# Simultaneous Spatialization and Synthesis of Sound with Nonlinear Functions

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We present an analysis of micro-compositional techniques used for the synthesis and spatialization of sound with a computer. Our discussion centers on the fact that a single algorithm, a particular implementation of iterative feedback FM, has been found to concurrently provide the basis for complex chaotic sound synthesis as well as for the definition of complex interaural time and intensity differences that permit an interesting diffusion of the sound in a stereo field. Both processes are generated in real-time from the same algorithm: as the sound is synthesized, the spatial characteristics are also synthesized. Original work was performed on the UPIC system at Les Ateliers UPIC (Unité Polyagogique d'Informatique de CEMAMu), and has since been expanded and further investigated by the implementation of an UPIC simulation in the MatLab programming environment on a personal computer. We present the results of various simulations, highlighting the effects of the iterative feedback FM technique as a chaotic sound synthesis tool, as well as its ability to provide spatialization cues to the listener in an acoustic space. The geometry of the listening space is evaluated (the loudspeaker-listener interface), linking the algorithm and the digitized sample stored in the computer to the interaural cues that are ultimately presented to the listener, and which are responsible for both the resulting timbre as well as for the spatialization of that timbre. Our discussion is presented within the context of what is described as "emergent composition", a term that we suggest could describe the manner in which micro-compositional methods (as those presented here), influence and induce the resultant listening experience as the sound is presented to the listener in an acoustic space.

Keywords: Chaotic Sound Synthesis; Emergent Composition; Frequency Modulation; Micro-Composition; Sound Spatialization

## Introduction

For humans, the perception of a sound's position in a listening space is in large part dependent upon what is described as lateralization cues. These lateralization cues include interaural intensity differences (IIDs) and interaural time differences (ITDs). The "differences" refer to the time of arrival delays between the left and right ears as an incident acoustic wavefront encounters a listener. Lateralization cues may be synthesized or applied to an existing sound by various means, such as through the application of carefully measured head related transfer functions (HRTFs), computer modeling and simulation of a sound source in a particular acoustic setting, or the application of measured impulse responses in the form of digital filters. Another method, to be discussed in this paper, describes the use of nonlinear functions such as frequency modulation (FM) for the characterization and definition of a sound's lateralization position in a stereo field, as well as for the simultaneous definition of the sound's timbre itself. We suggest that the process which defines a synthesized sound's characteristic timbre can, at the same time, describe and influence that sound's spatialization in a listening space, since the process being utilized operates at one of the lowest levels of the sound's representative architecture: the individual sample level.

An algorithm is applied to a set of stored wavetables, in this case a frequency modulation algorithm. The technique described in this article originated on an implementation of the UPIC system in early 1996, and has since been adapted and applied on a standard personal computer via a simulation of the UPIC's operation using the MatLab programming environment. This simulation has enabled a more thorough investigation of the mechanisms involved at the low-level micro-structure of the FM algorithm, and the ramifications of such mechanisms to the higher level macro-structure of the resulting sound as presented to the listener in an acoustic space. We will show that micro-composition at the level of the digitized sample (and even further down in the architecture of the acoustic environment, for example the air molecules themselves of the listening space), may define considerably more than just the timbre of a particular sound mass; indeed, the entire acoustic event that emerges in the listening space and in the listener is dependent upon the inter-relationships of the algorithm, the physics of the air, and the physiology of human hearing.

The sound synthesis and spatialization techniques described in this paper have both been discussed separately in the literature, however, what we present here is an integration and realization of both techniques via the same computer algorithm. In other words, the methods used to compose the sound itself and the spatialization of that sound are based on the same algorithm (in this case iterative feedback FM), operating simultaneously on digitized audio data. In [Kendall 1995], the spatialization technique is described as the decorrelation of the audio signal, where slight differences in phase and delays contribute to create the spatial imagery of the resulting sound. However, Kendall describes in detail the application of FIR and IIR digital filters to achieve the decorrelation of the audio signal; FM is mentioned as an alternate method, but not analyzed or described. In [Slater 1998], the synthesis of chaotic signals with feedback FM is described.

When first confronted with the UPIC system as a compositional tool, one of the author's original intentions was to use the system as a means to spatialize sampled audio signals in a stereo field, in addition to synthesizing the sounds themselves. Although an unlikely application for the UPIC system, it was thought that with the ability to precisely place a sound sample in the amplitude-time-frequency space as it is with the UPIC, a certain level of control could be obtained for the diffusion of the sound between the left and right loudspeakers, and therefore a spatialization of the sound would be possible. However, both these goals (the synthesis and the spatialization of the sound) were initially approached separately, following traditional paradigms. As work with the UPIC system continued, it became apparent that because it was possible to operate on the digitized sound at one of its lowest levels, the digital sample, it was learned that multiple processes were being applied to the digital data concurrently, and the results of these processes were emerging simultaneously with the sound once it was experienced in the listening space. Careful listening to the results of a long list of various tests and algorithms created with the UPIC made it apparent that the same algorithm used to synthesize the sound was responsible for the spatialization of that sound as well. In effect, both the chaotic sound synthesis described by Slater and the decorrelation of that same audio signal as described by Kendall was occurring at the same time. These techniques provide very interesting and non-standard methods for the synthesis and spatialization of sound. However, an analysis of the respective processes has always been relegated to separate domains, where the underlying processes were not analyzed as concurrent processes, or presented as gestalts of the same fundamental algorithmic process.

In this article, we will first present a brief description of the particular UPIC implementation used to realize the spatialization and synthesis techniques; for a more detailed discussion of the UPIC's operation and its capabilities, please refer to the listed references [Marino 1993, Xenakis 1992 ]. A detailed description and analysis of the resulting sound will then be presented, highlighting the way in which micro-compositional techniques influence the emergent composition as experienced by the listener.

**A Simulation of the UPIC System**

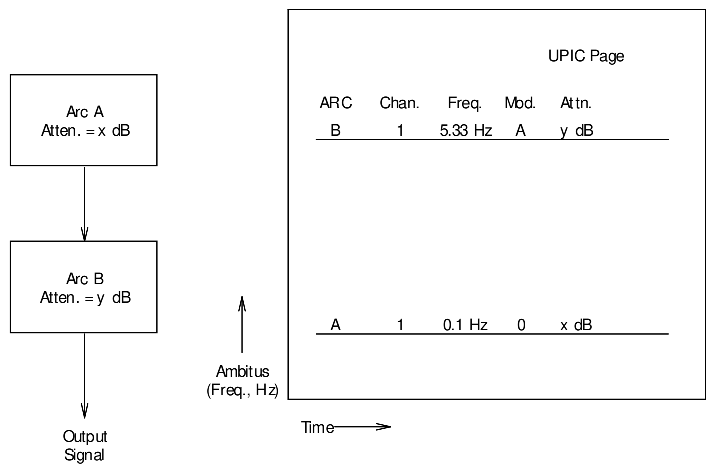
Sound is synthesized on the UPIC system via a bank of 64 real-time oscillators, which are based on a wavetable look-up procedure. The wavetables consist of 8192 sample points, and their frequency of oscillation can vary between .01 hertz to 22,050 hertz. The user of the UPIC can organize these wavetables, with respect to frequency of oscillation and their occurrence in time, through the use of a graphical display on the computer screen. A blank display (the UPIC page) offers to the user the ability to place lines and curves (called arcs) with the mouse-pointing device. The length of each line corresponds to a length of time, and the line's position on the vertical axis associates the arc to a particular oscillating frequency. Each arc, in turn, can have a particular wavetable assigned to it.

The UPIC system also offers frequency modulation as a sound processing and synthesis tool. Any two arcs, or groups of arcs, on the UPIC page can frequency modulate each other, as long as the arcs overlap each other along the time axis. In this way, the UPIC system can be viewed as an algorithmic tool for the FM synthesis and transformation of sound. Examples of a few FM algorithms implemented on the UPIC system are given in Figure 1 (a), (b), (c), and (d.). Figure 2 shows an organization of arcs comprised of 60 arcs total. There are 10 groups of 5 arcs each in the upper portion of the page (part of which are shown in the figure), and an additional group of 10 arcs located lower on the ambitus (frequency) scale (not visible in the figure). The lower group of arcs is used as modulators for each group of 5 arcs in the upper region. In turn, as indicated in the flow block diagram of Figure 1 (d.), one of the 5 arcs in each of the upper groups is used to modulate each of the lower arcs. This organization permits a nonlinear feedback system to develop in the form of iterative feedback FM. An important note should be made concerning the output channel assignments for each of the arcs. For the upper group, the output channel assignments for each arc are alternated, left – right – left – etc. Keeping in mind that the difference in oscillation frequency between each of the upper arcs is a fraction of a hertz (between 1/100$^{th}$ to 1/10$^{th}$), and that the oscillation frequency of the lower arcs is .01 hertz each, the phase relationships among the whole conglomeration of arcs takes precedence within the nonlinear feedback system implemented as FM.

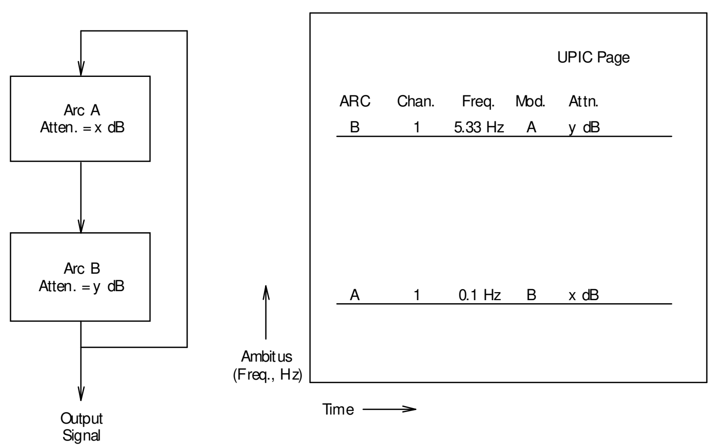
The aural result of such an organization of the algorithm, implemented with the number of oscillators as is done here, is one where the diffused sound in the listening space travels between the loudspeakers in a panning fashion. In addition, the sound gives the impression that it does not belong to a flat surface, as may be the case when simple panning techniques are used from a mixing console, for example. In this case, there appears to be several layers of movement, or at the least, the sound takes on a voluminous quality, seemingly jumping from the plane defined by the left and right speakers. This effect is also dependent upon the position of the listener in the diffusion space. As one moves about in the room, the movement of the sound between and among the loudspeakers takes on different forms (i.e., perhaps it is not as dramatic, or perhaps the movement may be slower or faster).

**Figure 1 (a.) through (d.)** These flow diagrams illustrate the various FM algorithms implemented on the UPIC system as described in the text. The figures on the right of the flow diagrams give a schematic representation of the UPIC page with the associated arcs' parameters given. In 1(a.), the basic modulator-carrier configuration is shown, without feedback. In 1(b.), the same algorithm as in (a.) is used, but with a feedback path added between the output and the modulation input of the modulator arc. Figure 1(c.) introduces a third oscillator arc, which is inserted at the output of the carrier arc of 1(b.). In this algorithm, the modulation input of oscillator arc 3 (which is now also acting as a carrier) receives the resulting complex output of the previous arcs. Figure 1(d.) describes an algorithm where a larger group of arcs are modulated by a single arc located lower on the ambitus range (this algorithm has been described extensively in this paper). The iterative feedback loop is completed via the arc number 10, a member of the larger group, which then simultaneously modulates the arc 11.

(a.)




(b.)




(c.)

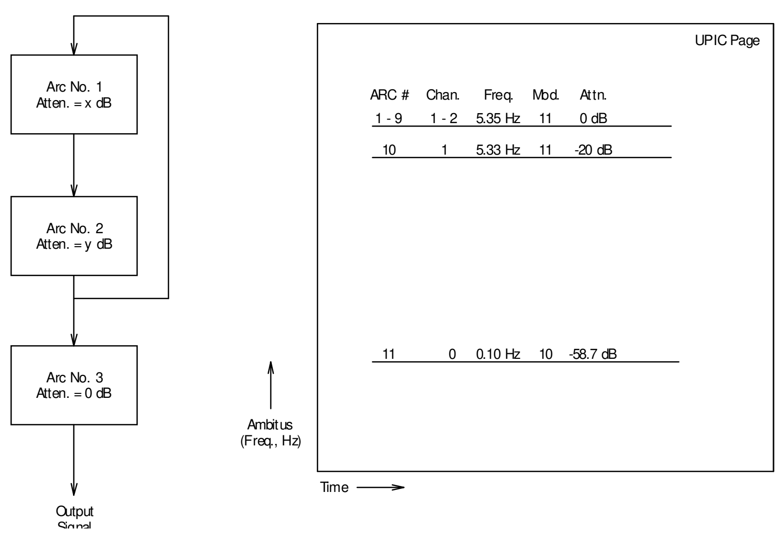
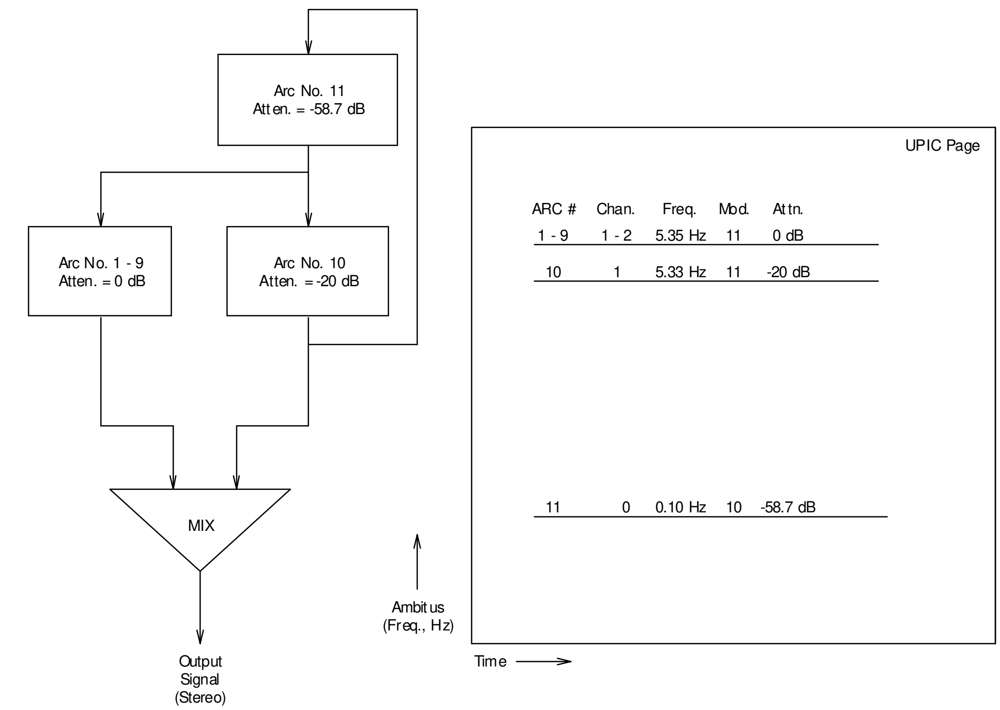


Figure 1, continued

(d.)




## Chaotic Synthesis and Decorrelation of Sound with FM on the UPIC System

We now present an analysis of the frequency modulation algorithm discussed above, as it is implemented in the UPIC system, aided by a MatLab algorithm that simulates the UPIC system's operation.

Consider two wavetables of 8192 sample points each, $W_c$ and $W_m$, consisting of extracts of a sampled complex (non-sinusoidal) waveform. Each waveform traces a curve defined by amplitudes $A_1$ and $A_2$, the values of which may be positive or negative values. For the purposes of our discussion, we shall normalize the amplitudes to +/- 1.0. Each waveform may be assigned a frequency of oscillation within the range of 0.01 Hz and 22,050 Hz, depending upon the position of their associated arcs on the UPIC page (i.e., their vertical position on the ambitus range). For a sampling frequency of Fs = 44100 Hz, then, 44100/8192 = 5.3833 Hz. This frequency, 5.3833 Hz, corresponds to the vertical ambitus position of an arc on the UPIC page that would produce an output signal which would be heard at a rate equal to the original sampled sound. In other words, for a sampled recording of a voice brought into the UPIC system, if an arc is placed at 5.3833 Hz on the ambitus, and is also associated to a waveform made of the sampled recording (8192 sample points of it), the output would be played back at a standard rate – the voice signal as heard would not be modified in pitch or speed. We essentially have a sample playback system. If the arc were placed at 10.7666 Hz, the rate of playback would be doubled, and the pitch of the sample is now an octave higher. Conversely, if the arc were placed at 2.69165 Hz, the rate of sample playback would be half the normal rate, and the pitch an octave lower.

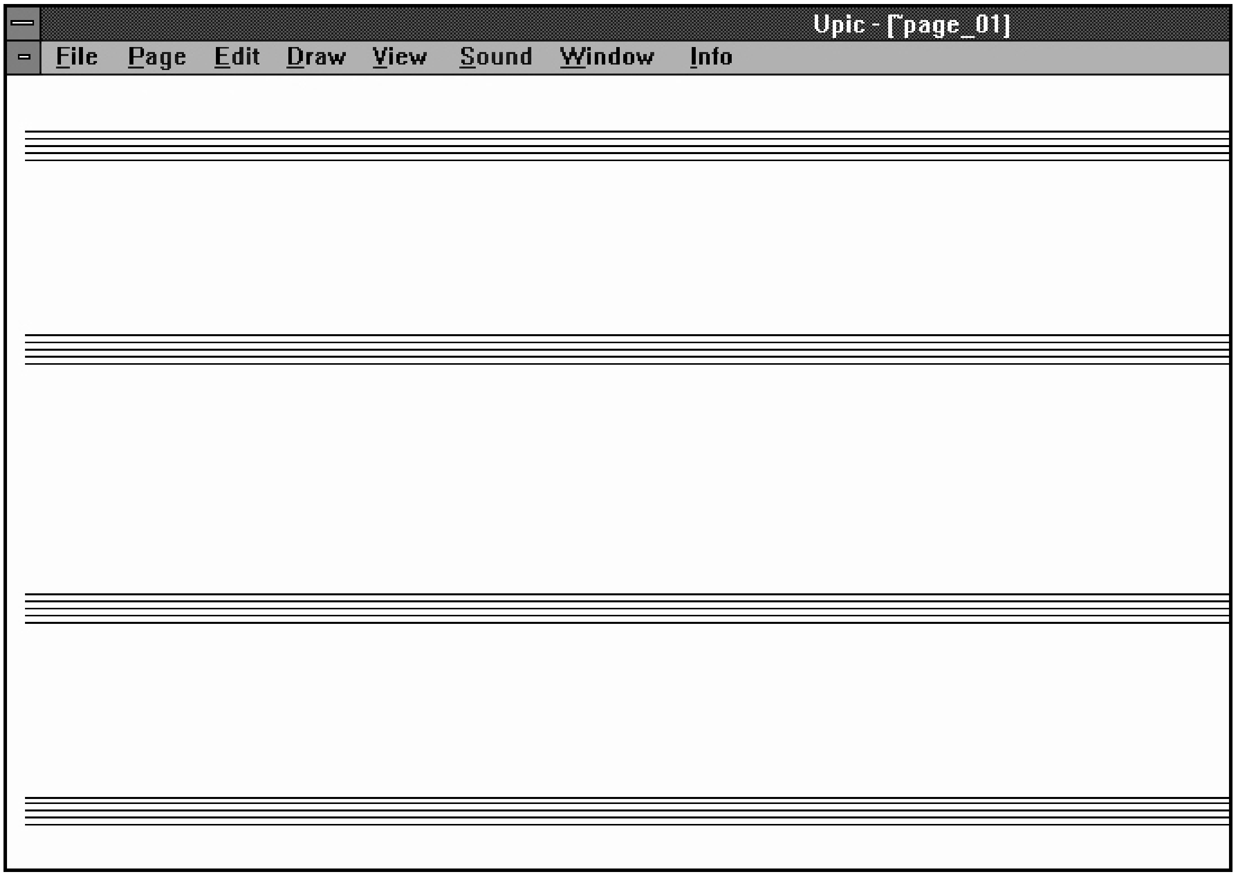



**Figure 2** This figure shows a grouping of arcs from an UPIC graphics page created by the author for the performance of the composition *Maya*. These four groups are part of a larger set of arc groups (this is an enlargement of an area of the UPIC graphics page), which contains an additional six groups identical to the ones shown above, for a total of ten (the additional groups extend in a similar manner below the ones shown above). In addition, a smaller group of arcs, positioned lower on the graphics page (lower on the frequency scale), are used to frequency modulate the larger group. The larger group, in turn, modulates the second group, creating the recursive feedback FM synthesis algorithm.

In order to replay the sample (wavetable) at a lower rate, say 1.00 Hz, we must interpolate the waveform in order that there are enough samples to accommodate the required pitch. If the playback rate is 100 Hz, we would need to perform a decimation operation on the waveform contained in the wavetable. The interpolation or decimation operation on the wavetable samples effectively changes the length of the waveform itself. In order to replay the waveform at a lower pitch, say, 1.00 Hz, a total of 5.3833*8192 samples must be inserted among the existing samples of the wavetable, changing its length to 44100+8192 sample points, and requiring 1 seconds of playback in order for one period of it to be heard. If the ambitus frequency were .01 Hz, we would need 100 seconds for an entire period, and 4,410,000 samples would need to be inserted among the original 8192 sample points of the wavetable.

With the UPIC system, one is able to frequency modulate one complex waveform with another complex waveform. In addition, one is capable of connecting a feedback loop from the output of one oscillator to the input of the same or other oscillator (see Figures 1(b.), (c.) and (d.)). To simulate such an operation in MatLab, a table lookup procedure is used. The waveforms are stored in arrays of length 8192 samples, and a third array is defined which consists of the modulated output waveform. Let $W_c$ be defined as the carrier waveform, $W_m$ the modulator waveform, and $W_o$ the modulated output waveform. $W_o$ will be dependent upon the amplitude values of $W_m$. So, using the following relation, we define the sampling increment, $SI_n$,

$$SI_n = 1.0 + A_n\left(\frac{f_m}{f_c}\right)$$

where $f_m$ is the modulating frequency, $f_c$ the carrier frequency, and $A_n$ is the amplitude value for sample n. Thus, the sample increment value is dependent upon the amplitude of the modulating waveform, and the ratio of the modulating frequency to the carrier frequency. In addition, we define a running sum, $\Sigma_n$, used to specify the particular value of the carrier waveform we are to send to the output array. The running sum is an accumulation of the $SI_n$ values, as they are calculated. For our particular value of our clock counter n, we shall have a corresponding value for $SI_n$ and $\Sigma_n$. $\Sigma_n$ shall be rounded to the nearest whole number value, and this whole number shall determine which of the 8192 sample points of the carrier we are to send to the output array. The output array will then consist of enough sample points to accommodate one period of the new modulated waveform, $W_o$.

The above description is the simple case when one waveform frequency modulates another. However, as also mentioned above, one may implement a feedback path among the waveforms. In this case, the modulated output will now have an influence upon the resulting output signal. We can rewrite the above equation to accommodate the new feedback path as follows:

$$SI'_n = 1.0 + A_{n-1}\left(\frac{f'_m}{f'_c}\right)$$

where $SI'_n$ is the new sample increment value used to determine the new associated modulated waveform. Since a nonlinear system has developed during this process, the outcome will vary with respect to time, and with respect to the initial conditions. At times, the system enters a mode of oscillation where the output seems to take on a shape and form of its own. This behavior has also been witnessed on the UPIC system, however, the UPIC permits real-time output of the resulting waveform, whereas the MatLab simulation does not. One is capable of "steering" and "guiding" the real-time output of the UPIC, much like a real-time instrument (the author's composition "*Maya*" was created in this manner, as well as the set of "*UPIC Etudes 2-8*").

The above scenario consists of a pair of waveforms only, $W_c$ and $W_m$. However, the UPIC provides a bank of 64 oscillators that may be oscillating at the same time. This fact was exploited, enabling the following multi-waveform arc configuration implemented within the FM algorithm previously described.

Ten groups of five arc/waveforms were organized, each waveform was specified to oscillate at a rate that was offset by $1/10^{th}$ of a Hertz with respect to the neighboring waveform. The group of five waveforms were then modulated by another waveform, set at .01 Hertz. In addition, one of the five waveforms from the group then acted as a modulator to the primary modulating waveform. A feedback system was again defined, but in this case, a larger complex of waveforms was part of the overall system that developed.

The small delta in oscillating frequency among the group of carrier waveforms provided the conditions necessary for the decorrelation of the sound to take place in the listening space. These differences in frequency correspond to a shift in phase among the group of five waveforms, and, as the modulating waveform (oscillating at .01 Hz) operates on the carriers in tandem, the subtle differences in phase provided a complex of shifting phases within the system. As the modulator itself is modulated by one of the carriers, an added dimension is injected, contributing an even more complex parameter into the feedback path. As the waveforms are being modulated, their resultant output is again modulating the system itself. The cycle is repeated until ultimately a chaotic mode of oscillation is encountered, perturbing the system into a sonic stratosphere.

What is interesting about the resultant sound generated by this process and algorithm is that the sound that is projected from the loudspeakers is presented binaurally, within a stereo field. No panning effects are implemented a posteriori. The complex coupling of the phase relationships among the waveforms' oscillating frequencies, and the nonlinear feedback system provided by the FM algorithm as implemented above, creates a sound field that is emergent and complex. Due to the nature of the chaotic system, one cannot precisely predict the sonic outcome, but one does have a general idea as to the direction the algorithm will lead the composer. What follows is a description of the complex binaural field that is constructed by the feedback algorithm, demonstrating the manner in which the differences in phase of the oscillating frequencies correlates to the acoustic output in the air of the listening space.

Non-standard synthesis methods rely heavily on the precise specification of the digital sample within an amplitude-time-frequency space. The organization of the digital sample in this amplitude-time-frequency space (and therefore the associated digitized waveforms used in the synthesis algorithms) bears a direct influence on the resulting macroscopic sound composition[1]. By bridging the

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[1] Winckel makes an interesting observation regarding this fact. He states "The perception of a stationary room excited by an acoustical event occurs in the microstructure of the statistical reflections which reach the conscious through a process of integration." (*Music, Sound and Sensation: A Modern Exposition*, p. 78) Concerning the macrostructure of a sound on the listener, he continues "The 'whole'

amplitude-time-frequency space used to describe the sample, with the large-scale emergent acoustic event in the listening space, we hope to present a means for more thoroughly understanding the inter-level relationships and processes present in such a micro-compositional approach to sound.

Begault suggests the source-medium-receiver model [Begault 1993] for a description of sound spatialization. Applying this model in our discussion, we can suggest that the sample data as manipulated by the computer and the algorithm is our "source material", the loudspeaker diaphragm and the air our diffusion medium, and the ears along with all associated physiological attributes of the listener as the receiver. Transformations on the sound, whether the sound is in the form of digital data in the computer, pressure waves in the air and on the eardrums, or electrical impulses in the listener's nervous system, all contribute to the perceived spatialization of the sound, and occur at every step of the model, not only in the transferal medium portion of the model. Indeed, it is the combination and interdependence of the transformations among and between the three sections that creates the emergent sound that is heard, along with all its properties and characteristics.

Table 1 presents a set of data that corresponds to a particular grouping of arcs as implemented in our simulation of the UPIC iterative feedback FM algorithm. The table also shows data that describes the physical configuration of the listener in the listening space with respect to the loudspeaker system diffusing the synthesized sound. Figure 4 presents the geometry of the listening space from which the data in Table 1 is calculated. And in figure 3, we show the resulting 5 time-pressure curves generated by the FM algorithm, using the same information as that used to characterize one of the group of 5 arcs in figure 2. The curves shown in figure 3 trace only the first 512 sample points of the resulting waveforms, however one can see the beginning of the shifting in time and phase of the individual time-pressure curves. Note that these curves are not the curves which are presented to the listener's ears in the acoustic space. Figure 4 shows the time-pressure curves that are presented to the ears of the listener, after the interaural time and intensity differences have been taken into account. One can see a slight difference in the trajectory that the curve traces, as well as a slight difference in phase of the left waveform with respect to the right waveform.

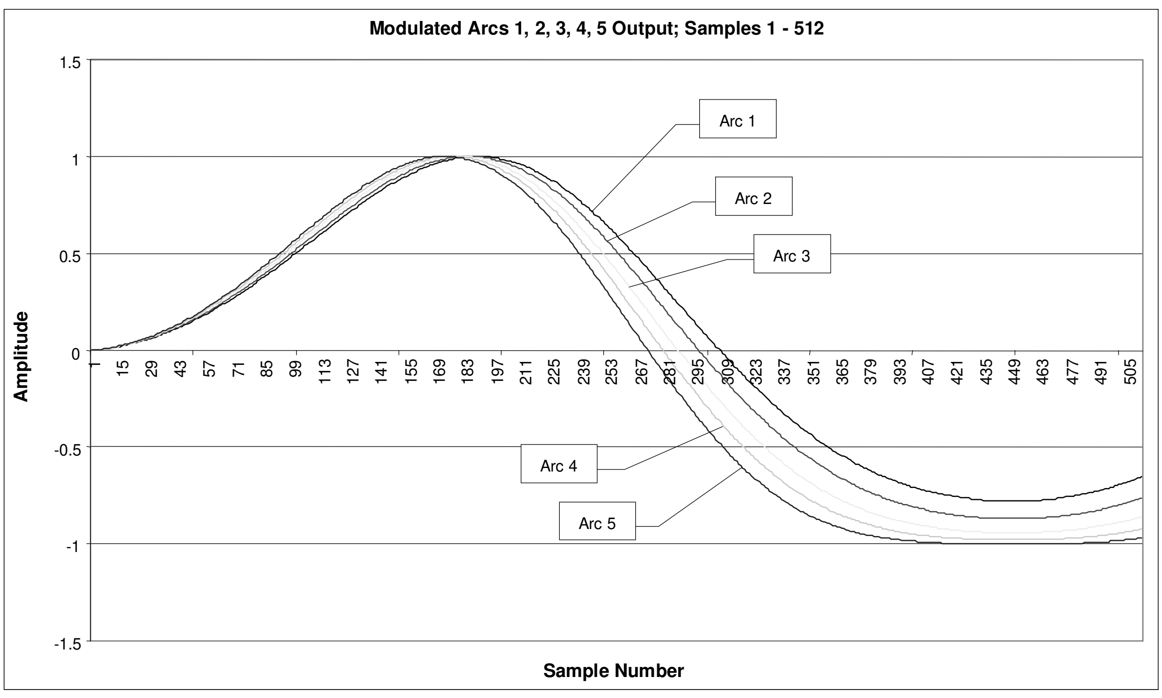


**Figure 3** A graphic plot of the resulting time-pressure curves of five waveforms produced with the FM synthesis algorithm of the UPIC system as described in this paper. The five waveforms shown here correspond to the five lines (arcs) of a single group in Figure 2 (the oscillating frequencies in this case for arcs 1-5 range from 5.40-5.60 Hz). The difference in phase and amplitude can be seen among the various waveforms, as they develop over time. Although all the waveforms begin their time-pressure trajectory at time 0, the effect of frequency modulation on the five waveforms (each of which has a slightly different oscillating frequency), produces a slight phase shift on the resulting sound, as well as a change in shape of the curve itself (which ultimately influences the timbre of the sound). These differences in phase directly influence the listener's perception of the spatiality of the sound in the air via the process of binaural hearing.

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is a 'gestalt' in the sense of Gestalt psychology, it is perceived instantaneously, independent of time, just as thought…" (ibid.).

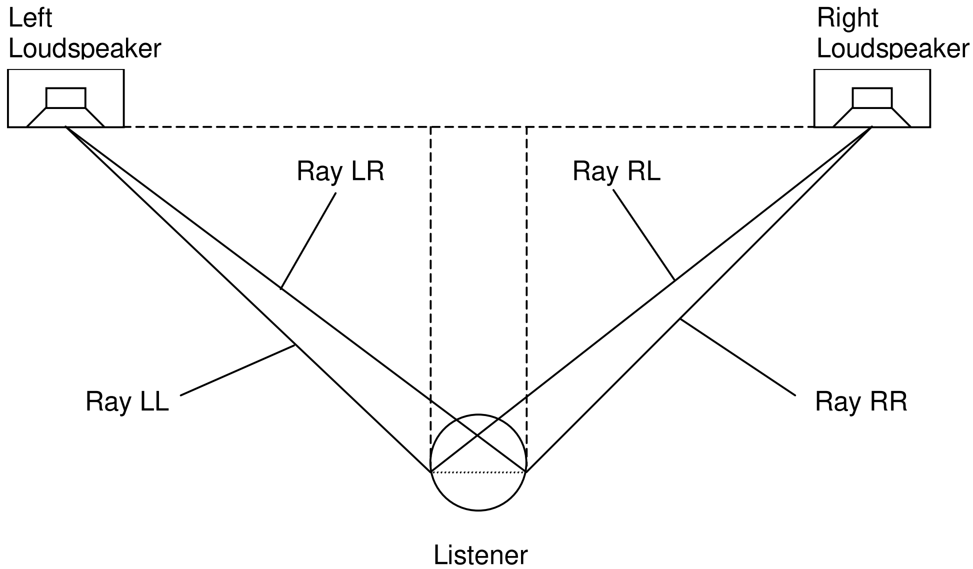


**Figure 4** The geometry of the binaural (or also transaural) diffusion of sound. The above illustration describes in a geometric schematic form the manner in which the various sound waves are propagated in a listening space and received by a listener in that space. Rays LL, LR, RL, and RR represent the distance a sound wave must travel before being received by the ears. The width of the listener's head introduces a slight time delay between the wavefronts of the time-pressure curve exiting from the same loudspeaker, as they impinge upon the listener's ears. This interaural time delay is the basis of binaural sound. As the listener's head is shifted left to right on the horizontal plane, the various rays will change in length as well, providing a different binaural listening experience depending on the position of the listener in the space. The technique is best experienced with the use of stereo headphones. Imagine the ball in the figure swinging like a pendulum in front of the loudspeakers.

### Interaural Time Differences and Phase Analysis of FM Algorithm

Distance of listener from L-R loudspeakers plane (centered) = 4.5m (~15 ft.)
Distance between left and right loudspeaker = 6.0m (~19ft. 8in.)
Speed of sound in air (normal conditions, 21 degrees C.) = 340 m/sec (1115.2 ft/sec)
Duration of sound in air (from loudspeaker to center of head of listener) = 17.9357/1115.2 = 0.01608294476 sec. (~16.083ms)
Distance between left and right ear of listener = ~18 cm (~7in.)

Distance from left/right speaker to center of listener's head = sqrt($15^2 + 9.8333^2$) = 17.9357 ft.
Ray LL = distance from left speaker to left ear = sqrt($15^2 + 9.54^2$) = 17.777 ft.
Ray LR = distance from left speaker to right ear = sqrt($15^2 + 9.8317^2$) = 18.0974 ft.
Ray RL = distance from right speaker to left ear = sqrt($15^2 + 9.\ 8317^2$) = 18.0974 ft.
Ray RR = distance from right speaker to right ear = sqrt($15^2 + 9.54^2$) =17.777 ft.

### Table 1

| Arc Number | Frequency (Hz) | Frequency Δ (Hz) | Period[1] (sec) | Δ Period (Sec) | Δ Distance (m) | Beat Frequency |
|---|---|---|---|---|---|---|
| Arc 1 | 5.38 | - | 0.1858736 | 0.0006885 | 0.234090 | 1/0.0006885 |
| Arc 2 | 5.40 | 0.02 | 0.1851851 | 0.0006833 | 0.232322 | 1/0.0006833 |
| Arc 3 | 5.42 | 0.02 | 0.1845018 | 0.0006783 | 0.230622 | 1/0.0006783 |
| Arc 4 | 5.44 | 0.02 | 0.1838235 | 0.0006734 | 0.228956 | 1/0.0006734 |
| Arc 5 | 5.46 | 0.02 | 0.1831501 | | | |
| Arc 6 | 0.01 | - | 100 | | | |

[1]This period represents that of an oscillator cycling at 5.38Hz (1/5.38 sec.). The wavetable oscillating at this frequency and QW1Athat contains the waveform used in the FM algorithm (8192 sample points), may in fact be comprised of several if not tens of periods of the sampled sound itself. It is at 5.38Hz (the UPIC oscillating and ambitus frequency of the digital oscillators) where the playback of the sample will be heard as natural – as it was when it was digitally recorded - without a shift in pitch or duration. I.e., 44100 Hz / 8192 = 5.38330078125 Hz.

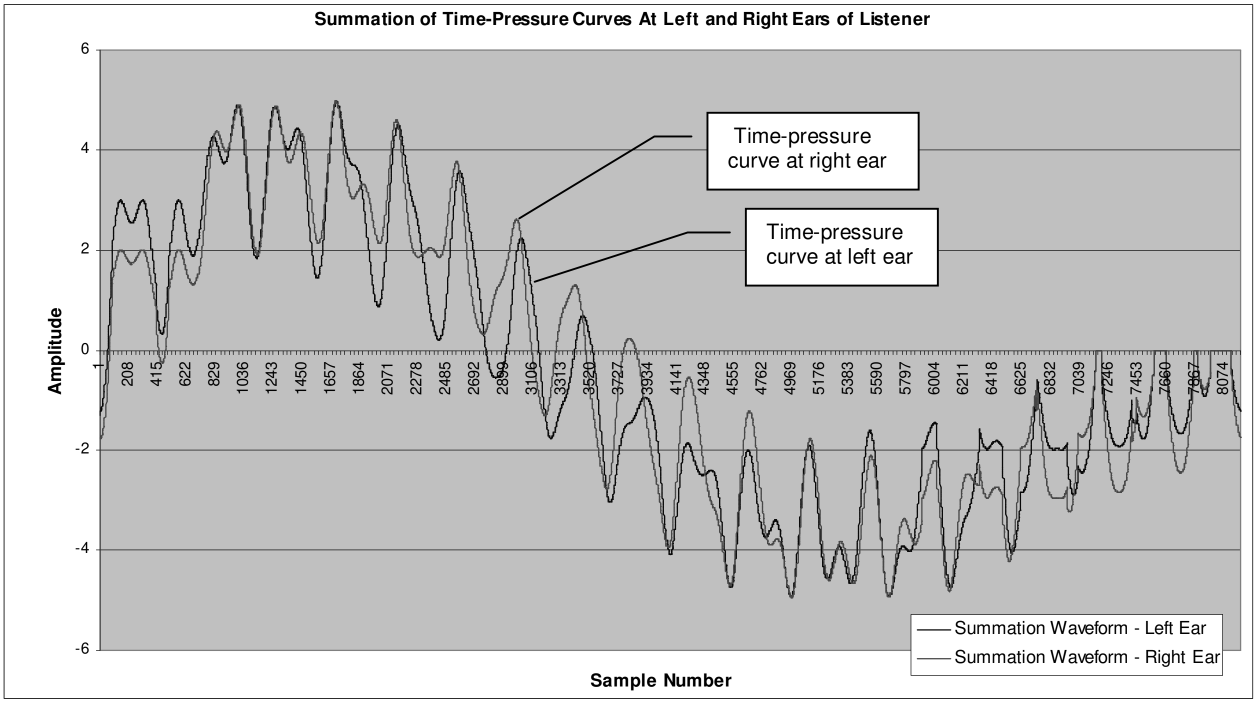


**Figure 5** The resulting time-pressure curve as delivered by the loudspeakers, and received at the position of the listener's ears. These waveforms represent the summation of the individual synthesized curves, as generated by the FM algorithm, and shown in Figure 3. The difference in the shape of the two curves is due to the fact that the time-pressure curves, at the position of the listener's ears, will add together differently because of the delay caused by the interaural time difference (the distance between the listener's ears). Although the left loudspeaker delivers arcs 1, 3, and 5, and the right loudspeaker delivers arcs 2 and 4, all five time-pressure curves add constructively in the air. The interaural time difference, inherent in binaural hearing, contributes to the difference in timbre and directional cues detected by the listener.

## A Temporal-Spatial Composition: Phase Yields Time, Space, and Timbre

In traditional sound spatialization techniques, the movement of sound is typically achieved through the use of complex panning techniques, where the audio signal is "pushed", "moved", or "routed" within the electronic circuits that comprise whatever external equipment is used to perform such signal manipulation (normally an elaborate mixing console often found in most studios). With the advent of computer techniques, the traditional model of space was retained, with the added benefit of delegating control of the sound to the high-speed number crunching capabilities of the computer. Instead of the composer having to turn knobs, press buttons, or push sliders, the computer could be programmed to direct the signal in an electronic circuit, and ultimately out of a pair of loudspeakers. Or, alternatively, carefully measured head-related transfer-functions (HRTFs) can be implemented in the form of digital filters: such digital filters are applied to digitized sound, resulting in the specific positioning of the sound in the listening space. However, space, or spatial movement, is *added,* a-posteriori to the sound, after the sound has been created. Within the context of the UPIC system, or within the microstructure of the originating sonic material (in the form of the samples of the particular waveforms used in the FM algorithms), the concepts of space and timbre, or movement in the space of a timbre, becomes and emergent aspect of the non-linear process used to create the digital samples in the computer. Timbre and spaciality are inextricably linked in the synthesis process. In other words, timbre and spatiality do not exist until the acoustic time pressure curves are created by the diaphragms of the loudspeakers and delivered to the listener.[2] The sound becomes the space, and the space is then defined by the sound, only through the interaction of the time-pressure curves and their complex interdependent relationships. External "panning knobs" are not required in order for sound to "move in space". What *is* required, within the UPIC framework (or, more generally, within a micro-

[2] Of course, our notion of space does not originate from a conception that is based on the internal architecture of electronic circuits or the particular designs of a loudspeaker enclosure. What we emphasize here is that the timbre and the spatiality of sound in the context of micro-composition does not rely on the established model of "moving" the represented sound with external equipment, be it a computer or other electronic device.

compositional framework), is the definition of specific phase relationships among the various waveforms in the FM algorithms[3] (figures 3, 5). In this case, the sound is not treated as a separate entity, dislocated from the actual process of composition (or listening, for that matter). The composer, in real-time, may directly manipulates the samples, which comprise the waveform in the wavetables, while the algorithm is in process. The effects of such manipulations directly influence the macrostructure of the resulting sound. The physics of sound in the air, the physiological mechanisms in the listener, and the "black box machine" in the form of the general purpose computer coupled with a synthesizer, all form a dynamic, interactive listening environment where each element plays a role in the overall emergent development and impression of an acoustic event (a sound) on the listener. One might suggest that conventional composition ends with the score, micro-composition ends with the speaker membrane (in the context of spatial music), and "emergent composition" ends with the listener.

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[3] These phase relationships are an integral part of the phenomenon of binaural hearing. See [Winckel 1967] pp. 72-74, and [Begault 1993] pp. 39-45.