DIGITAL MORPHOPHONE ENVIRONMENT. COMPUTER RENDERING OF A PIONEERING SOUND PROCESSING DEVICE

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ABSTRACT

This paper introduces a digital reconstruction of the *morphophone*, a complex magnetophonic device developed in the 1950s within the laboratories of the GRM (*Groupe de Recherches Musicales*) in Paris. The analysis, design, and implementation methodologies underlying the *Digital Morphophone Environment* are discussed.

Based on a detailed review of historical sources and limited documentation – including a small body of literature and, most notably, archival images – the core operational principles of the *morphophone* have been modeled within the MAX visual programming environment. The main goals of this work are, on the one hand, to study and make accessible a now obsolete and unavailable tool, and on the other, to provide the opportunity for new explorations in computer music and research.

1. INTRODUCTION

The advent of magnetic tape-based recording in the early 1950s led to the development of sound-processing systems aimed at generating echo and reverberation, achieved through tape loops, multiple recording and playback heads. Such devices were in wide use in musical applications and in different technical contexts. One notable example is the Philips *delay wheel*, a system developed at the Eindhoven research laboratories for the purpose of acoustic correction of concert rooms [1].

The activities initiated by Pierre Schaeffer (1910-1995), particularly the experiments on *musique concrète* at the GRM (*Groupe de Recherches Musicales*) in the late 1940s and the early 1950s, were relevant in defining technological infrastructures as an aid to composition. In this context the *morphophone* (Figure 1) was developed, a very unique device enabling magnetophonic transformations.

By intervening in the normal operation of magnetophones, Schaeffer and Jacques Poullin (1920-2014) devised new tools for processing sound. In 1952 Poullin made the *tri-pistes*, the ancestor of multichannel magnetophones, based on the synchronous sliding of three magnetic tapes. The following year the *phonogène* was made, which used a small musical keyboard to modulate the sliding speed and transpose the signal recorded on the tape. Following the same principle Poullin made a variant, the *phonogène à coulisse*, which allowed continuous signal transposition, independent of the tempered chromatic scale. In 1953 Poullin also developed the *morphophone*, this time with some input from the physicist Abraham Moles (1920-1992)[1].

Since at least the 1990s, there has been much research aimed at digital modeling of analog audio systems (*Virtual Analog*) from creating circuit models of generic audio equipment, to more specific sound synthesis and processing devices [2, 3, 4, 5, 6, 7, 8, 9, 10, 11]. The literature related to modeling magnetophonic processing devices is of special interest for us here, although very limited in quantity [12, 13, 14].

The present work aims to analyze the behavior of an obsolete device and to make possible using it in current musical contexts.

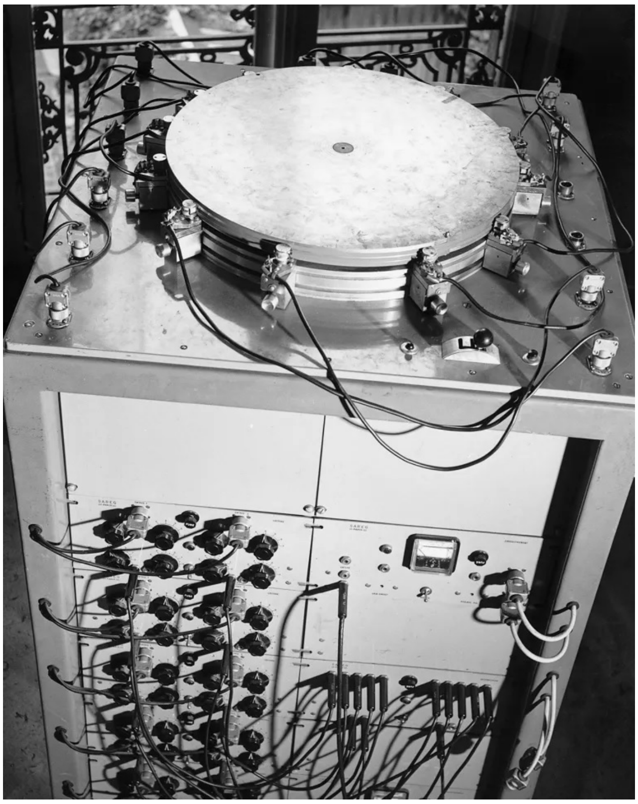

Figure 1: *Overview of the morphophone [15].*

[1] The *phonogene* was patented by Schaeffer and made in collaboration with the *Tolana* company. Prototypes of the *phonogène à coulisse* and the *morphophone* were produced in collaboration with an outside company called *S.A.R.E.G.*

2. MORPHOPHONE

2.1. Overview

The *morphophone* was designed to construct complex sound forms by means of multiple delays at prescribed time intervals. It consisted of a rotating tape loop and twelve magnetophone heads arranged around the central wheel: one recording head, ten playback heads, and one erasing head. While the position of the recording and erasing heads were fixed, the playback heads could be repositioned, providing control over the delay time of each repeat based on its distance from the recording head. Each playback head had its own preamplifier and its own band-pass filter, allowing both the level and frequency response of individual repeats to be independently adjusted. The preamplified signals were then mixed and routed to the output.

The maximum achievable delay time depends on two factors: the overall diameter of the wheel and the rotation speed. Available sources indicate a diameter of approximately 50 cm [16, 17], with selectable rotation speeds of 19, 38 and 76 cm/s. Therefore, assuming a diameter of 50 cm (circumference: 157.8 cm) and a rotational speed of 19 cm/s, the maximum achievable delay time would be around 8.3 seconds.[2]

Playback signals could also be routed back to the input, thus establishing feedback loops. The feedback gain and decay time depended on the current settings of preamplifiers and band-pass filters. Furthemore, the erasing head could be removed allowing overdubbing of the signal, thus eventually accumulating additional material in the tape loop and achieving complex sonic textures.

2.2. Processing Modes

The *morphophone* could lend itself to several processing modes. For instance, to create artificial reverberation, one could adjust the delay times and amplitude levels of the playback heads, emulating the natural reflections of a sound source in an enclosed space. Alternatively, it could function as a multi-tap delay effect, repeating a given sound event with varying delays and timbral characteristics. The use of feedback and overdubbing may result in dense and complex sonic material, characterized by signal accumulation and build-up phenomena. According to the available sources, the *morphophone* was conceived by Moles, Schaeffer and Poullin as a means of transforming the spectrum of sustained sounds; this is evident, for example, in an article by Poullin explaining, "a homogeneous element of stable dynamics that passes several times in front of the reading heads will be able to undergo dynamic modulation and secondarily, if desired, spectral modifications." [18]. Moles similarly describes the device as "an apparatus whose purpose is to make the spectrum evolve in the course of time, that is, to create a finished sound object, distinct [...] from the original sound form, endowed with a particular attack, a body and an extinction..."[3] [19].

There are no certain sources on the use of the *morphophone* in the creation of musical works, a fact probably justified by the operational difficulty of the device: according to Daniel Teruggi, in fact, "[t]his was a very complex and rich machine, however, it was almost impossibile to make it work. It was very difficult

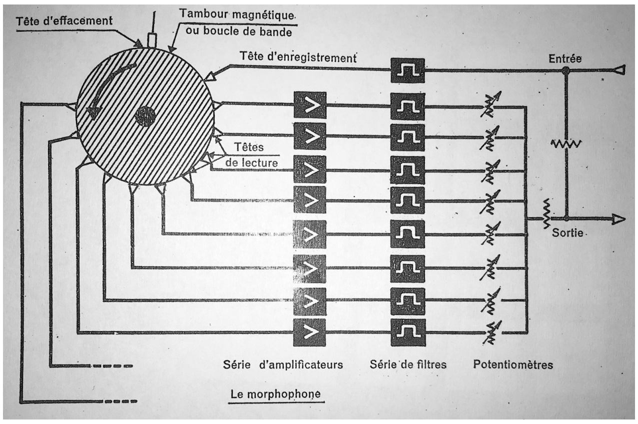

Figure 2: *Signal flow diagram in the morphophone.*

to stick a looped tape on the turning disk and since the magnetic heads had to be in close contact with the tape after a few turns, a head would detach the tape. Some experiments were done on the machine but it was never really musically used" [16]. According to some authors [20, 21], the *morphophone* was used by, among others, Iannis Xenakis in some of his work at GRM. Evidence of its use in the making of one of Xenakis's very early pieces was recently provided in [22].

3. DIGITAL MORPHOPHONE ENVIRONMENT

At the state of the art, there appear to be no digital models of the *morphophone*. Some applications are based on its operating principles, with varying degrees of reinterpretation[4]. However, after careful analysis, none of the latter seemed directly useful for our purposes. In particular, those implementations lack a faithful correspondence between their features and the documented historical sources. This restricts their use for any investigation into the operational practices allowed by the original device. The present implementation aims to provide a standalone, fully integrated application that is independent of more general frameworks, while remaining linkable with a widely adopted environment such as MAX. Furthermore, the open-source nature of the *Digital Morphophone Environment* (DME) ensures full access to the internal architecture, making it expandable, editable, and analyzable. Work is still in progress and developments are documented online at the GitHub platform[5].

DME offers two modes of operation, philological and extended. It was deemed useful to distinguish between the two modes in order to obtain, on the one hand, a reconstruction that closely adheres to the original prototype[6], and on the other, a more general tool capable of exploiting currently available computational resources and processing methods to extend its possibilities. The focus of this paper is exclusively on the philological version.

The implementation was preceded by an analysis of available

[2] In fact, the uncovered portion of the tape present between the erasing and writing heads (about 3 cm [17]) should also be considered in the calculation, which results in a reduction of the overall circumference.

[3] Translations from French by the author of this article.

[4] Examples include *Delays* [23], a plugin included in the famous *GRM Tools* [24], and *Morphophone* [25], created in the REAKTOR environment by Jonathan Tremblay.

[5] https://github.com/danielscorranese/DigitalMorphophoneEnvironment

[6] Clearly we refer here to the available historical sources: the original *morphophone* prototypes (apparently two different ones were built) were lost or destroyed [26].

documentation, comprising a limited amount of bibliographic material – often containing contradictory information – and, more significantly, a number of photographs found in published papers and online archives. These sources have proved useful in understanding some of the main structural features of the *morphophone*. Among the most important is a functional diagram found in a rare volume by Abraham Moles [27] (see Figure 2)[7].

3.1. Structure and Control Panel

The overall configuration of the *morphophone* reflects a modular approach. Internal and external wiring connects the various modules, allowing users to adapt the system to their specific needs and enabling a wide range of possibile configurations.

As shown in Figure 1, the top of the device features the rotating wheel with the twelve heads positioned around it, as well as the controls for the motor drive and rotation speed.

3.1.1. Readout Modules: Amplification and Filtering

Figure 3 shows a portion of the *morphophone* front panel, including three of the five readout modules (referred to as *lecture* in French, indicated by the letter A), each equipped with preamplification and band-pass filter controls. Each module covers two playback heads (5 modules $\times$ 2 heads = 10 heads). From top to bottom, we find the playback head signal input (1 - *entree*), preamplifier potentiometers (2 - *gain*) and band-pass filter controls (3 - *correction*, 4 - *f.filtre*, 5 - *q.filtre*), plus two output jacks for each of the two playback heads (6 - *sorties*).

Regarding the band-pass filters, the *f.filtre* potentiometer selects the center frequency from seven discrete values, while *q.filtre* controls the bandwidth. The function of the *correction* parameter is less clear. Footage from 1958, filmed at the GRM laboratories [28], suggest that this control may alter the frequency scaling of the filter. It also appears to be a stepped potentiometer. Based on these observation, it is plausible that the seven frequency values selected by *f.filtre* are rescaled according to the three positions of *correction* control, although the exact scaling factors remain unknown. In the DME implementation, the center frequency ranges visible in the aforementioned footage (Figure 4) were adopted. The three frequency sets (in Hz) are as follows:

[7]Note that the direction of the arrow indicating the direction of rotation is incorrect.

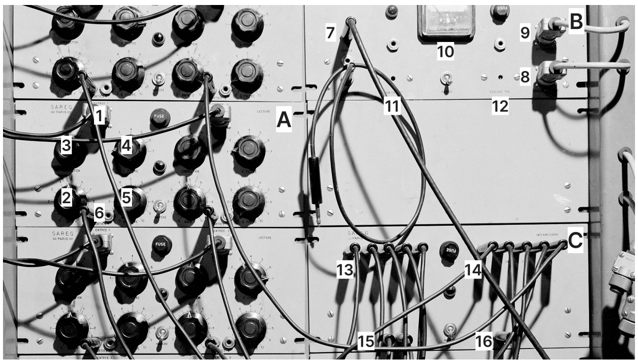

Figure 3: *Detail of the front panel of the morphophone (A: Lecture. B: Enregistrement. C: Intermediaire.)*

Figure 4: *Snapshot of BBC film depicting a filtering module [28].*

1. Range I: 32, 64, 128, 256, 512, 1024, 2048;
2. Range II: 37.5, 75, 150, 300, 600, 1200, 2400;
3. Range III: 50, 100, 200, 400, 800, 1600, 3200.

As for the *Q* factor, it appears that the *morphophone* provided a scale of ten values (1 to 10), with the value 0 presumably used to disable the filter. The same logic is applied in the DME environment.

3.1.2. Recording Module: Signal Input

On the *morphophone* right half front panel (Figure 3), there is an *enregistrement* module[8] (B) and an *intermediaire* module (C), responsible for managing audio signal input and routing (effectively acting as an audio mixer). The *enregistrement* module has three audio inputs, one output for routing the signal to the recording head, and a second output presumably intended for the erasing head (9 - *effacement*). A VU meter (10) is also present for monitoring the input signal. Additionally, two small input holes labeled *gain enreg* (11) and *reglage pol* (12) are visible on the panel, which were presumably used to adjust the input signal gain and the bias level, respectively. The presence of three input sockets suggests that multiple sound sources could be processed simultaneously, an essential requirements for implementing feedback loops.

3.1.3. Intermediate Module: Signal Routing

In the absence of reliable information, it can be assumed that the signal from the ten playback heads are grouped into two sets of five (13, 14) and routed to two corresponding output pairs (15, 16), either to be returned to the inputs and/or to feed other external devices. Overall, the use of this module in conjunction with the dual playback head output sockets and the multiple *enregistrement* inputs, enable a variety of feedback configurations.

3.2. Implementation and Methodological Choices

In developing the DME application, the goal was to design a modular architecture in which each unit would be functionally self-contained and capable of interfacing with other modules. To this

[8]From the available photos, there is a second *enregistrement* module too, placed at the bottom of the panel. Yet, it always appears to be unwired. It can be assumed this was related to a second recording head, not actually available in the photographed prototype.

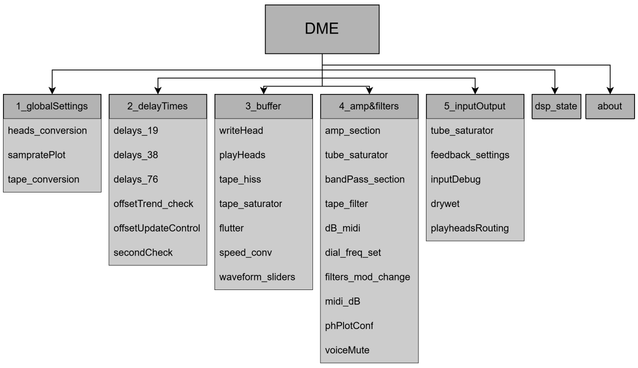

Figure 5: *Diagram of the patches that make up the DME environment.*

end, the *bpatcher* function found in MAX[9] was employed, which allows the contents of a patch to be encapsulated within in a user-defined area of the interface. Figure 5 provides an overview of the patches that make up the DME environment, while Figure 6 displays the main graphical user interface. Each box in the diagram of Figure 5 represents an individual patch. The dark-colored boxes constitute the GUI elements, which in turn invoke various subpatches (represented by the lighter boxes).

While the implementation aims to recreate the *morphophone* functional architecture with attention to historical sources, certain aspects of analog modeling are not fully addressed at this stage. Different from other contributions (e.g. [29]), in this research we cannot rely on the original technical datasheets or other fully reliable details of analog implementations. The solutions so far included should be considered generic: they are meant to achieve the intended sound behavior, but not to replicate the analog signal process in fully exact detail. We head towards a more precise emulation of credible analog behaviors to be addressed in further work.

3.2.1. *Processing Stages and Extra Functions*

Figure 7 illustrates the complete signal processing chain implemented in DME. The following sections provides further detail about the main processing stages. It should be noted that certain functions not present in the original *morphophone* have been introduced. These include a multichannel recorder, a dry/wet control, a input mute switch, an output amplitude control, and a overall feedback gain control (referred to as the output trimmer and feedback trimmer in Figure 7). Furthermore, the "Global settings" submenu (Figure 8) contains a subset of analog modeling parameters, which may be utilized for creative applications (more such parameters may be implemented in future developments).

[9] Version 8.6.5 (https://cycling74.com/).

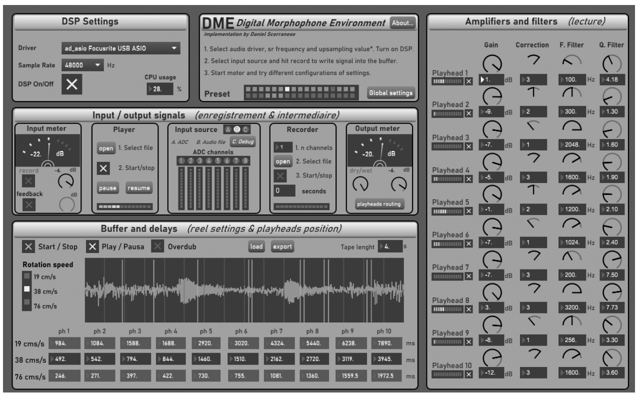

Figure 6: *Main interface of the DME environment.*

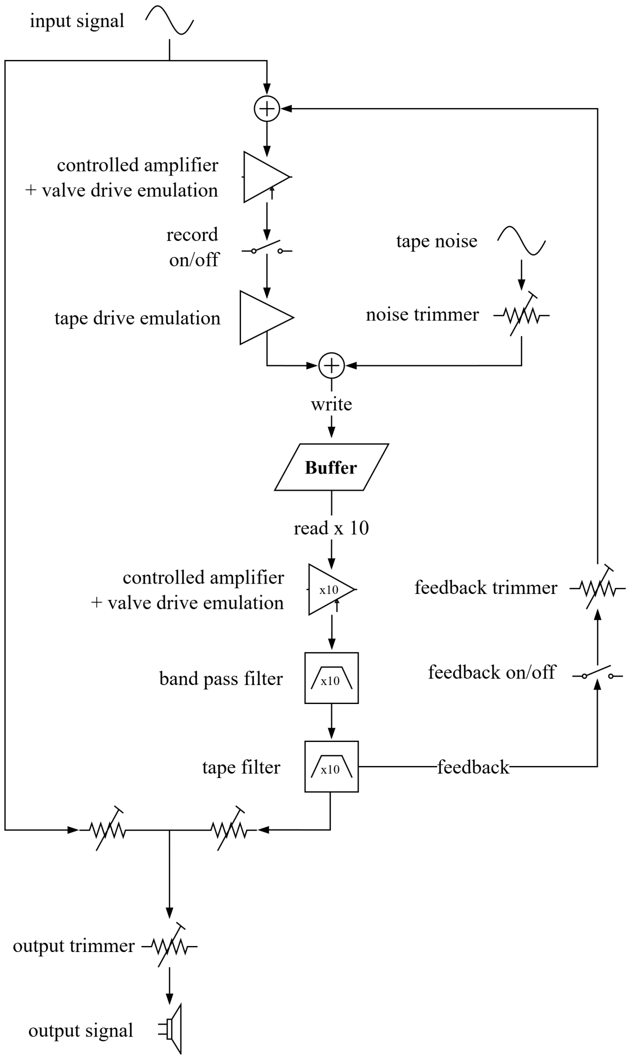

Figure 7: *DME: signal processing chain.*

3.2.2. *Input and Output Management*

The implementation provides a high degree of flexibility in managing inputs and outputs, even extending beyond the original limitations of the *morphophone*. Users can select multiple audio sources for processing simultaneously, either by accessing the input channels of the audio interface in use (up to eight input channels) or by loading an audio file. Since the system processes monophonic signals, any multi-channel input is automatically mixed down to

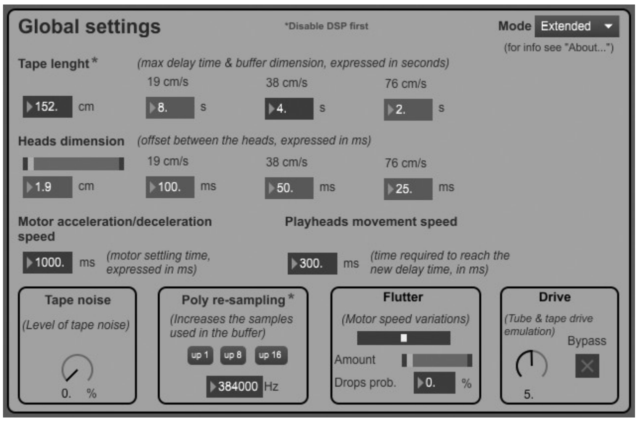

Figure 8: *"Global settings" control panel.*

mono. Regarding output management, the signal from each individual playback head can be assigned to any output channel (up to ten), or routed back to the input for use in feedback loops. These routing configurations are managed via a graphical interface. Output assignments are not exclusive – multiple signals can be directed to the same output channel, and a single playback head can be routed to multiple destinations.

3.2.3. Analog Preamplification Modeling

In an effort to approximate the analog behavior of the original *morphophone* technology, harmonic distortion stages were introduced at several points in the signal path. Specifically, three stages of saturation were modeled: first, a (presumably tube-based) preamplifier at the input stage (module B, *Enregistrement*), which typically causes saturation at high input levels, resulting in waveform compression or 'squashing'; second, a playback amplifier (likely a power amplifier) responsible for setting the playback head output level; third, magnetic tape saturation during recording, caused by high current levels that fully align the magnetic domanins in the tape medium. Although such distortion phenomena are generally considered undesirable, they significantly contribute to the overall sonic character of the system. For this reason, a dedicated signal processing module simulating these three types of distortion was integrated into the DME[10]. This was implemented using an external *LADSPA* plugin within the *gen*∼ environment in MAX[11]. It is important to note that the degree of magnetic saturation depends on the gain level of the input preamplifier. As a result, this control simultaneously influences the behavior of both the valve and magnetic distortion stages.

It should be noted that the current model does not simulate the full input-output transfer function of the magnetic recording process. In real tape systems, signal distorsion can occur even at low magnetization levels due to the intrinsic non-linearity of the medium, typically characterized by a double-S curve. Moreover, it remains uncertain whether the original *morphophone* adopted any form of biasing which would significantly affect both the linearity and the noise floor. These aspects are not implemented at this stage, as the main focus has been on reconstructing the device's architecture rather than its analog behavior. More accurate modeling of magnetic recording are left for future developments.

3.2.4. Tape Loop and Circular Buffer

The tape loop is simulated using a circular buffer, which is cyclically written and read via designated pointers (corresponding to the write and playback heads). In MAX, the write and read indices are implemented through the *ipoke*∼[12] and *peek*[13] objects, respectively. Both are driven by sawtooth oscillators, which provides continuous and linear access to the buffer contents. The frequency of the oscillators determines the read/write speed and is directly linked to the sampling rate. For example, at a sampling rate of 44.1 kHz, a 1 Hz oscillator processes 44100 samples per second. Consequently, a buffer containing 44100 samples will be written and read once per second with a 1 Hz sawtooth wave. Doubling or halving the oscillator frequency results in halving or doubling the playback duration, respectively. In analogy with the *morphophone*, the three different motor speeds are obtained by acting on the frequency of the oscillators. The buffer size, which can be set by the user, defaults to 4 seconds in duration (at a virtual speed of 38 cm/s), regardless of the current sampling rate. The overdubbing function, which corresponds to the *morphophone*'s erase head bypass, is also implemented via *ipoke*∼. When overdubbing is enabled and the write and read speeds differ, the signal is read at a rate that is either faster or slower than the rate at which it was written to the buffer. This results in a playback duration that differs from the original, leading to a corresponding transposition in pitch. It is also possible to load a prerecorded audio file into the buffer or export its contents to a new file. These features reproduce some of the typical workflows of magnetic tape technology, which consist of using prerecorded tapes as processing sources or for performing overdubbing operations.

Both the buffer and the write/read pointers are defined within *poly*∼ instances, which allow efficient replication of the read pointer algorithm (ten instances in this case), as well as independent upsampling and downsampling for the processing. Upsampling was particularly important to mitigate aliasing artifacts caused by modulations applied to the oscillator driving the write pointer, especially during motor speed variation, such as start-up or acceleration and deceleration phases. In fact, the transition from a stationary to motion, as well as the stabilization of the motor at a new rotational speed, does not occur instantaneously. Instead, it requires a variable amount of time, depending on the physical characteristics of the specific device (such as motor response and inertia of the medium). To simulate this behavior when needed, the frequency of the oscillators controlling the write and read indices is dynamically modulated over time, so that it reaches a target value within a predetermined, user-adjustable duration. During the debugging phase, it was found that modulating the frequency of the write pointer could introduce spurious frequencies components that were not originally present in the recorded signal. However, using a higher sampling rate and/or applying upsampling significantly reduces the energy of these artifacts, rendering them negligible, particularly for complex or broadband input signals.

[10] Harmonic distorsion is automatically enabled when operating in philological mode.

[11] The plugin used to simulate these processes via waveshaping functions is *TabTubeWarmth*, part of the *Tap-Plugins* library by Tom Szilagvi (https://tomscii.sig7.se/tap-plugins/ladspa/tubewarmth.html). A version ported to the *gen*∼ environment was later released by an anonymous user (https://cycling74.com/tools/stkr-waveshaping).

[12] https://www.no-tv.org/MaxMSP/.

[13] Both functions implement sample interpolation algorithms: linear for writing, and cubic for reading.

3.2.5. Flutter and Tape Hiss Effect Modeling

Modulations in the frequencies of the oscillators driving the write and read indices also occur when the flutter function is enabled. This feature simulates mechanical imperfections in the rotation motor that prevent it from maintaining a constant speed. Such instabilities can take various forms, including periodic fluctuations or more irregular, unpredictable behavior. The implementation in DME focuses on the latter scenario: random values for frequency and amplitude values are generated and used to control a sinusoidal oscillator. The absolute value of this oscillator is then used to modulate the frequency of the write and read oscillators. Importantly, these modulations remain within the sub-audio domain, thereby imitating realistic mechanical deviations.

As shown in Figure 7, a built-in noise generator simulates the increased noise floor typical of degraded magnetic tape. The type of magnetic tape used in the original *morphophone* remains unknown. Future research may investigate which formulations were available at the GRM during the 1950s. Different tape types, manufacturers, or aging states may yeld distinct noise and response characteristics. For now, the noise simulation is based on a generic profile and should not be interpreted as specific to any historical tape formulation. The simulation is based on a two-stage amplitude modulation process: first white noise is modulated by pink noise; then, the result is again modulated using the same pink noise signal. The resulting signal is amplitude-scaled, low-pass filtered, and mixed back into the main signal path, thus mimicking the spectral and dynamic profile of analog tape hiss.

3.2.6. Delay Lines

Once acquired and appropriately conditioned by the preceding processing stages, the signal is written into a buffer and read by ten independent read pointers, each configured with a different delay time. These delays correspond to the distance between each playback head and the write head. To simulate this delay mechanism, the read indices are offset relative to the current write index, thereby introducing the desired time shifts. The oscillators that control both the write and read pointers must operate synchronously – that is, they must share the same frequency and phase – so that all pointer indices progress uniformly over time. This synchronization ensures that any change in the motor's state (e.g., start/stop, play/pause, or speed variation) affects all oscillators equally in terms of frequency and phase. From this synchronized condition, a delay can be introduced at any given time t by subtracting a specific amount of samples from the read index. This offset creates a temporal gap between the read and write pointers, allowing the read pointer to access earlier buffer contents and thus reproduce the intended delay effect.

3.2.7. Delay Line Management

In the *morphophone*, delay times are determined by the physical placement of the playback heads around the rotating wheel. In DME, these delays can be adjusted through dedicated numeric controls, which define the time (in milliseconds) between the writing and reading of the signal, further categorized according to the selected motor speed. However, this flexibility is constrained by a set of rules designed to replicate the behavior of the original system. First, the movement range for any playback head is limited by the positions of adjacent heads. This means that the delay time associated with each head must remain within a specific range.

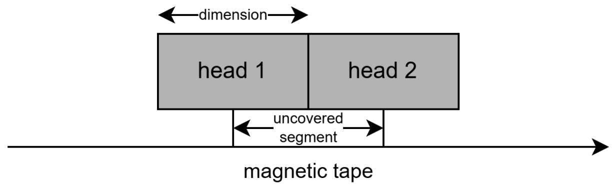

Figure 9: *Uncovered portion of tape between two adjacent heads.*

Second, the physical size of each head occupies a portion of the tape surface, resulting in an uncovered segment between any two adjacent heads[14]. This segment can only be accessed when one of the heads is repositioned. As a result, the uncovered portion of tape introduces a temporal gap (or offset) that depends on both the physical size of the head and the rotational speed of the tape. For example, if a head occupies 1.9 cm, and assuming that the uncovered segment roughly equals this length (see Figure 9), then at a speed of 38 cm/s, the resulting time offset is 50 ms (since time = space ÷ speed). Consequently, the maximum achievable delay time equals the buffer size minus this offset, while the minimum delay time is exactly equal to the offset itself.

3.2.8. Band-Pass Filter

The signal recorded in the buffer and simultaneously read by the ten read pointers results in ten delayed replications of the original input. Each of these undergoes independent amplification and band-pass filtering. As previously discussed in Section 3.2.3, the amplification stage includes a simulation of analog saturation. For filtering, the implementation employs the *reson∼* object in MAX, which applies a digital filter based on finite-difference equations – specifically, a second-order IIR filter with an FIR component.

In addition, a final-stage filter is included at the end of the processing chain to emulate the general frequency response of a magnetophonic system [30]. The simulated frequency response takes into account the typical equalization stages found in analog tape systems, applied during both recording and playback. These corrections aim to flatten the overall response, compensating for the low and high-frequency deviations introduced by the tape medium and head characteristics. This filter is automatically activated when operating in philological mode, and bypassed in extended mode. In either case, users can enable or disable the filter manually, regardless of the selected operational mode.

It should also be noted that the frequency response of a magnetophonic system is affected by tape speed due to gap losses at the playback head. In particular, the upper cutoff frequency is approximately proportional to the ratio between tape speed and gap lenght. Consequently, different motor speeds – such as the three selectable in the *morphophone* – would result in different spectral behaviors. This dependency is not currently implemented in DME, but may be considered in future updates aimed at refining analog accuracy.

3.3. Sound Processing Applications

Beyond its historical reconstruction goals, the DME environment lends itself to a variety of creative and experimental uses in sound

[14]This occurs because the point of transduction – where energy is converted (from electrical to magnetic in write and erase heads, and the reverse in playback heads) – is generally located at the center of the head.

design and music composition. The system can be configured in various ways to achieve both conventional and unconventional sound processing tasks.

Among the most immediate applications is the simulation of classic delay-based effects. Alternatively, as introduced in Section 2.2, multiple playback heads can be adjusted with increasing delay times and decreasing amplitude levels to obtain artificial reverberation, thus recreating the early attempts at artificial reverberation pioneered by 1950s studio engineers. More complex results can be achieved by using the feedback and overdub functions. These allow the progressive accumulation of material in the circular buffer, resulting in a densification of the sonic content. The result is often a layered texture with continuously evolving spectral and dynamic properties. In this context, the use of differential write/read speeds introduces pitch transposition. This kind of configuration opens to interesting performative possibilities. In addition to timbral and temporal transformations, the DME supports spatialization strategies when used with multichannel audio systems. Each playback head can be independently routed to a specific output channel, enabling the projection of delayed and spectrally distinct signal components toward distinct spatial locations via separate loudspeakers. This setup allows for the simulation of sound movement, the construction of spatial textures, or the generation of immersive echo patterns across the listening space. Such configurations can be further enriched using feedback and/or overdub functions to create complex spatial textures that evolve over time. Some of the latter are easily integrated into live electronic performances. In the DME main window, real-time control of playback head position, filter settings, feedback amount and motor speeds enables dynamic interaction and spontaneous sonic shaping, enhancing expressive and performative possibilities.

In summary, the DME provides a fertile environment where historical approaches intersect with new creative possibilities afforded by today's digital domain, whether used as a pedagogical tool offering insight into early electroacoustic practice or as a creative platform for composers and sound artists.

4. CONCLUSIONS AND FUTURE WORK

DME represents an attempt to digitize the core processes of the *morphophone*, aiming to replicate the device's characteristic operational modes and behaviors. Particular attention was devoted to a faithful reproduction of the original signal processing functionalities. Developing an accurate and detailed model presents several challenges, chief among them the lack of definitive technical documentation (e.g., circuit schematics) and the impossibility of conducting direct experimental analysis of the original hardware. Recent work [22] proved the DME very useful in reconstructing segments of Xenakis's early tape music *Diamorphoses* (1957)[15]. By implication, that suggests the DME's behavior being realistically consistent to the original *morphophone*: in the absence of other historical sources, there is possibly no other way to evaluate the DME implementation safe by the regeneration of sounds once resulting from the GRM *morphophone*. The extended version was also succesfully used in the author's performance practice. However improvements are still possible and necessary – both in terms of computational efficiency and the development of more precise, albeit generalized, models of analog circuitry distinctive qualities. Future work may also involve expanding the system's control architecture, allowing for more sophisticated signal processing strategies and further signal processing effects, however removed from the actual *morphophone*.

[15] Sound examples from that contribution are available at https://github.com/danielscorranese/SCIAMIGLISSANDO_audioExamples.

5. ACKNOWLEDGMENTS

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The DME source code is available on the GitHub repository (see note 5). A folder within the repository contains a set of audio examples that illustrates the algorithm's processing behavior under different configurations.

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