LEM: a sound object that performs Live Electronic Music and proposes a new way to compose and distribute music

Spyros Polychronopoulos
Department of Informatics and Telcommunications, University of Athens, Greece
spyrospoly@uoa.di.gr

ABSTRACT

Since the invention of the phonograph the majority of the musical releases are made in such a way that same audio at every playback is executed. With this work I propose a sound object (Live Electronic Music: LEM) that is programmed to interpret the musical piece, always with a different variation at every iteration. LEM is a stand-alone, iterative composition device. A small plastic box houses a single board computer running an algorithm that plays random audio samples selected from a sample bank. Further, the algorithm is programmed to alter the variables of the audio filters randomly and in random instances, within defined limits. This flexibility is an important compositional factor as the decision, with this practice, is not single and concrete (i.e., the composer specifically determines the way for the piece to be played) but probabilistic (i.e., the composer determines the likelihood of more than one way for the piece to be played, as in aleatoric music). This work proposes a new way of music distribution shaping the way a composer makes a decision, and the audience perceives live and prerecorded music. The piece reveals itself endlessly, exposing new dimensions and dynamics drawn from a table of acoustic elements.

1. INTRODUCTION

1.1 Inspiration

Technologies of audio reproduction (i.e., vinyl, digital formats etc.) allow the composers not just to lay their signature via the work's notation (score) but also through the concrete sound-recording of its performance, thus, rendering the work's precise reproduction. Prior to the emergence of the possibility of sound-recording and reproduction, the composer was accountable mainly to the score and secondarily to its performance [1], [2] – i.e. the case when he/she was physically present during a performance or when the composed musical piece was presented by an exclusive performance from specific musicians. A new condition of musical reception has been constituted due to both the possibility of music recording and the listener's familiarization with prerecorded musical works. Thus, the composer is forced to submit his work in the form he/she believes it should be heard and not to hand down written instructions as to how it should be performed. Even in the hypothetical scenario in which a score is so analytically written to the extent that no margin of interpretative freedom is left for the performer, the sound-product will have a unique timbre every time, simply because each performance requires a specific instrument being played in a specific space. When music is read as a text, it is perceived differently than when it is heard. The score has to be interpreted, to be performed and, finally, to be sounded as music through the instruments. Even those who are most familiar with semiography – those who proclaim that when they read the score, in essence they listen to music - should reconsider what they exactly mean, since nowadays one has access to a variety of different recorded performances based on the exact same instructions. Having in mind certain versions of piano timbre and believing that by reading a score for solo piano one actually hears with precision the composer's work, is exactly as if one is mentally interpreting the musical piece. In that way the receiver operates like a computer program, which performs or executes a score by retrieving from its memory the same piano note that the subject has once heard and calling for that note to "sound" mentally. Based on the frequency that corresponds to each note, the surrounding overtones are also mentally shaped, by means of a previous piano hearing. The significance of a musical performance is also highlighted by the distinctive statement of Honore de Balzac's fictional character (Gambara, the composer) in his short-story [3]: "Well, monsieur, a composer always finds it difficult to reply when the answer needs the cooperation of a hundred skilled executants. Mozart, Haydn, and Beethoven, without an orchestra would be of no great account."

Under this condition, the concrete sound-recording that bears the composers' signature is essentially their work and all the performances that may follow are simply considered as re-performances of this work. For instance, when a composer releases an album today, they essentially certify that the particular recording, constitutes the specific form they would like the work to be heard. This recording may be distributed in multiple copies, either in physical form (e.g., vinyl disc or cd) or as digital files (e.g., wav or mp3 sound files). During the listening of prerecorded music, one gets the impression that the uniqueness of the performance is absent. In other words, what is seemingly lost, is what Benjamin would describe as the aura [4] emanating from the uniqueness of each distinct performance of the same musical work. However, one can claim that today the recorded musical work's aura lies hidden in its timbre and is unveiled only to the initiated listener as a reward for his dedication. To the frivolous listener's ears, the timbre sounds mute and convinces them that the overtones surrounding the central frequency that they perceive is merely a parameter devoid of any content. Thus, for them, aura will remain invisible during the listening of prerecorded music [5]. While the spreading of musical reproduction devices has condemned the aura, to its own death, looming out of the listening of prerecorded music, the special attention which is nowadays paid to timbre has come, almost after a whole century, to resurrect that aura, so that it can finally gain its real immortality with the aid of its own executioner.

At this point it is worth pausing for a moment in order to clarify as much as possible, in the following few lines, the concept of aura which was the main influence to come up with the proposed device and is borrowed from Walter Benjamin. The latter, reflecting on the alterations that technology has induced both to the works of art themselves as well as to their reception, claims that "what is altered in the age of the mechanical reproduction of the work of art is its 'aura'" [4]. Aura's reception moves along two main axes: far and close – uniqueness and repetition. Following Benjamin's description: "Namely, the desire of contemporary masses to bring things "closer" spatially and humanly, which is just as ardent as their bent toward overcoming the uniqueness of every reality by accepting its reproduction". Even though nowadays there is the possibility of a direct musical listening through devices of musical reproduction, the audience is still attracted by a concert's unrepeatability. The perception of live music is not accomplished through representation; on the contrary, it requires the listener's physical presence, the "here and now" (hic et nunc) of the performance. This experience feels as if it is not originated from this world, while the closer one approaches toward the locus of the performance (the musical stage) the more unworldly the event itself appears.

During the listening of contemporary music, the work's timbre occupies the most eminent position. The fundamental parameters which determine the timbre of a recorded musical piece are the room of the listening, the sound system, the position of the speakers and the position of the listener [6]. When a sound emits in a specific space certain essential features are revealed, for example, its volume. Everyone can have a sense of the space in which they are in, even with their eyes closed, solely by hearing. For example, a whistle or a gunshot will sound differently in a recording studio than in a large enclosed stadium. Each space has unique acoustics. The room in which each sound emits or is being recorded has an impact on it and co-constitutes its timbre. In that way we can perceive a room's dimensions, when a familiar sound is heard. The sound familiarity is of great importance, in order for a certain reference to an already familiar timbre to exist, so that we will be able to perceive space's impact on it.

The listening of prerecorded instrumental music can be considered as such a case of hearing familiar sounds, a case in which the listeners may picture the room that each instrument was recorded, exactly because they have a general sense of the natural instruments' timbre. If we listen to a prerecorded piano piece, for instance, we are able to make an assertion about the size of the room in which the recording took place, e.g., if it was actually big or small. It is possible that the listener may get the impression that the room in which the instrument's recording took place has no reverberation and, consequently, when the recording sounds from the speakers, the instrument is heard as if it is located within the same room in which one finds oneself i.e., within the room where the listening to the music takes place. Nevertheless, even though an instrument's timbre may sound as if it was produced within the same room in which the listener is currently in, they never have the impression that the performance is carried out live in front of them. And that happens because in the opposite case one would essentially have the possibility to actually see someone in person, in his/her physical presence, performing the musical work live. This possibility of perceiving the space, in which recording of instruments takes place, opens up many novel paths to prerecorded music.

On the other hand, during the listening of electronic music, the auditor is not capable of associating directly each sound with a timbre familiar to him/her, of which he/she can imaginatively project its development in time, based on the specific space in which it is heard. However, there is always a chance that an electronic sound's timbre may be similar to one of a natural sound. In that case, with the condition that it is not included within the context of a kitsch creation but resonates in a truly convincing manner as exhibiting such a similarity, it falls under the category of the reception of familiar sounds, which we discussed above. Yet, in any other case, electronic sounds will have no reference to something familiar and thus the listener will not be capable of defining the acoustical parameters of the space in which it resonates. In other words, when electronic sounds - that do not struggle to imitate natural sounds - are heard within a room, it is not feasible to perceive the sound on the ground of the alternation that a timbre, which functions as a primary point of reference, undergoes, just because it sounded within a particular room. Considering the above, one must acknowledge a difference between prerecorded instrumental music and electronic music. Listening to instrumental music from a reproduction device is always accompanied by perceiving the idea that the recording took place in a certain point of space and time which differs from the space and time of the current hearing, i.e., it is situated not in a "hic et nunc" but in an "illic et tunc". Reception of electronic music differs considerably from the reception of acoustic instrumental music. One of the reasons is that one always listens to electronic music through speakers, thus confusing live performance with the reproduction of prerecorded material. This confusion escalates by the fact that the person who listens to electronic music has not associated each and every sound with a particular movement of the performer. For example, the performer of electronic music is in a position to change a different parameter each time by using the same button, in contrast to instrumental music, in which case, when one hears a violin producing a tremolo, they have already associated the sound with the corresponding movement of the violinist. Moreover, instrumental music is performed recorded and then the auditor who listens to the recording experiences the absence of the aura, in contrast to a live performance. In electronic music this contradistinction is absent, on the one hand, for the simple reason that electronic music is always heard through speakers, on the other hand, due to the fact that there is no immediate association with the per-

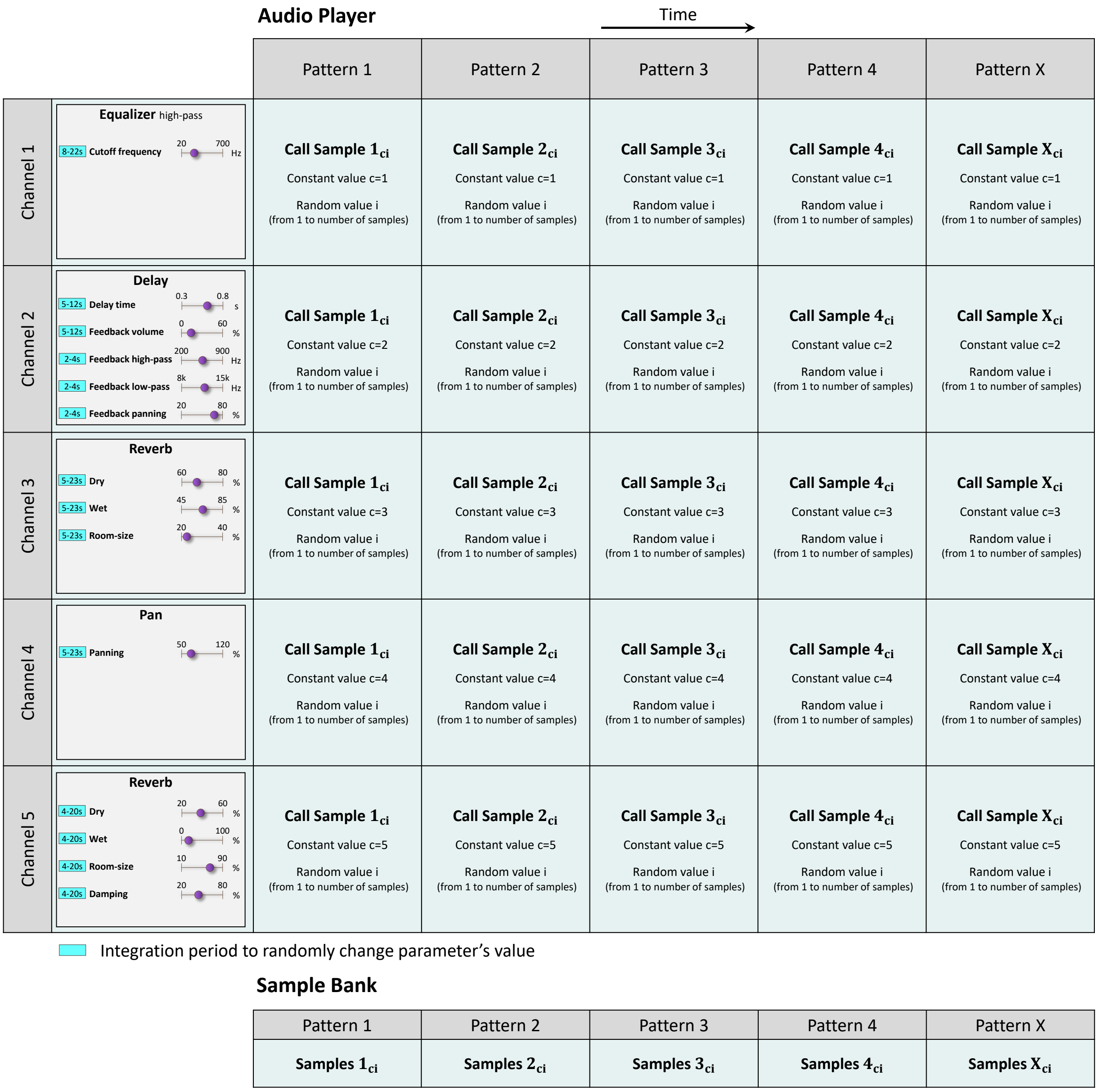

Figure 1. The block diagram of the algorithm showing the audio channels with their filters, the time periods (teal boxes) that they will alter their values (purple dots) and the samples called from a sample bank at every pattern/channel.

former.

Electronic sounds do not have a reference to any known timbre and this results to an elimination of the strict distinction between the space of recording and the space of hearing. The audience's lack of association both gestural and audible regarding the live performance of electronic music was the key factor to use electronic sounds in this project and try to challenge the perception of live electronic music.

1.2 Background

The feeling of live music performance can be explored with modern technology via algorithms programmed to randomly make decisions regarding some parameters of the musical piece's way of execution. The concept of randomness in music composition is not new. Musikalisches Würfelspiel (German for "musical dice game") [7] was a popular system, throughout Western Europe in the 18th century, that the composer would use a dice to randomly generate music from precomposed options. In the mid-20th century the French composer Pierre Boulez introduced the term Aleatoric Music or Aleatoric Composition [8] for music where some elements of the composition are left to chance. The performer of such compositions was given certain liberties regarding the order and repetition of parts of a musical work.

Algorithmic music composition, since more than half of a century now, is a pole of attraction of many musicians [9]. Brian Eno, exploring the technology trends at the time, introduced the term "generative music" to describe music that is ever different and changing. Generative music was put into practice in his work released in 1996 that was written for the Koan software. This work influenced modern composers to set goals and create music in real-time that can express different mood-states, which they achieved through a unique combination of a graph traversal-based chord sequence generator, a search-based melody generator, a pattern-based accompaniment generator, and a theory for mood expression [10]. Antoine Schmitt and Vincent Epplay released “Missing System - Infinite CD For Unlimited Music” in 1999, distributed in the form of software on a CD-ROM for both Mac and Windows operating systems. The authors state that their reason to work in this direction was to search for new ways of composing and thinking music, where the computational process of the machine can be part of the piece’s rendering [11]. A similar project is “The Morpheus CD-ROM” released in 2001, which was initiated by John Eacott and included five compositions written in SuperCollider code [12] that would be interpreted in real-time. In 2003, Runar and Thorr Magnusson collaborated on a generative music project called SameSameButDifferent.

Another approach was Rand()% Radio by Tom Betts and Joe Gilmore from 2003, which was an internet radio station that would stream generative music created by a large group of composers, running their software in real time. In 2010 Tristan Perich released “1-Bit Symphony” [13] which was a low-fi electronic composition in five movements on a single microchip. A complete electronic circuit utilizes on and off electrical pulses, synthesized by assembly code, to manifest data as sound. Icarus CD release of generative music in 2012, on which all the pieces were generated by a software and mixed down to a fixed-media audio file, when the album was purchased [11]. Other distribution formats housed generated releases, such as mobile apps (i.e., Brian Eno - Bloom (2008), Björk - Biophilia (2011), Radiohead - PolyFauna (2014), Massive Attack - Fantom (2016), etc.), Web Audio API (i.e., Patrick Borgeat’s interactive music applications) and collaborative music environments (i.e., Bronze environment: a compositional framework that supports generative composition, with a generative player). However, an iterative composition apparatus, distributed to the audience in a form of a stand-alone sound object, is never been proposed.

2. MATERIALS AND METHODS

The focus of this project is to propose a new way to distribute music: a device that will play endless variations of a musical piece. LEM was released and distributed by Room40 in 2016. The system’s essential parts are the hardware (i.e., a single board computer) and the software (i.e., the algorithm). First step was to carefully design the block diagram of the algorithm. This will define the needs of the hardware, software and furthermore, the compositional flexibilities and restrictions. As can be seen in Figure 1 there are 5 independent audio channels with different filters and filter parameters each. The value of each parameter will be altered (within specified limits) at random instances (within specified limits) defined by the composer (see teal box, Figure 1). At every pattern and channel, the algorithm will randomly select to play an audio sample from a unique subcategory from a sample bank. This system provides new compositional flexibility as the decision regarding a filter’s parameter value, or a sample does not have to be just one. For example, the composer can choose two extreme room-size values for a reverberation filter that he/she wants for the audio track and let the algorithm explore the resilience of the decision, or even load various samples (e.g., different piano notes) for the same pattern and channel and let the algorithm randomly select one of them to play. Moreover, in Figure 1 the filters, their parameters extreme values, and the time restrictions for the algorithm to alter them for this specific project (LEM) are illustrated.

2.1 Hardware

The block diagram defined the requirements regarding the hardware. Although the most popular single board computer at the time was (and still is) Raspberry Pi [14] its digital to analog converter did not meet the needs of this project and furthermore, it had unnecessary features (such as hdmi socket) which raised the cost of the product. After some research the best option for the needs of the project was c.h.i.p. next thing co (Figure 2), which is no longer available (company closed at 2017). This single board computer provided a 1 GHz Allwinner R8 ARMv7 Cortex-A8 processor, 256 MB of Nanya Technology DDR3 SDRAM, Onboard NAND 8GB storage, pre-installed Linux OS (Debian) and peripherals, USB port and stereo audio port via mini TRRS. The micro-USB socket would power the device, the USB was useful to transfer the data (audio samples and the algorithm) and store them on the 8GB onboard solid state drive and the stereo 3.5mm jack socket to output the audio signal.

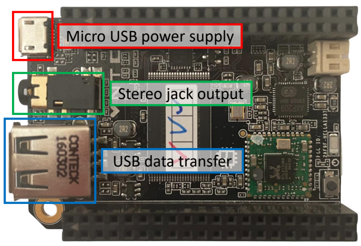

Figure 2. The single board computer running the algorithm and outputting the audio signal though the stereo jack output, a micro-USB port to supply power and a USB port for data transfer.

2.2 Software - Code

The first algorithm to put the block diagram (Figure 1) in practice was compiled in Mathworks MATLAB 2015a. However, the code could not run on the single board computer as ARMv7 Cortex-A8 processor did not support MATLAB and furthermore, it was relatively slow (approximately two minutes to run on a MacBook Pro, Intel Core i5 5th Gen Processor, 16GB RAM). Cycling ’74 Max Msp [15] is a popular visual programming language for music and multimedia but again it is not supported to run on the aforementioned processor.

Pure Data (Pd) [16] on the other hand runs on this processor and there was a stable version (Pure Data 0.44) running on Linux OS. There was a bottleneck to generate a different random number at every boot of the Pd algorithm, that would trigger the generation of more random values. It should be noted here that the single board computer should

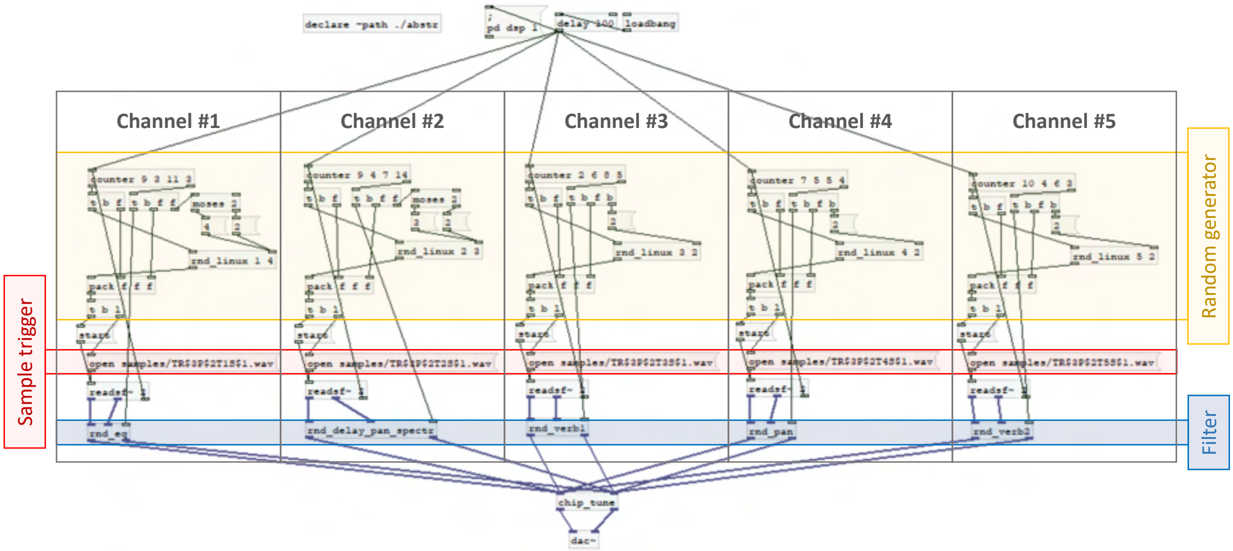

Figure 3. The PureData algorithm showing 5 audio channels the random generator (yellow) necessary for the filters' parameters' values, the instances that these values will be altered and the samples' selection, the sample trigger (red) and the channels' filter (blue).

not be connected to the internet to obtain a random number using for example atmospheric noise data. Although the Pd random number generator always outputted the same value at every boot, the OS's random generator, using the precise boot-time, outputted a different value at every boot, that was then introduced to the algorithm. The algorithm, illustrated in Figure 3, autoruns after the boot of the OS and as can be seen its comprised by sub-patches (abstractions) in order to keeping the code clean. The audio channels are illustrated in green, the random values generation and distribution are illustrated in yellow, the triggering of samples (wav format, PCM stereo, 44100 sample rate, 16bit) are illustrated in red and the channels' filters are illustrated in blue. All the signals go though a specially designed audio filter that helps the specific music to sound better considering the single board computer's DAC.

As the composition of the piece was taking place, I thought that it would be interesting to create Pd code for an ancient musical instrument [17], [18] simulating the reverberation of specific acoustic spaces [19] or even arbitrary audio filters [20], [21] and work with them during the process. However, as the composition took place in another piece of software it would be rather complicated to combine them both. Thus, I decided to use the samples generated from the software used to compose the music and I created simple Pd filters (delay, equalizer and panning) and used a freeware reverb from the Pd's external libraries (Freeverb , Schroeder/Moorer).

2.3 LEM for the end-user

In the first version (2016) the single board computer was housed in a bespoke hand-crafted polycarbonate box (on the left in Figure 4) and in the second version (2017) it was housed in a plastic box made using a bespoke injection molding (on the right in Figure 4). The unnecessary sockets of the board such as the USB data socket were hidden. Both versions were released by, an experimental Australian record label, Room40 and were sold out in less than a minute after the announcement (2016 version was 150 pieces and 2017 version was 350 pieces). When the user powers LEM the unique reproduction of the piece, that lasts approximately 22 minutes, starts in less than 30' and when it is finished there is 10 seconds of silence and then it starts again for a new and unique execution of the piece. Some of the people that purchased LEM used headphones but most of them connected it to speakers. Few of them sent me an email letting me know that they left LEM play in their sound system, in the background, for several days.

Figure 4. LEM sound object the first and the second edition released on Room40 records 2016 and 2017, respectively.

3. CONCLUSIONS

This project proposes a new way to compose and distribute music. The composer writes the musical piece to be played from an algorithm and sets the flexibilities parameters. The algorithm, with this controlled (set parameters) randomness, executes the piece always in a unique way at every iteration. LEM is a sound object that by using electronic means creates the effect of a musician trapped in a box of persistently morphing timbral and temporal paradoxes. In electronic music the audience has not associated the live performance with specific gestures from the musician or a specific instrument (e.g., piano) with a specific sound as the timber of the sound generated is unique and it is always played though speakers. The idea behind LEM is to provoke the concept of live performance in electronic music. During a live performance the audience feels the "here and now" of the work of art. Maybe LEM is a way to bring this feeling to an audience which is listening to prerecorded music by endless variations of a piece of electric music.

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