A General Model for Spatial Processing of Sounds
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Introduction

We perceive sounds in a spatial context. Without visual cues, we can often tell the direction or distance from which a sound comes. We also perceive things about the apparent acoustic environment of sounds, such as whether they seem to come from a reverberant cave or a padded cell. Multichannel recordings can portray the spatial characteristics of recorded sounds independent of listening conditions. In ways analogous to looking through windows, we can discern things about one acoustic environment through headphones or loudspeakers while we move about in another.

Ideally, spatial processing of sounds would allow us to have complete control over the acoustic environment heard through the loudspeakers. Each sound located within this heard environment could have a specified "size," direction, distance, and apparent motion. We can use computers to gain such control over the spatial characteristics of sounds, but for musical applications we must always specify the acoustic processing we believe will produce the intended psychological effect. Spatial processing therefore involves the simultaneous consideration of two sets of problems: the physical characteristics of a space to be simulated and the psychological characteristics of sounds presented to listeners over loudspeakers.

The work described in this article consists of (1) a conceptual model for representing the problem of spatial processing and (2) a description of an implementation of this model in the context of the Cmusic sound synthesis program (Moore 1982).

This work was described by the author in a talk presented at the International Computer Music Conference in Venice, Italy, in September 1982.

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Localization

Much attention has been paid to our ability to localize sounds. Roederer (1975) points out that, especially at high frequencies, intensity cues (amplitude differences between the sound waves arriving at our two ears) help us to determine the direction from which a sound comes. At lower frequencies, time cues also contribute to localization (Molino 1974). What distinguishes high from low? At a speed of about 335 m per second, it takes a sound wavefront about 500 μsec to travel the 17 cm or so between our ears. A 2000-Hz tone therefore has a wavelength about equal to our interaural distance. At frequencies below this, interaural time delay can be an important factor in localization.

For interaural time differences to be important, something must exist in our neural mechanism that *correlates* the signals coming into our two ears. Neural models of such crosscorrelators were proposed as early as 1959 (Licklider 1959), and physiological evidence for such mechanisms has since been found (Rose et al. 1969). Roederer also points out that the existence of such a crosscorrelation mechanism has implications for spatial control:

> It is easy to see that the location . . . will depend on the interaural time delay, which in turn depends on the direction of the incoming sound. Two tones, a mistuned interval apart, fed into *separate* ears, may "foul up" the crosscorrelator: The gradually shifting phase difference between the two tones . . . will be interpreted by this mechanism as a changing difference in the *time of arrival* of the left and right auditory signals, hence signaling to the brain the sensation of a (physically nonexistent) cyclically changing sound direction! This is why two pure tones forming a mistuned consonant interval, presented di-

chotically with headphones, gives the eerie sensation of a sound image that seems to be "rotating inside the head" [Roederer 1975].

For localization to occur, we must know not only the direction of a sound source but also its distance. In research based on earlier work by Gardner (1962) and Wendt (1961), Chowning (1977) demonstrated that the relative mixture of direct-to-reverberant sound is a powerful cue for determining the distance between sound source and listener. By combining simulated cues for angular location, distance, and velocity (i.e., Doppler shift) with artificial reverberation (Schroeder 1962) via the Music V program (Mathews et al. 1969), Chowning was able to create convincing illusions of moving sound sources.

Improvements in our understanding of the acoustics of rooms (Schroeder, Gottlob, and Siebrasse 1974) allowed Moorer (1979) to synthesize a concise yet powerful model for artificial reverberation. Following Schroeder's suggestions, Moorer based his processing model on a tapped-delay line filter (also known as a *finite-impulse response* [FIR] filter) to simulate the "early echo" response of a room followed by a bank of recirculating filters (also known as *infinite-impulse response* [IIR] filters) to produce the effect of dense global reverberation. Moorer used the data gathered by Gottlob (1975) and Schroeder to obtain "reasonable-sounding" values for the tap-delay and gain parameters of the delay-line filter, together with acoustic data on sound absorption, to obtain similar values for the comb filters. These all led to a loose but useful simulation of Boston's Symphony Hall, with suggestions for alternatives.

Psychophysics Versus Performance

To gain more general control over spatial characteristics of musical sound, we need a general way to obtain reasonable-sounding values for the processing algorithm. The work of Schroeder, Gottlob, and Siebrasse, on which Moorer's values are based, was oriented toward improving the subjective impression of the sound of concert halls. With spatial processing, however, we wish not to analyze but rather to synthesize the sound of a concert hall or some other acoustic environment. In terms of the *tapped-delay-plus-recirculating* (TDR) filter model, we must find ways to synthesize the gain, delay, and recirculation parameters according to a specified musical intent.

Direct manipulation of such psychophysical parameters as interaural time delay can lead to compelling illusions of sounds in space. Using headphones or biteboards (which a listener grasps with the teeth) to control relative head position, interaural time delays can be used to obtain TDR filter parameters. We would expect the results of such an approach to produce strong impressions of localization.

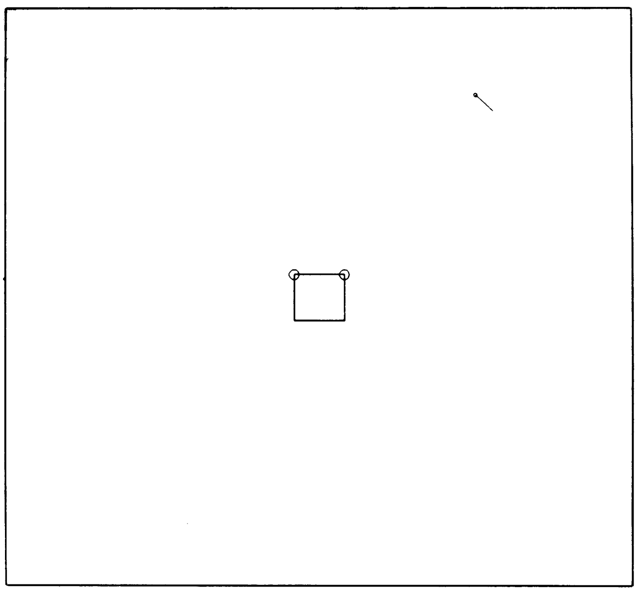
Unfortunately, the relative positions of listeners and sound sources in a concert situation is unpredictable. No two listeners in a concert hall hear exactly the same sound, rendering such factors as interaural time delay useless as control parameters for music intended to be heard under concert conditions. Even the relative amplitude of sound entering the two ears of each listener is likely to vary throughout the performance space.

Even though no two listeners in a concert hall hear the same thing, there is an invariance in their subjective perception—they all hear the same music, if from different vantage points. We are clearly able to compensate for our own vantage points, except under unusual conditions that fool the perceptual mechanism. (The failure of this compensation is the basis of most so-called illusions.) In the visual realm we compensate readily for the difference between size and distance, and we make similar sonic distinctions between loudness and intensity.

Since we cannot control psychophysical parameters directly in a concert, a practical model for spatial processing must be based on physical characteristics of the real or imaginary space or spaces to be simulated. This suggests modeling the playback situation itself, rather than the details of the listener's perception of it—the approach taken in this study. The model described here is based on the following elements:

The relevant characteristics of the listening

Fig. 1. An outer room enclosing an inner room. The circles on the periphery of the inner room represent holes in its walls (loudspeaker positions). A sound source is shown in the upper right quadrant. The small circle represents the base of the radiation vector associated with this source, and the line points in the direction of greatest radiation (the length of the vector is proportional to the amplitude of radiation in that direction).



space, such as its size and shape, as well as the number and placement of an arbitrary number of loudspeakers along its perimeter

The acoustic properties of the illusory space, including its size, shape, and sound-absorbing characteristics, specified independently of the characteristics of the listening space

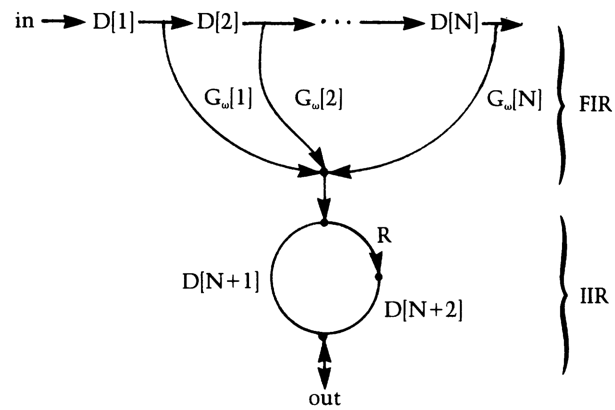
The radiating characteristics of the sound sources themselves, including their positions within the illusory space and their directional characteristics, again specified independently of the properties of the space or spaces in which they occur

Uncontrollable aspects of the concert situation, such as the listeners' head positions, present no greater problem from the standpoint of this model than they do in a traditional concert. Each listener hears something different, but all listeners gain individual perspectives on the same illusion.

Overview of the Model

The present model represents the performance and illusory acoustic spaces as a room within a room (Fig. 1).

Fig. 2. Signal flow diagram of a basic TDR filter. A characteristic set of delays (D[·]) and frequency-dependent gains (G_ω[·]) determine the operation of the tapped-delay (FIR) part of the filter. The summed output of the delay taps is then further processed by a recirculating (IIR) filter R, which provides dense-echo reverberation.



The *outer room* represents the illusory acoustic space from which the sounds emanate. The *inner room* represents an actual or intended performance space that holds listeners and loudspeakers. Loudspeakers are modeled as acoustic "windows" communicating with the illusory space outside the perimeter of the listening space. A particular set of "important" paths taken by the sound from sources located in the outer room to the loudspeakers is used to obtain time-varying TDR filter parameters. The TDR filter structure is extended to allow a separate tapped-delay section for every source location and for every sound channel (Figs. 2 and 3).

In practice, the numerous FIR stages can be collapsed into a single long delay with dynamic, movable taps, each with frequency-dependent gain.

For example, the inner room might be a living room with loudspeakers placed in each corner, while the outer room's specifications suggest the acoustic properties of a reverberant cathedral. From within the listening space, we listen through "holes" in the walls to sounds that exist in the outer room. Depending on proximity to the loudspeakers, each listener will hear these sounds from a slightly different perspective. Information is presented at each loudspeaker about all sound sources in the virtual outer room. The differences in perception among listeners in the inner room are analogous to perspective distortion. A sound moving along a circular path centered around the inner

Fig. 3. Extended TDR filter structure for spatial processing. Time-varying ($FIR_i(t)$) filter stages (one per speaker channel per sound source location) with both dynamically movable taps and frequency-dependent gains form the multichannel "early echo image" of the sound. Each sound source transmits to each loudspeaker along a direct path plus as many reflected paths as there are distinct surfaces in the outer room. A monophonic version of the multichannel early echo image is used to drive the global reverberator, which has decorrelated multichannel output.

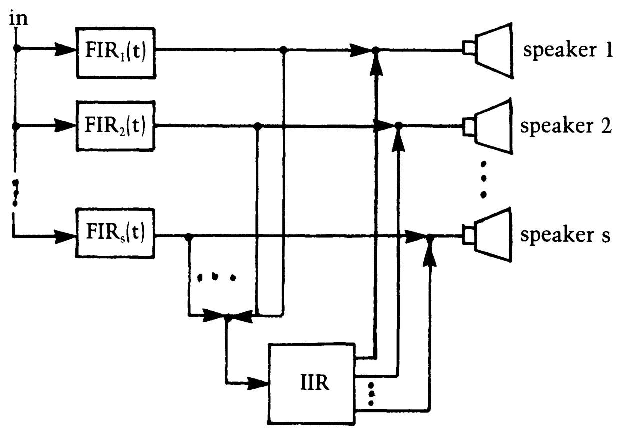
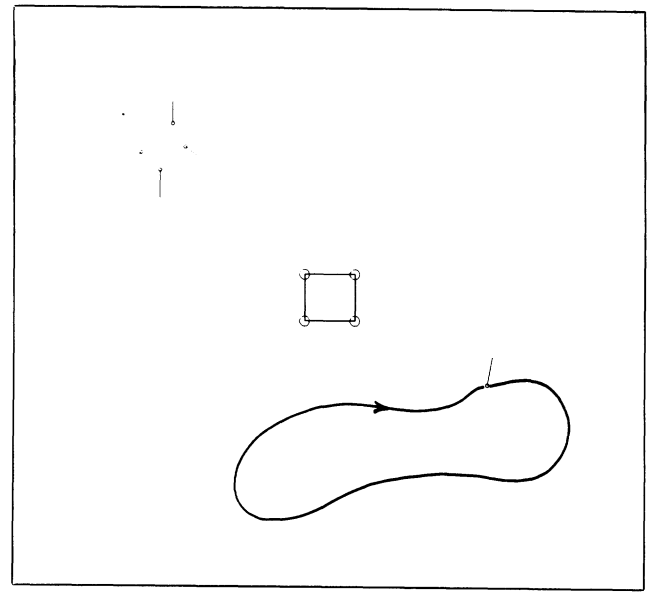


Fig. 4. Multiple radiation vectors and moving sources. The group of four radiation vectors in the upper left quadrant represents a radiating "surface" comprised of several directional sources (each source radiates, in general, a different version or component of the "same" sound). In the lower right quadrant, a sound path for a single moving source is depicted.



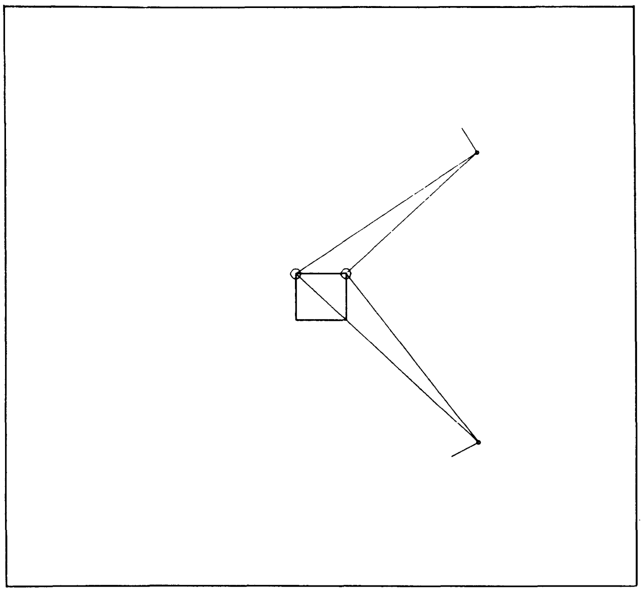
room would pass more closely to someone sitting close to a wall than to someone sitting in the center. After listening a while, however, all listeners should be able to agree on sound source locations regardless of where they sit in the inner room, at least as well as they could by listening through holes to actual sound sources moving about in a real outer room.

Sound sources may in general be located at any position in the outer room. Having more loudspeakers leads to a clearer depiction of the outer room. But even a small number of speakers can give the listener a great deal of information about the entire outer room and all sound source locations in it. Imagine, for example, that a performer in the outer room is walking in a circle around the inner room while beating on a tambourine. If we listen to the tambourine performance through two holes in the front two corners of the inner room (stereo), our ability to locate the sound will be excellent when it is in the front (the azimuth—or angle between the front-back line and a line pointing toward the sound source—lies between the two loudspeakers), less good when it is to the sides (outside the "cone" described by the lines drawn between the listener and the two loudspeakers), and ambiguous at the single point when it is directly behind the listeners. Thus even two loudspeakers would do a fair job of representing the entire two-dimensional plane of locations in the outer room. More loudspeakers would sample the acoustic field of the outer room with greater spatial frequency, lessening source location ambiguity.

By specifying sound paths in the outer room separately from the characteristics of the listening space, we separate the intended percept from its manner of presentation. Thus a given spatial composition might exist in several versions: one for two loudspeakers in a small room (a living-room-stereo version), one for headphones, and one for eight channels of sound in a large room (a concert version). The sound paths themselves would be identical in each version, that is, the composition itself would be *invariant*. Only the inner room specification would vary according to intended presentation.

A sound source in the outer room is modeled as one or more radiation vectors, each with an adjustable position, directionality, magnitude, and field shape. A single radiation vector suffices for most sound sources. Multiple radiation vectors (Fig. 4) may be used to describe sound-radiating surfaces to an arbitrary degree of precision. Individual radiation vectors may have time-varying characteristics, allowing sounds to move in arbitrary paths through-

Fig. 5. Front-back distinctions in stereo. The direct sound from one source is obstructed in the left channel, while the other is not. While this case is too simple to allow an auditory distinction to be made, more complex information in the early echo image of the sound gives perceptible front-back cues, even in stereo.



out the illusory space. In a real room, the number of radiated sound paths between source and destination is infinite. Restricting the model to two dimensions and including only the "important" sound paths results in the inclusion of only the direct paths from a source to each loudspeaker, plus one reflection from each wall of the outer room to each loudspeaker.

If the outside wall of the inner room is made absorptive (acoustically opaque), it is possible to minimize undersampling effects, emphasizing, for example, front-back distinctions even when only two front stereo loudspeakers are used (Fig. 5).

The outer room and radiation-vector specifications allow a complete description of the *perceptual effect* intended by the musician. Given the inner room specifications, computations for the requisite delays, attenuations, Doppler shifts, and so on can be made automatically from the specification of the desired perceptual effect.

The extent to which real loudspeakers are as acoustically transparent as holes in walls is questionable, of course. The infinite baffle loudspeaker model (which is basically just a hole in an infinite wall), allows sound to radiate hemispherically (on one side). This requires a hole that is small compared to the transmitted wavelengths, making it a purely refractive source. Higher frequencies would tend to "beam" through the hole if it were large compared to a wavelength of the transmitted sound. Good loudspeaker design minimizes this effect, however, producing as faithfully as possible a far field (i.e., nondirectionally differentiated) representation of the acoustic field sampled at a point by an ideal microphone.

Details of the Model

The major features modeled are the following:

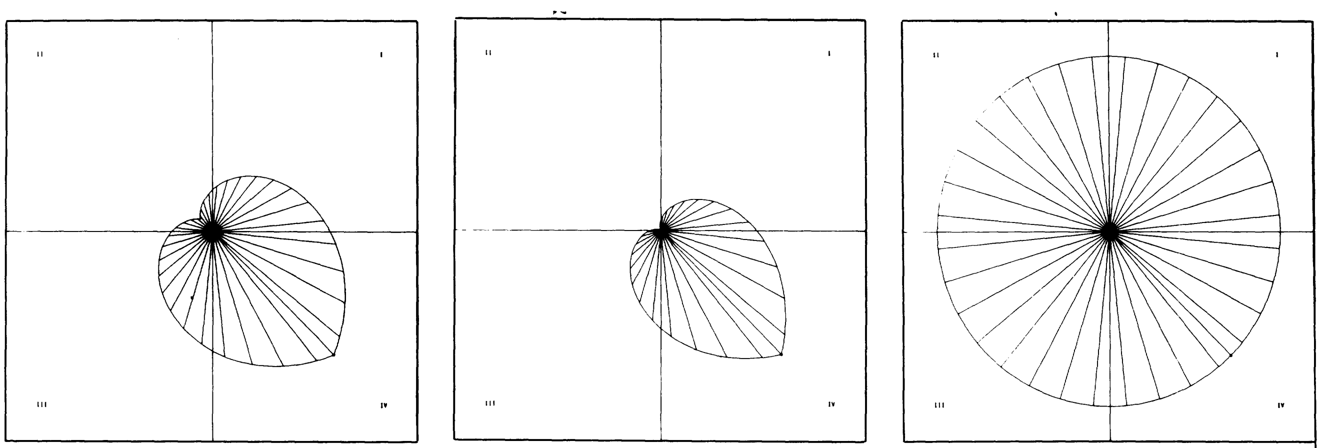
Loudspeaker placement in the listening space
Geometry of the virtual acoustic space
Radiation of sound sources into the virtual space
Early echo response of the virtual space
Global response of the virtual space

Loudspeaker Placement

Under normal circumstances it is impossible to create the illusion that the sound source is closer to the listener than the loudspeakers. Therefore, the shape of the inner room is determined by the shape of the performance hall and the location of the loudspeakers within it. For quadraphonic sound, the loudspeakers are normally considered to be at the four corners of a square (or other quadrilateral); for stereo they are normally at the front two corners of a square. For stereo headphones, the two loudspeakers are considered to be in the middle of the two side walls of a very small listening space (the size of one's head). In general, the loudspeakers are considered to lie at the vertices or along the perimeter of an arbitrary, closed polygon.

Virtual Acoustic Space

In two dimensions, the geometries of the inner and outer rooms are those of arbitrary, closed planar polygons. Each surface of the outer room is made an effective part of the acoustic space by being allowed to contribute to the early echo response. Because of the crudity of our spatial perception,



(a) (b) (c)

Fig. 6. Detailed radiation pattern for (x=0, y=0, θ=315°, amp=1, back=.1), in which the amplitude of the radiation in the direction opposite to 315° is one-tenth that of the forward radiation (−20 db) (a). *Detailed radiation pattern for (x=0, y=0, θ=315°, amp=1, back=0), in which the amplitude of the radiation in the direction opposite to 315° is zero (−∞ db)* (b). *Detailed radiation pattern for (x=0, y=0, θ=315°, amp=1, back=1), in which the amplitude of the radiation in the direction opposite to 315° is equal to that of the forward radiation (−0 db)* (c).

simple shapes such as squares and rectangles should usually suffice for the outer room.

The inner room polygon is determined not by the actual shape of the performance hall but by the location of the loudspeakers within it. Four speakers at the corners of a square 10 m on a side therefore define a square inner room 10 m on a side, regardless of the actual size and shape of the listening room. No particular allowance is made in this model for reverberant or other properties of the listening space, since matching and/or compensating for these is largely independent of the spatial characteristics of the illusory outer room.

Radiation Vectors

Sound sources are injected into the space by means of radiation vectors. A radiation vector **RV** is completely defined by the quintuple

$$\mathbf{RV} = (x, y, \theta, amp, back), \qquad (1)$$

where

- x and y are the base of the vector (all coordinates are given in meters, with the origin (0, 0) in the center of the inner room);
- θ is the direction of the vector (an angle of 0 rad points to the right as viewed from above);
- *amp* is the length of the vector and is used to scale the amplitude of the source sound; and
- *back* is the relative amplitude of the radiation in the direction opposite to that of the vector.

Sound is considered to be radiated in a supercardiodal pattern principally in the direction of the vector but with smaller amplitude to the sides and back (see Fig. 6).

The *back* value given in the specification of the radiation vector varies between 0 and 1. A *back* value of 0 implies no back radiation and a strongly directional radiation pattern. A *back* value of 1 implies an omnidirectional radiation pattern. The supercardiodal shape of the radiation pattern is given by:

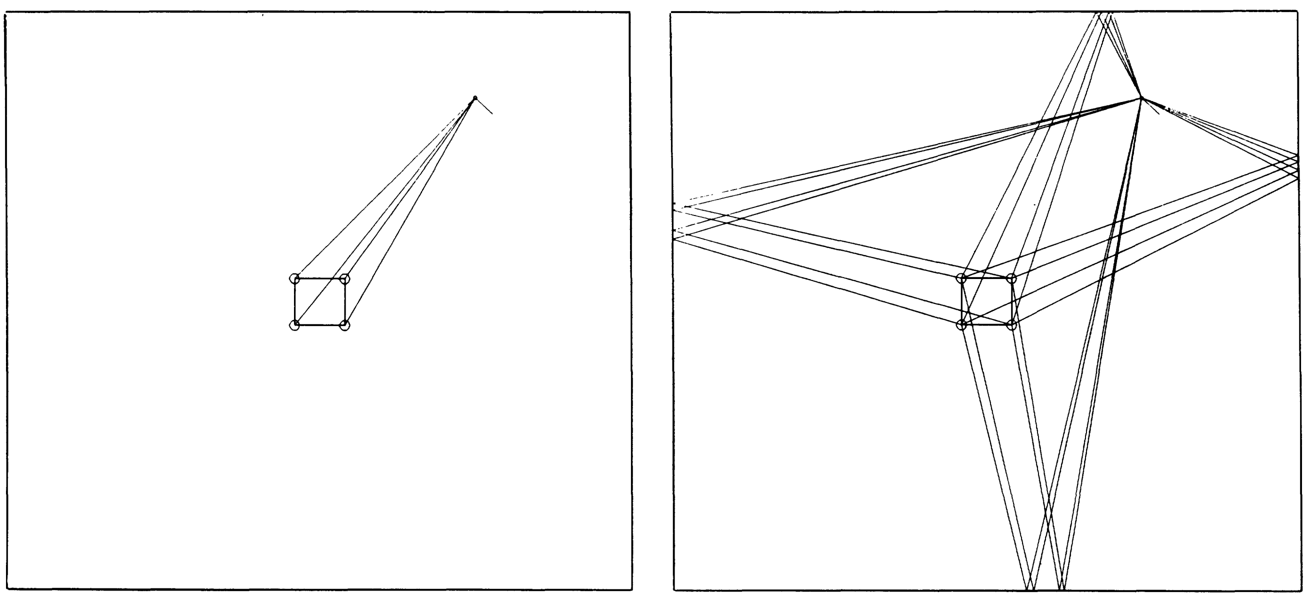
$$r(\phi) = \textit{scaler}\text{ for }\textit{radiation in ray direction } \phi = \left[1 + \frac{(back - 1)\,|\theta - \phi|}{\pi}\right]^2, \qquad (2)$$

where θ and *back* are defined as in Eq. (1). A single sound source emanates from one or more radiation vectors. Each radiation vector may be located anywhere outside the inner room in the space described by the outer room. Each radiation vector represents a source of sound in the virtual sound space.

Sound Paths

Sound sources radiate to each speaker channel in two ways: (1) by direct paths and (2) by reflected paths. There is exactly one potential direct path between each source and each speaker channel. For each source, there is also one potential reflected

Fig. 7. Direct radiation between a single source and four loudspeakers (note the cut path) (a). Reflected radiation between a single source and four loudspeakers (note cut paths) (b).



(a)

(b)

path to each speaker channel from each wall of the outer room. Thus a single source radiating sound to quad loudspeakers in a square outer room is modeled with four potential direct paths (one to each loudspeaker) and sixteen potential reflected paths (one from the source to each wall to each loudspeaker) (Fig. 7).

The shape of each path determines the following parameters:

- Attenuation along the path due to distance
- Frequency-dependent attenuation due to air absorption
- Frequency-dependent attenuation due to reflection (absorption by the reflecting surface)
- Absorption due to collision with the outer walls of the inner room (these are modeled as being completely absorptive)
- Time delay due to the finite speed of sound transmission

The paths are calculated in the following manner. Each direct sound path is simply a straight line between the source and the loudspeaker. One reflection path is used for each wall of the outer room. Of the possible reflection paths from a given wall, the one chosen for this model is the *principal* reflection path, that is, the one that results in the greatest amplitude at the loudspeaker position. The reflection point chosen is therefore the one that results in the shortest distance from source to wall to loudspeaker. This is the point at which the angles of sound incidence and reflection are equal. Such reflections are easily modeled by standard acoustic techniques involving virtual sources located on the opposite side of the reflecting wall.

Cut Factors

Modeling the outer wall of the inner room as completely absorptive yields good directional distinctions among various source locations. Therefore, a sound path is potential until it is determined whether it is obstructed by the inner room. An obstructed sound path is considered to be "cut" by the barrier, and to be completely absorbed at that point. We can define the *cut factor* for a sound path to have a value of 0 when the path is obstructed by a barrier and 1 when it is unobstructed. The cut factor can then be combined multiplicatively with the overall amplitude of the sound on a particular path.

Since the cut factors may dynamically vary for moving sound sources, flipping back and forth between 0 and 1 as ray paths change with changing source locations, some mechanism must be used to avoid clicks (due to stopping or starting the sound too abruptly). A simple method for dealing with this problem involves a linear interpolation between the 0 and 1 values as necessary to avoid clicks. A more sophisticated approach would be to model the refraction of the path around the edge of a cutting surface, but it seems unlikely that the computation involved would be justified on perceptual grounds.

Early Echo Pattern

The early echo pattern for each sound source is a collection of delays and frequency-dependent gains. At each moment, for each radiation vector, one potential direct path exists between the base of the vector and each loudspeaker. The cut factors determine whether the path is actually present or not. In addition, one potential path exists from each radiation source to each reflecting surface of the outer room to each loudspeaker. Thus the total number of sound paths included in this model is

$$N_{path} = N_{vec}N_{chan}(1 + N_{surf}) \quad (3)$$

sound paths modeled, where

N_{path} is the total number of paths;
N_{vec} is the number of radiation vectors;
N_{chan} is the number of speaker channels; and
N_{surf} is the number of reflecting surfaces in the outer room.

For each of these paths, P_i, $i = 1, 2, \ldots, N_{path}$, we define a delay $D[P_i]$ and a frequency-dependent gain $G_\omega[P_i]$:

$$D[P_i] = \frac{Dist[P_i]}{c} \quad (4)$$

and

$$G_\omega[P_i] = \frac{Amp[P_i]Rad[P_i]Cut[P_i]}{1 + Dist[P_i]}, \quad (5)$$

where

Dist[·] is the length of the path (the symbol [·] stands for the path argument);
c is the speed of sound;
Amp[·] is the amplitude scalar given in the radiation vector that is the source of the path;
Rad[·] is a "radiant," that is, an amplitude scaler for the direction of radiation (= $r(\phi)$ for the path); and
Cut[·] is the cut factor for the path (0 if cut and 1 if not).

A changing source position would likely cause a changing delay parameter, which would result in pitch shift as a side effect (shrinking delay values would shift the pitch up and vice versa). The magnitude of this pitch shift is precisely the same as that of a Doppler shift for a moving sound source. Since the TDR filter (with properly interpolated delay taps) provides such shifts automatically, no specification of Doppler shift is necessary for moving sounds with this model (they happen automatically!).

The complexity O_{dir} of the direct path computation is proportional to

$$O_{dir} \propto N_{vec}N_{chan}. \quad (6)$$

The complexity of the reflected path computation is then

$$O_{refl} \propto N_{vec}N_{chan}N_{surf}, \quad (7)$$

where the factors are defined as above. The overall complexity of the computation involved is proportional to the total number of paths:

$$\begin{aligned} O_{tot} &= O_{dir} + O_{refl} \\ &\propto N_{vec}N_{chan}(1 + N_{surf}) = N_{path}. \end{aligned} \quad (8)$$

Since for each path P_i we must compute both $D[P_i]$ and $G_\omega[P_i]$ as defined above, the total amount of computation for this model is significant. If all radiation-vector elements are allowed to be time varying, for example, new values must in general be calculated at every sample.

Global Reverberation

The output of the dynamic TDR filters is further processed by the global reverberator, which simulates the dense reverberation of the outer room after the first 60 msec or so. The characteristics of the global reverberation may be obtained from specifi-

Fig. 8. Cmusic score example. This score produces a sound that moves in a 10-m-radius circle centered about the point (32,22).

```
#include <carl/cmusic.h>
set stereo;
{
{The following are all default values which are repeated
  here only to show what they are}
{specify outer room 100 meters square}
  set space = 50,50 -50,50 -50,-50 50,-50;
{specify inner room 6 meters square}
  set room = 3,3 -3,3 -3,-3 3,-3;
{specify speakers in front corners}
  set speakers = 3,3 -3,3;
{global reverb time = 3 seconds}
  set t60 = 3;
{overall scale factor for global reverb}
  set revscale = .15;
{stop computing when reverb tail under cutoff}
  set cutoff = -60dB;
}
ins 0 circ;
  seg b4 p5 f4 d 0;   {b4 = envelope}
  osc b2 p7 p8 f2 d;  {b2 = x}
  osc b3 p7 p8 f3 d;  {b3 = y}
  adn b3 b3 p10;      {p10 = y-offset}
  adn b2 b2 p9;       {p9 = x-offset}
  osc b1 b4 p6 f1 d;  {b1 = carrier}
{specify 1 radiation vector with x = b2, y = b3,
  theta = 0, amp = 1, and back = 1 (omni source)}
  SPACE(b1,1) b2 b3 0 1 0dB;
end;
SAWTOOTH(f1);
SINE(f2);
COS(f3);
ENV(f4);
{play a note on instrument circ:} note 0 circ 4
{p5 = main amplitude:} 0dB
{p6 = carrier frequency:} 1000Hz
{p7 = circle radius in meters:} 10
{p8 = circular motion period:} 2sec
{p9,p10 = x,y center of circle:} 32,22
;
sec;
{allow 3 seconds for reverb tail to die away}
ter 3;
```

cations of the overall size and shape of the outer room and the reflective properties of its walls.

For the purposes of this model, the global reverberator accepts a single (monophonic) signal input consisting of the mixed outputs of all TDR filter signals. Parallel comb filters with frequency-dependent loop gains may be used to achieve dense echo. Output channels of the global reverberator (one for each loudspeaker) must be statistically decorrelated from each other in order to produce a good subjective effect (Schroeder 1980).

Implementation in Cmusic

The model has been implemented as a special unit generator in the Cmusic sound synthesis program. A special macro called SPACE allows an especially simple and direct control over the space unit generator. The initial implementation allows for simple room geometries and as much automatic operation as possible. Figure 8 shows a sample Cmusic score that moves a sound in a circle about an arbitrary point.

Conclusion

The Cmusic implementation described here is not a complete implementation of the spatial model, but it has been sufficient to test the hypothesis that the model yields convincing results. It produces convincingly localized sound images when used conservatively, such as with front sources in stereo. It also performs as well as expected under adverse conditions, such as with rear sources in stereo.

The main advantage of this model lies in its generality: from a physical specification of an intended perceptual effect, the model provides a tool with which to realize this effect computationally under given playback constraints. Either a change in the constraints, such as a change in the number or location of loudspeakers, or a more fundamental change in the method by which the computational structure is realized does not affect the intention or its specification. The problems of specification of the intended effect, realization of the computational means by which to produce this effect, and the playback constraints imposed in a given performance situation are neatly separated.

The model could readily be extended to three (or more) dimensions. Improvements in the computational structure are likely to follow from improvements in our understanding of both room acoustics and psychoacoustics.

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