

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

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(Dated: 19 September 2013)

Notes, ornaments and intervals are examples of basic elements of music, their representation as discrete-time digital audio signal 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 sound samples. Such a compendium is important since it enables scientific experiments in precise and trustful ways. In this paper, a comprehensive description of the musical elements within a unified approach is presented. 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 as means to achieve variation within a note. Moreover, the generation of musical structures such as rhythmic meter, pitch intervals and cycles, are reached canonically with notes. The availability of these resources in scripts (MASSA - Music and Audio in Sequences and Samples) is also provided in public domain. 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 of basic musical elements was confirmed by the synthesis of small musical pieces. As shown, it is possible to synthesize whole albums through collage of scripts and parametrization.

PACS numbers: \*43.66.-x, 43.66.+y, 05.10.-a

Keywords: psychophysics, acoustics, statistics, signal processing, digital audio, music

## 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  $20\text{Hz}$  and  $20\text{kHz}$  is appreciated by the human hearing system with boundaries dependent on the person, climate conditions and sonic characteristics themselves<sup>?</sup>. Considered the speed of sound as  $\approx 343.2\text{m/s}$ , these limits imply wavelengths of  $\frac{343.2}{20} = 17.16\text{m}$  and  $\frac{343.2}{20000} = 17.16\text{mm}$ .

Human perception of sound involves captivation by bones, stomach, ears, transfer functions of head and torso, and processing by the nervous system. 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, seen 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 audio (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,  $f_s \geq$

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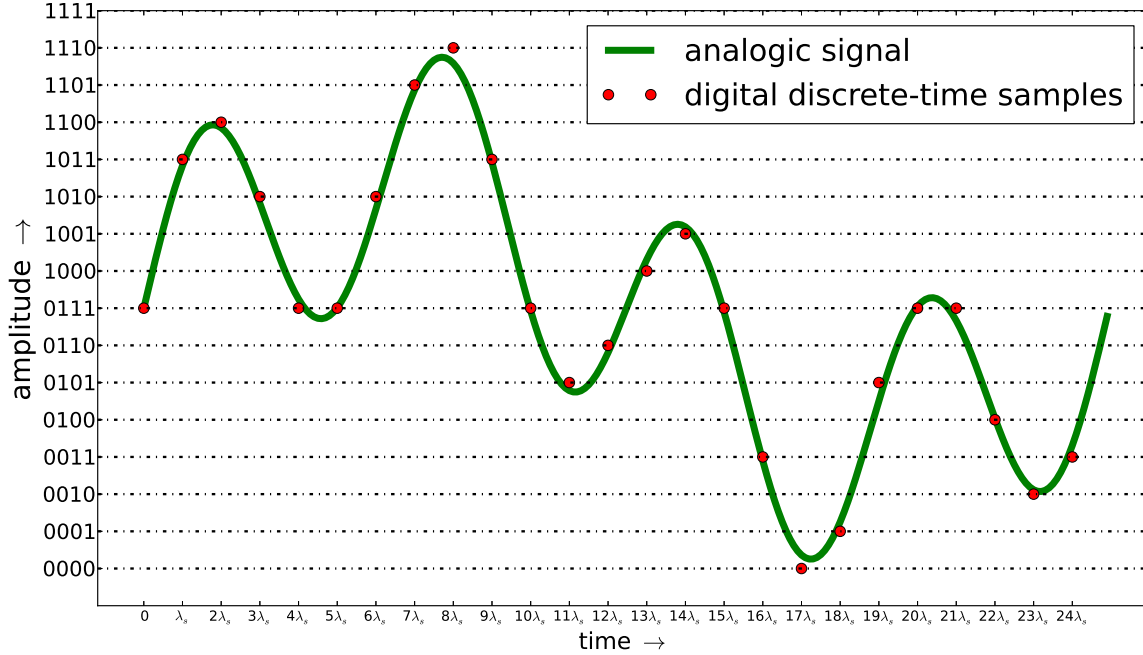


FIG. 1. Pulse Code Modulation (PCM) audio: an analogical signal is represented by 25 samples with 4 bits each.

$2 \times 20kHz = 40kHz$ . 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 characterized. 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>3</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 ??) 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>?</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 a 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>?</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>???</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? ? .

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; internals of the musical note is further developed in section III; section IV tackles the organization of musical notes in higher levels of musical structure<sup>1? ? ? ?</sup>. As these descriptions embody topics such as psycho-acoustics, cultural traditions, formalisms and protocols, the text points to external complements as needed<sup>? ? ?</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 though as being constituted by units called notes, and these units taken as "atoms" that constitute music itself<sup>1? .2</sup>. (\*\*\*\*nao entendi\*\*\*\*, texto ruim) 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>? ?</sup>. Canonically, the principal properties of a musical note are duration, volume, pitch and timbre<sup>? ?</sup>. These qualities can be managed quantitatively, dictated by the evenly time spaced sound 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 of  $\lfloor \Delta \cdot f_s \rfloor$  samples. That is, taken the integer part of the multiplication, it is admitted an error of at most  $-\lambda_a$  seconds, which are usually fine for musical purposes. Considering  $f_s = 44.1kHz \Rightarrow$

$\lambda_s = \frac{1}{44100} \approx 23$  microseconds. It is reasonable to state that:

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

Being  $\Lambda = \lfloor \Delta \cdot f_s \rfloor$  the number of samples in the sequence, thus  $T_i = \{t_i\}_0^{\Lambda-1}$ .

### B. Volume

The sensation of sound volume depends on reverberation and harmonic distribution, among other characteristics described on section ???. One can get volume variations through the potency of the wave<sup>? ?</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". Handy references are the  $10dB$  for each step in the intensity scale: *pianissimo*, *piano*, *mezzoforte*, *forte* e *fortissimo*. Other useful references are that equivalent in  $dB$  of doubling 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 ( $10dB$ ) is:

$$\begin{aligned} 10 \log_{10} \frac{pot(T'_i)}{pot(T_i)} &= 10 \Rightarrow \\ \Rightarrow \sum_{i=0}^{\lfloor \Delta \cdot f_s \rfloor - 1} t_i'^2 &= 10 \sum_{i=0}^{\Lambda-1} t_i^2 = \sum_{i=0}^{\Lambda-1} (\sqrt{10} \cdot t_i)^2 \quad (6) \\ \therefore t'_i &= \sqrt{10} t_i \Rightarrow t'_i \approx 3,16 t_i \end{aligned}$$

Thus, it is necessary a little bit more than triplicate the amplitude 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 explained in the previous subsections, the musical particle (note) is a sequence  $T_i$  in which duration and volume corresponds to the size of sequence and the amplitude of its samples. 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 an integer. If  $T_i^f$  is a sonic sequence of fundamental frequency  $f$ , then:

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

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

### D. Timbre

While the wave period corresponds to the fundamental frequency, the trajectory of the wave inside the period - called the waveform - defines a harmonic spectrum and, thus, a timbre<sup>4</sup>. Musically, it matters that sonic spectra with minimum differences can result in timbres with crucial expressive differences and, consequently, different timbres can be produced by using different spectra<sup>5</sup>.

The simplest (and most important case, as we will see in the following) 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'. Let  $S_i^f$  be 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  $\lambda_f = \frac{f_s}{f} = \frac{\delta_f}{\lambda_a}$  is the number of samples in the period.

In a similar fashion, other waveforms are applied in music considering their spectral qualities and simplicity. While the sinusoid is an isolated point in the spectrum, these additional waves present a succession of harmonic components of the sinusoid. These waveforms are specified in equations ??, 10, 11 and ?? and illustrated in Figure 2. Although the artificial waveforms are traditionally used in music for synthesis and oscillatory control of variables, they are also useful outside music context<sup>6</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 following:

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

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 sawtooth is a common starting point for a subtractive synthesis, because it has both odd and even harmonics with high energy. For musical intention, these waveforms are 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 square wave preservers 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.

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

The spectrum of each basic waveform is in Figure ??. The isolated and exactly harmonic components of the spectrum is a consequence of the fixed period usage. The sinusoid consists of a one and only node in spectrum (\*\*esquisito o texto\*\*), pure frequency. As we said before, 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.

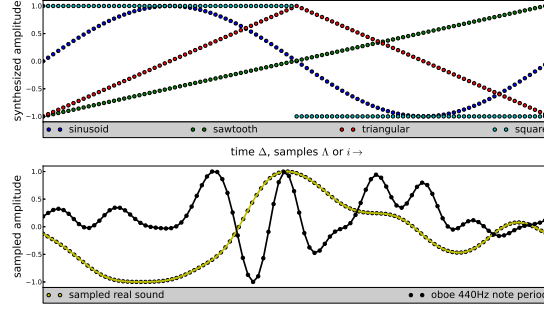


FIG. 2. Basic musical waveforms: (a) the synthetic waveforms; (b) the real waveforms.

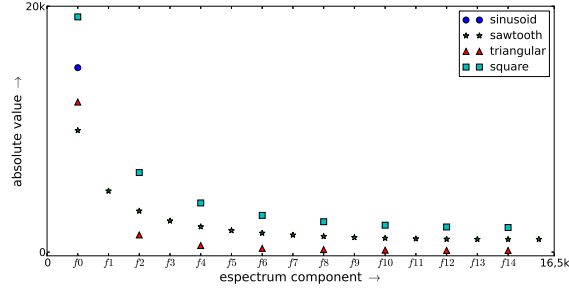


FIG. 3. Spectrum of basic artificial musical waveforms.

The harmonic spectrum is formed by frequencies that are multiple from the fundamental frequency  $f_n = (n + 1)f_0$ . As the human linear perception of pitch follows a geometric progression of frequencies, the spectrum has notes different from the fundamental frequency. Additionally, the number of harmonics will be limited to the maximum frequency  $f_s/2$  (by Nyquist's theorem).

From a musical perspective, 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  $f_n$ , is separated by the ear to enter in a cognitive level of processing (this separation is done in many species with mechanisms similar to the human cochlea<sup>?</sup>).

The sinusoidal components are usually the main responsible for the timbre quality. If they are not presented in harmonic proportions (small number relations), the sound is perceived as noisy or dissonant, in opposite of a sonority with a unequivocally established fundamental frequency when they are present. Furthermore, the notion of absolute pitch in a complex sonority is based relies on the similarity of the spectrum to the harmonic series<sup>?</sup> (\*\*nao entendi o q vcs querem dizer\*\*).

In the case of a fixed length period and waveform, the spectrum is perfectly harmonic and static. Each waveform is compound of specific proportions of harmonic components and the greater the curvature of the part, the greater the contribution of the part to the energy on the high harmonics. This aspect can be seen on real

sounds. The wave labeled as “sampled real sound” in Figure 2 has a period of  $\Lambda_f = 114$  samples, extracted from a relatively well behaved real sound. The oboe wave was sampled for A4 also in  $44.1kHz$ . The chose period for sampling was a relatively short one, with 98 samples, and corresponds to the frequency  $\frac{44100}{98} = 450Hz$ . It can be noticed 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 sound. 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 and whose samples consists of the repetition of cycle of Figure 2. The natural spectrum exhibits variations in the frequencies of the harmonics, in their intensities and some noise. The note made from the sampled period has a perfectly harmonic spectrum.

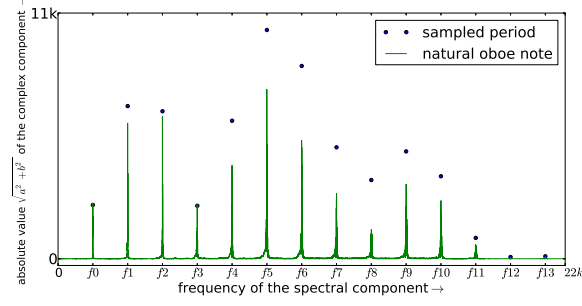


FIG. 4. Spectrum of the sonic waves of a natural oboe note and one made from a sampled period. The natural sound has fluctuations in the harmonics and in its noise, while the sampled period note has a perfectly harmonic spectrum.

### 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 frequencies components as time samples<sup>5</sup>:

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

where  $c_k = a_k + j.b_k$  defines the amplitude and phase of each frequency:  $w_k = \frac{2\pi}{\Lambda} k$  in radians or  $f_k = w_k \frac{f_a}{2\pi} = \frac{f_a}{\Lambda} k$  in Hertz, taking into account the respective limits in  $\pi$  and in  $\frac{f_a}{2}$  given by the Nyquist Theorem.

Equation 14 shows how the imaginary term  $c_k$  adds a phase to the real sinusoid. For example, the imaginary terms  $b_k$  from the Fourier spectral decomposition make possible the phase sweep  $[-\frac{\pi}{2}, +\frac{\pi}{2}]$  given by  $tg^{-1}(\frac{b_k}{a_k})$  which has this image. The signal from  $a_k$  specifies the right and left side of the trigonometric circle, complementing the phase sweeping:  $[-\frac{\pi}{2}, +\frac{\pi}{2}] \cup [\frac{\pi}{2}, \frac{3\pi}{2}] \equiv [2\pi]$ .

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

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

Figure 5 shows two samples and its spectral components. In this case, the Fourier decomposition has one unique pair of coefficients  $\{c_k = a_k - j.b_k\}_0^{\Lambda-1=1}$  relatives to frequencies  $\{f_k\}_0^1 = \left\{ w_k \frac{f_a}{2\pi} \right\}_0^1 = \left\{ k \frac{f_a}{\Lambda=2} \right\}_0^1 =$

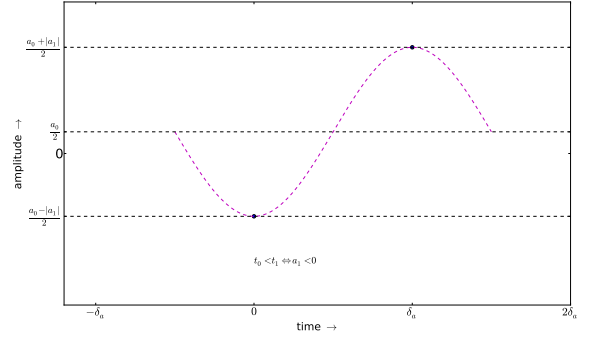


FIG. 5. Oscillation for 2 samples (maximum frequency for any  $f_a$ ). The first coefficient reflects a detachment (*offset* or *bias*) and the second coefficient specifies the oscillation amplitude.

$\{0, \frac{f_a}{2} = f_{\max}\}$  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 detachment<sup>6</sup> and  $\frac{a_1}{2}$ , oscillation own amplitude given by the relation  $f_k = k \frac{f_a}{\Lambda=2}$ . This case have special relevance. It is necessary the of 2 samples to represent an oscillation and it yields the Nyquist frequency  $f_{\max} = \frac{f_a}{2}$ . That is the maximum frequency in a sound sampled with  $f_a$  samples per second<sup>7</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, e.g. 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 relevance<sup>8</sup>. 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:  $f_k = \frac{f_a}{2} - (f_k - \frac{f_a}{2}) = f_a - f_k =$

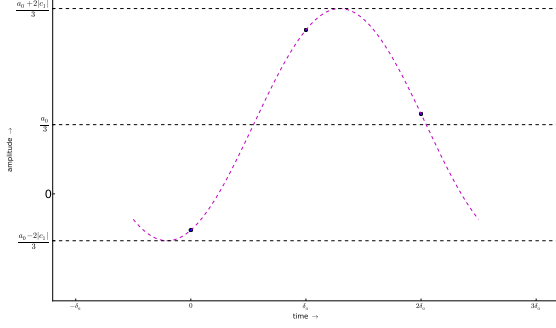


FIG. 6. Three fixed samples presents only one non-null frequency.  $c_1 = c_2^*$  and  $w_1 \equiv w_2$ .

$$f_a - k \frac{f_a}{\Lambda} = (\Lambda - k) \frac{f_a}{\Lambda} \Rightarrow f_k \equiv f_{\Lambda-k}, \forall k < \Lambda.$$

The same could be observed in  $w_k = f_k \cdot \frac{2\pi}{f_a}$  and considering the periodicity  $2\pi$ , it results in  $w_k = -w_{\Lambda-k}$ . Given the cosine as an even function and the inverse tangent as an odd function, the components in  $w_k$  and  $w_{\Lambda-k}$  sums up in the equation that reconstructs the real samples, yielding equation 14.

In other words, in a decomposition of  $\Lambda$  samples, the  $\Lambda$  given frequencies components  $\{c_i\}_0^{\Lambda-1}$  are equivalents in pairs. Except for  $f_0$ , and when having an even  $\Lambda$ , from  $f_{\Lambda/2} = f_{\max} = \frac{f_a}{2}$  both components are isolated, e.g. there is any other component in frequency  $f_0$  or  $f_{\Lambda/2}$  (if even  $\Lambda$ ) regarding itself. It is true because  $f_{\Lambda/2} = f_{(\Lambda-\Lambda/2)=\Lambda/2}$  and  $f_0 = f_{(\Lambda-0)=\Lambda} = f_0$ . Furthermore, these two frequencies (zero and maximum frequency) are not represented having phase variation, being 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 make clear visible the equivalences 17, 18 and 19:

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

Having  $a_k = a_{\Lambda-k}$  and  $b_k = -b_{\Lambda-k}$ :

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

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

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

The equation 15 for the real signal reconstruction, the modules and phases equivalences 18 and 19, the number

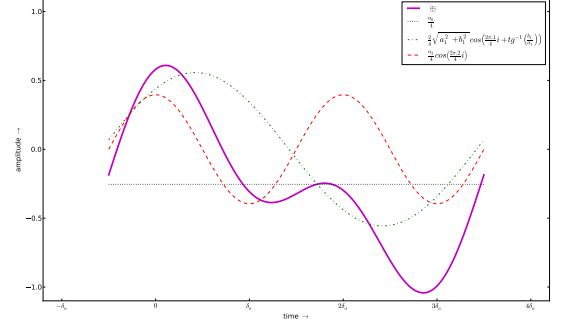


FIG. 7. Frequential components for 4 samples.

of paired coefficients 16, and equivalence of paired frequencies 17 expose a general case for components combination in each sample  $t_i$ :

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

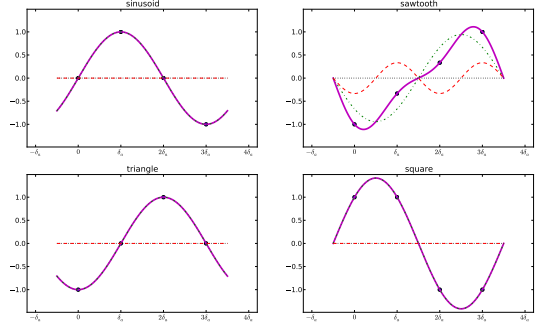


FIG. 8. Basic wave forms for 4 samples.

Therefore, as in Figure 6, the Fourier transform of 3 samples have 2 frequencies coefficients with same amount of energy in the same frequency.

With 4 samples it is possible to represent 1 or 2 frequencies with any proportions. Figure 7 depicts wave forms for 4 samples and its respective two components. The individual contributions sums together yielding the original wave form and a brief inspection reveals the major curvatures resulting from the higher frequency, while a fixed detachment of component summation results from the zero frequency component.

Figure 8 shows the harmonics for 4 samples in basic wave forms of equations 9, ??, 11 and 12. Their conjunction results in only 1 sinusoid, with exception to sawtooth wave, which have even harmonics.

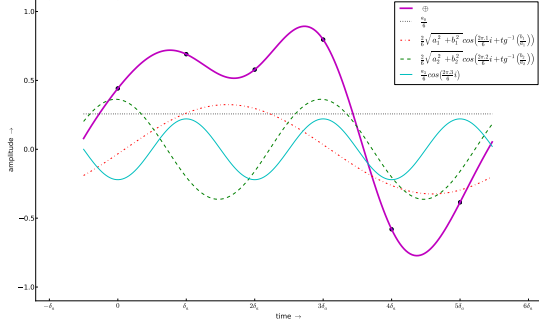


FIG. 9. Frequencies components for 6 samples: 3 sinusoid sums with *bias*.

Figure 9 presents a sinusoidal decomposition for the 6 samples case and Figure 10 presents the decomposition of the basic wave forms. In this case all wave forms are different in spectrum: square and triangular ones have the same components but with different proportions, while the sawtooth have an extra component.

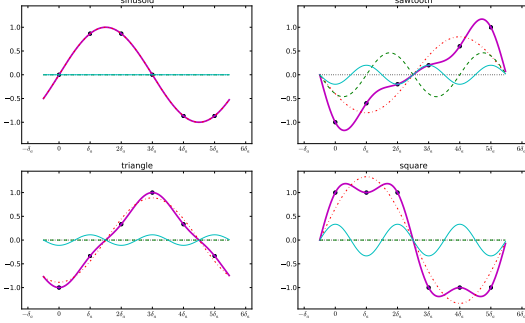


FIG. 10. Basic wave forms for 6 samples: triangular and square wave forms have odd harmonics, with different proportions and phases; the sawtooth wave also has even harmonics.

## F. The basic note

Consider  $f$  presented in a such way that divides  $f_a^9$ . A sequence  $T_i$  of sound samples are separated by  $\delta_a = 1/f_a$  and describes a musical note with a frequency of  $f$  Hertz and  $\Delta$  seconds of duration if, and only if, it has the periodicity equal to  $\lambda_f = f_a/f$  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 specifies a timbre. Nevertheless, it is necessary to choose the right waveform so that the samples  $t_i$  have a value established individually.

A 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, ??, 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. In this context, the existence of an environment that reverbs the played note it is the matter of 'spatialization'. Both fields, spatialization and spatial localization, are widely valued by audiophiles and music industry<sup>?</sup>.

### 1. Spatial localization

It is believed that the perception of spatial localization of sound occurs in our nervous system by three information: 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>?</sup> ? ? .

Considering only the direct incidences in each ear the equations are quite simple. Given the distance between ears  $\zeta^{10}$ , an object placed at  $(x, y)$  as in Figure 11 is distant of each ear by:

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

and immediate calculations results in the Interaural Time Difference:

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

and in the Interaural Intensity Difference:



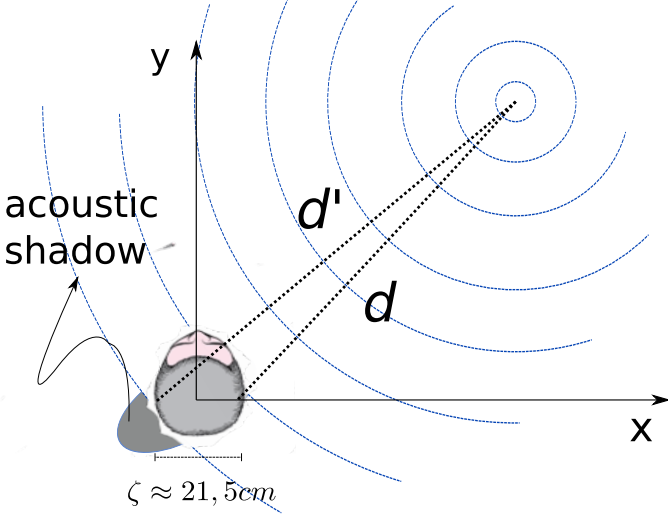


FIG. 11. Detection of sound source spatial localization: schema used to calculate Interaural Time Difference (ITD) and Interaural Intensity Difference (IID).

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

Converting to amplitude it gives  $IID_a = \frac{d}{d'}$ . The  $IID_a$  can be used as a multiplicative constant to the right channel of a stereo sound signal:  $\{t'_i\}_0^{\Lambda-1} = \{IID_a \cdot t_i\}_0^{\Lambda-1}$ . It is possible to use IID together with ITD as a time advance for the right channel in relation with the left one. It is a crucial entail to localization in bass sounds and percussive sonorities<sup>?</sup>. Considering  $\Lambda_{ITD} = [ITD \cdot f_a]$ :

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

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  $\Lambda'_{ITD} = |\Lambda_{ITD}|$ .

Although simple until here, spatial localization depends considerably of other steps. Using ITD and IID it is possible to specify only 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  $\theta$  focuses in the so called "cone of

confusion": where the same pair of ITD and IID results in a large number of points inside the cone. On those points the inference of azimuthal angle depends specially of the attenuate filtering for high frequencies, since the head interferes much more in the treble mechanical waves than in the bass waves<sup>?</sup>. It is also relevant to the hearing of source side, where the sound has enough low frequencies there is a diffraction and the wave comes to the ear with a delay of  $\approx 0,7ms$ .<sup>?</sup>

Figure 11 also shows an acoustic shadow from 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. This makes it hard to conceive given the filtering and diffraction dependent of the sound spectrum. In this way, the cone of confusion can be understood as a cone with its top side placed in the middle of head and growing out in the direction of each ear<sup>?</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>?</sup>. There are well known open databases of HRTF as CIPIC and it is possible to apply those transfer functions in a sound signal by convolution (see equation 42)<sup>?</sup>. The human body considerably changes the filtering and there are techniques to generate HRTFs to be universally used, as expected<sup>?</sup>.

## 2. Spatialization

Spatialization results from sound reflections and absorptions by room/environment surface where the note is played. The sound propagates throw the air with a speed of  $\approx 343,2m/s$  and can be sent from a source without any directionality pattern. When a sound pulse encounters a surface there is reflection. In this reflection occurs: 1) the component inversion of propagation speed whose is normal to the surface, as 2) the energy absorption, specially in trebles. The sound waves propagate until they reach inaudible levels. The moment when some sound pulse reach the human ear can be described as the moment it reaches the ear and the absorption filters of each surface it has reached. It is possible to simulate reverberations that are impossible in real systems. To experimentation, 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 now found in real systems.

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

- 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: bass has some resonance frequencies relatively spaced.
- Second band: mid and treble have progressive decay and smooth statistical fluctuations.

Smith III points out 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 suggests 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 a tangled form whose incidences are less distinct by hearing. These first reflections are particularly important to space 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 dependents of room dimension, the separation between the first reflection is at 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  $\approx k.n^2$ . A discussion about the use of convolutions and filtering to favor these implementations is provided in subsection III F, specially at paragraphs about reverberation.

## H. Musical use

Once we have well defined the basic note, it is time to build musical structures with sequences based on those 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 overlapped spectral contents in a process called sound mixing:

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

Figure 12 illustrates this overlapping process of discretized sound waves, disposing 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. Someone just need to complete the sequence with zeros to sum up sequences with different sizes.

The mixed notes are generally separated by the ear according to the physic laws of resonance and by the nervous system<sup>?</sup>. This musical notes mixing process results

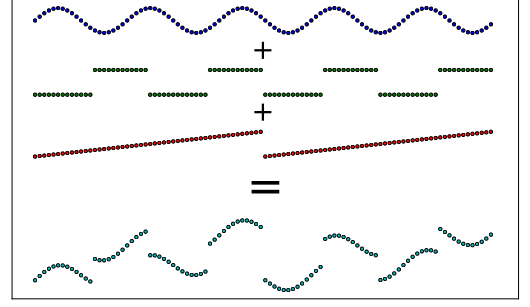


FIG. 12. Mixing of three sound sequences. The amplitudes are directly overlapped.

in musical harmony whose intervals between frequencies and chords of simultaneous notes guide subjective and abstract aspects of music and its appreciation<sup>?</sup>.

The 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 musical sequence or melody:

$$\{t_i\}_0^{\sum \Delta_k-1} = \{t_{l,i}\}_0^{\sum \Delta_k-1}, \quad l \text{ smallest integer} : \quad \Lambda_l > i - \sum_{j=0}^{l-1} \Lambda_j \quad (30)$$

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

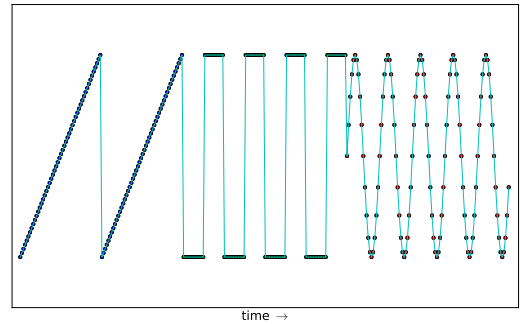


FIG. 13. Concatenation of three sound sequences by temporal overlap of its samples.

The musical piece *reduced-fi* explores the temporal overlapping of notes, resulting in a homophobic piece. The vertical principle is demonstrated at the *sound boards*, static sounds with peculiar spectrum. Both pieces

were written in Python at Appendices ?? and ??, and are available as part of the *MASSAtoolbox*.<sup>?</sup>

Now that the basic digital music note is described, the next section develops the temporal evolution of its contents as in *glissandi* and volume envelopes. The filtering of spectral components and noise generation complements the definition of 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.

### III. VARIATION IN THE BASIC MUSICAL NOTE

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

At first place, the characteristics of the note are modified along the duration of the note itself<sup>?</sup>. For example, a note played in a piano with duration of 3 seconds has abrupt intensity at its beginning time of execution. Then, it behaves with progressively decays, spectrum variations, presenting harmonics decaying before others and presenting harmonics showing up along the time. Those variations are not mandatory but they are guidance from sound synthesis to musical uses, because it suitable reflects how sounds are represented in nature<sup>11</sup>. To explore all the possible ways which those variations occur is out of the scope of this study, given the considerably sensibility of the human ear and the complexity of our sound cognition. In the following we point out primary resources to produce those characteristic variations in the basic note. It is worthwhile to mention that all the described relations in the following section are currently implemented in Python at Appendix ??. The music pieces *Transita para metro*, *Vibra e treme*, *Tremolos*, *vibratos e a frequencia*, *Trenzinho de caipiras impulsivos*, *Ruidosa faixa*, *Bela rugosi*, *Chorus infantil*, *ADa e SaRa* are listed in Appendices ??, ??, ??, ??, ??, ??, ?? and ??. The code is also part of the toolbox MASSA, available online.<sup>?</sup>

#### A. Lookup Table

The *Lookup Table* (or simply LUT), is a data structure array which substitutes continuous and repetitive calculation by indexed access. It is frequently used to reduce computational complexity and to make possible the use of functions without the possibility of direct calculation, as the data sampled from nature. In music its usage transcends the former, simplifying operations and making possible to use a single wave period to synthesize sounds in the whole audible spectrum, no matter what kind of wave was originally sampled.

Being  $\tilde{\Lambda}$  the wave period size and  $\tilde{L}_i = \{\tilde{l}_i\}_0^{\tilde{\Lambda}-1}$  the elements  $\tilde{l}_i$  of any wave period (refer to Equation 22), a

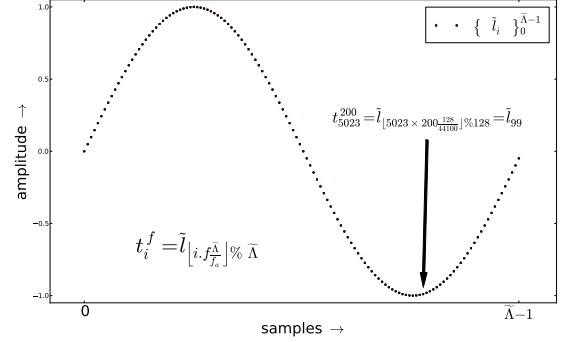


FIG. 14. Search procedure in an indexed array (known as *Lookup Table*) to synthesize sounds in different frequency using an unique waveform with high resolution.

sequence  $T_i^{f, \Delta}$  with samples of a sound with frequency  $f$  and duration  $\Delta$  can be obtained by means of  $\tilde{L}_i$  as:

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

In other words, with the right LUT indexes ( $\gamma_i \% \tilde{\Lambda}$ ) it is possible to synthesize a sound with any frequency. Figure 14 illustrates the calculation of a  $\{t_i\}$  sample from  $\{\tilde{l}_i\}$  for a frequency of  $f = 200Hz$ ,  $\tilde{\Lambda} = 128$  and considering the sample rate as  $f_a = 44.1kHz$ . 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  $f_a = 44.1kHz$  the standard guidelines suggest the use of  $\tilde{\Lambda} = 1024$  samples, since it does not produce relevant noise considering the audible spectrum. The round or interpolation method are not decisive in this process<sup>?</sup>.

The expression defining the variable  $\gamma_i$  can be understood as  $f_a$  being summed to  $i$  at each 1 second. If  $i$  is divided by the sample frequency, it results in  $\frac{i}{f_a}$ , which is incremented in 1 at each 1 second. Multiplied by the period length it results in  $i \frac{\tilde{\Lambda}}{f_a}$  that look up the period in 1 second. Finally, with frequency  $f$  it results in  $i \cdot f \frac{\tilde{\Lambda}}{f_a}$  which complements  $f$  lookups of the period  $\tilde{\Lambda}$  in 1 second, i.e., the resulting sequence presents the fundamental frequency  $f$ .

There are important considerations here: it is possible to use any value for  $f$ , the limits exist only in the loud frequencies when the size of table  $\tilde{\Lambda}$  is not sufficient enough considering the sample rate  $f_a$ . The lookup table procedure is computationally costless and replaces

the 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 music note quasi-periodic<sup>?</sup> ? .

## B. Incremental Variations of Frequency and Intensity

Following the Weber and Fechner<sup>?</sup> law, the human perception has a logarithmic relation with the causing stimulus. In other words, a stimulus following an exponential progression is perceived as linear. For didactic reasons and given its use in AM and FM synthesis (refer to subsection III E), the linear variation is discussed first.

Considering the note with duration  $\Delta = \frac{\Lambda}{f_a}$ , the frequency  $f = f_i$  varies linearly from  $f_0$  to  $f_{\Lambda-1}$ . In this way, it is possible to define the following:

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

$$\Delta_{\gamma_i} = f_i \frac{\tilde{\Lambda}}{f_a} \Rightarrow \gamma_i = \left[ \sum_{j=0}^i f_j \frac{\tilde{\Lambda}}{f_a} \right] = \left[ \sum_{j=0}^i \frac{\tilde{\Lambda}}{f_a} \left[ f_0 + (f_{\Lambda-1} - f_0) \frac{j}{\Lambda - 1} \right] \right] \quad (33)$$

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

where  $\Delta_{\gamma_i} = f_i \frac{\tilde{\Lambda}}{f_a}$  is the LUT increment between two samples given the sound frequency in the first sample.

Thus, it is possible to calculate the elements  $t_i^{f_0, f_{\Lambda-1}}$  having the period  $\left\{ \tilde{t}_i \right\}_0^{\Lambda-1}$ .

The equations 32, 33 and 34 are related with the linear progression of the frequency. We recall that, for the general case, the frequency progression *perceived* as linear follows an exponential progression<sup>12</sup>. It is possible to write this such that:  $f_i = f_0 \cdot 2^{\frac{i}{\Lambda-1} n_8}$  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 transition equations of linear frequency for the human ear are given by:

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

$$\Delta_{\gamma_i} = f_i \frac{\tilde{\Lambda}}{f_a} \Rightarrow \gamma_i = \left[ \sum_{j=0}^i f_j \frac{\tilde{\Lambda}}{f_a} \right] = \left[ \sum_{j=0}^i f_0 \frac{\tilde{\Lambda}}{f_a} \left( \frac{f_{\Lambda-1}}{f_0} \right)^{\frac{j}{\Lambda-1}} \right] \quad (36)$$

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

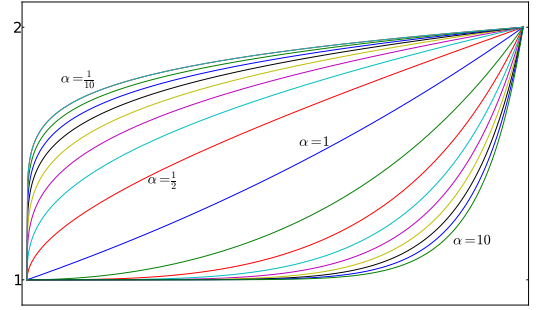


FIG. 15. Intensity transitions to different values of  $\alpha$  (see equations 38 and 39).

The term  $\frac{i}{\Lambda-1}$  looks up 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 of a vibratory wave to change the volume intensity<sup>13</sup>. It is sufficient to multiply the original sequence (being it generated or predetermined) 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 aimed value to change until the end of transition.

Thus, for amplitude variations:

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

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

It is possible to take  $a_0 = 1$  in order to start a new sequence with the original amplitude and then keep changing it with the following samples. This restriction makes  $a_{\Lambda-1}$  the amplitude variation term. If  $\alpha = 1$ , the amplitude variation follows the geometric progression whose

defines the linear progression. Figure 15 shows the transitions between values 1 and 2 and for different values of  $\alpha$ , a gain of  $\approx 6dB$  as given by equation 4.

An special attention should be driven 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 of 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* ( $a_{\Lambda-1} = 10^{-4}$ ).

For linear amplification – but not linear to 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 conversation 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>?</sup>

The linear transitions will be used for AM and FM synthesis and for applications of logarithmic transitions for tremolos and vibratos. A non-oscillatory exploration of those variations is in the music piece *Transita para metro*, which code is in Appendix ?? and online at MASSA.<sup>?</sup>

### C. Application of Digital Filters

This subsection is limited to a description of the procedure in sequences by convolution and differential equations, and the immediate applications. Its complexity exceeds to the scope of this study<sup>14</sup>. The filters application can be part of the synthesis process or made individually as part of processes commonly called sound treatment.

- Convolution and finite impulse response (FIR) filters

Filters applied by means of convolution are known by acronym FIR (Finite Impulse Response) and are characterized by having a time finite sample representation. This sample representation is called 'impulse response'  $\{h_i\}$ . The FIR filters are applied to the time domain of the digital sound by means of convolution with the respective impulse response of the filter<sup>15</sup>. For this study, the convolution is defined as:

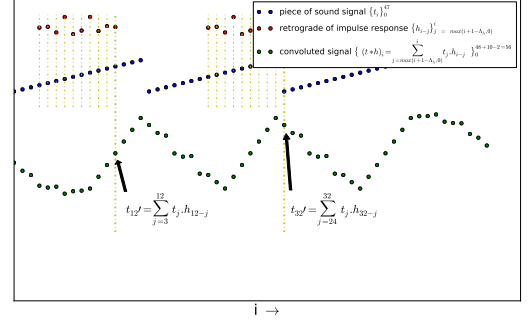


FIG. 16. Graphical interpretation of convolution. Each resulting sample is the sum of the previous samples of a signal, with each one multiplied by the retrograde sequence of samples from the other signal.

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

where  $t_i = 0$  for the samples not given beforehand. In other words, the sound  $\{t'_i\}$  resulting from the convolution of  $\{t_i\}$  with the given 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 another impulse response  $\{h_i\}_0^{\Lambda_h-1}$ . This procedure is illustrated in Figure 16, where the impulse response  $\{h_i\}$  is traveled by 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 resulting signal always has the length of  $\Lambda_t + \Lambda_h - 1 = \Lambda_{t'}$ .

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

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



The inverse Fourier transform of an envelope even and real is equivalent to a impulse response of a FIR filter. It performs a frequency filtering with the envelope. The greater the number of samples, the higher the envelope resolution and the computational complexity as well. The convolution is recognized as computationally 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 to each relapse of sound. In Figure 17 it is possible to observe the shifting caused by convolution with each impulse. Depending on the intensity of the impulses, the result is perceived in rhythm (around 20 impulses by second) or in pitch (around 20-40 impulses by second). In the last case, the processes resemble granular synthesis, delays, reverbs and equalization.

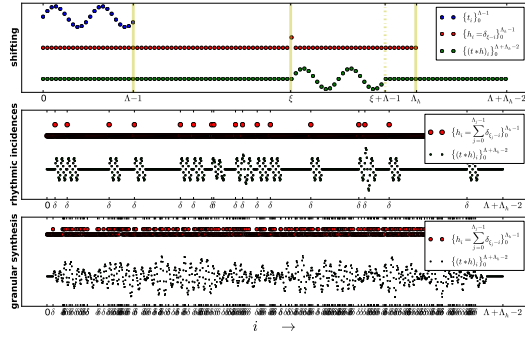


FIG. 17. Convolution with impulse: shifting (a), delay lines (b) and granular synthesis (c). Disposed in increasing order of its pulse density.

- Infinite impulse response (IIR) filters

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

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

with  $b_0 = 1$  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 between the original samples  $\{t_i\}$  and the previous resulting ones  $\{t'_{i-k}\}$ .

There are many methods and tools to prepare IIR filters. The following texts list a selection of them

for didactic purpose and to make easy future consulting. They are well behaved filters whose aspects are described in Figure 18.

Considering filters with simple order, the cutoff frequency  $f_c$  is the position where the filter performs an attenuation of  $-3dB \approx 0.707$  in the original amplitude. For band-pass and band-reject (or 'notch') filters, this attenuation results on two specifications:  $f_c$  (in this case, referred as 'center frequency') band width  $bw$ , in both frequencies  $f_c \pm bw$  there is an attenuation of  $\approx 0.707$  in 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 shift in the expected waveform, expanding the envelope to the bass side of the band.

For filters whose frequency responses have other envelopes (for the module), it is possible to create chains and apply them successively. Another possibility is to use some biquad 'filter receipt'<sup>16</sup> or procedures for the calculation of coefficients of Chebichev filters<sup>17</sup>. Both possibilities are explored by the referenced studies, in special<sup>?</sup>?, and the collection of filters maintained by the *Music-DSP* community from University of Columbia.<sup>?</sup>?

1. Low-pass with simple pole and module of the frequency response in the upper left corner of Figure 18. The general equation has as reference by the cutoff frequency  $f_c \in (0, \frac{1}{2})$ , fraction of the sample frequency  $f_a$  that there is approximately an attenuation of  $3dB$ . The coefficients  $a_0$  and  $b_1$  of the IIR filter are given by means of the intermediate variable  $x \in [e^{-\pi}, 1]$ :

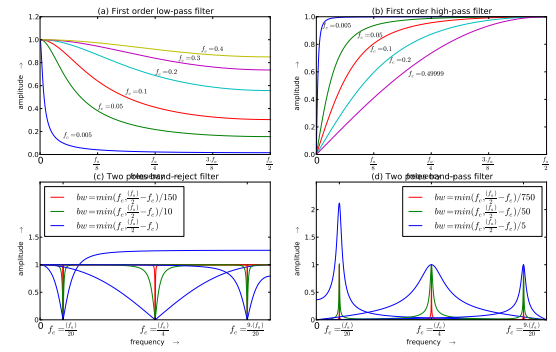


FIG. 18. Modules for the frequency response (a), (b), (c) and (d) for IIR filters of equations 44, 45, 47 and 48 respectively, considering different cutoff frequencies, center frequencies and band width.

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

2. High-pass filter with simple pole and modules 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  $x \in [e^{-\pi}, 1]$ :

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

3. Notch filter. This filter is parametrized by a center frequency<sup>18</sup>  $f_c$  and band width  $bw - f_c \pm bw$  that results in 0.707 of the amplitude, i.e. attenuation of 3dB – both given as fractions of  $f_a$ , therefore  $f, bw \in (0, 0.5)$ .

In order to simplify, consider the auxiliary variables  $K$  and  $R$  defined as:

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

The band-pass filter in the lower left corner of Figure 18 has the following coefficients in equation 43:

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

The coefficients of band-reject filter are:

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

with the module of its frequency response represented in the lower left corner of the Figure 18.

## D. Noise

Generally, sounds without a defined pitch are called noise<sup>7</sup>. They are important parts of the musical sounds with undefined pitch like the noise present in piano notes, violin, etc. Besides that, the majority of percussion instruments do not have defined pitch and their sounds are generally comprehended as noise<sup>7</sup>. In electronic music, including electro-acoustic and dance genres, noise has diversified use, and commonly characterizes the music style<sup>7</sup>.

The absence of a defined pitch is due to the absence of a harmonic organization in the sinusoidal components that compose the sound. In this way, there are many ways to generate noise. The use of random values to generate the sound sequence  $T_i$  is an attractive method but the result is generally not useful because it tends to the white noise<sup>7</sup>.

Another way is to generate noise using the desired spectrum, from which it is possible to perform the inverse Fourier transform. The spectral distribution should be done with care because, if it uses the same phase or phases with strong correlation, the synthesized sound will include considerably amount of concentrated energy in some periods of its duration.

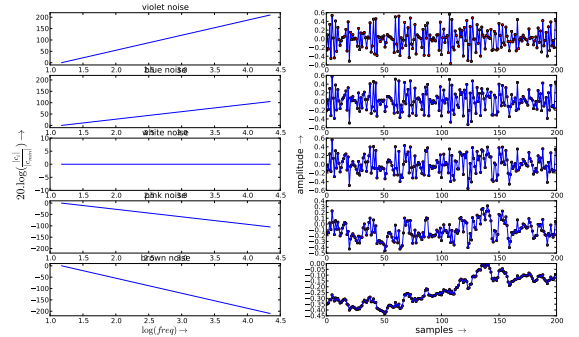


FIG. 19. Colors of noise generated by equations 49, 50, 51, 52 and 53: resulting spectrum and waveforms.

Some noise with static spectrum are listed below. They are called *colors of noise* since they are associated with colors. Figure 19 shows the spectrum profile and the corresponding sound sequence side-by-side. All the noise were generated with a same phase, making it possible to observe the contributions of different regions of the spectrum.

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

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

The value  $c_i$ , calculated by the exponential, is only an artifice to obtain unitary module and random phase. Besides that,  $c_{\Lambda/2}$  is always real (as discussed in the previous section).

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

$$\begin{aligned}
f_{\min} &\approx 15Hz \\
f_i &= i \frac{f_a}{\Lambda}, \quad i \leq \frac{\Lambda}{2}, \quad i \in \mathbb{N} \\
\alpha_i &= \left(10^{-\frac{3}{20}}\right)^{\log_2\left(\frac{f_i}{f_{\min}}\right)} \\
c_i &= 0, \quad \forall i : f_i < f_{\min} \\
c_i &= e^{j \cdot x} \cdot \alpha_i, \quad j^2 = -1, \\
x \text{ random} &\in [0, 2\pi], \quad \forall i : f_{\min} \leq f_i < f_{\lceil \Lambda/2 - 1 \rceil} \\
c_{\Lambda/2} &= \alpha_{\Lambda/2} \quad (\text{se } \Lambda \text{ par}) \\
c_i &= c_{\Lambda-i}^*, \quad \text{for } i > \Lambda/2
\end{aligned} \tag{50}$$

The minimum frequency  $f_{\min}$  can be chosen based on the human hearing limit, since no one listens to a pitch with a sound component which frequency is bellow  $\approx 20Hz$ .

Other noises can be made from the pink noise procedure by simply modifying some details, specially the equation that defines  $\alpha_i$ .

- The brown noise received this name after Robert Brown, who described the brownian movement<sup>19</sup>

What characterizes this noise is the decreasing of  $6dB$  by octave. In this way,  $\alpha_i$  in the set 50 is defined as:

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

- In the blue noise there is a gain of  $3dB$  by 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:

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

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

$$\alpha_i = \left(10^{\frac{6}{20}}\right)^{\log_2\left(\frac{f_i}{f_{\min}}\right)}, \quad f_{\min} \approx 15Hz \tag{53}$$

- 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 imperative to obtain  $\alpha_i$ . An implementation of ISO 226, that is the last revision of those curves, is in the toolbox MASSA.<sup>?</sup>

This subsection exposed only noises with static spectrum. There are also characterizations for noises with dynamic spectrum during the time, or 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 noise, but a fast scan of some given frequency band<sup>?</sup>.

The noise from equations 49, 50, 51, 52 and 53 are presented in Figure 19. The spectrum were build 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 period variation in pitch (frequency) and tremolo is a period variation in volume (intensity)<sup>20</sup> For the general case, the vibrato is described as follows:

$$\gamma'_i = \left[ i f' \frac{\tilde{\Lambda}_M}{f_a} \right] \tag{55}$$

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

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



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

$$= \left\lfloor \sum_{j=0}^i \frac{\tilde{\Lambda}}{f_a} f \left( \frac{f+\mu}{f} \right)^{t'_j} \right\rfloor = \left\lfloor \sum_{j=0}^i \frac{\tilde{\Lambda}}{f_a} f \cdot 2^{t'_j \frac{\nu}{12}} \right\rfloor \quad (58)$$

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

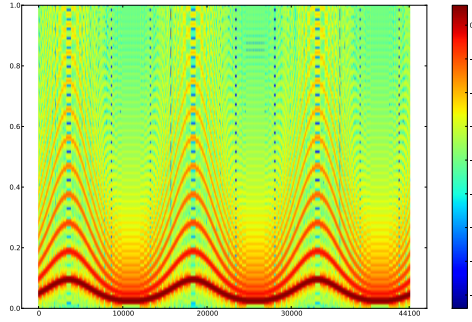


FIG. 20. Spectrogram of a sound with sinusoidal vibrato of 3Hz and one octave depth in a 1000Hz sawtooth wave (considering  $f_a = 44.1kHz$ ).

For the properly realization of vibrato, it is important to pay attention on the following tables (quais tabelas? seriam termos?) 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 vibrato intensity:  $\mu$  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 pikes 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 diminish, but the semitones variations remains.

Figure 20 illustrates the spectrogram of an artificial vibrato for a note with 1000Hz (between a B and a C), and which frequency deviation reaches one octave above and one below. Any waveform can be used to generate a sound and a vibrato oscillation pattern given any oscillation frequency and pitch deviation<sup>21</sup>. Those oscillations with precise forms 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:

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

$$T_i^{tr(f')} = \left\{ t_i^{tr(f')} \right\}_0^{\Lambda-1} = \{t_i \cdot a_i\}_0^{\Lambda-1} =$$

$$= \left\{ t_i \cdot 10^{\frac{V_{dB}}{20} t'_i} \right\}_0^{\Lambda-1} = \left\{ t_i \cdot a_{\max}^{t'_i} \right\}_0^{\Lambda-1} \quad (61)$$

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  $f_a = 44,1kHz \Rightarrow \text{duration} = \frac{t_{\max} = 82000}{f_a} = 2s \Rightarrow \frac{3 \text{ oscillations}}{2s} = 1,5$  oscillations for second (Hz).

The music piece *Vibra e treme* explores those 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<sup>22</sup>. Aiming a qualitative appreciation, the piece also develops a comparison between vibratos and tremolos in logarithmic and linear scales. Its source code is in the Appendix ?? and available online as part of the *MASSA toolbox*.

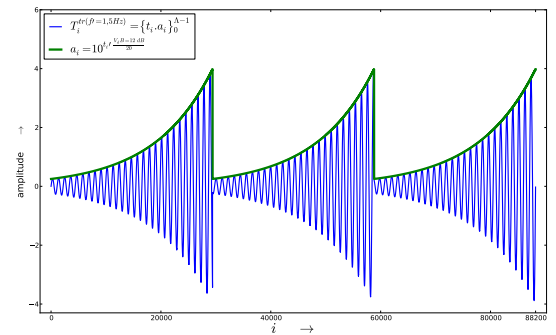


FIG. 21. Tremolo with a depth of  $V_{dB} = 12dB$  with a sawtooth waveform as its oscillatory pattern with  $f' = 1.5Hz$  in a sine of  $f = 40Hz$  (considering a sample frequency of  $f_a = 44,1kHz$ ).

With progressive increasing of  $f'$  for a linear proximity of frequency aiming to hear the phenomena as pitch ( $\approx 20Hz$ ) it (o que?) generates roughness to both tremolos

and vibratos. Those roughness are largely appreciated both in traditional classical music and current electronic music, specially in the *Dubstep* genre. Roughness is also generated by spectral content that produce beating? ? . The sequence *Bela Rugosi* explores the threshold with concomitants of tremolos and vibratos at the same voice, with different intensity and waveforms. The respective code is presented in Appendix ?? and available online in MASSA.

Increasing even more the frequency makes those oscillations no longer remains noticeable. In this case, the oscillations are audible as pitch. Then,  $f'$ ,  $\mu$  and the waveform changes together the frequency 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 to data transferring by means of electromagnetic waves (e.g. AM and FM radios).

For musical ends, it is possible to understand FM based on the case of sines and to decompose the signals into their respective Fourier components (i.e. sinusoidal) to more complex cases. 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  $f'$ :

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

where

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

is the Bessel function? ? which specifies the amplitude of each component in FM synthesis.

On those equations, the frequency variation introduced by  $\{t'_i\}$  does not respect the geometric progression that follows the pitch perfection but reflects the equation 32. Equations 57 are used for FM and are described in Appendix ??, where the spectral content of the FM synthesis is calculated and obtained with oscillations in the

logarithmic scale. In fact, what is attractive in FM is its simple behavior (only with linear variations 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 a distance of  $f'$  from  $f$ . Again, this is obtained by variations in the linear scale of the amplitude. Appendix ?? has an exposition of the spectrum considering an AM performed with an oscillation in the amplitude logarithmic scale. It also loses the simple behavior.

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

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

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

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

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

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

$$T_i^{f, FM(f', \mu)} = \left\{ t_i^{f, FM(f', \mu)} \right\}_0^{\Lambda-1} = \left\{ \tilde{l}_{\gamma_i \% \tilde{\Lambda}} \right\}_0^{\Lambda-1} \quad (68)$$

where  $\tilde{l}$  is the waveform period with a length  $\tilde{\Lambda}$  for the carrier signal.

To perform AM someone just needs to modulate the signal  $\{t_i\}$  with  $\{t'_i\}$  using the following equations:

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

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

## F. Musical usage

At this point, the musical possibilities have been exploded. All characteristics like pitch (given by frequency), timbre (given by the waveform and filters), volume (given by intensity) and duration (given by the number of samples) can be considered by themselves or treated during their duration (with the exception of the duration itself).

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

### 1. Relations between characteristics

Another interesting possibility is to use the relations between parameters of tremolo and vibrato, and some parameters of the basic note like frequency. In this way, it is possible to say that vibrato frequency is proportionally different from pitch, or the tremolo depth is inversely proportional to pitch. Therefore, with equations 55, 57 and 60, it is possible to define:

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

with  $f^{vbr}$  and  $f^{tr}$  as  $f'$  in the referenced equations. They can also be associated with the vibrato and tremolo oscillation frequency in equation 55.

Besides that,  $\nu$  and  $V_{dB}$  are the respective depth values of vibrato and tremolo. Functions  $func_a$ ,  $func_b$  and  $func_c$  are arbitrary and dependents on the musical intentions. The music piece *Tremolos, vibratos e a frequencia* explores such characteristics and shows variations in the oscillation waveform with specific relations with the purpose of building a *musical language*<sup>23</sup>. The corresponding code is on Appendix ?? and it is also available online as part of the MASSAtoolbox.

About convolution it is possible to use a musical pulse as duration – like a BPM pulse – and to distribute impulses during this original pulse, aiming at establishing metrics and rhythms<sup>24</sup>. For example, two impulses equally spaced builds a binary division into the pulse. Two signals, one with 2 pulses and another with 3 pulses, both with equally spaced impulses in the pulse duration, results in the pulse maintenance with a rhythmic mark that is possible to use as binary or ternary divisions in many ethnic or traditions music styles<sup>?</sup>. The absolute values of the impulses results in proportions among the amplitudes of the convoluted signals. Using convolution with impulses as a metric is explored in the music piece *Trenzinho de caipiras impulsivos*. The features embrace the creation of ‘sound amalgam’ based on granular synthesis and this piece provides a link to the next section.

Refer specially to the Figure 24. The source code of the music piece is included in the section ?? and online as well (MASSAtoolbox).

### 2. Moving audio source and receptor, Doppler effect

Resuming the exposition in subsection II G: when an audio source and a receptor are moving, their characteristics are ideally updated at each sample of the digital signal. The speeds are decomposed in relation of each ear direction. In this way, given the audio source speed (or velocity)  $v_s$ , that is positive if the source moves away from receptor, and receptor speed  $v_r$ , that is positive when it gets closer of the audio source, the frequency is given by the well known Doppler effect:

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

With this frequency and relations given by the new 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 relative speed that, in each second, adds the traversed period, by means of a waveform with potency:  $\Delta P = P_0 \left( \frac{v_r - v_s}{343,2} \right)$ , where  $P_0$  represents the signal potency.

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  $y_i = y_0 + v_s - v_r$  if considering  $v_s$  and  $v_r$  both in horizon. Amplitude changes with the distance and with the potency factor cited above (see subsection II B for potency to amplitude conversion).

$$A_i = \left\{ \frac{z_0}{d_i} A_{\Delta P} \right\}_0^{\Lambda-1} = \left\{ \frac{z_0}{\sqrt{y_i^2 + z_0^2}} \sqrt{\frac{v_r - v_s}{343,2} + 1} \right\}_0^{\Lambda-1} \quad (73)$$

Observe that the factor which changes the amplitude (caused by the distance) is even, while the factor 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 becomes steep if both source and receptor intersects each other exactly at the same point, otherwise, there is a monotonic progression. In the given case, where there is a static height  $z_0$ , it is necessary to observe the speed component in the direction between the observer and audio source:

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

In the Appendix ?? there is a Python implementation of the Doppler effect as describe above, also considering the intersection between audio source and receptor.

### 3. Filter and noises (subsections III D and III C)

Using filters the possibilities are many. It is possible to convolve a signal to reverberate it, to remove its noise, to generate distortions or to treat the audio with aesthetically manipulation in mind. For example, it is possible to simulate sounds from and old television if a band-pass filter is applied to accept just sound between  $1kHz$  and  $3kHz$ . Or if someone removes (with some precision) just the frequency of electric oscillation (usually  $50Hz$  or  $60Hz$ ) and the harmonics, it will remove noises caused by audio devices. 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>?</sup>. Chains of those filters can perform complex and more accurate filtering routines. The music piece *Ruidosa faixa* explores those features by using filters and many kinds of noise synthesis. The source code is in Appendix ?? and is available online as part of MASSA.

When used together the features can create an effect known as *chorus*. Based on what happens in a singers choir, in this effect 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 listed in Appendix ?. The script is also available in the MASSAtoolbox.

### 4. Reverberation

Using the same spatialization terms of subsection II G, the late reverberation can be modeled as a convolution with a period of pink, brown or black noise, with exponential decay of amplitude and time related. In this way, the treble attenuation and irregular smooth in the frequency response are contemplated with great success. Delay lines can be added as prefix to 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 localization of the last surface where each wave

front reflected before reach the ear in the fairest 100–200 milliseconds. Then, one could apply a low-pass filter as described in subsection III C. The colored noise can be gradually introduced since the initial moment gives direct incidence (i.e. without any reflection and given by ITD and IID), with *fade-in*, reaching its maximum at the beginning of the 'late reverberation', when the geometric incidences loses their importance to the statistics properties of the noise decay.

As an example, consider  $\Delta_1$  as the duration of the first period and  $\Delta_R$  as the duration of total reverberation ( $\Lambda_1 = \Delta_1 f_a$ ,  $\Lambda_R = \Delta_R f_a$ ). It is possible to add a probability  $p_i$  of a sound reincidencing in the  $i$ -th sample with amplitude with exponential decay. Following subsection II G, the reverberation  $R_i^1$  of the first period can be described as:

$$R_i^1 = \{r_i^1\}_0^{\Lambda_1-1} : \quad r_i^1 = \begin{cases} 10^{\frac{V_{dB}}{20} \frac{i}{\Lambda_R-1}} & \text{with probability } p_i = \left(\frac{i}{\Lambda_1}\right)^2 \\ 0 & \text{with probability } 1 - p_i \end{cases} \quad (75)$$

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 decay:

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

like:

$$R_i = \{r_i\}_0^{\Lambda_R-1} : r_i = \begin{cases} 1 & \text{if } i = 0 \\ r_i^1 & \text{if } 1 \leq i < \Lambda_1 - 1 \\ r_i^2 & \text{se } \Lambda_1 \leq i < \Lambda_R - 1 \end{cases} \quad (77)$$

the reverberation represented by  $R_i$  can be applied as 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 reverb space consists of a tricking (almost a *cliché*) to induce surprise and interest in the listener.

### 5. 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>?</sup>.

The stochastic ADSR envelope is characterized by 4 parameters: attack duration (time to sound reaches its maximum volume), decay duration (follows the attack immediately), level of sustain volume (in which the volume remains stable after the decay) and release duration (after sustain, this is the duration until the volume decays to zero). 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.

It is possible to apply 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, as a sound sequence  $t_i$  defined as:

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

with  $\Lambda_X = \lfloor \Delta \cdot f_a \rfloor \quad \forall \quad X \in (A, D, R, )$  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 terms in the right side of the equation 78 attend for both introduction and ending of sound from the zero intensity because they are linear. Schematically, Figure 22 shows the ADSR envelope, 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 stays. Another common example is the use of more elaborated tracings to attack or decay. The music piece *ADa e SaRa*, available both in Appendix ?? and in MASSA, explores many configurations of the ADSL envelope.

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

#### IV. NOTES ORGANIZATION IN MUSIC

Consider  $S_j = \{s_j = T_i^j = \{t_i^j\}_{i=0}^{\Lambda_j-1}\}_{j=0}^{H-1}$  as the sequence  $S_j$  of  $H$  musical events  $s_j$ . In addition,  $S_j$ , called

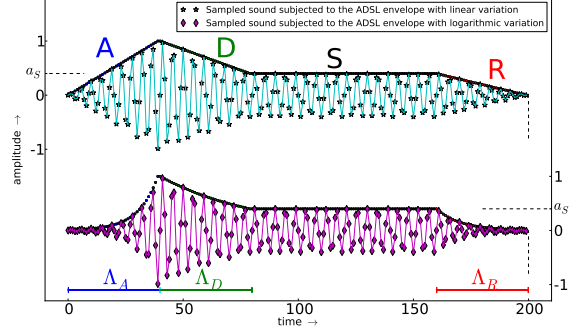


FIG. 22. An ADSR envelope (*Attack, Decay, Sustain, Release*) applied to an arbitrary sound sequence. The linear variation of the amplitude is above. Below the amplitude variation is exponential.

a 'musical structure', as composed by events  $s_j$  which are musical structure 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, building intervals and chords. This scenery reflects the 'vertical thought' in music. On the other hand, the concatenation of  $S_j$  as in equation 30 and in Figure 13, builds melodic sequences and rhythms, which are associated with the musical 'horizontal thought' in music. The fundamental frequency  $f$  and the beginning moment (attack) are generally the most important characteristics of the elements in  $S_j$ . Such characteristics make 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

Here we describe characteristics in music related to harmony and melody.

##### 1. Tuning

Doubling the frequency is equivalent to ascending one octave ( $f = 2f_0$ ). The octave division in twelve notes is the canon of classical western music, although its usage has also been observed outside western tradition, like ceremonial/religious and ethnic context<sup>?</sup>.

The factor given by  $\varepsilon = 2^{\frac{1}{12}}$  defines a semitone. It builds a note grid along the hearing spectrum. Fixing a frequency  $f$ , all the possible fundamental frequencies stay separated by intervals that are multiples of  $\varepsilon$ . Twelve semitones (or half steps), equidistant to the human ear, make an octave. 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. For the real music instruments, however, there are deviations of those frequencies in order to make the harmonics of corresponding notes more compatible. Besides that, the fixed reference  $\varepsilon = 2^{\frac{1}{12}}$  characterizes an equally tempered tuning. There are tunings which respective intervals are ratios of low-order integers, originally based on observations of physical behaviors. First of these tunings were formalized around two thousand years before the existence of an equal temperament<sup>?</sup>.

Two iconic tunings:

- The **just intonation** is defined as notes of the diatonic scale that are associated with ratios of low-order integers that defines the harmonic series. The basic ratios are the same in the Ionian mode (white piano keys from to (see below in subsection IV A 3): 1, 9/8, 5/4, 4/3, 3/2, 5/3, 15/8, 2/1). The intervals are considered as notes in a scale together with the following: 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.
- The **Pythagorean tuning** is based on the interval 3/2 (perfect fifth). The Ionian mode becomes: 1, 9/8, 81/64, 4/3, 3/2, 27/16, 243/128, 2/1. The intervals are also considered as notes in a scale. Besides the intervals in the mode, it is also used 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 attends for micro-tonality<sup>25</sup>. It is possible to use non-integer real values as pitch sequences or even to maintain the integer values and change the factor  $\varepsilon = 2^{\frac{1}{12}}$ . For example, a tuning really near of the harmonic series itself is proposed with the division of the octave in 53 notes:  $\varepsilon_2 = 2^{\frac{1}{53}}$ <sup>?</sup>. In that division the notes are related by means of integers with  $\varepsilon_2$ . Note that if  $S_i$  is a pitch sequence related by means of  $\varepsilon_1$ , a mapping for all the notes related by  $\varepsilon_2$  defines a new sequence  $S'_i = \{s'_i\} = \left\{s_i \frac{\varepsilon_1}{\varepsilon_2}\right\}$ . The music piece *Micro tom* explores this micro-tonal features and its code is in Appendix ?? and is part of MASSA.

## 2. Intervals

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

The nomenclature is based both in impositions and for convenience of the tonal system and practical aspects of note manipulation and can be specified as<sup>?</sup> ? :

- Intervals as number of ratios between notes: first (unison), second, third, forth, fifth, sixth, seventh and eighth. The ninth, tenth, eleventh, etc., are compound intervals made by one or more octaves + a interval inside the octave, which characterizes the compound interval. The intervals are represented by the numeric digits, e.g. 1, 3, 5 are a unison, a third and a fifth, respectively.
- Qualities of each interval: the 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 is both perfect consonance or dissonance according to the context and theoretical background. As general rule, it can be considered as consonance except when it is used as passage to resolve a fifth or a third.
- Tritone is dissonance in Western music because it characterizes the “dominant” in the tonal system (see subsection IV B) and represents the instability. Some cultures chant the interval as consonance.
- 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: considering 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 below, the interval is inverted. When summed to its inversion an interval results in 9 (e.g. m7 is inverted to M2:  $m7 + M2 = 9$ ). An inverted major interval results

TABLE I. The music intervals together with their traditional notation, basic classification for dissonance and consonance, and number of semitones. Perfect (P) unison, fifths and octaves are the perfect 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 the first special case, being a perfect consonance when considering it as the 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 searches for its resolution in a third or sixth. Due to this instability it is considered a dissonance interval.

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

in a minor interval and vice-versa. An inverted augmented interval results in a diminished interval and vice-versa (like a doubly augmented results in a doubly diminished and vice-verse). 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 specified 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. The scale degrees (subsection IV A 3) generally determine 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 far from the 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, consequence of the first degree be between the near two degrees: the second and the leading note<sup>?</sup>.

The descriptions presented here summarizes the traditional theory of the music intervals<sup>?</sup>. The music piece *Intervalos entre alturas* explores these intervals in a independently and conjunction manner. The reader is invited to see the source code in Appendix ?? and available on-line in the MASSAtoolbox<sup>?</sup>.

### 3. Scales

Basically speaking, scale is a set of ordered pitches. Usually, scales repeat at each octave. The ascending sequence with all notes from the octave division in 12 equal intervals, separated by ratio  $\varepsilon = 2^{\frac{1}{12}}$ , is known as the chromatic scale with equal temperament. There are 5 perfect symmetric divisions of the octave within the chromatic scale. These divisions are considered as scales by the easily and peculiar use they provide. Considering the integers that the factor  $\varepsilon = 2^{\frac{1}{12}}$  is powered to multiply  $f_0$ , the scales are the following:

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

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

The diatonic scales are:

#### 4. Chords

$$\begin{aligned}
\text{natural minor} &= \text{aeolian mode} = \\
&= E_i^m = \{e_i^m\}_0^6 = \{0, 2, 3, 5, 7, 8, 10\} \\
\text{locrian mode} &= E_i^{mlo} = \{e_i^{mlo}\}_0^6 = \{0, 1, 3, 5, 6, 8, 10\} \\
\text{major} = \text{ionian mode} &= E_i^M = \{e_i^M\}_0^6 = \{0, 2, 4, 5, 7, 9, 11\} \\
\text{dorian mode} &= E_i^{md} = \{e_i^{md}\}_0^6 = \{0, 2, 3, 5, 7, 9, 10\} \\
\text{phrygian mode} &= E_i^{mf} = \{e_i^{mf}\}_0^6 = \{0, 1, 3, 5, 7, 8, 10\} \\
\text{lydian mode} &= E_i^{ml} = \{e_i^{ml}\}_0^6 = \{0, 2, 4, 6, 7, 9, 11\} \\
\text{mixolydian mode} &= E_i^{mmi} = \{e_i^{mmi}\}_0^6 = \{0, 2, 4, 5, 7, 9, 10\}
\end{aligned} \tag{81}$$

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

All the diatonic scales follow a pattern of successive intervals *tone, tone, semitone, tone, tone, tone, semitone*. Thus, it is possible to write:

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

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

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

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

The ascending and descending contour of the melodic minor scale is necessary for the existence of the leading tone (seventh and last degree, separated by one semitone of the octave, enhances the tonic polarization) in the ascending trajectory, which is not necessary in the descending mode, since it recovers the normal form. The harmonic scale presents the leading tone but does not avoid the augmented second interval between the sixth and seventh degrees. In addition, it does not need to cover a melodic trajectory, just having to present the leading tone, crucial to the tonal system (the leading tone tends to the tonic, asserting it)<sup>?</sup>. Other scales can be represented in a similar form, like the pentatonic and the modes of limited transposition by Messiaen<sup>?</sup>.

The simultaneous occurrence of three or more notes is observed by means of chords. Their base of tonal music are the triads. 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 referred as inverted. A close position is where any note of chord fits between two consecutive notes<sup>?</sup>. All the non-inverted triads in the close position form and with root in 0 are described as following:

$$\begin{aligned}
\text{major triad} &= A_i^M = \{a_i^M\}_0^2 = \{0, 4, 7\} \\
\text{minor triad} &= A_i^m = \{a_i^m\}_0^2 = \{0, 3, 7\} \\
\text{diminished triad} &= A_i^d = \{a_i^d\}_0^2 = \{0, 3, 6\} \\
\text{augmented triad} &= A_i^a = \{a_i^a\}_0^2 = \{0, 4, 8\}
\end{aligned} \tag{84}$$

To consider 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. The inversions and open positions can be obtained with a selective addition of 12 to the components.

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

In the following we present general recommendations:

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



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

Omission of the basic chaining is the key to obtain modal and atonal harmonies. In case of absence of these minimal tonal structures, we say that the harmony is modal if the notes match with some diatonic scale (see equations 81) or if the notes are presented in a small number. If basic tonal chaining are absent, the notes do 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 are not tonal neither atonal. The modal harmony 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?

### 1. Atonal harmony

In fact, the techniques around atonal music comprehend structures for avoiding the hearing relation with modes and tonality. This difficulty imposes the use of dodecafonism. 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 (no sei se ficou mt claro). In this context, the tonic becomes difficult to settle. Nevertheless, the Western hearing searches for tonal traces in musics and easily finds them by unexpected paths and even tortuous.

The use of dissonant intervals (specially tritones without resolution), together with ninths, seconds and sevenths, reinforces the absence of tonality. Considering this context, it is possible to account for the following considerations while creating a music piece:

- Repeating notes. Considering the immediate repetition as an extension of the previous incidence, the use of same notes in sequence does not add relevant information.
- It is interesting to play adjacent notes at the same time, making harmonic intervals and chords.
- Present durations with liberty, respecting the notes order.
- In order to gain variation, one can use 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. Most of them

were written by great dodecafonic composers like Alban Berg and even Schoenberg. Many pieces create purposeful links between tonal and atonal techniques.

### 2. Tonal harmony

In the XX century, rhythmic music with emphasis on sonorities/timbres, extended the concepts about tonality and harmony. Nevertheless, the tonal harmony has strong influence in the artistic movements and commercial venues. In addition, the dodecafonism is considered having tonal nature because it was created to deny the tonal characteristics of polarization.

In tonal or modal music, the base of the harmonic filed are chords – like the ones listed in equations 84– built with a root note for each degree of a scale – like the ones in equation 81. In this context, music harmony aims at studying the observation of incident 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 harmony', a scale has its corresponding tonic (first scale degree) and can be major (with the same notes of the Ionian mode) or minor (notes from Eolian mode form the 'natural minor', which has both a harmonic and a melodic version, see equation 83). The scale is base on triads, each one with its root and representing a degree into the scale:  $\hat{1}, \hat{2}, \hat{3}, \hat{4}, \hat{5}, \hat{6}, \hat{7}$ . To build triads, the third and the fifth note above the root is considered together with the root (or fundamental). It is possible to write  $\hat{1}, \hat{3}, \hat{5}$  as the first degree chord, built by the first degree of scale and the center for a tonal music. The chords of the fifth degree  $\hat{5}, \hat{7}, \hat{2}$  ( $\hat{7}$  sharp when a minor scale) and of the forth degree  $\hat{4}, \hat{6}, \hat{1}$  are secondary. After that, other degrees are considered. The traditional harmony comprises conventions and stylistic techniques to create chord chains, made in each scale degree?

The 'functional harmony' ascribes functions to these three central chords and tries to understand their use by means of these functions. The chord built under the first degree is named the **tonic** chord ( $T$  or  $t$  for a major or minor tonic, respectively) and its (role) function consisting of preserving a center, usually referred as a "ground" for the music. The chord built under the fifth degree is the **dominant** ( $D$ , the dominant is always major) and its function is call for the tonic (we say that the dominant chord asks for a conclusion and this conclusion is usually the tonic chord). Thus, the dominant chord guides the music to the tonic. The triad built under the forth degree is the **subdominant** ( $S$  or  $s$  for a major or minor subdominant, respectively) and its basically function is to distance the music from the tonic. The process aims at confirming the tonic using tonic-subdominant-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 asso-

ciated relative (relative tonic  $Tr$ , relative subdominant  $Sr$  and relative dominant  $Dr$ ) is the triad built with 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 with a third above. In the minor scale the same happens, but the triad with a third below is called counter-relative ( $tA$ ,  $sA$ ) and the triad with one third above is called relative ( $tR$ ,  $sR$ ). The precise functions and musical effects of these chords are controversial. Table II shows the relation between the triads built in each degree of the major scale.

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: $\hat{4}, \hat{6}, \hat{1}$	$\hat{6}, \hat{1}, \hat{3}$

The dominant counter-relative should form a minor chord. It explains the change in the fourth degree by a semitone above  $\hat{7}\#$ . The diminished chord  $\hat{7}, \hat{2}, \hat{4}$ , is generally considered a 'dominant seventh chord with no root'? . In the minor mode, there is the change in  $\hat{7}$  by an ascending semitone to make possible the existence of an unique semitone separating the tonic, and making also possible the dominant (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 as 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 dominant or individual subdominant, which is the chord based in the third formed by a fifth above or by a fifth below, respectively. These dominants and individual subdominants, in the same way, have also subdominants and individual dominants that can be used. 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 present in each chord in two ways: the upper chromatic mediant, formed by the root at the third of original chord; and the lower chromatic 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, the mediant is called 'doubly chromatic mediant'. Again, there are two forms for each chord: the upper form, with a third in the fifth of the original chord; and the lower form, with a third in the root of the original triad. It is interesting to observe that a major chord has upper chromatic mediant and upper doubly chromatic mediant. A minor chord has lower chromatic mediant and lower doubly chromatic mediant. This relation between chords is considered as advanced tonalism, sometimes even considered as expansion and dissolution of tonalism, and, as consequence, it has strong and outstanding effects although perfectly consonant. The chromatic mediant are used since the end of Romanticism by Wagner, Liszt, Richard Strauss, among others, and are quite simple to be realized? ? .

### 4. Modulation

Modulation is the change of the key (tonic, or tonal center) in a music. It can be characterized by observing the key at the start and at end of the chord transitions. Keys are always conceived as related by fifths and their relative and counter-relatives. Forms to perform modulation include:

- 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.
- Using carefully an individual dominant and perhaps the individual subdominant, to confirm the change in key and harmonic field.
- Using chromatic alterations to reach a chord in the new key starting from a chord in the previous key. It is called chromatic modulation.
- Featuring a unique note, possibly repeated or suspended with no accompaniment, common to start and end a key, 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. Key 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? ? ? ? . The music piece *Acorde cedo* explores these chord relations. Its code is available both in Appendix ?? and online as part of MASSA? .

### C. Counterpoint

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

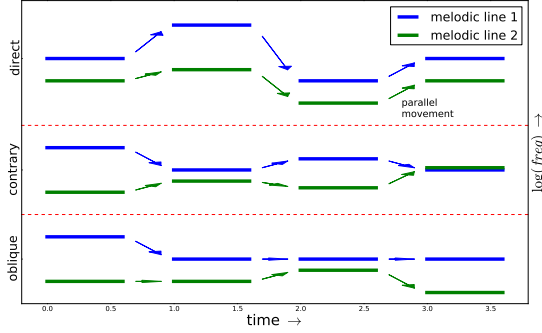


FIG. 23. Different motions of counterpoint aiming to preserve independence between voices. There are 3 types of motions: direct, contrary and oblique. The parallel motion is a type of direct motion.

The purpose of counterpoint is to conduct voices in a way they sound independent. In order to do that, the relative motion of voices (in pairs), is crucial and categorized as: direct, oblique and contrary motion (see Figure 23). The parallel motion is an oblique motion. The gold rule here is to take care with the direct motions, avoiding making them lead to a perfect consonance. The parallel motion should occur only between imperfect consonances and no more than three consecutive times. The 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 a note to a neighbor sound more coherent. When having 3 or more voices, the melodic importance lies on the higher and lower voices, in this particular order? ? ? .

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

### D. Rhythm

Basically speaking, the rhythm notion is dependent on events separated by durations? . These events can be heard individually if spaced by at least 50 – 63ms. To make the temporal separation between them be perceived as duration, the period should be large enough, around 100ms? . It is possible to summarize the duration transitions heard as rhythm and pitch, in the following way? ? .

The duration region marked as transition is presented 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 sound characteristics? ? .

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 cognitively studies, it is common to associate these range of pulse frequencies to the durations of the heart beat, movements of inspiration and expiration and walk steps? ? .

The pulse is subdivided into equal parts and repeated in sequence. These relations (division and concatenation) usually follow relations of small order integers<sup>26</sup>. A schematic exposition is shown in Figure 24.

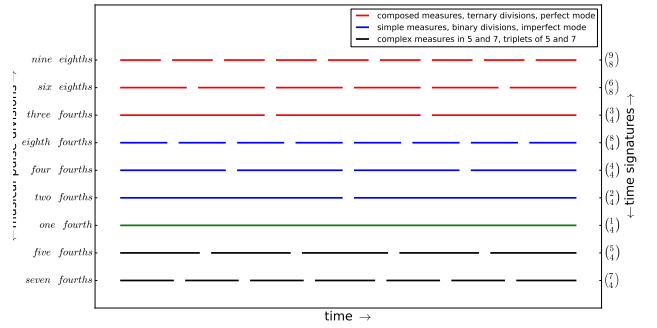


FIG. 24. Divisions and grouping in bars for the music pulse to establish a rhythmic metric. Divisions of the quarter note, established as the pulse, is presented in the left. The time signature whose specifies the same metrics but having as scale the groupings of the music pulse is presented in the right.

Dual relations (common time and binary divisions) are frequent in dance rhythms and celebrations. Ternary relations are common in ritualistic music and is related to the sacred. Dual relations are called imperfect, while and the ternary relations are called perfect.

The strong unities (accents) are the ones that fall in the 'head' of bars, the downbeats. The head of a unity is the first part of the subdivision. In binary divisions (2, 4 and 8, for example), the unities originally strong turn into weak (no sei se isso ficou claro) (e.g. division in 4 is: strong, weak, strong medium, weak). In ternary divisions (3, 6 and 9) two weak unities succeeds the downbeat (e.g. division in 3 is: strong, weak, weak)<sup>28</sup>

The accent in the weak beat is known as the backbeat, whereas a note starting in a weak beat which duration coming across the strong beat is known as syncopé.

Notes can occur inside and outside of these 'music metric' divisions. In most behaved cases, notes occur exactly on these divisions, with broader incidence on attacks of strong beats. In extreme cases, the time metric cannot be perceived? . Small grid variations help to build musical interpretation and differences between styles? .

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

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

Special attention should be given to the limits of  $j$ : pulse divisions are durations sensible as rhythm. Furthermore, the pulse joints sums, 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 few hours.

Each level  $j$  has some indexes  $i$ . A index  $i$  always has three different values (or multiple of three) that occur in a perfect relation. When  $i$  has only multiple of two, four or eight different values, than an imperfect relation occurs, as shown in Figure 24.

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

$$P_{\{i_k\}}^{\{j_k\}} \quad (85)$$

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

As 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  $P_2^1$ . Each unit or unit set  $P_i^j$  can be associated with a sequence of temporal samples  $T_i$  that forms a music note.

The music piece *Poli Hit Mia* explores different metrics (available in Appendix ?? and as part of MASSA).

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

Given the basic music structures, either frequential (chords and scales) or rhythmic (beat divisions and simple, compound and complex grouping), it is natural to organize these structures in a cohesive way<sup>7</sup>. To this end, the concept of arcs are fundamental: we make an arc by departing from a place (music unit) and going to another one. The audition of melodic and harmonic lines is pervaded as musical arcs thanks to the cognitive nature of the musical hearing. It is possible to consider the note as the smaller arc, and each motif (definir motif aqui? nao foi falado ainda) and melody as another arc. Each time, beat subdivision, measure and music section constitutes an arc. A music in which the arcs do not present consistency by themselves, can be understood as a music with no cohesion. The coherence feeling comes, mostly, from the skilled manipulation of arcs in a music piece.

Musical arcs are abstract structures and liable of basic operations. An 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  $S_j = \left\{ s_j = T_i^j = \{t_i^j\}_0^{\Lambda_j-1} \right\}_0^{H-1}$  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 period) 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 is a displacement  $\tau$  of a material for a different pitch  $\Xi' = \Xi + \tau$ . It is a variation that changes pitch localization of material:  $\{s'_j\} = \{s_j^{\Xi'}\} = \{s_j^{\Xi+\tau}\}$  where  $\Xi$  is the pitch of a period  $S_j$  or of an event  $s_0$  of the original structure  $S_j$ . Observe that  $\tau$  is the pitch offset of the displacement. If  $\tau$  is given in semitones, the displacement in frequency is  $\tau_f = f_0 \cdot 2^{\frac{\tau}{12}}$  where  $f_0$  results from the adoption of some reference:  $f_0 = \Xi_{f_0} Hz \sim \Xi_0$  absolute value of pitch. For the frequency of any pitch value:  $\Xi_f = \Xi_{f_0} \cdot 2^{\frac{\Xi - \Xi_0}{12}}$ . For the pitch of any frequency value:  $\Xi = \Xi_0 + 12 \cdot \log_2 \left( \frac{\Xi_f}{\Xi_{f_0}} \right)$ . In the MIDI protocol,  $\Xi_{f_0} = 55 Hz$  while pitch  $\Xi_0 = 33$  marks a *A 1*, another reference point is  $\Xi_{f_0} = 440 Hz$  which corresponds to  $\Xi_0 = 69$ . The unit difference is equal for semitones. It is not possible to say that the unit is the semitone because  $\Xi = 1$  is not a semitone, but a note with an audible frequency as rhythm, with less than 9 occurrences each second (see Table III).
- Interval inversion is the inversion traversed by the material. The inversion is strict if the number of

semitones is being used as a reference for the operation:  $S'_j = \{s'_j\}_0^{H-1} = \left\{s_j^{-\varepsilon_j \cdot f_0}\right\}$ , where  $\varepsilon_j$  is the factor between the event frequency  $s_j$  and the frequency of  $s_0$ . The inversion is said tonal if the distances are considered in terms of degree number of

the scale  $E_k$ :  $S'_j = \{s'_j\}_0^{H-1} = \left\{s_j^{-\varepsilon^{(e_{j_e})} \cdot f_0}\right\}_0^{H-1}$

where  $j_e = e_{j_e}^{inv}$  is the index  $k = j_e$  in  $E_k$  of the note of event  $s_j$ .

- Rotation of musical elements is the transference of the last element to the position of the first element, and the displacement of that to the penult, one position ahead. It is possible to define rotation of  $\tilde{n}$  positions by  $S'_n = S_{(n+\tilde{n})\%H}$ . If  $\tilde{n} < 0$ , it is enough to use  $\tilde{n}' = H - \tilde{n}$ . It is reasonable to associate  $\tilde{n} > 0$  with the clockwise rotation and  $\tilde{n} < 0$  with the anti clockwise rotation. More information about rotations is presented in subsection IV G.
- The insertion and removal of materials into structure  $S_j$  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 the empty instant. 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  $S_j$  or making it the prefix or 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.

Other processings can be derived together with the presented ones, as, for example, the retrograde inversion, temporal contraction with an external suffix, etc. The music structures resound in the human cognitive system due to the own nature of thought. In its many veneers, an idea read with the same number of elements and connective aspects between them. The music, when tuned with those mental structures, raises impressions. In this way, a whole process of mental and neurological resonance is unleashed, responsible by the feelings, memories and imaginations, typical to a mindful music listening. This cortical activity helps the musical therapy, known by its utility in cases of depression and neurological injury. It is known that regions of the human brain responsible by hearing processing, are also used for other activities, even linguistic and mathematical? .

The paradigmatic structures guide the creation of new music material. One of the most central of them is the dipole tension/relax. Traditional dipoles are relate to tonic/dominant, repetition/variation, consonance/dissonance, coherence/rupture, symmetry/asymmetry, equality/difference, arrival/departure,

near/far, stationary/moving, etc. The ternary relations tend to relate to the circular and 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.

## F. Directional structures

The arcs can be decomposed in two convergent sequences: the first arc reaches the apex and the second arc returns from the apex to the 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 parameter of variation does not exist, thus the arc consists only of the reference structure? .

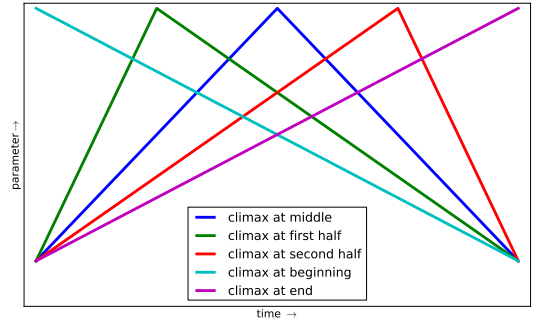


FIG. 25. Canonical distinctions of musical climax in a given melody and other domains. The considered possibilities are: climax at beginning, climax at first half, climax at middle, climax at second half and climax at the end. The y-axis is not properly specified since there exist no real parametric variation and, therefore, the structure is a reference.

Consider  $S_i = \{s_i\}_0^{H-1}$  as an increasing sequence. The sequence  $R_i = \{r_i\}_0^{2H-2} = \{s_{(H-1-|H-1-i|)}\}_0^{2H-2}$  is a sequence that presents a perfect profiteer symmetry, i.e. the second half is the mirrored version of the former. Following the musical concepts, the climax is localized in the middle of the sequence. It is possible to modify this fact using sequences with different sizes. All the mathematical theory of sequences, already well established and taught routinely in calculus courses, can be used to generate those arcs? . Theoretically, when applied to any characteristic of music events, these sequences produce arcs, since they imply the deviation and return of an initial parametrization. Henceforth, it is possible to have a number of distinct arcs, with different sizes and climax for a same sequence of events. This is an interesting and useful resource, and the correlation of arcs results in coherence? .

Historically (and practically nowadays), the golden ratio and the Fibonacci sequence have special importance. The Lucas sequence allows the generalization of Fibonacci sequence, making it easy to understand. 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 occurring in the first values with relation 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 about the same error progression but starts right at the second step since  $\frac{1}{1} \approx \frac{101}{100}$ .

The music piece *Dirracional* exposes the use of arcs into directional structures. Its source code is included in Appendix ?? and available online as part of MASSA<sup>?</sup>.

### G. Cyclic structures

The philosophical understanding that the human thought is based on the perception of similarities, differences, stimuli and objects, places the symmetries at the core of the cognitive process<sup>27</sup>. Mathematically, symmetries are algebraic finite groups, and a finite group is always isomorphic to a permutation group. It is possible to say that permutations represent any symmetry in a finite system (precisa de uma referencia aqui). In music, permutations are ubiquitous and considerably present in techniques, which confirms their central role. The successive application of permutations generates cyclic arcs<sup>?</sup> <sup>?</sup> <sup>?</sup>. Two academic studies were dedicated to this end, aiming at generating music structures<sup>?</sup> <sup>?</sup>.

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

$$\begin{aligned}
 \forall p_1, p_2 \in G &\Rightarrow p_1 \bullet p_2 = p_3 \in G \\
 &\quad \text{(closure property)} \\
 \forall p_1, p_2, p_3 \in G &\Rightarrow (p_1 \bullet p_2) \bullet p_3 = p_1 \bullet (p_2 \bullet p_3) \\
 &\quad \text{(associativity property)} \\
 \exists e \in G : p \bullet e &= e \bullet p \quad \forall p \in G \\
 &\quad \text{(identity element property)} \\
 \forall p \in G, \exists p^{-1} : &p \bullet p^{-1} = p^{-1} \bullet p = e \\
 &\quad \text{(inverse element property)} \quad (86)
 \end{aligned}$$

From the first property it is possible to conclude that any permutation can be operated with another permutation. In fact, it is possible to apply a permutation  $p_1$  and another permutation  $p_2$ , and, comparing both initial and final orderings, results in 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 called the element order. Thus, a finite permutation  $p$ , successively applied, reaches the initial ordering of its elements, making a cycle. This cycle, if used as parameter of music notes, implies in a cyclic musical arc.

These arcs can be established by using a set of different permutations. As a historic example, the tradition called *change ringing* conceives the 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 represents one traditional *peal* 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, the music structure consists of permutations by itself and some different permutations operate due to the cyclic behavior.

TABLE IV. Change Ringing: Standard *peal* for 3 bells. Permutations occur between each line. Each line is a bell ordering and each ordering is played a line 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: consider  $S_i = \{s_i\}$  as 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. The permutations can be written based on two notations: cyclic or 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  $[0 \ 1 \ 2 \ 3 \ 4 \ 5 \ \dots]$ , the permutation is noted by the sequence it produces (ex.  $[1 \ 3 \ 7 \ 0 \ \dots]$ ). In the cyclic notation, permutation is the changing of one element by the posterior one, and the last element by the first one.

It is not necessary to permute elements of  $S_i$ , but only some characteristics. In this way, being  $p^f$  a permutation in frequency and  $S_i$  a sequence of basic notes as exposed in the end of section IIF, the new 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  $p^f$  presents.

It is worthwhile to mention two subtleties of this procedure. First, the permutation  $p$  does not have to involve all elements of  $S_i$ , i.e. it can operate in subsets of  $S_i$ . Second, the elements  $s_i$  do not have to be entirely executed at each state access. To exemplify, consider  $S_i$  as a se-

quence 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 set aside those notes to the first four notes of  $S_i$ .

Each one of the permutations  $p_i$  described above relates the note dimensions where it operates (frequency, duration, *fades*, intensity, etc) and the period of incidence (at how much access a permutation is applied). During realization of notes in  $S_i$ , an easy and coherent form is to execute the first  $n$  notes<sup>?</sup>.

Appendix ?? and MASSA present the computational implementation about permutations<sup>??</sup>.

## H. Musical idiom?

There are many studies intending to model and explore a 'musical language', or 'linguistics applied to music' or even to discern between different 'musical idioms'<sup>???</sup>. In a simple way, a musical idiom is the result of a chosen material together with repetition of elements and repetition of relations between elements present along the music. The related dichotomies are outstanding, as explained at subsection IV E: repetition and variation, relax and tension, equilibrium and instability, consonance and dissonance, etc.

## I. Musical usages

The basic note was defined and characterized in quantitative terms (section ??). Next, the internal note composition was addressed and both internal transitions and immediate treatment were understood (section ??). Finally, this section aims at organizing these notes in music. The gamma of resources and consequent infinitude of possibilities is a typical situation and highly relevant for art contexts<sup>1?</sup>.

There are studies dedicated to each one of the presented resource. For example, it is possible to obtain the 'dirty' triadic harmonies (with notes without the triad) by superpositions of dirty fourths. Another interesting example is the simultaneous presence of rhythms in different metrics, constituting what is called *polyrhythms*. The music piece *Poli-hit mia* explores these simultaneous metrics by impulse train (??) convolved with notes that compound the line. Its source code is included in Appendix ?? and available online as part of MASSA.

The microtonal scales are important for the music of the 20th century<sup>?</sup> and present diverse outstanding results, like the 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 presented in Appendix ?? and available online in MASSA.

As pointed in subsection III F, the relations between parameters are powerful ways to acquire music pieces. The number of permuted notes can vary during the music, revealing relationship with piece duration. Harmonies can be made of triads (eqs. 84) 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 presented symmetries at octave divisions (eqs. 80) and the symmetries presented as permutations (Table IV and eqs. 86) can be used together. In the music piece *3 trios* this association is done in a systematic way in order to enable the required listening. This is a instrumental piece, not included as a source code<sup>?</sup>.

The *PPEPPS* (Pure Python EP: Projeto Solvente) is an EP synthesized using the resources presented in this study. With minimal parametrization, the program generates complete musics, allowing the easy composition of musics and sets of musics. Using few lines of code it is possible to obtain a directory with the generated musics. This facility and technological demystification create aesthetically possibilities for both sharing and education.

## V. CONCLUSIONS AND FUTURE WORKS

We have described a concise system that explores the musical elements and relate them to the digital signal. The reader is invited to access *Scripts* and MASSA, where these relations are computationally implemented. We hope the didactic report along the paper and the supplied scripts facilitates and encourages the use of the proposed framework.

The open possibilities provided by the techniques and results discussed in this paper involve psychoacoustic experiments and the creation of interfaces for the generation of music, noise and other music applications in a high fidelity (*hi-fi*). It is worthwhile to mention the benefit of these results for artistic and didactic purposes. The incorporation of programming skills is facilitated by the provided visual aids. Initial *live-coding* practices and guided courses based on this work have already been realized with successful. Examples are Puredata and Chuck.

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 (as shown in Appendix ??).

Besides, the free software license and online availability of the exposed content as hypertext, in conjunction with the respective codes and sound examples, strongly facilitates future collaborations and the generation of sub-products in a co-authorship fashion. As consequence, the expansion of MASSA is feasible and straightforward, making it possible new developments and implementations of musical context.

In addition, this work permitted the formation of



groups with common interests, like 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 (no devo ter traduzido certo), georeferencing techniques, artistic and educational activities. Some of these reports are available in Appendix ?? and made online. There are more than 700 videos, written documentations, original softwares and contributions in well-known external softwares such as Firefox, Scilab, LibreOffice, GEM/Puredata, to name a few? ? ? .

Future works include the application of the reported results in machine learning and artificial intelligence methods for the generation of appealing artistic materials.

<sup>1</sup>Anton Webern, *The Path To The New Music* (Theodore Presser Company, 1963).

<sup>2</sup>William Lovelock, *A Concise History of Music* (Hammond Textbooks, 1962).

<sup>3</sup>Timbre is a subjective and complex characteristic. Physically, the timbre is multidimensional and given by the temporal dynamics behavior 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>4</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>5</sup>Also called *bias* or *offset*.

<sup>6</sup>Any sampled signal has this property, not only the digitalized sound.

<sup>7</sup>Equal real part and imaginary with inverse order:  $a_{k1} = a_{k2}$  and  $b_{k1} = -b_{k2}$ . As consequence the modules are equal and phases have inverse order.

<sup>8</sup>As pointed before, this limitation simplify the explanation without losing generality, and will be overcome in the next section.

<sup>9</sup>It is known that  $\zeta \approx 21,5cm$  to an adult human.

<sup>10</sup>The gold tip here is: to make a sound that incites interest by itself, do internal variations on it? .

<sup>11</sup>In other words, a geometric progression of frequency is perceived as an arithmetic progression of pitch.

<sup>12</sup>The volume change (psychophysical quality) is consequence of different sound characteristics like reverberation and concentration of treble harmonics, among which is wave energy. Wave energy is the easiest one to modify (see Equation 2) and it can vary in many ways. The most simple way consists of modifying the amplitude by multiplying the whole sequence by a real number. The increased energy without amplitude variation is the *sound compression*, useful nowadays for music production? .

<sup>13</sup>The implementation of filters comprehends an area of recognized complexity with dedicated literature and software implementations. We recommended the reader to consult the bibliography? ? .

<sup>14</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>15</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>16</sup>Butterworth and Elliptical filters can be considered as specific cases of Chebichev filters? ? .

<sup>17</sup>Attention with the cutoff frequency  $f_c$  in low-pass and high-pass filters.

<sup>18</sup>Although its origin is disparate given its association with the brown color, this noise become established with this specific name. Anyway, this association is satisfactory once the white and pink noises are more strident and associated with intense colors? ? .

<sup>19</sup>Some musical instruments and contexts use different terms. For example, in piano, the called tremolo is a vibrato and a tremolo is 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 an specific instrument, practice or musical tradition? ? .

<sup>20</sup>The pitch deviation is called 'vibrato depth' and is generally given as semitones or cents, as convenience.

<sup>21</sup>Tremolos and vibratos occur many times together in a traditional music instrument and voices.

<sup>22</sup>Details in the next section.

<sup>23</sup>It is important to remember that the convolution with a impulse results in a sound shifted to the moment that the impulse occurred.

<sup>24</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 it is also a *convention* adopted even by western classical music. Other tunings are present. To cite one example, the traditional Thai music uses a octave division in seven notes equally spaced ( $\varepsilon = 2^{\frac{1}{7}}$ ), resulting in intervals that have little in common with the intervals in the diatonic scale? .

<sup>25</sup>In ascending order of occurrence in written and ethnic music, these indicate the musical pulse divisions and their sequential grouping at time: 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 divisions 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 commonly compositions and decompositions of 1-9 equal parts? ? .

<sup>26</sup>Division in 6 is considered compound but can also occur as a binary division. A binary division which suffers a ternary division results in 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 applying ternary division that suffers a binary division, resulting in: a strong unit (subdivided in strong and weak) and two weak units (subdivided in strong and weak for each).

<sup>27</sup>G. Deleuze, *Difference and Repetition* (Continuum, 1968).

<sup>28</sup>The execution of disjoint notes of  $S_i$  equals to modify the permutation and to execute the first notes.

<sup>29</sup>F. de Saussure, *Course in General Linguistics* (Books LLC, 1916).

<sup>30</sup>A. Papoulis S. U. Pillai, *Probability, Random Variables and Stochastic Processes* (McGraw Hill Higher Education, 2002).

<sup>31</sup>R. A. Johnson D. W. Wichern, *Applied Multivariate Statistical Analysis* (Prentice Hall, 2007).

<sup>32</sup>C. W. Therrien, *Discrete Random Signals and Statistical Signal Processing* (Prentice Hall, 1992).

<sup>33</sup>R. O. Duda P. E. Hart D. G. Stork, *Pattern Classification* (Wiley-Interscience, 2000).

<sup>34</sup>L. da F. Costa R. M. C. Jr., *Shape Analysis and Classification: Theory and Practice (Image Processing Series)* (CRC Press, 2000).

<sup>35</sup>D. Papineau, *Philosophy* (Oxford University Press, 2009).

<sup>36</sup>Bertrand Russel, *A History of Western Philosophy* (Simon and Schuster Touchstone, 1967).

<sup>37</sup>F. G. G. Deleuze, *What Is Philosophy?* (Simon and Schuster Touchstone, 1991).

<sup>38</sup>Alan Walker, *Hans von Bülow: a life and times* (Oxford University Press, 2010).



- <sup>39</sup>Roy Bennett, *History of Music* (Cambridge University Press, 1982).
- <sup>40</sup>Aaron Kozbelt, "Performance time productivity and versatility estimates for 102 classical composers," *Psychology of Music* **37**, 25–46 (2009).
- <sup>41</sup>Aaron Kozbelt, "A quantitative analysis of Beethoven as self-critic: implications for psychological theories of musical creativity," *Psychology of Music* **35**, 144–168 (2007).
- <sup>42</sup>Dean Keith Simonton, "Emergence and realization of genius: The lives and works of 120 classical composers," *Journal of Personality and Social Psychology* **61**, 829 – 840 (1991), ISSN 0022-3514.
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- <sup>45</sup>Peter van Kranenburg, "On measuring musical style – the case of some disputed organ fugues in the J.S. Bach (BWV) catalogue," *Computing In Musicology* **15** (2007-8).
- <sup>46</sup>Joe H Ward Jr, "Hierarchical grouping to optimize an objective function," *Journal of the American Statistical Association* **58**, 236–244 (1963).
- <sup>47</sup>Hal Varian, "Bootstrap tutorial," *The Mathematica Journal* **9** (2005).
- <sup>48</sup>Christian P. Robert, "Simulation in statistics," in *Proceedings of the 2011 Winter Simulation Conference* (2011) arXiv:1105.4823.
- <sup>49</sup>J. Cohen *et al.*, "A coefficient of agreement for nominal scales," *Educational and psychological measurement* **20**, 37–46 (1960).
- <sup>50</sup>D.D. Boos, "Introduction to the bootstrap world," *Statistical Science* **18**, 168–174 (2003).