, at most, a



TABLE III. Transition of durations heard individually until it turns into pitch.

|                | perception zone of duration as rhythm      | transition                                       | -   |
|----------------|--|--|---|
| duration (s)   | ... 32, 16, 8, 4, 2, 1, 1/2, 1/4, 1/8,     | $\frac{1}{16} = 62, 5ms$ , $\frac{1}{20} = 50ms$ | $\frac{1}{40}$ $\frac{1}{80}$ $\frac{1}{160}$ $\frac{1}{320}$ $\frac{1}{640}$ ... |
| frequency (Hz) | ... 1/32, 1/16, 1/8, 1/4, 1/2, 1, 2, 4, 8, | 16, 20   | 40 80 160 320 640 ...   |
|                | -  | transition                                       | perception zone of duration as pitch  |

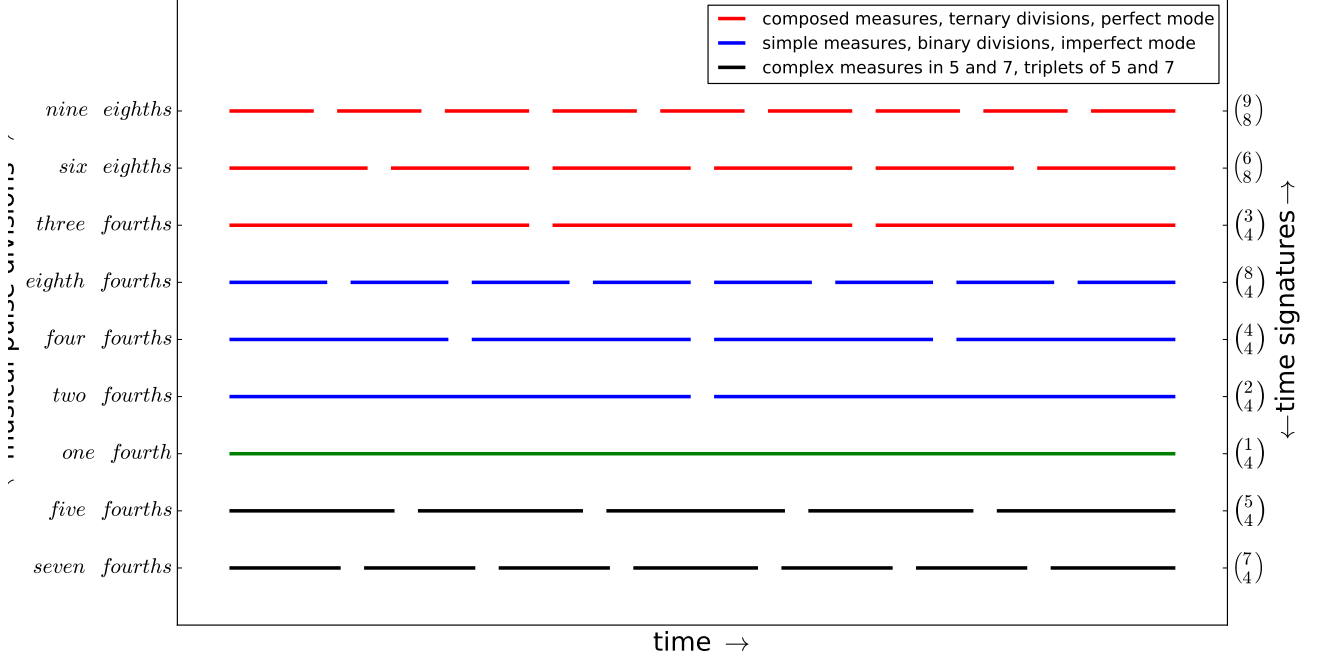


FIG. 24. Divisions and groupings of the musical pulse for establishing a metric. Divisions of the quarter note, regarded as the pulse, is presented in the left. The time signature yielded by groupings of the music pulse is presented in the right.

few hours.

Each level  $j$  has some sections  $i$ . When  $i$  has three different values (or multiple of three) there is a perfect relation. When  $i$  has only two, four or eight different values, than there is an imperfect relation, as shown in figure 24.

Any unit (note) of a given musical sequence with a time metric can be unequivocally specified as:

$$P_{\{i_k\}}^{\{j_k\}} \quad (86)$$

where  $j_k$  is the grouping level and  $i_k$  is the unit itself.

As an example, consider  $P_{3,2,2}^{-1,0,1}$  as the third subdivision  $P_3^{-1}$  of the second pulse  $P_2^0$  and of the second pulse group  $P_2^1$  (possibly second measure). Each unit  $P_i^j$  can be associated with a sequence of temporal samples  $T_i$  that comprehends a note.

The music piece *Poli Hit Mia* explores different metrics (available as part of MASSA toolbox).

### E. Repetition and variation: motifs and larger units

Given the basic musical structures, both frequential (chords and scales) and rhythmic (simple, compound and complex beat divisions and agglomerations), it is natural to present these structures in a coherent and meaningful way<sup>64</sup>. For such, the concept of arcs is fundamental: by departing from a place and returning, an arc is made. The audition of melodic and harmonic lines is permeated by musical arcs due to cognitive nature of the musical hearing. The note can be considered the smaller arc, and each motif and melody as an arc as well. Each beat and subdivision, each measure and music section, constitutes an own arc. A music in which the arcs do not present consistency with one another, can be understood as a music with no coherence. Coherence impression comes, mostly, from the skilled handling of arcs in a music piece.

Musical arcs are abstract structures and liable of basic operations. A spectral arc, like a chord, can be inverted, enlarged and permuted, to cite some examples. Temporal arcs, like a melody, a motif, a measure or a note, are also capable of variations. Remembering that

$S_j = \{s_j = T_i^j = \{t_i^j\}_0^{\Lambda_j-1}\}_0^{H-1}$  is a sequence of  $H$  musical events  $s_j$ , each event with its  $\Lambda_j$  samples  $t_i^j$  (refer to the beginning of this section IV), the basic techniques can be described as:

- Temporal translation is a displacement  $\delta$  of a specific material to another instant  $\Gamma' = \Gamma + \delta$  of the music. It is a variation that changes temporal localization in a music:  $\{s'_j\} = \{s_j^{\Gamma'}\} = \{s_j^{\Gamma+\delta}\}$  where  $\Gamma$  is the duration between the beginning of the piece (or considered section) and the first event  $s_0$  of original structure  $S_j$ . Observe that  $\delta$  is the time offset of the displacement.
- Temporal expansion or contraction is a change in duration of each arc by a factor  $\mu$ :  $s_j^{\Delta} = s_j^{\mu_j \cdot \Delta}$ . Possibly,  $\mu_j = \mu$  is constant.
- Temporal reversion consists of generating a sequence with elements in reverse order of the original sequence  $S_j$ , thus:  $S'_j = \{s'_j\}_0^{H-1} = \{s_{(H-j-1)}\}_0^{H-1}$ .
- Pitch translation, or transposition, is a displacement  $\tau$  of the material to a different pitch  $\Xi = \Xi_0 + \tau$ . It is a variation that changes pitch localization of material:  $\{s'_j\} = \{s_j^{\Xi'}\} = \{s_j^{\Xi+\tau}\}$  where  $\Xi_0$  is a reference value, such as the pitch of a section  $S_j$  or of the first event  $s_0$ . If  $\tau$  is given in semitones, the displacement in frequency is  $\tau_f = f_0 \cdot 2^{\frac{\tau}{12}}$  where  $f_0$  is the reference frequency value:  $f_0 = \Xi_{f_0} Hz \sim \Xi$  absolute value of pitch. The frequency of any pitch value is  $f = f_0 \cdot 2^{\frac{\tau}{12}}$  s; and the pitch of any frequency is:  $\Xi = \Xi_0 + 12 \log_2 \left( \frac{f}{f_0} \right)$ . In the MIDI protocol,  $\Xi_{f_0} = 55 Hz$  while pitch  $\Xi_0 = 33$  marks a *A 1*. Another good MIDI reference is  $\Xi_{f_0} = 440 Hz$  and  $\Xi_0 = 69$ . The difference  $(\Xi_1 - \Xi_2)$  is measured in semitones.  $\Xi$  is not a measure in semitones:  $\Xi = 1$  is not a semitone, it is a note with an audible frequency as rhythm, with less than 9 occurrences each second (see table III).
- Interval inversion is: 1) the inversion of note pitch order, within octave equivalence, such as exposed in IV A 2, and  $S'_j = \{s'_j\}_0^{H-1} = \{s_j^{\varepsilon_j \cdot f_0}\}$  with selective  $s_j = 2, 1/2$ , or 1; or 2) the inversion of interval orientation. In the former case, the number of semitones is preserved in the “strict inversion”:  $S'_j = \{s'_j\}_0^{H-1} = \{s_j^{-\varepsilon_j \cdot f_0}\}$ , where  $\varepsilon_j$  is the factor between the frequency of event  $s_j$  and the frequency of  $s_0$ . The inversion is said tonal if the distances are considered in terms of the diatonic scale  $E_k$ :  $S'_j = \{s'_j\}_0^{H-1} = \left\{ s_j^{2^{\left( \frac{12 - e(7-j_e)}{12} \right)} \cdot f_0} \right\}_0^{H-1}$  where  $j_e$  is the index in  $E_k$  of the note  $s_j$ .

- Rotation of musical elements is the translation of all elements a number of positions ahead or behind, with the care to fill empty positions with events out of convenient slots. Thus, a rotation of  $\tilde{n}$  positions is  $S'_n = S_{(n+\tilde{n}) \% H}$ . If  $\tilde{n} < 0$ , it is enough to use  $\tilde{n}' = H - \tilde{n}$ . It is reasonable to associate  $\tilde{n} < 0$  (events advance) with the clockwise rotation and  $\tilde{n} < 0$  (elements delay) with the anti-clockwise rotation. More information about rotations is presented in subsection IV G.
- The insertion and removal of material in  $S_j$  can be ornamental or structural:  $S'_j = \{s'_j\} = \{s_j \text{ if condition A, otherwise } r_j\}$ , for any music material  $r_j$ , including silence. Elements of  $R_j$  can be inserted at the beginning, like a prefix in  $S_j$ ; at the end, as a suffix; or at the middle, dividing  $S_j$  and making it the prefix and suffix. Both materials can be mixed in a variety of ways.
- Changes in articulation, orchestration and spatialization,  $s'_j = s_j^{*j}$ , where  $*j$  is the new characteristic incorporated by the element  $s'_j$ .
- Accompaniment. Both orchestration and melodic lines presented when  $S_j$  occurs can suffer modifications and be considered as a variation of  $S_j$  itself.

With these presented processes, others are derived, as, for example, the inverted retrograde, a temporal contraction with an external suffix, etc. Music structures resonate in the human cognitive system due to the own nature of thought. In its many facets, an idea is observed with the same number of elements and connective aspects between them. The music, when tuned with mental structures, raises related impressions. In this way, a whole process of mental and neurological activity is unleashed, responsible by the feelings, memories and imaginations, typical of a diligent music listening. This cortical activity is critical to musical therapy, known by its utility in cases of depression and neurological injury. It is known that regions of the human brain responsible by sonic processing, are also used for other activities, such as linguistic and mathematical<sup>1,65</sup>.

Paradigmatic structures guide the creation of new musical material. One of the most central of them is the tension/relaxation dipole. Traditional dipoles are relate to tonic/dominant, repetition/variation, consonance/dissonance, coherence/rupture, symmetry/asymmetry, equivalence/difference, arrival/departure, near/far, stationary/moving, etc. Ternary relations tend to relate to the circle and to unification. The lucid ternary communion, ‘modus perfectus’, opposes to the passionate dichotomic, ‘modus imperfectus’. Hereafter, there is an exposition dedicated to directional and cyclic arcs.

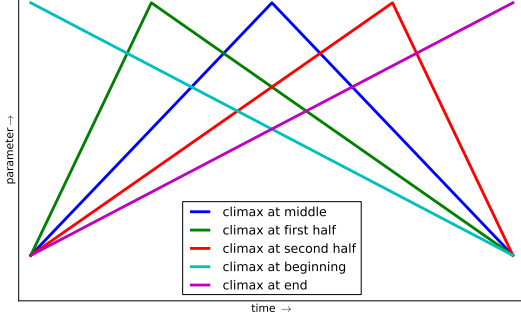


FIG. 25. Canonical distinctions of musical climax in a given melody and other domains. Considered possibilities are: climax at beginning, climax at first half, climax at middle, climax at second half and climax at the end. The x and y-axis is not properly specified since the parameters can be non-existent, as in a reference structure.

### F. Directional structures

The arcs can be decomposed in two convergent sequences: the first reaches the apex and the second returns from apex to start region. This apex is called climax by traditional music theory. It is possible to differ between arcs whose climax are placed at the beginning, middle, end, and at the first and second half of the considered duration. These structures are shown in figure 25. The varying parameter can be unexistent, a case in which the arc consists only of a reference structure<sup>16</sup>.

Consider the  $S_i = \{s_i\}_0^{H-1}$  increasing sequence. The sequence  $R_i = \{r_i\}_0^{2H-2} = \{s_{(H-1-|H-1-i|)}\}_0^{2H-2}$  presents perfect specular symmetry, i.e. the second half is the mirrored version of the first. In musical terms, the climax is in the middle of the sequence. It is possible to modify this by using sequences with different sizes. All the mathematics of sequences, already well established and taught routinely in calculus courses, can be used to generate these arcs<sup>16,66</sup>. Theoretically, when applied to any characteristic of musical events, these sequences produce arcs, since they imply a deviation and return of an initial parametrization. Henceforth, it is possible for an only sequence to have numerous distinct arcs, with different sizes and climax. This is an interesting and useful resource, and the correlation of arcs yields coherence<sup>58</sup>.

In practice, and historically, there is special incidence and use of the golden ratio. The Lucas sequence allows the generalization of Fibonacci sequence, making it easy to follow. Given any two numbers  $x_0$  and  $x_1$ , the Lucas sequence can be obtained as:  $x_n = x_{n-1} + x_{n-2}$ . The greater  $n$  is, the more  $\frac{x_n}{x_{n-1}}$  approaches the golden ratio: 1.61803398875.... The sequence converges fast even with high discrepant initial values. Being  $x_0 = 1$  and  $x_1 = 100$ , and  $y_n = \frac{x_n}{x_{n+1}}$  an auxiliary sequence, the error for the fraction of first values with respect to the golden ratio

is, approximately,  $\{e_n\} = \{100 \frac{y_n}{1.61803398875} - 100\}_1^{10} = \{6080.33, -37.57, 23, -7.14, 2.937, -1.09, 0.42, -0.1601, 0.06125, -0.02338\}$ . The Fibonacci sequence presents exactly the same error progression, but starts at the second step of the most discrepant case ( $\frac{1}{1} \approx \frac{100+1=101}{100}$ ).

The music piece *Dirracional* exposes the use of arcs into directional structures. Its source code is available online as part of MASSA toolbox<sup>32</sup>.

### G. Cyclic structures

The philosophical understanding that human thought is founded on the concept of similarities and differences (e.g. perceived in stimuli and objects), places symmetries at the core of cognitive processes<sup>67</sup>. Mathematically, symmetries are algebraic groups, and a finite group is always isomorphic to a permutation group (by Cayley's theorem). In a way, this states that permutations represent any symmetry in a finite system<sup>7</sup>. In music, permutations are ubiquitous and present in scholastic techniques, which confirms their central role. The successive application of permutations generates cyclic arcs<sup>15,68,69</sup>. Two academic studies were dedicated to generating musical structures<sup>70,71</sup>.

Any permutation set can be used as a generator of algebraic groups<sup>69</sup>. The properties defining a group  $G$  are:

$$\begin{aligned}
 \forall p_1, p_2 \in G &\Rightarrow p_1 \bullet p_2 = p_3 \in G && \text{(closure property)} \\
 \forall p_1, p_2, p_3 \in G &\Rightarrow (p_1 \bullet p_2) \bullet p_3 = p_1 \bullet (p_2 \bullet p_3) && \text{(associativity property)} \\
 \exists e \in G : p \bullet e = e \bullet p \quad \forall p \in G &&& \text{(identity element existence)} \\
 \forall p \in G, \exists p^{-1} : p \bullet p^{-1} = p^{-1} \bullet p = e &&& \text{(inverse element existence)}
 \end{aligned} \tag{87}$$

From the first property follows that two permutations act as one permutation. In fact, it is possible to apply a permutation  $p_1$  and another permutation  $p_2$ , and, comparing both initial and final orderings, observe another permutation  $p_3$ .

Every element  $p$  operated with itself a sufficient number of times  $n$  reaches the identity element  $p^n = e$  (otherwise the group would be infinite, generated by  $p$ ). The lower  $n : p^n = e$  is the element order. Thus, a finite permutation  $p$ , successively applied, reaches the initial ordering of its elements, and makes a cycle. This cycle, if used for parameters of musical notes or structures, yields a cyclic musical arc.

These arcs can be established by using a set of different permutations. As a historic example, the *change ringing* tradition conceives music through bells played one after other and then played again, but in a different order. This process repeats until it reaches the initial ordering.

The set of different traversed orderings is a *peal*. Table IV presents a traditional *peal*, named “*Hunt*”, for 3 bells (1, 2 and 3), which explores all possible orderings. Each line indicates one bell ordering to be played. Permutations occur between each line. In this case, music structure consists of permutations itself and some different permutations operate in the cyclic behavior.

TABLE IV. Change Ringing: Traditional *peal* for 3 bells. Permutations occur between each line. Each line is a bell ordering and each ordering is played at a time.

```

1 2 3
2 1 3
2 3 1
3 2 1
3 1 2
1 3 2
1 2 3

```

The use of permutations in music can be summarized in the following way: with  $S_i = \{s_i\}$  a sequence of musical events  $s_i$  (e.g. notes), and a permutation  $p$ .  $S'_i = p(S_i)$  comprises the same elements of  $S_i$  but in a different order. Permutations has two notations: cyclic and natural. The natural notation basically comprehends the order of the resulting indexes from the permutation. Thus, agreed the original ordering given the sequence of its indexes [0 1 2 3 4 5 ...], the permutation is noted by the sequence it produces (ex. [1 3 7 0 ...]). In the cyclic notation, a permutation consists of swaps by elements and its posteriors, and the last element by the first one. E.g. (1, 2, 5)(3, 4) in cyclic notation is equivalent to [0, 2, 5, 4, 3, 1] in natural notation.

In the auralization of a permutation, it is not necessary to permute elements of  $S_i$ , but only some characteristic. In this way, being  $p$  a permutation and  $S_i$  a sequence of basic notes as exposed in the end of section II F, the sequence  $S'_i = p^f(S_i) = \{s_i^{p(f)}\}$  consists of the same music notes, respecting same order and maintaining the same characteristics, but with the fundamental frequencies permuted following the pattern that  $p$  presents.

Two subtleties of this procedure. First, a permutation  $p$  is not restricted to involve all elements of  $S_i$ , i.e. it can operate in a subset of  $S_i$ . Second, not all elements  $s_i$  need to be executed at each access to  $S_i$ . To exemplify, be  $S_i$  a sequence of music notes  $s_i$ . If  $i$  goes from 0 to  $n$ , and  $n > 4$ , at each measure of 4 notes it is possible to execute the first 4 notes. The other notes of  $S_i$  can occur in measures where permutations allocate those notes to the first four events  $s'_i$  of  $S'_i$ .

It is inalienable to each permutations  $p_i$  described above: 1) note dimensions where it operates (frequency, duration, *fades*, intensity, etc); and 2) the period of incidence (how much consults before a permutation is applied). During realization of notes in  $S_i$ , an easy and coherent form is to execute the first  $n$  notes, the execution of disjoint sets of  $S_i$  equals modifying the permutation and executing the first  $n$  notes.

The MASSA toolbox presents a computational implementation that isolates permutation application to sonic characteristics, in order to deliver musical structures<sup>32,70,71</sup>.

## H. Musical idiom?

In numerous studies, there are models and explorations of a ‘musical language’, many can somewhat be considered ‘linguistics applied to music’ and some even discern different ‘musical idioms’<sup>31,58,63,72</sup>. In a simple way, a musical idiom is the result of chosen material together with repetition of elements and of relations between elements, along a music piece. In these matters, dichotomies are prominent, as explained in subsection IV E: repetition and variation, relaxation and tension, equilibrium and instability, consonance and dissonance, etc.

## I. Musical usages

First, the basic note was defined and characterized in quantitative terms (section II). Next, internal note composition was addressed and both internal transitions and immediate sonic treatment were understood (section ??). Finally, this section aims at organizing these notes in music. The gamma of resources and consequent infinitude of praxis possibilities is a typical and highly relevant situation for artistic contexts<sup>12,31</sup>.

There are studies for each of the presented resources. For example, it is possible to obtain ‘dirty’ triadic harmonies (with notes out of the triad) by superposition of perfect fourths. Another interesting example is the simultaneous presence of rhythms in different metrics, constituting what is called *polyrhythm*. The music piece *Poli-hit mia* explores these simultaneous metrics by impulse trains convolved with notes. Its source code is online as part of MASSA toolbox<sup>32</sup>.

Microtonal scales are important for 20th century music<sup>54</sup> and present diverse remarkable results, e.g. fourths of tone in the Indian music. The musical sequence *MicroTom* explores these resources, including microtonal melodies and microtonal harmonies with many notes in a very reduced note scope. Its code is available online in MASSA toolbox<sup>32</sup>.

As in subsection III F, relations between parameters are powerful ways to acquire musical pieces. The number of permuted notes can vary during the music, revealing relationship with piece duration. Harmonies can be made from triads (eqs. 85) with replicated notes at each octaves and more numerous as minor the depth and frequency of vibratos (eqs. 55, 56, 57, 58, 59), among other uncountable possibilities.

The symmetries at octave divisions (eqs. 80) and the symmetries presented as permutations (table IV and eqs. 87) can be used together. In the music piece *3 trios*

this association is done in a systematic way in order to enable a dedicated listening. This is an instrumental piece, not included as a source code but available online<sup>73</sup>.

*PPEPPS* (Pure Python EP: Projeto Solvente) is an EP synthesized using resources presented in this study. With minimal parametrization, the scripts generate complete musical pieces, allowing easy composition of sets of musics. A simple script of a few lines specifies music delivered as 16 bit 44.1kHz PCM files (WAVE). This facility and technological demystification create aesthetical possibilities for both sharing and education.

## V. CONCLUSIONS AND FURTHER DEVELOPMENTS

This concise presentation relates basic musical elements to digital sound. The reader is invited to access *Scripts* and *MASSA*, where these relations are computationally implemented. The didactic report along the paper and the supplied scripts eases and encourages the use of the proposed framework.

The possibilities provided by these results include psychoacoustic experiments, interfaces for the generation of noise and other high fidelity sounds and musical structures, and for artistic and didactic purposes. The incorporation of programming skills is facilitated by visual aids, which was explored by *live-coding* practices and courses based on this framework, with success.

This work systematically investigates the parameterization issues (like the tremolo, ADSR, etc.) in a high fidelity, which has significant artistic utility. Such detailed analytical descriptions, together with the computational implementations, have not been covered before in the literature, such as reviewed in <sup>3</sup>.

The free software license and online availability of the exposed content as hypertext, with the respective codes and sonic examples, strongly facilitates future collaborations and the generation of sub-products in a co-authorship fashion. As consequence, the expansion of *MASSA* is favoured and straightforward, easing new implementations and musical pieces development.

In addition, this framework permitted the formation of interests groups, with topics such as music creativity and computer music. Specially, the project *labMacambira.sf.net* groups Brazilian and foreign co-workers in order to offer relevant contributions in diverse areas like Digital Direct Democracy, georeferencing techniques, artistic and educational activities. Some of these reports are available online<sup>3</sup>. There are more than 700 videos, written documents, original software applications and contributions in well-known external softwares such as Firefox, Scilab, LibreOffice, GEM/Puredata, to name a few<sup>74-76</sup>.

Future work includes application of these results in machine learning and artificial intelligence methods for the generation of appealing artistic materials. Some psychoacoustical effects were detected, which needs validation and should be reported, specially with<sup>7</sup>. Other foreseen

advances are: JavaScript version of *MASSA* toolbox, better hypermedia deliverables of this framework, use guides for different ends (musical composition, psychophysical experiments, sound synthesis, education, etc), creation of musical pieces and open experiments to be studied with EEG recordings and further analytical specification of musical elements in the discrete-time representation of sound as community feedback corroborate. .

<sup>1</sup>Juan G. Roederer, *The Physics and Psychophysics of Music: An Introduction*, fourth edition ed. (Springer, 2008) ISBN 978-0387094700.

<sup>2</sup>Music from the twentieth century enlarged this traditional comprehension of music. This occurred in the concert music, specially in the concrete, electronic and electroacoustic styles. On the last decade of the century, it became evident that popular music had also incorporated sounds without defined pitch and temporal organization, to name a few of simple characteristics.

<sup>3</sup>R. FABBRI, *Musica no udio digital : descrição psicofísica e caixa de ferramentas [online]*, Ph.D. thesis, Instituto de Física de So Carlos, Universidade de So Paulo (2013).

<sup>4</sup>S. Chacon, J.C. Hamano, and S. Pearce, *Pro Git*, Vol. 288 (Apress, 2009).

<sup>5</sup>T.E. Oliphant, *A Guide to NumPy*, Vol. 1 (Trelgol Publishing USA, 2006).

<sup>6</sup>"Audiolab, a python package to make noise with numpy arrays," "<http://www.ar.media.kyoto-u.ac.jp/members/david/softwares/audiolab/>"% BibitemShutNoStop

<sup>7</sup>G. Van Rossum and F. L. Drake Jr, *Python tutorial* (Odense Universitet, Institut for Matematik og Datalogi, 1995).

<sup>8</sup>G. Van Rossum and F.L. Drake Jr, *Python reference manual* (Centrum voor Wiskunde en Informatica, 1995).

<sup>9</sup>E.S. Raymond, *The art of Unix programming* (Addison-Wesley Professional, 2004).

<sup>10</sup>L. Lessig, "Free culture," Retrieved February 5, 2006 (2002).

<sup>11</sup>Jos Miguel Wisnik, *O som e o sentido* (Companhia das Letras, 1999) ISBN 8571640424.

<sup>12</sup>Anton Webern, *The Path To The New Music* (Theodore Presser Company, 1963).

<sup>13</sup>Perry R. Cook, *Real sound synthesis for interactive applications* (A K Peters, Natick, Massachusetts, 2002) ISBN 1-56881-168-3.

<sup>14</sup>Osvaldo Lacerda, *Compndio de Teoria Elementar da Msica*, 9th ed. (Ricordi Brasileira, 1966).

<sup>15</sup>Joaquim Zamacois, *Curso de formas musicales: Con numerosos ejemplos musicales* (Barcelona : Idea Books, 2002) ISBN 84-8236-235-6.

<sup>16</sup>A. Schoenberg and E. Seincman, *Fundamentos da composição musical* (Edusp, 1990).

<sup>17</sup>C. Roads, *Microsound* (MIT press, 2004).

<sup>18</sup>William Lovelock, *A Concise History of Music* (Hammond Textbooks, 1962).

<sup>19</sup>John M. Chowning, "Digital sound synthesis, acoustics and perception: A rich intersection," Proceedings of the COST G-6 Conference on Digital Audio Effects (DAFX-00)(Dec 2000).

<sup>20</sup>Timbre is a subjective and complex characteristic. Physically, the timbre is multidimensional and given by the temporal dynamics of energy in the spectral components that are harmonic or noisy. Beyond that, the word timbre is used to designate different things: one same note has different timbres, a same instrument has different timbres, two instruments of the same family have, at the same time, the same timbre that blends them in the same family, and different timbres as they are different instruments. It is worth to mention that timbre is not always manifested in spectral traces, since cultural or circumstantial aspects alter our perception of timbre.

<sup>21</sup>A.V. Oppenheim, R.W. Schaffer, J.R. Buck, *et al.*, *Discrete-time signal processing*, Vol. 2 (Prentice hall Englewood Cliffs, NJ, 1989).

<sup>22</sup>It is important to note that the factor  $\frac{1}{\lambda}$  could be distributed

among the Fourier transform and its reconstruction, as preferred.

<sup>23</sup>Also called *bias*.

<sup>24</sup>Any sampled signal has this property, not only the digitalized sound.

<sup>25</sup>As pointed before, this limitation simplify the explanation without losing generality, and will be overcome in the next section.

<sup>26</sup>Floraldo Menezes, *A Acstica Musical em Palavras e Sons* (Ateli Editorial, 2004).

<sup>27</sup>C.I. Cheng and G.H. Wakefield, "Introduction to head-related transfer functions (hrtfs): Representations of hrtfs in time, frequency, and space," *Watermark* 1 (2012).

<sup>28</sup>"Perception lecture notes: Auditory pathways and sound localization," <http://www.cns.nyu.edu/~david/courses/perception/lecturenotes/localization/localization.html>.

<sup>29</sup>V.R. Algazi, R.O. Duda, D.M. Thompson, and C. Avendano, "The cipc hrtf database," in *Applications of Signal Processing to Audio and Acoustics, 2001 IEEE Workshop on the* (IEEE, 2001) pp. 99–102.

<sup>30</sup>B. carty and V. lazzarini, "binaural hrtf based spatialisation: new approaches and implementation," in *dafx 09 proceedings of the 12th international conference on digital audio effects, politecnico di milano, como campus, sept. 1-4, como, italy* (Dept. of Electronic Engineering, Queen Mary Univ. of London., 2009) pp. 1–6.

<sup>31</sup>A. Schoenberg and M. Maluf, *Harmonia* (Ed. UNESP, 1999).

<sup>32</sup>"Massa: msica e udio em sequencias e sries amostrais," <http://labmacambira.git.sourceforge.net/git/gitweb.cgi?p=labmacambira/massa;a=blob;f=README.txt>;

<sup>33</sup>Gnter Geiger, "Table lookup oscillators using generic integrated wavetables," *Proc. of the 9th Int. Conference on Digital Audio Effects (DAFx-06)*(September 2006).

<sup>34</sup>R. Bristow-Johnson, "Wavetable synthesis 101, a fundamental perspective," in *Proc. AES Convention*, Vol. 101 (1996).

<sup>35</sup>S. Dehaene, "The neural basis of the weber–fechner law: A logarithmic mental number line," *Trends in cognitive sciences* 7, 145–147 (2003).

<sup>36</sup>The volume (psychophysical quality) is consequence of different sound characteristics, like reverberation and concentration of high harmonics, among which is wave total energy. Wave energy is the easy to modify (see Equation 2) and it can vary in many ways. The simplest way consists of modifying the amplitude by multiplying the whole sequence by a real number. The increase of energy without amplitude variation is the *sound compression*, quite popular nowadays for music production<sup>37</sup>.

<sup>37</sup>P. Guillaume, *Music and acoustics: from instrument to computer* (Iste, 2006).

<sup>38</sup>The implementation of filters comprehends an area of recognized complexity, with dedicated literature and software implementations. The reader is welcome to visit the bibliography for further discussion<sup>21,43</sup>.

<sup>39</sup>It is possible to apply the filter in the frequency domain multiplying the Fourier coefficients of both sound and the impulse response, and then performing the inverse Fourier transform in the resulting spectrum.<sup>21</sup>.

<sup>40</sup>Short for 'biquadratic': its transfer function has two poles and two zeros, i.e. its first direct form consists of two quadratic polynomials forming the fraction: 
$$\mathbb{H}(z) = \frac{a_0 + a_1 \cdot z^{-1} + a_2 \cdot z^{-2}}{1 - b_1 \cdot z^{-1} - b_2 \cdot z^{-2}}.$$

<sup>41</sup>Butterworth and Elliptical filters can be considered as special cases of Chebichev filters<sup>21,43</sup>.

<sup>42</sup>Julious O. Smith III, *Mathematics of the discrete fourier transform (dft) with audio applications*, 2nd ed. (2012) [https://ccrma.stanford.edu/~jos/log/FM\\_Spectra.html](https://ccrma.stanford.edu/~jos/log/FM_Spectra.html).

<sup>43</sup>S.W. Smith, "The scientist and engineers guide to digital signal processing, 1999," (2009).

<sup>44</sup>B.D. Class, FFT Java, A. Wah, L.F. Crusher, P. Waveshaper, S. Enhancer, and S.F.R.V.T. Matrix, "musicdsp.org source code archive," .

<sup>45</sup>Although its origin is disparate with its color association, this noise became established with this specific name, in musical

contexts. Anyway, this association can be considered satisfactory once violet, blue, white and pink noises are more strident and associated with more intense colors<sup>13,37</sup>.

<sup>46</sup>Some musical instruments jargon and contexts use different terms. For example, in piano, the called tremolo is a vibrato in the classification used here. The presented definitions are common in contexts regarding music theory and electronic music. In addition, they are based on a broader literature than the one used for a specific instrument, practice or musical tradition<sup>14,31</sup>.

<sup>47</sup>The pitch deviation is called 'vibrato depth' and is generally given as semitones or cents, as convenient.

<sup>48</sup>TA Porres, J. Manzolli, and F. Furlanete, "Análise de dissonância sensorial de espectros sonoros," in *Congresso da ANPPOM*, Vol. 16 (2006).

<sup>49</sup>A.T. Porres and A.S. Pires, "Um external de aspereza para pure data & max/msp," in *Proceedings of the 12th Brazilian Symposium on Computer Music 2009* (2009).

<sup>50</sup>"An introduction to fm (snd manual)," <https://ccrma.stanford.edu/software/snd/snd/fm.html>.

<sup>51</sup>J.E. Gramani, *Rítmica viva: a consciência musical do ritmo* (UNICAMP, 1996).

<sup>52</sup>C. Roads, *The computer music tutorial* (MIT press, 1996).

<sup>53</sup>Micro-tonality is the use of intervals less than one semitone and have ornamental and structuring functionalities in music. The division of the octave in 12 notes is motivated by physical fundamentals but also a *convention* adopted too by western classical music. Other tunings are incident. To point only one example, the traditional Thai music uses an octave division in seven notes equally spaced ( $\varepsilon = 2^{\frac{1}{7}}$ ), which allows intervals that have little in common with the intervals in the diatonic scale<sup>11</sup>.

<sup>54</sup>S.R. Wilkinson, K. Laubach, R.S. Schiff, and J. Eiche, *Tuning in: Microtonality in Electronic Music: a Basic Guide to Alternate Scales, Temperaments, and Microtuning Using Synthesizers* (H. Leonard Books, 1988).

<sup>55</sup>J. Clough, N. Engebretsen, and J. Kochavi, "Scales, sets, and interval cycles: A taxonomy," *Music Theory Spectrum*, 74–104(1999).

<sup>56</sup>S.M. Kostka, J.P. Clendinning, R. Ottman, and J. Phillips, *Tonal Harmony: With an Introduction to Twentieth* (McGraw-Hill, 1995).

<sup>57</sup>H.J. Koellreutter, "Harmonia funcional," (1986).

<sup>58</sup>F. Salzer, *Structural hearing: Tonal coherence in music*, Vol. 1 (Dover publications, 1962).

<sup>59</sup>E. Aldwell, C. Schachter, and A. Cadwallader, *Harmony & voice leading* (Wadsworth Publishing Company, 2010).

<sup>60</sup>J.J. Fux and A. Mann, *The study of counterpoint from Johann Joseph Fux's Gradus ad Parnassum*, Vol. 277 (WW Norton & Company, 1965).

<sup>61</sup>L. Tragtenberg, *Contraponto: uma arte de compor* (Edusp, 2002).

<sup>62</sup>A. Schoenberg and L. Stein, *Preliminary exercises in counterpoint* (Faber & Faber, 1963).

<sup>63</sup>Gustavo Oliveira Alfaix de Assis, *Em busca do som* (Editora UNESP) ISBN 8539302079.

<sup>64</sup>*A Msica Hoje*.

<sup>65</sup>O. Sacks, *Musicophilia: Tales of Music and the Brain, Revised and Expanded Edition*. (Knopf Doubleday Publishing Group, 2008) ISBN 9781400033539.

<sup>66</sup>H.L. Guidorizzi, *Um curso de cálculo* (Livros Técnicos e Científicos Editora, 2001).

<sup>67</sup>G. Deleuze, *Difference and Repetition* (Continuum, 1968).

<sup>68</sup>R. Duckworth and F. Stedman, *Tintinnologia, or, the Art of Ringing* (Echo Library, 2007).

<sup>69</sup>F.J. Budden and FJ Budden, *The fascination of groups* (Cambridge University Press London, 1972).

<sup>70</sup>Renato Fabbri e Adolfo Maia Jr., "Applications of group theory on granular synthesis," *SBCM 2007 Proceedings*(2007).

<sup>71</sup>Renato Fabbri e Adolfo Maia Jr., "Applications of group theory on sequencing and spatialization of granular sounds," *Anais do 6o Congresso de Engenharia de udio da AES Brasil*(2008).

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- <sup>73</sup>“3 trios para obo, flauta e fagote,” .
- <sup>74</sup>“Página principal do lab macambira,” <http://labmacambira.sourceforge.net>.
- <sup>75</sup>“Wiki do lab macambira,” [http://wiki.nosdigitais.teia.org.br/Lab\\_Macambira](http://wiki.nosdigitais.teia.org.br/Lab_Macambira).
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