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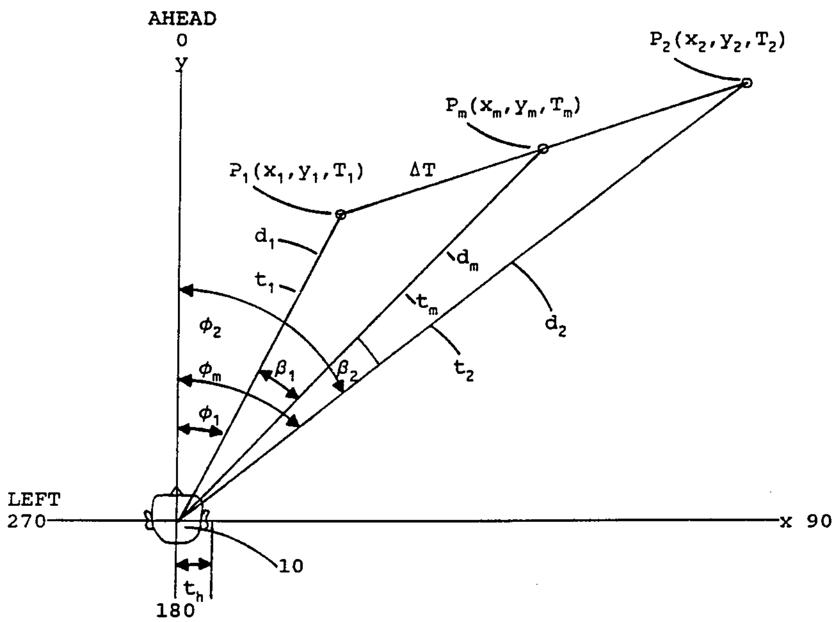

(57) Abstract

A method for producing three-dimensional sound associated with an object that is moving from a first position (P_1) to a second position (P_2) with respect to the listener (10). The method includes the effects of doppler shifting, head shadowing, distance on frequency components of the sound as well as the volume of the sound, and the natural sensitivity of the human ear in the 7-8 kHz range. The method provides a sequence of digital sound samples which when converted into analog waveforms and for production of audio signals will provide an audio signal which will provide sound queues to the listener for the location of the sound in three-dimensional space.

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METHOD FOR GENERATING THREE DIMENSIONAL SOUND

CROSS-REFERENCES TO RELATED APPLICATIONS

This application is related to:

PCT Patent Application Serial No. PCT/US92/09349, entitled AUDIO/VIDEO COMPUTER ARCHITECTURE, by inventors Mical et al., filed concurrently herewith, Attorney Docket No. MDIO4222, and also to U.S. Patent Application Serial No. 07/970,308, bearing the same title, same inventors and also filed concurrently herewith;

PCT Patent Application Serial No. PCT/US92/09342, entitled RESOLUTION ENHANCEMENT FOR VIDEO DISPLAY USING MULTI-LINE INTERPOLATION, by inventors Mical et al., filed concurrently herewith, Attorney Docket No. MDIO3050, and also to U.S. Patent Application Serial No. 07/970,287, bearing the same title, same inventors and also filed concurrently herewith;

PCT Patent Application Serial No. PCT/US92/09350, entitled METHOD FOR CONTROLLING A SPRYTE RENDERING PROCESSOR, by inventors Mical et al., filed concurrently herewith, Attorney Docket No. MDIO3040, and also to U.S. Patent Application Serial No. 07/970,278, bearing the same title, same inventors and also filed concurrently herewith;

PCT Patent Application Serial No. PCT/US92/09462, entitled SPRYTE RENDERING SYSTEM WITH IMPROVED CORNER CALCULATING ENGINE AND IMPROVED POLYGON-PAINT ENGINE, by inventors Needle et al., filed concurrently herewith, Attorney Docket No. MDIO4232, and also to U.S. Patent Application Serial No. 07/970,289, bearing the same title, same inventors and also filed concurrently herewith;

PCT Patent Application Serial No. PCT/US92/09460, entitled METHOD AND APPARATUS FOR UPDATING A CLUT DURING HORIZONTAL BLANKING, by inventors Mical et al., filed concurrently herewith, Attorney Docket No. MDIO4250, and also to U.S. Patent Application Serial No. 07/969,994, bearing the same title, same inventors and also filed concurrently herewith;

PCT Patent Application Serial No. PCT/US92/09461, entitled IMPROVED METHOD AND APPARATUS FOR PROCESSING IMAGE DATA, by inventors Mical et al., filed concurrently herewith, Attorney Docket No. MDIO4230, and also to U.S. Patent Application Serial No. 07/970,083, bearing the same title, same inventors and also filed concurrently herewith; and

PCT Patent Application Serial No. PCT/US92/09384, entitled PLAYER BUS APPARATUS AND METHOD, by inventors Needle et al., filed concurrently herewith, Attorney Docket No. MDIO4270, and also to U.S. Patent Application Serial No. 07/970,151, bearing the same title, same inventors and also filed concurrently herewith.

The related patent applications are all commonly assigned with the present application and are all incorporated herein by reference in their entirety.

BACKGROUND OF THE INVENTION

Field of the Invention

The invention relates to a method for generating three dimensional binaural sound from monaural digital sound samples that are associated with an object where that object is moving with respect to the listener.

Description of the Related Art

Over the past twenty years much work has been done in the area of sound processing to create the sensation that the sound being generated is in three dimensional space and not located from the loud speakers generating the sound. It is well understood in the field of acoustics that there are sound queues which allow a listener to locate the source of the sound in three dimensional space. Much of the work has been directed towards sound processing of pre-recorded sounds on records, tapes, laser discs, etc., to give the listener the illusion that the sound is located in three dimensional space and not solely to the speakers generating the sound. Such art, by way of example, can be found in U.S. Patent Number 4,817,149, entitled "Three Dimensional Auditorial Display Apparatus and Method Utilizing Enhanced Bionic Emulation of Human Binaural Sound Localization", Inventor: Peter H. Meyers, Issued March 28, 1989.

The article "Active Localization of Virtual Sounds", Jack M. Loomis, Chick Herbert, Joseph G. Cicinelli, Journal of the Acoustical Society of America, Volume 88(4), p. 1757, October 1990, describes a system in which monaural sound is generated and then sound queues are added to the sound such that the person listening to the generated sound through a headset has the sensation that the sound is being generated in three dimensional space. This article also describes and accounts for the movement of a persons head to aid in the location of a sound source.

U.S. Patent entitled "Sounds Imaging Process", U.S. Patent Number 5,046,097, Inventors: Danny D. Lowe et al., Issued September 3, 1991, describes a digital processing system for adding sound queues to produce the illusion of distinct sound sources being distributed throughout three dimensional space while using conventional stereo playback equipment.

Work is presently being done in the field entitled "Virtual Reality" which includes both three dimensional visual displays as well as three dimensional sound. Further, with the advent of home computers and interactive visual communication systems using home television sets as a video display means, it has become desirable to be able to generate a three dimensional sound or sounds associated with an object or objects appearing on the television screen and further to allow the listener and viewer to make interactive decisions with what is being displayed on the screen.

For example, if in a given video game situation the player observes a train moving from his right to left and towards him, it would be desirable to have the sound associated with the train not only give the queues as to the location of the train as it moves between the two locations, but to also include the doppler shift associated with the movement of the train as it moves toward or away from the player. Further, if the listener has the means of controlling his relative position with regard to what is being observed on the screen, the sound being generated must reflect the relative movement made by the listener. In that the relative position of the sound source, i.e. object on the screen, and the listener is no longer fixed, a method must be used to produce the sound being generated on a real time basis.

SUMMARY OF THE INVENTION

Accordingly, it is an object of the present invention to provide a method for generating sound associated with an object as that object travels from a first location to a second location.

It is a further object of the invention for the method to include the doppler shift associated with the relative movement of the object and the listener.

It is another object of the invention to provide a method which incorporates sound queues to aid the listener in locating the object in three dimensional space.

The preferred embodiment of the invention is to be implemented by a computer program. The method calls for input data indicative of the location (X and Y coordinates) and the time associated with a start point and end point of travel of the object. Inputs to the method can also comprise a descriptor of the amount of reverberation in the environment in which the action is taking place and, secondly, the relative loudness of the sound associated with the object. One set of input data is called a segment. The user continuously processes segments to define the relative movement of the object and the listener. The segments must be short enough in duration to allow the method to produce the proper sound as the player interacts with the system.

The method first determines if the input segment to the system meets the segment requirements of the system. If the segment is too large the segment is broken into subsegments until all subsegments meet the criteria of the method. The subsegments are ordered sequentially so they define the initial segment to the system.

Each subsegment is then processed sequentially. From the input data associated with the segment or subsegment being processed, ratios are formed for both ears as well as the value for various multipliers used in the reverberation, frequency shaping, and amplitude control portion of the method. The method uses monaural digital sound samples stored in the memory. These monaural sound samples have been sampled at the compact disc (CD) audio rate of 44.1 kHz. The method will generate digital output sound samples at the same rate, i.e. 44.1 kHz. A tick is the period of the frequency 44.1 kHz and is used as a basic time unit in the method.

The method uses the ratio for each ear to control the rate at which monaural sound samples for each ear are taken from memory. The source sound samples are taken consecutively from the memory. By this method the sound represented by the source sound samples can be compressed or elongated in time to provide the effect of the doppler shift caused by the object moving towards or away from the listener. During each tick one digital output sound sample is generated for each ear. The generated sound samples for each ear are processed separately.

The generated sound samples for each ear are processed for reverberation and passed through a combined notch and low pass filter for frequency shaping. The samples are then processed for amplitude adjustment which is a function of the distance between the listener and the object and the relative loudness of the sound. The processed digital output sound sample for each ear for each tick is stored in memory. Samples for each ear are taken from memory at the rate of 44.1 kHz and passed through a digital to analog converter. The resulting analog signal for each ear is fed to respective sides of a set of earphones.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will be described with respect to particular embodiments thereof and reference will be made to the drawings, in which:

FIGURE 1 is a diagram indicating the movement of an object from point P_1 to P_2 and depicts the parameters and orientation used in the invention.

FIGURE 2 is a logic diagram representative of the method for generating a digital sound sample as a function of the ratio.

FIGURE 3 is a logic diagram representative of the method used for interpolation.

FIGURE 4 is a graph exemplifying 22 monaural sound samples used by the method.

FIGURE 5 is a graph showing the sound samples generated by the method using the ratio of 1.25 with reference to the center of the head.

FIGURE 6 is a graph showing the sound samples generated by the method using the ratio of 0.9 with reference to the center of the head.

FIGURE 7 is a graph showing the sound samples generated by the method for the off ear using the ratio of 1.23.

FIGURE 8 is a graph showing the sound samples generated by the method for the near ear using the ratio of 1.27.

FIGURE 9 is a logic diagram depicting the method practiced by the invention for introducing reverberation for both ears.

FIGURE 10 is a logic drawing depicting the combination notch filter and low pass filter used for providing waveshaping.

FIGURE 11 is a logic diagram depicting the function of volume adjustment as a function of distance, the relative loudness of the sound to be generated and the storage of the final digital sound sample in memory which is connected to a digital to analog converter to provide an analog output for each ear.

FIGURE 12 is a graph depicting alpha and beta values for the left ear as a function of the angle of the object at an average distance of 500 units.

FIGURE 13 is a graph depicting alpha and beta values for the right ear as a function of the angle of the object at an average distance of 500 units.

FIGURE 14 is a graph depicting the conversion tables for alpha and beta from decibels to units.

FIGURE 15 is a graph depicting the relationship of the volume adjust multiplier as a function of average distance and the relationship of the left and right ear reverberation multipliers as a function of distance in accordance with the methods described.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

The method of this invention is used to generate the sound that an object associated with that sound would make as the object travels through three dimensional space. The method generates the sound on a real time basis thereby allowing the sound generated to be responsive to the interaction between the listener and the system the listener is using. One use for this method is in computer games which allows the viewer to interact with the computer system the viewer is using to play the game. However, the use of this method is not limited simply to computer games and may be used wherever virtual sound is desired.

The method is carried out by a computer program stored within a processor that has a computer having the capacity and speed to perform the method included within the program.

This method is to be used with other programs which will provide segment data to the method containing necessary parameters to generate the given sound. Table 1 lists the segment data provided to the method and the constants used by the method. Table 2 lists the parameters that will be calculated in order to practice the method. All of the parameters to be calculated as show in Table 2 are well known in the art and can be found in any physics text.

The basic unit of distance is 0.1 meters. The basic unit of time is derived from the CD recording rate of 44.1 kHz. The basic unit of time is one cycle of that frequency and is referred to as a tick. Therefore there are 44,100 ticks per second. The time between ticks, 226 μsec, is the amount of time that the processor has to perform the method practiced by this invention.

The method has five major parts. The first part is segment determination for determining if the segment given to the system meets the requirements and criteria of the method. The second part is the generation of a digital sound sample for each ear as a function of the position of the object and the listener for each tick of the segment. This portion also adjusts for the doppler effect caused by the object moving relatively toward or away from the listener. The third portion is the addition of reverberation to the generated sound sample. The user of the method can define the reverberation characteristics of the environment in which the object exists. The fourth portion is frequency shaping for the purpose of inserting queues to the listener for positioning the object in three dimensional space. The fifth portion is volume adjusting to account for the decrease in the loudness of the sound as the sound travels the distance between the listener and the object and for the relative initial loudness of the sound generated by the object. For example, a jet engine at 1,000 yards will be heard very clearly while a human voice at that same distance will not. Thus the method defines and accounts for the variable in initial loudness or power of the sound which is to be defined by the user.

Figure 1 illustrates the listener 10 observing an object going from point P_1 to point P_2 and the various parameters used by the method. It is understood that the location of the object is purely for exemplary purposes and the method operates for an object moving between any two points in three dimensional space around the listener. In the method the listener is always at the center of the coordinate system and the object moves relative to the listener. Where the listener can interact with the systems such that the listener can change the relative motion, such a change in relative motion is dealt with by rotating the axes of the system in an appropriate manner. The rotation of the coordinate system is not included within this method. Such a motion change would be reflected in the next segment of data sent to the method for sound generation.

A segment is defined as a start point P_1 and an end point P_2. Both points are defined by X,Y coordinates in units of distance of the system. The time, T_1 and T_2, at which the object is at P_1 and P_2 are also provided.

Segment Determination

When a new segment is defined for the method, the method will first determine if the segment defined meets the criteria of the method. The criteria are:

1. If the value of the midpoint distance, d_m, is less than 10 units (1 meter), then use the segment set as presented.

2. If angle B_1 is less than 5°, and if angle B_2 is less than 5°, and if the difference between distance d_1 and d_m is less 5% of d_m, and if the difference between d_2 and d_m is less 5% of d_m, then use the segment as presented otherwise divide the segment into subsegments.

If the conditions above are not met such that the segment has to be divided, then the segment is divided at the midpoint generating two new segments. The first portion of the first subsegment would have a start point P_1, a stop point P_m, and a start time T_1, and an end time T_m. The second segment would have as its parameters the

Table 1

INPUTS TO METHOD (Segment Data)	
Symbol	Description
$P_1(X_1,Y_1)$	Location of object at start of segment.
T_1	Time of start of segment.
$P_2(X_2,Y_2)$	Location of object at end of segment.
T_2	Time of end of segment.
CONSTANTS	
Symbol	Description
t_h	One-half the time for sound to travel the width of the head (12.5 ticks).
S	Speed of Sound in medium (for air = 315 m/sec).
R	Sample Rate of Stored Sound Sampler (14,100 kHz).
Rev	Reverberation characteristic of location.
Lou	Comparative Loudness setting for object.

Table 2

CALCULATED VALUES

Symbol	Description
P_m	Midpoint between P_1 and P_2.
T_m	Midpoint between T_1 and T_2.
d_1	Distance from P_1 to the center of the listener's head.
d_2	Distance from P_2 to the center of the listener's head.
d_m	Distance from P_m to the center of the listener's head. (Average distance.)
t_1	Time for sound to travel the distance d_1.
t_2	Time for sound to travel the distance d_2.
t_m	Time for sound to travel the distance d_m.
ϕ_1	Angle between Y axis and point P_1.
ϕ_2	Angle between Y axis and point P_2.
ϕ_m	Angle between Y axis and point P_m. (Average angle.)
β_1	Angle between P_1 and P_m.
β_2	Angle between P_2 and P_m.

start point P_m and the end point P_2, the start time of T_m and the end time of T_2. Each of those subsegments wouldthen be tested against the criteria to assure that the subsegment meets the criteria. If a subsegment meets the criteria then the subsegment is used. If the subsegment does not meet the criteria it is divided. This process is continued until all subsegments meet the criteria. The resulting subsegments are kept in order such that the final number of subsegments, when taken in sequence, will continuously form the segment as originally defined by the user.

Sample Generation

For the sake of discussion it will be assumed that the parameters shown in Figure 1 meet the criteria of a segment and, therefore, the segment did not have to be divided. The method next generates sound samples at the rate of 44.1 kHz that will correspond to the sound generated by the object as the object moves between P_1 and P_2.

The time that sound would be generated by the object is ΔT, the difference between time T_1 and T_2. Since the object is moving away from the listener in Figure 1, the time t_2 for sound to reach the listener from the end point P_2 will be greater than the time t_1 for sound to reach the listener from the start point P_1. Therefore the listener will hear sound for a longer period of time than the time sound was actually generated by the object. Conversely, the listener will hear sound for a shorter period of time than the time sound was generated by the object as it moves from P_1 to P_2. This gives rise to a change in pitch of the sound which is commonly referred to as the doppler effect. The method further takes into account the difference in time for the sound to be heard by the right and left ear of the listener as a function of the location of the sound in the coordinate system as shown in Figure 1. Clearly if the object is located as shown in Figure 1, the listener's right ear will receive sound waves earlier than the left ear. The method generates for each ear a ratio between the length of time that the sound would have been generated by the object as it moved from P_1 to P_2, to the length of time that the listener hears the sound that would have been generated as the object moved from P_1 to P_2. The ratio includes a correction factor to adjust for the location of the object with reference to the listener's ears.

An assumption is made that the time for sound to travel the distance between the listener's ears ($2t_h$) is very small when compared to ΔT, the time that the sound was generated. Further, since the time that the sound is heard by the listener is of the same magnitude as ΔT, then the time for sound to travel between the listener's ears, $2t_h$, is much smaller than the time sound is heard by the listener.

The ratio is first derived for the center of the head as follows:

Time sound would be generated:

$$\Delta T = T_2 - T_1 \tag{1}$$

Time sound would be heard by the listener using the center of the listener's head as a reference:

$$\Delta t = (T_2 + d_2/s) - (T_1 + d_1/s) \tag{2}$$

$$= (T_2 - T_1) + (d_2/s - d_1/s) \tag{3}$$

however, d_2/s are in units of decimeters/sec.

Converting the speed of sound to dm/tick, yields

$$S = 3150 \frac{\text{dm}}{\text{sec}} \times \frac{\text{sec}}{44{,}100 \text{ ticks}} = 0.0714285 \frac{\text{dm}}{\text{tick}}.$$

Thus,

$$t_2 \text{ (ticks)} = d_2/s = \frac{d_2 \text{ dm}}{0.07142285 \text{ dm/tick}} = 14d_2 \text{ (ticks)}.$$

In a similar manner t_1 (ticks) = $14d_1$ (ticks)

Thus,

$$\Delta t = (T_2 - T_1) + (14d_2 - 14d_1) \quad (4)$$

$$\text{Ratio (center of head)} = \frac{\Delta T}{\Delta t} \quad (5)$$

$$\text{Assume } t_h << \Delta T \text{ and } \Delta t \quad (6)$$

Correction for center of head to each ear is:

$$\delta = t_n(\sin\phi_1 - \sin\phi_2) \quad (7)$$

Ratio (right ear) is:

$$R_R = \Delta T/(\Delta t - \delta) \quad (8)$$

Ratio (left ear) is:

$$R_L = \Delta T/(\Delta t + \delta) \quad (9)$$

Since δ can have a maximum value of 25 ticks, we can assume that $\delta << \Delta t$. Further assume that the speed of the object is small when compared to the speed of sound. Under such assumption:

$$R_R \approx (\Delta T + \delta)/\Delta t \quad (10)$$

$$R_L \approx (\Delta T - \delta)/\Delta t \quad (11)$$

Equations (10) and (11) can be used when the criteria for segment determination has been used to limit the size of the segment. If the criteria for segment determination has not been used or made less stringent, then equations (8) and (9) should be used for the ratios.

The ratios for the right and left ear are generated once for each segment and is used throughout the segment for the purpose of generating sound samples. A sound sample is generated for each ear for each tick in the segment. It is envisioned that a segment will be changed once to twice a second. Therefore a segment will have 22,000 to 44,100 ticks and thus 22,000 to 44,100 sound samples will be generated for each ear for each segment.

The method at this stage is divided for the right and left ear. The description hereinafter will be with regard to the right ear, but it should be understood that the same processing is done for the left ear.

Ratio for the right ear, R_R, will be composed of two parts, an integer part and a fraction part. Where the object is going away from the listener, the integer part will be 0 and if the object is coming towards the listener, then the integer portion of the ratio will be 1. For each cycle of operation, i.e. a tick, the ratio R_R is added to the fractional portion of the summation ratio of the previous cycle. The results of this addition leads to a summation ratio for the right ear. Thus, by adding a fractional portion of a previous number to a number having an integer and fractional portion, the resulting summation ratio can have the integer portion to be equal to 1 if originally 0, and 2 if originally 1.

The integer portion is then used to select the samples from the monaural digital samples for the sound that has been stored in memory for use with this process.

It should be understood that the user of this method can store any monaural digital sampled sound in the memory. It is further well understood in the art that the time necessary for the monaural digital sound samples to describe the actual sound may be short. Therefore the monaural digital sound samples associated with the sound are looped so as to give a continuous source of digital sound samples of the sound to be generated. In the event there is a requirement that the monaural digital sound samples are greater in number than the allocated memory space, it is well within the art for the user to update the memory with new monaural digital sound samples as they are needed.

As previously stated, the integer portion of the summation ratio is used to control the number of monaural sound samples withdrawn from memory. The last two monaural sound samples that have been retrieved from the memory are used to generate the digital sound sample for the present tick. Interpolation is done between the two values of the sound sample using the fractional portion of the summation ratio. The interpolated value then becomes the generated digital sound sample for that tick for further processing during the tick.

Figure 2 is a logic diagram which logically depicts this portion of the method. Memory 21 stores the fractional portion D of the summation that results from adder 22. Adder 22 has as its inputs the ratio for the right ear R_R and the fractional portion D of the previous summation from the previous cycle stored in memory 21. After a new summation is done by adder 22, the new fractional portion D is stored in memory 21 to be used during the next cycle or tick. The integer portion I is sent to a comparator 23 to be tested. If the integer I is equal to 1 then the next monaural sound sample value is taken from memory 24 and stored in a two-stage FIFO 25. If the integer is equal to 2 then the next two monaural sound samples are taken from memory 24 and transferred to FIFO 25. If the tick has an integer value equal to 0 then the previous two fetched monaural sample values will still exist in FIFO 25. If one sample was fetched then FIFO 25 would have the previous sample that was fetched from memory 25 plus the present sample that has been fetched during this cycle. If two new samples were fetched from the memory 24, FIFO 25 will have the two samples stored this cycle. Interpolation is then done by interpolator 26 between the values of the two samples in FIFO 25 using the fractional portion D of the summation ratio. The interpolator 26 generates a digital sound sample for further processing by the remaining portion of the method.

Figure 3 is a logic diagram of the interpolator 26 of Figure 2. The interpolation is straightforward. The digital values stored in the second stage of FIFO 25 are subtracted from the digital values stored in the first stage of FIFO 25 yielding the difference between those two values. This difference (A - B) is then multiplied by multiplier 32 by the fractional portion D of the summation ratio to yield D(A - B). The output of multiplier 32 is then added together with the digital value of the first stage of FIFO 25 by adder 33 yielding the interpolated value for the sound sample for that tick.

Figure 4 depicts 22 monaural sound samples stored in memory 24. In this example it is assumed that the samples of Figure 4 are the middle of the segment being processed. The numbers assigned to the sample numbers are for convenience.

Table 3 demonstrates the generation of output value of the sound samples where the ratio to the center of the head is equal to 1.25. The summation ratio (SUM RATIO) illustrates the addition of the ratio 1.25 being added to the fractional portion of the preceding summation ratio. It is assumed that the value of the fractional portion of the preceding summation ratio prior to the start of this example was 0. The table further shows the monaural sample values used and the output value for the sound sample after interpolation.

Figure 5 is a graph of the output values of Table 3. It should be understood that at the start of this example FIFO 25 would have included samples 1 and 2 within that FIFO and, therefore, in Figure 4, samples 1 and 2 have already been read from memory 24. The next value that would be read from memory 24 would be sample 3. At the end of the example, sample 22 has been read from the memory store and is stored in FIFO 25.

It therefore can readily be realized by comparing Figure 4 with Figure 3 that samples 3 through 22 of Figure 4 have now been compressed into the 16 sound samples of Figure 5.

Table 4 is another example for the generation of sound samples in accordance with the method. Table 4 assumes the ratio R_R to be 0.9, that is the object is moving away from the listener rather than toward the listener. Figure 6 illustrates the sound samples generated by the method from the monaural sound samples of Figure 4. Again, sound samples 1 and 2 of Figure 4 were stored in FIFO 25 at the start of the example and it was assumed that the fractional portion D of the resulting summation ratio was equal to 0 at the start of the example. Table 4 indicates that for the first cycle of operation the integer portion of the summation ratio is 0, thus no new samples will be extracted from memory 24 and the existing values of samples 1 and 2 in FIFO 25 will be used. Since the fractional portion of the summation ratio is 0.9, the interpolation will be performed generating an output value of 5.80. Comparing Figure 6 with Figure 4 shows that sound samples 3 through 22 of Figure 4 have been expanded or stretched out to give generated sound samples 1 through 22 of Figure 6.

The previous discussion has used the ratio to the center of the head where the method uses the ratio which has been corrected to the right and left ear. Figures 7 and 8 depict the resulting sound samples if we allowed the correction to the ratio for the center of the head to be +/-0.02. Again this is being used for exemplary purposes only and it is anticipated that the differences between the right and left ear will not be of the magnitude of 0.02. However, for exemplary purposes, Figures 7 and 8 show the results of the method as

TABLE 3

Ratio = 1.25

SAMPLE NUMBER	SAMPLE VALUE	SUM RATIO	SAMPLE VALUES USED		OUTPUT VALUE	CYCLE NUMBER
1	4					
2	6	1.25	6	7	6.25	1
3	7	1.50	7	5	6.00	2
4	5	1.75	5	5	5.00	3
5	5	2.00	3	6	3.00	4
6	3	1.25	6	5	5.75	5
7	6	1.50	5	9	7.00	6
8	5	1.75	9	8	8.25	7
9	9	2.00	7	6	7.00	8
10	8	1.25	6	3	5.25	9
11	7	1.50	3	1	2.00	10
12	6	1.75	1	2	1.75	11
13	3	2.00	1	5	1.00	12
14	1	1.25	5	4	4.75	13
15	2	1.50	4	5	4.50	14
16	1	1.75	5	6	5.75	15
17	5	2.00	8	6	8.00	16
18	4					17
19	5					18
20	6					19
21	8					20
22	6					21
						22
						23

Table 4

Ratio = 0.9

SAMPLE NUMBER	SAMPLE VALUE	SUM RATIO	SAMPLE VALUES USED		OUTPUT VALUE	CYCLE NUMBER
1	4					
2	6	0.90	4	6	5.80	1
3	7	1.80	6	7	6.80	2
4	5	1.70	7	5	5.60	3
5	5	1.60	5	5	5.00	4
6	3	1.50	5	3	4.00	5
7	6	1.40	3	6	4.20	6
8	5	1.30	6	5	5.70	7
9	9	1.20	5	9	5.80	8
10	8	1.10	9	8	8.90	9
11	7	1.00	8	7	8.00	10
12	6	0.90	8	7	7.10	11
13	3	1.80	7	6	6.20	12
14	1	1.70	6	3	3.90	13
15	2	1.60	3	1	1.80	14
16	1	1.50	1	2	1.50	15
17	5	1.40	2	1	1.60	16
18	4	1.30	1	5	2.20	17
19	5	1.20	5	4	4.80	18
20	6	1.10	4	5	4.10	19
21	8	1.00	5	6	5.00	20
22	6	0.90	5	6	5.90	21
		1.80	6	8	7.60	22
		1.70	8	6	6.60	23

previously described for the near ear and for the off ear. Figures 7 and 8 can be compared with Figure 5. For the near ear (Figure 7) the number of ticks generated are the same but the magnitude of each of theticks are different. With regard to the off ear (Figure 8) one additional tick was necessary for the method than for the near ear. Again, each of the ticks are of a different magnitude than that for the center of the head. Thus, the near and the off ear will have a different set of generated sound samples for the same segment.

<u>Reverberation Portion</u>

It is desirable to add reverberation to the generated sound since it is desired to emulate sound in three dimensional space. Figure 9 logically depicts the method for introducing reverberation. The generated sound sample for the right ear is first multiplied by multiplier 91 and then added together with the output of multiplier 93. The values of the multiplication factors for multipliers 91 and 93 when added together are equal to 1. The input to multiplier 93 is the output of the reverberation buffer 94. The reverberation buffers 93 and 94 are analogous to a FIFO buffer where the oldest sample stored in the buffer is the sample that will next be multiplied by multiplier 93 and added by adder 92 to the output of multiplier 91. The input of reverberation buffer 94 is the output of adder 98 which is the adder associated for the left ear. Thus there is cross-coupling between the two ears in the reverberation section of the method. It has been found that reverberation buffers 94 and 95 should be of different lengths. In the present embodiment reverberation buffer 94 is 2,039 ticks and reverberation buffer 95 is 1,777.

This delay equates to a 46 and 40 millisecond delay respectively for buffers 94 and 95.

The upper limit of the multiplication factor for the multiplier 96 and 93 is 0.5. In the present embodiment of the invention the maximum reverberation level is set to 100 units on a scale of 256 units, or a decimal value of 0.391. It has further been found that it is desirable to not only have the delay for each ear different but also the amount of reverberation should be different for each ear. To this end, the method calls for a 5% reduction in the reverberation level between the two ears.

Reverberation is a function of distance from the listener to the object that would be generating the sound. The greater the distance the greater the reverberation for a given set of reverberation characteristics. It is desirable to have some reverberation in all cases and, therefore, a minimum reverberation level is set to 20 on a scale of 256 or 0.078. Figure 15 is a graph which shows the reverberation levels settings for the two ears as a function of distance. It has been found convenient in this method to use a scaling factor of 256 for calculating the various values of multiplied functions used throughout the method. For example, if the right ear reverberation level for a given distance was determined to be 50 units, then the reverberation level for the left ear would be 5% less, or 47.5 units. The multiplication factor associated with the multiplication function as illustrated by multiplier number 93 would be 0.195. This would cause the setting of multiplier 91 to be equal to 0.805 in that the multiplication factors for the multiplication steps illustrated by multipliers 90 and 91 must equal 1. The multiplier factors for the left ear would be set slightly different in that multiplier factor associated with multiplier 96 is set at a value of 95% of the multiplication factor represented by multiplier 93. The resulting value of the multiplication factor associated with the multiplier 96 would be 0.185, which in turn would cause the multiplication factor associated with multiplier 97 to be 0.815. Again, the summation of the multiplication factors associated with multipliers 96 and 97 must equal 1.

The method defines the reverberation level to equal the MINIMUM REVERBERATION plus the sum of the average distance d_m minus the NEARBY value divided by the REVERBERATION STEPS. The MINIMUM REVERBERATION is 20, NEARBY is 100 units of distance, REVERBERATION STEP is 10 and MAXIMUM REVERBERATION is 100. Therefore the settings of the reverberation multiplication factor is a function of the distance between the listener and the object, taking into account maximum and minimum reverberation values allowed. The user of the system is given the prerogative of setting the values for MAXIMUM REVERBERATION level, MINIMUM REVERBERATION, NEARBY and REVERBERATION STEP. In this means the user has the freedom to control the reverberation characteristics which the user wishes to have associated with the sound being generated. The user can, by adjusting these values, have the reverberation sound like the object and listener are in a tunnel or in a deadened sound-proof room.

<u>Frequency Shaping</u>

It is known that a sound generated in three dimensional space has its frequency components filtered by the media through which it is travelling such that the sound heard by the listener has the frequency components of the original sound substantially altered. Besides the doppler effect, as previously addressed, there are other well known phenomena that affect the frequency component of the sound and act as location queues for the listener.

The first phenomenon is the greater the distance sound has to travel, the greater the high frequency components of the sound are attenuated. A second phenomenon is that of head shadowing of the sound where low frequencies easily go around the head while the high frequencies are blocked or attenuated.

Another phenomenon is the sensitivity peak for the normal ear in the range of 7 - 8 kHz. Since it is envisioned that the listener will be wearing earphones, all sound will be subject to this phenomenon. It has been understood that the effects of this phenomenon is a function of the location of the object making the sound with respect to the listener such that it is maximum when the object making the sound is perpendicular to an ear of the listener and minimum when the object making the sound is in front or behind the listener. Since earphones are used any sound queue with regard to this phenomenon that would have been attainable are destroyed because the earphones are directly perpendicular to the listener's ear. Therefore the method adjusts for this phenomenon by having a notch filter at 7 - 8 kHz where the depth of the notch is a function of the object's location relative to the listener. When the object is perpendicular to the listener's ear then the notch of the notch filter is approximately 0, leaving the phenomenon to exist in its natural state. As the object is located around the listener's head, the depth of the notch of 7 - 8 kHz is increased to a maximum level of 5db when the object is either directly in front of or to the rear of the listener. By this method the sound queues associated with the phenomenon are again provided to the listener.

To this end, a combination of a notch and low pass digital filter has been employed. Digital filters are commonly known and a discussion of them will not be provided herein. A reference for digital filters is the text entitled "Digital Audio Signal Processing by John Strawm, published by William Kaufman, Inc., 1985, ISBN 0-86576-0H2-9.

Figure 10 is a logic illustration of the combined digital notch and low pass filters used for waveshaping for location queueing. The notch filter is comprised of a three sample delay 101, two multipliers, 102 and 103, and adder 104. The low pass filter is shown as a one sample delay 106 and multiplier 105. The output of multiplier 105 is added to the output of multipliers 103 and 102 by adder 104. In determining the multiplication factor associated with multipliers 102, 103 and 104, methodology has been established to emulate the frequency shaping that is provided in the natural environment for sound generated in three dimensional space.

To this end a first value (beta) is generated. A value for beta is calculated for the right and for the left ear as follows:

Beta left ear = ((d_m - NEARBY) 100) +
$10|\sin\phi_m|$ (if ϕ_m is in the range of 90° to 180°) +
$10|\sin\phi_m|$ (if ϕ_m is in the range of 0° to 180°)

Beta right ear = ((d_m - NEARBY) 100 +
$10|\sin\phi_m|$ (if ϕ_m is in the range of 90° to 180°) +
$10|\sin\phi_m|$ (if ϕ_m is in the range of 180° to 360°)

Beta is used to account for distance roll-off, rear head and side head shadowing.

A second value (alpha) is calculated for each ear as follows:

Alpha left ear = Beta left ear 2 + $5|\cos\phi_m|$ - 2.5
Alpha right ear = Beta right ear 2 + $5|\cos\phi_m|$ - 2.5

It has been found that the combination of the notch filter with the low pass filter provides the best results with regard to the desired frequency shaping. Alpha values depend on the beta values thereby allowing flatter and lower knee of the high frequency roll-off characteristics of the filters. The term $5|\cos\phi_m|$ controls the notch of the notch filter as a function of the position of the object and limits the notch to be 5 decibels.

Most earphones have designed compensation for the frequency at 7 - 8 kHz. A factor of -2.5db has been included to offset this design characteristic of the earphone such that the listener receives truer sound queues. The value of alpha and beta are in decibels and it is necessary to convert those values into multiplication factors for the various multiplication functions to be performed within the digital filters. A scale of 256 has again been used and by experimentation the following tables have been generated.

As can be seen from Table 6, the maximum decibel level that would be allowed in the alpha table is 20db and, therefore, if the value of alpha should be greater than 20, it would be limited to the value for 20db. In a similar fashion Table 7 shows that if the value of beta should be greater than 9db, the value will be limited to 9db. Where the resulting decibel values of alpha and beta are not whole integers, alpha and beta is obtained by interpolation. The results of the interpolation are always rounded to the nearest whole number.

Because the filters used were combined the alpha value must be adjusted. The alpha value is mapped into the remaining scale units not used by beta such that alpha will have the same percentage of the remaining units that it had in the original scale. The value of alpha is obtained as follows:

$$\text{Alpha(new)} = (256 - \text{Beta}) \times (\text{Alpha(old)}/256)$$

Table 6

Alpha Table

Decibel	Scale Value	Decibel	Scale Value
0	0	11	92
1	15	12	96
2	27	13	102
3	38	14	103
4	48	15	105
5	56	16	107
6	64	17	110
7	72	18	112
8	77	19	113
9	82	20	115
10	88		

Table 7

Beta Table

Decibel	Scale Value
0	0
1	15
2	30
3	46
4	59
5	73
6	87
7	102
8	112
9	127

Let us assume for the sake of example that beta equaled 3db for a scale value of 46, and alpha equaled 9db for a scale value of 82. In the example given, alpha (new) would become 67. The beta value is used to set the multiplication factor associated with multiplier 105 and in this given example would be 0.180.The alpha value of 67 would be used to create the multiplication factor associated with multiplier 103 which in our example would be 0.262. It is required that the multiplier functions of multipliers 102, 103 and 105 equal one or unity and, therefore, multiplier 102's value would be 0.558.

Figure 14 is a plot of the conversion scale of Tables 6 and 7 for alpha and beta.

Figure 12 is a graph showing the values for alpha and beta in decibels as a function of the average angle ϕ_m for a constant average distance d_m, of 500 units for the left ear. Figure 13 shows the value of the alpha and beta in decibels as a function of the average angle ϕ_m for a constant average distance d_m of 500 units for the right ear. As can readily be seen by Figures 12 and 13, the resulting values for the alpha and beta for the right and left ear will be different for the same average distance d_m for the sum average angle ϕ_m.

Each of the generated samples, after being altered for reverberation, are processed through the digital filters using the values for the multiplication functions as herein described. The output of the filters is a filtered sound sample.

<u>Volume Adjust Portion</u>

The filtered sound sample is then adjusted in volume to account for the distance between the listener and the object as well as for the relative strength of the original sound.

Figure 11 is a logic diagram illustrating the method of the invention. The digital sound sample is first multiplied by multiplier 111. Multiplier 111's multiplication factor is determined as a function of the distance from the listener to the object. Once again a scale from 0 - 256 is used. The equation for adjusting the volume is:

$$\text{Volume} = (256 \times \text{NEARBY})\ d_m$$

If the mid or average distance d_m is less than NEARBY then the volume is full at 256. It is desirable to have the volume always set at some value greater than zero and, therefore, a minimum setting for the volume is VERY QUIET which has a value of two. Figure 15 is a graph which depicts the volume setting as a function of distance in units. For example, if the average distance d_m was 1,000 units (100 meters), the resulting volume adjust level would be 25.6. When converted to a decimal value the resulting multiplication factor would be 0.1.

The output of multiplier 111 is then multiplied by multiplier 112. The multiplier factor associated with multiplier 112 is set by the user and determines the relative loudness or strength of the sound for the various sounds being generated by the method.

It is anticipated that more than one sound will be generated at the same time. This can be done by parallel processing of the sound or sequential processing of the sounds where the computing machine is fast enough to generate the digital samples for each of the sounds for each of the ears during the period of one tick.

The output of the multiplier 112 or the multiplication function associated therewith is then stored in a memory 113. The digital sound samples are taken from memory 113 at the rate of 44.1 kHz. The output of the memory 113 is in turn sent to a digital-to-analog converter 114. The output of digital analog converter 114 will be an analog signal which when processed by earphones will generate sound where the sound will be representative of the sound that would have been generated by the object as that object moves from P_1 to P_2. Again, it is understood that there is a channel for generating and processing sound samples for each ear and that the resulting analog signal for each ear will be different.

A fully computerize implementation of the invention as described heretofore uses known digital software implementations. A program written in programming language C is provided in Appendix A. This program practices the method of the invention consisting of the five portions as set forth above.

While the preferred embodiment of the invention as described herein included five portions, it should be understood that the segment determination portion, reverberation portion, frequency shaping portion, and the volume adjust portion may be deleted. The omission of one or more of these portions will effectively lose some sound queues as to the location of the object and will decrease the quality of the sound produced.

A specific embodiment of the sound generation method has been described for the purpose of illustrating the manner in which the invention may be practiced. It should be understood that implementation of the other variations and modifications of the invention in its various aspects will be apparent to those skilled in the art and that the invention is not limited thereto by the specific embodiment described. The present invention is therefore contemplated to cover any and all modifications, variations and equivalents that fall within the true spirit and scope of the underlying principles disclosed and claimed herein.

```
*********************************************************
*                                                       *
* Copyright © 1992 The 3DO Company                      *
*                                                       *
*                                                       *
*                                                       *
* 33D sound processing routine.  Two channels           *
*                                                       *
* Register twiddling set up for 4/8/4 mapping.          *
* Odd-numbered RBASE landrumizations not used           *
*********************************************************


*********************************************************
*                                                       *
* RBASE R0-R3   R4-R7   R8-R11
*   0   0-3     104-107 200-203     General input/control            *
*   2   8-B     10C-10F 208-20B     Left ear, doppler/picker          *
*   4   10-13   114-117 210-213     Left ear, filter                 *
*   6   18-1B   11C-11F 218-21B     Right ear, doppler/picker        *
*   8   20-23   124-127 220-223     Right ear, filter                *
*   A   28-2B   12C-12F 228-22B     Final volume/mixing for both ears *
*                                                       *
*********************************************************

RBASE_GENERAL           EQU     $0
RBASE_LEFT_DOPPLER      EQU     $2
RBASE_LEFT_FILTER       EQU     $4
RBASE_RIGHT_DOPPLER     EQU     $6
RBASE_RIGHT_FILTER      EQU     $8
RBASE_MIX               EQU     $A

*********************************************************
*                                                       *
* INPUT-FIFO                    Purpose                         *
*                                                       *
* F0                            Left ear, raw sounds            *
* F1                            Right ear, raw sounds           *
* F2                            Left ear, reverb input          *
* F3                            Right ear, reverb input         *
*                                                       *
*********************************************************

LEFT_RAW_IN             EQU     $F0
RIGHT_RAW_IN            EQU     $F1
LEFT_REVERB_IN          EQU     $F2
RIGHT_REVERB_IN         EQU     $F3

*****************************************************************
*                                                               *
* Working-memory locations, normally mapped into filters' R11   *
* space, which are accessed directly during final mixing        *
*                                                               *
*****************************************************************/

LEFT_OUT                EQU     $213
RIGHT_OUT               EQU     $223

*********************************************************
*                                                       *
* OUTPUT-FIFO                   Purpose                         *
*                                                       *
* 330                           Left ear, reverb output         *
* 331                           Right ear, reverb output        *
* 332                           Left ear, final sound           *
* 333                           Right ear, final sound          *
*                                                       *
*********************************************************
```

```
LEFT_REVERB_OUT         EQU     $3F0
RIGHT_REVERB_OUT        EQU     $3F1
LEFT_DAC_OUT            EQU     $3F2
RIGHT_DAC_OUT           EQU     $3F3

/****************************************************
*                                                   *
* RBASE=0 register allocation                       *
*                                                   *
* R0    DSPP clock, memory mapped                   *
* R1    unused     *
* R2    unused                                      *
* R3    unused                                      *
* R4    == 0 if first time, == 1 otherwise          *
*                                                   *
****************************************************/

*****************************************************
*                                                   *
* RBASE=RBASE_MIX register allocations for final mix *
*                                                   *
* R0    left-DAC volume for 33D left ear sound      *
* R1    left-DAC volume for 33D right ear sound     *
* R2    right-DAC volume for 33D right ear sound    *
* R3    right-DAC volume for 33D left ear sound     *
*                                                   *
****************************************************/
        PAGE
* Main routine.

START:  R4
*       SYNC
        BNE     NOT1ST

*
* Initial program bootstrap.  Walk through the sound channels' register
* banks, and preinitialize the working registers with whatever's necessary.
*
        MOVE    R4<-#1
        RBASE   RBASE_LEFT_DOPPLER
        JSR     ZEROWRK
        MOVE    R8<-#LEFT_RAW_IN
        MOVE    R9<-#LEFT_REVERB_IN
        RBASE   RBASE_LEFT_FILTER
        JSR     ZEROWRK
        NOP                     ; keep following RBASE from executing
        RBASE   RBASE_RIGHT_DOPPLER
        JSR     ZEROWRK
        MOVE    R8<-#RIGHT_RAW_IN
        MOVE    R9<-#RIGHT_REVERB_IN
        RBASE   RBASE_RIGHT_FILTER
        JSR     ZEROWRK
        NOP                     ; wait for it...
        SLEEP
*
* Subroutine to zero out all working registers
*

ZEROWRK MOVE    R4<-#0
        MOVE    R5<-#0
        MOVE    R6<-#0
        MOVE    R7<-#0
        MOVE    R8<-#0
        MOVE    R9<-#0
        MOVE    R10<-#0
        MOVE    R11<-#0
        RTS
        NOP
        PAGE
```

```
 *
 *      Up and running.
 *

NOT1ST:
        RBASE   RBASE_LEFT_DOPPLER       ; Doppler/interpolate left ear
        JSR     INTERP
        NOP
        LEFT_REVERB_OUT-A                ; write accume to reverb out FIFO
        RBASE   RBASE_LEFT_FILTER
        JSR     FILTER                   ; filter it
        NOP                              ; wait for it
        RBASE   RBASE_RIGHT_DOPPLER      ; Doppler/interpolate right ear
        JSR     INTERP
        NOP
        RIGHT_REVERB_OUT-A               ; write accume to reverb out FIFO
        RBASE   RBASE_RIGHT_FILTER
        JSR     FILTER                   ; filter it
        NOP                              ; wait for it
        RBASE   RBASE_MIX                ; prepare to mix
        R0*LEFT_OUT                      ; mult left ear sound, left DAC
        LEFT_DAC_OUT-R1*RIGHT_OUT+A     ; mult/mix right ear sound, left DAC
        R2*RIGHT_OUT                     ; mult right ear sound, right DAC
        RIGHT_DAC_OUT-R3*LEFT_OUT+A      ; mult/mix left ear sound, right DAC
        SLEEP                            ; done


*******************************************************
*                                                     *
* Phase 1 - handle Doppler sampling and reverb processing *
*           with linear interpolation during sample   *
*           skipping and doubling.                    *
*                                                     *
* Input registers:                                    *
*                                                     *
*       R0      reverb level                              *
*       R1      nonreverb level (1.0 - reverb level)      *
*       R2      Doppler pick rate, integer portion        *
*       R3      Doppler pick rate, fractional portion     *
*                                                     *
* Work registers:                                     *
*                                                     *
*       R4      Doppler sample time, integer, high part   *
*       R5      Doppler sample time, integer, low part    *
*       R6      Doppler sample offset, integer            *
*       R7      Doppler sample offset, fractional         *
*       R8      input-sample DMA pointer                  *
*       R9      reverb-input DMA pointer                  *
*       R10     "this" sample from input                  *
*       R11     "prev" sample from input                  *
*                                                     *
* Output registers:                                   *
*                                                     *
*       None used.                                        *
*                                                     *
* Memory outputs:                                     *
*                                                     *
*       Dopplered and reverbed sample returned in ALU accume. *
*       Hasn't yet been written to reverb-output DMA.     *
*                                                     *
*******************************************************/

INTERP:
        NOP
        NOP
        NOP
 *      Add R3 to R7, write result back to R7   # fractional part of doppler
        @R7+R3
```

```
*       Add carry bit to R2, write to R6        # real part of doppler
        R6=R2+C
*       Add result to R5, write back to R5          # update time/low
        @R5+A
*       Add carry to R4, write back to R4       #  update time/high
        @R4+C
*       Subtract 1 from R6, >> 4 arith          # see if it rolled over
        R6-#1>>>'4
*       Branch if negative to LININT            # quit if not
        BMI LININT
*       Sanity buffer
        NOP

DOPPL2:
*       Decrement accume, 20 bit                #  see if need to skip later
        --A
*       Move R10 to R11                         # propagate
        MOVE R11<-R10
*       Move (R8) to R10                        # fetch a sample
        MOVE R10<-[R8]
*       Branch back if need to skip this sample
        BGE DOPPL2

LININT:
*       ALU xfer R7 >> 1, write to R6           # convert to pos fraction
        R6=R7>>'1
*       ALU add 0x8000 to accume            #  calc F - 1.0
        A+#$8000
*       Mult accume * R11, ALU pass             #   prev*(F-1)
        A*R11
*       Mult R6*R10, subtract accume            #    this*F + prev*(1-F)
        R6*R10-A

REVERB2:
*       Multiply accume * R1, pass through ALU  # Weight the interp sample
        R1*A
*       Multiply (R9) * R0, add, return in ALU  #  Weight reverb, update bfr
        [R9]*R0+A
*
*       No reverb for now, assume a zero value
*
*       #0*R0+A
*       Return                                  # Done
        RTS
        PAGE
*******************************************
*                                         *
* Phase 2 - handle filtering and mixing   *
* -------------------------------------   *
*                                         *
* Input registers:                        *
*                                         *
*       R0      alpha level (1-sample feedback)    *
*       R1      beta level (3-sample feedforward)  *
*       R2      feed level (1 - |alpha| - |beta|)  *
*       R3      unused                             *
*                                         *
* Work registers:                         *
*                                         *
*       R4      feedforward 1                      *
*       R5      feedforward 2                      *
*       R6      feedforward 3                      *
*       R7      feedback                           *
*       R8      unused                             *
*       R9      unused                             *
*       R10     input sample                       *
*       R11     output sample                      *
*                                         *
* Output registers:                       *
```

```
*                                                    *
*        none                                        *
*                                                    *
********************************************/

FILTER:

*        Move accume to R10                          ; save input
         R10=A
*        Multiply accume * R2, ALU passthrough       ; weight input
         R2*A
*        Multiply R0 * R7, ALU add                   ;  weight alpha feedback
         R0*R7+A
*        Multiply R1 * R6, ALU add, write R11        ;
         R11=R1*R6+A
*        Move R5 to R6                               ; advance feedforward
         MOVE R6<-R5
*        Move R4 to R5                               ;  advance feedforward
         MOVE R5<-R4
*        Move R10 to R4                              ;  advance feedforward
         MOVE R4<-R10
*        Move R11 to R7                              ; feed it back
         MOVE R7<-R11
*        Return                                      ; done
         RTS
         NOP
         END
```

```
#define SAMPLESPERSEC 44100L
#define SECONDS 1
#define BASEFREQ 60

#define MAXALPHACOUNT 5
#define MAXALPHA 127
#define MAXBETA 127
#define MAXDELTA 25
#define MAXVOLUME 255

/* MAXREVERBTIME must be a power of 2, minus 1! */

#define MAXREVERBTIME 2047
#define MAXREVERBLEVEL 100

/*
    SAMPLESPERUNIT defines the number of sound samples per measuring unit
    (currently 14 samples per decimeter unit... this misestimates the speed
    of sound by about 4%).
*/

#define SAMPLESPERUNIT 14

/*
    A sound source located within NEARBY units of the observer will be heard at
    full volume.  If further away than NEARBY, the volume will be reduced in
    proportion to the distance.  Minimum volume is set by VERYQUIET.
*/

#define NEARBY 100
#define VERYQUIET 2

/*
    A sound source located within NEARBY units of the observer will have a
    reverberation level of MINREVERB.  If further away than NEARBY, the
    reverb level will be increased to a maximum of MAXREVERBLEVEL, with an
    increase of one reverb unit per REVERBSTEP units of distance.  Reverb time
    constants are set by REVERBTC1 and REVERBTC2... prime numbers which correspond
    roughly to 50 and 40 milliseconds.
*/

#define MINREVERB 20
#define REVERBSTEP 10
#define REVERBTC1 2039
#define REVERBTC2 1777

/*
    intBits and fractBits define the portions of a 32-bit word to be used
    for a fixed-point number.  The integer portion must be long enough to hold
    a signed number twice the magnitude of MAXDELTA.  fractMask must correspond
    to the rightmost fractBits in the word.
*/
```

```
/*
   Utility.c - utility functions
*/

#include <math.h>
#include "CSounderParms.h"
#include "Utility.h"

#define PI              3.14159265

/*
   Alpha cuts are available from 0 dB (flat) to 20 dB.
*/

int alphaTable[NUMALPHAVALUES] = {0, 15, 27, 38, 48, 56, 64, 72, 77, 82, 88, 92, 96,
                                  102, 103, 105, 107, 110, 112, 113, 115};

/*
   Beta cuts are available from 0 dB (flat) to 9 dB.
*/

int betaTable[NUMBETAVALUES] = {0, 15, 30, 46, 59, 73, 87, 102, 112, 127};

/*
   Scrambler table (gray-code)
*/

int scrambleTable[NUMSCRAMBLES] = {0, 1, 3, 2 /* 6, 7, 5, 4,
                                   12, 13, 15, 14, 10, 11, 9, 8 */};

Point origin = {0,0};

int PtDist(Point p1, Point p2)
{
    long int d1, d2;
    d1 = p1.h - p2.h;
    d1 *= d1;
    d2 = p1.v - p2.v;
    d2 *= d2;
    return (int) (.5 + sqrt(d1 + d2));
}

int AbsAngle(Point p)
{
/*
   note that the point P is being represented with the Y axis reversed from
   the QuickDraw coordinate system... v > 0 is up, not down!
*/
    Fixed slope;
    int angle;
    if (p.h == 0) {
        if (p.v > 0) {
            return 0;
```

```
        } else {
            return 180;
        }
    }
    slope = FixRatio(p.h, -p.v);
    angle = AngleFromSlope(slope);
    if (p.h < 0) {
        return angle - 180;
    } else {
        return angle;
    }
}

Boolean SharpTurn(int angle1, int angle2, int criticalAngle)
{
    int deltaAngle;
    deltaAngle = angle1 - angle2;
    if (deltaAngle < 0) {
        deltaAngle = -deltaAngle;
    }
    if (deltaAngle < criticalAngle) {
        return FALSE;
    }
    return TRUE;
}

int AzimuthDelay(int theAngle)
{
    double realAngle;
    if (theAngle == 0 || theAngle == 180) {
        return 0;
    } else if (theAngle > 90) {
        theAngle = 180 - theAngle;
    } else if (theAngle < -90) {
        theAngle = -180 - theAngle;
    }
    realAngle = (theAngle * PI) / 180;
    return 123 * (realAngle + sin(realAngle)) /  22.675736961451; /* chuckle */
}

long int TimeToSamples(Fixed theTime)
{
    long int integerPart;
    unsigned int fractionalPart;
    integerPart = HiWord(theTime);
    fractionalPart = LoWord(theTime);
    return integerPart * SAMPLESPERSEC +
            ((unsigned long) fractionalPart) * SAMPLESPERSEC / 65536;
}
```

```
static int Lookup(int theTable[], int tableSize, double theValue)
{
    int theIndex, below, above, invert;
    double partial;
    if (theValue < 0.0) {
        invert = -1;
        theValue = -theValue;
    } else {
        invert = 1;
    }
    if (theValue > tableSize-1) {
        return invert * theTable[tableSize-1];
    }
    theIndex = theValue;
    below = theTable[theIndex];
    above = theTable[theIndex+1];
    partial = theValue - theIndex;
    return invert * (below + partial * (above - below));
}

void CSounderDoc::Sweep(CSoundPoint *startPoint, CSoundPoint *endPoint,
                        DistantSound *leftEar,
                        DistantSound *rightEar,
                        TimeCheck *now)
{
    long int dStart, dEnd;
    long int tStart, tEnd;
    long int hearStart, hearEnd;
    long int lagStart, lagEnd;
    long int deltaT, deltaH, runRate;
    static long int safePositive = 0x3FFFFFFF, safeNegative = 0xC0000000;
    int shiftCount;
    int startAngle, endAngle, deltaAngle, startDelta, endDelta;
    long int earDelta;
    long int earDiff, leftEarDiff, rightEarDiff;
    Fixed avgDistance;
    double avgAngle, rearRolloff, distanceRolloff,  notchCut, volumeDown;
    int volume;
    long int reverbLevel;
    dStart = PtDist(startPoint->where, origin);
    dEnd = PtDist(endPoint->where, origin);
    if (doppler) {
        lagStart = dStart * SAMPLESPERUNIT;
        lagEnd = dEnd * SAMPLESPERUNIT;
    } else {
        lagStart = lagEnd = 0;
    }
    tStart = TimeToSamples(startPoint->when);
    tEnd = TimeToSamples(endPoint->when);
    leftEar->startSound = rightEar->startSound = tStart;
    leftEar->finishSound = rightEar->finishSound = tEnd;
    if (now) {
        hearStart = now->time;
```

```
} else {
    hearStart = tStart + lagStart;
}
hearEnd = tEnd + lagEnd;
startAngle = AbsAngle(startPoint->where);
endAngle = AbsAngle(endPoint->where);
deltaAngle = startAngle - endAngle;
avgAngle = ((startAngle + endAngle) / 2) * PI / 180;
if (deltaAngle > 180 || deltaAngle < -180) {
    if (avgAngle > 0) {
        avgAngle = -PI + avgAngle;
    } else {
        avgAngle = PI + avgAngle;
    }
}
if (timePan) {
    if (now) {
        startDelta = (now->rightEarClock - now->leftEarClock) / 2;
    } else {
        startDelta = AzimuthDelay(startAngle);
    }
    endDelta = AzimuthDelay(endAngle);
} else {
    startDelta = endDelta = 0;
}
leftEar->startToHear = hearStart + startDelta;
leftEar->finishHearing = hearEnd + endDelta;
rightEar->startToHear = hearStart - startDelta;
rightEar->finishHearing = hearEnd - endDelta;
deltaT = tEnd - tStart;
deltaH = hearEnd - hearStart;
if (deltaT == deltaH) {
    runRate = 1L << fractBits;
} else {
    shiftCount = fractBits;
    while (shiftCount > 0 && (deltaT <= safePositive && deltaT >= safeNegative)) {
        deltaT = deltaT << 1;
        shiftCount --;
    }
    runRate = deltaT / deltaH;
    runRate = runRate << shiftCount;
}
earDelta = endDelta - startDelta;
if (deltaH == 0) {
    earDiff = 0;
} else {
    earDiff = earDelta << fractBits;
    earDiff /= deltaH;
}
leftEar->clockRate = runRate - earDiff;
rightEar->clockRate = runRate + earDiff;
if (earDiff > 0) {
    leftEar->clockStart = 1L << (fractBits - 1);
```

```
        rightEar->clockStart = 0;
    } else {
        leftEar->clockStart = 0;
        rightEar->clockStart = 1L << (fractBits - 1);
    }
    avgDistance = dStart / 2 + dEnd / 2;
    if (avgDistance <= NEARBY) {
        volume = 256;
        reverbLevel = MINREVERB;
    } else {
        volume = (256L * NEARBY) / avgDistance;
        if (volume < VERYQUIET) {
            volume = VERYQUIET;
        }
        reverbLevel = MINREVERB + (avgDistance - NEARBY) / REVERBSTEP;
        if (reverbLevel > MAXREVERBLEVEL) {
            reverbLevel = MAXREVERBLEVEL;
        }
    }
    if (!distance) {
        volume = 256;
    }
    if (volumePan) {
        volumeDown = sin(fabs(avgAngle));
        if (avgAngle > 0) {
            leftEar->volume = volume * (1.0 - .9 * volumeDown);
            rightEar->volume = volume;
        } else {
            leftEar->volume = volume;
            rightEar->volume = volume * (1.0 - .9 * volumeDown);
        }
    } else {
        leftEar->volume = rightEar->volume = volume;
    }
    if (reverb) {
        rightEar->reverbLevel = reverbLevel;
        leftEar->reverbLevel = reverbLevel - reverbLevel / 20; /* 5% reduction */
        leftEar->reverbTime = REVERBTC1;
        rightEar->reverbTime = REVERBTC2;
    } else {
        leftEar->reverbLevel = rightEar->reverbLevel = 0;
        leftEar->reverbTime = rightEar->reverbTime = 0;
    }
    if (freqShape) {
        distanceRollcff = (avgDistance - NEARBY) / 100.0;        /* .1 dB/meter */
        if (avgAngle > 0) {
            if (avgAngle > PI/2) {
                rearRolloff = 10 * sin(avgAngle - PI/2);
            } else {
                rearRolloff = 0;
            }
            leftEar->hfRolloff = 10 * sin(avgAngle) + rearRolloff + distanceRolloff;
            rightEar->hfRolloff = rearRolloff + distanceRolloff;
```

```
        } else {
            if (avgAngle < -PI/2) {
                rearRolloff = 10 * sin(-PI/2 - avgAngle);
            } else {
                rearRolloff = 0;
            }
            leftEar->hfRolloff = rearRolloff + distanceRolloff;
            rightEar->hfRolloff = 10 * sin(-avgAngle) + rearRolloff + distanceRolloff;
        }
        leftEar->notch = leftEar->hfRolloff / 2;
        rightEar->notch = rightEar->hfRolloff / 2;
        if (avgAngle >= -PI/2 && avgAngle <= PI/2) {
            notchCut = 5 * cos(avgAngle);
        } else {
            notchCut = -5 * cos(avgAngle);
        }
        leftEar->notch += notchCut - 2.5;
        rightEar->notch += notchCut - 2.5;
        leftEar->alphaLevel = Lookup(alphaTable, NUMALPHAVALUES, leftEar->notch);
        rightEar->alphaLevel = Lookup(alphaTable, NUMALPHAVALUES, rightEar->notch);
        leftEar->beta = Lookup(betaTable, NUMBETAVALUES, leftEar->hfRolloff);
        rightEar->beta = Lookup(betaTable, NUMBETAVALUES, rightEar->hfRolloff);
    } else {
        leftEar->alphaLevel = leftEar->beta = 0;
        rightEar->alphaLevel = rightEar->beta = 0;
    }
    leftEar->alphaCount = rightEar->alphaCount = 3;

}


OSErr CSounderDoc::SendCommand(SndChannelPtr theChannel, int cmd,
                                int param1, long int param2)
{
    SndCommand theCommand;
    theCommand.cmd = cmd;
    theCommand.param1 = param1;
    theCommand.param2 = param2;
    return (SndDoCommand(theChannel, &theCommand, FALSE));
}

void CSounderDoc::CleanTimeline()
{
    int         n;
    CSoundPoint *pointN;
    n = 1;
    while (n <= timeline->GetNumItems()) {
        pointN = (CSoundPoint *) timeline->NthItem(n);
        if (pointN->GetType() == intermediatePoint) {
            timeline->Remove(pointN);
            pointN->Dispose();
        } else {
            n++;
```

```
            }
        }
    }

    void CSounderDoc::SliceTimeline()
    {
        int          n;
        CSoundPoint *pointN, *pointNpl, *pointNew;
        Point        pN, pNpl, pMid;
        int          angleN, angleNpl, angleMid;
        int          dN, dNpl, dMid;
        int          dNtoMid, dMidtoNpl, dTotal;
        Fixed        criticalDistanceRatio;
        int          criticalAngle;

        criticalDistanceRatio = FixRatio(5, 100); /* change of 5% triggers cut */
        criticalAngle = 5; /* change of 5 degrees triggers cut */

        CleanTimeline();

        n = 1;
        while (n < timeline->GetNumItems()) {
            pointN = (CSoundPoint *) timeline->NthItem(n);
            pointNpl = (CSoundPoint *) timeline->NthItem(n+1);
            pN = pointN->GetWhere();
            pNpl = pointNpl->GetWhere();
            pMid.h = (pN.h + pNpl.h) / 2;
            pMid.v = (pN.v + pNpl.v) / 2;
            dTotal = PtDist(pN, pNpl);
            dN = PtDist(pN, origin);
            dNpl = PtDist(pNpl, origin);
            dMid = PtDist(pMid, origin);
            dNtoMid = dN - dMid;
            if (dNtoMid < 0) {
                dNtoMid = -dNtoMid;
            }
            dMidtoNpl = dMid - dNpl;
            if (dMidtoNpl < 0) {
                dMidtoNpl = -dMidtoNpl;
            }
            angleN = AbsAngle(pN);
            angleNpl = AbsAngle(pNpl);
            angleMid = AbsAngle(pMid);
            if (dTotal >= 10 && (SharpTurn(angleN, angleMid, criticalAngle) ||
              SharpTurn(angleMid, angleNpl, criticalAngle) ||
              FixRatio(dNtoMid, dMid) >= criticalDistanceRatio ||
              FixRatio(dMidtoNpl, dMid) >= criticalDistanceRatio)) {
                pointNew = new(CSoundPoint);
                pointNew->ISoundPoint();
                pointNew->SetType(intermediatePoint);
                pointNew->SetWhere(pMid);
                pointNew->SetWhen((pointN->GetWhen() + pointNpl->GetWhen()) / 2);
                timeline->InsertAfter(pointNew, pointN);

            } else {
                n++;
            }
        }
    }
```

CLAIMS

What is claimed is:

1. A method of generating sound that would be associated with an object moving respectively to the listener comprising the steps:

a) generating a ratio between the length of time that said object would generate such a sound and the length of time that the listener would hear said sound; and

b) generating a series of digital sound samples as a function of said ratio for the period of time that said listener would have heard said sound.

2. The method of Claim 1 wherein said ratio is a digital value having an integer portion and a fraction portion.

3. The method of Claim 2 wherein each one of said series of generated sound samples is generated by the steps of:

c) forming a summation ratio by combining said ratio and the fractional portion of the summation ratio generated for the immediately preceding sound sample of said series of sound samples; and

d) generating a digital sound sample having a value which is a function of said summation ratio for said sound sample being generated.

4. The method of Claim 3 further comprising the steps of:

e) providing a plurality of digital monaural sound samples representing the sound of said object when said object is at a constant distance from the listener.

5. The method of Claim 4 wherein the step of generating each digital sound sample further comprises the steps of:

f) selecting two of said provided digital monaural sound samples as a function of said integer portion of said resulting ratio for the sound sample to be generated; and

g) interpolating between the values of said two selected digital sound samples as a function of said fractional portion of said resulting ratio for the sample to be generated, the resulting value of said interpolation being the value of the digital sound sample being generated.

6. A method for generating three dimensional binaural sound that a listener would hear from an object generating that sound where said object is moving with respect to the listener comprising the steps of:

a) storing a plurality of digital monaural sound samples associated with said object, said monaural samples having been sampled at a sample rate;

b) storing a segment which comprises data for describing the relative movement of said object to said listener in both space and time for said segment;

f) generating from said segment data a first ratio between the length of time that said object would generate sound and the length of time that said generated sound would be heard by the listener's right ear, said first ratio comprising an integer and fraction portion;

g) generating a series of digital sound samples for the length of time said generated sound would be heard by the listener's right ear as a function of said first ratio;

h) generating from said segment data a second ratio between the length of time that said object generates sound and the length of time that said generated sound is heard by the listener's left ear, second ratio comprising an integer and fraction portion;

i) generating a series of digital second sound samples for the length of time said generated sound would be heard by the listener's left ear as a function of said second ratio.

7. The method of Claim 6 wherein the steps of generating each one of said series of first sound samples comprises the step of:

j) forming a first summation ratio by combining said ratio and the fractional portion of the first summation ratio generated for the immediately preceding sound sample of said series of said sound samples; and

wherein the step of generating each one of said series of second sound samples comprises the step of:

k) forming a second summation ratio by combining said ratio and the fractional portion of the second summation ratio generated for the immediately preceding sound sample of said series of said sound samples.

8. The method of Claim 7 comprising the further steps of:

c) storing segment criteria;

d) determining if said segment meets the requirements of said segment criteria;

e) dividing said segment into subsets of said segment where each said subset meets the requirement of said segment criteria if said segment did not meet the requirement of said segment criteria.

9. The method of Claim 8 wherein said segment criteria of step d is:

if 1) $|d_1 - d_m| < .05\, d_m$ and

2) $|d_2 - d_m| < .05\, d_m$ and

3) $1 < 5°$ and

4) $2 < 5°$ are all met

or if:

$d_m < 10$ units

then said segment meets requirement otherwise divide said segment being processed into subsegment.

10. The method of Claim 6 wherein:

said first ratio (R_R) of step f is generated in accordance with the mathematical formula

$$R_R = \frac{\Delta T}{\Delta t - t_h(\sin\phi_1 - \sin\phi_2)}; \text{ and}$$

said second ratio (R_L) of step h is generated in accordance with the mathematical formula

$$R_L = \frac{\Delta T}{\Delta t + t_h\,(\sin\phi_1 - \sin\phi_2)}.$$

11. The method of Claim 7 wherein:

said first ratio (R_R) of step f is generated in accordance with the mathematical formula

$$R_R = \frac{\Delta T}{\Delta t - t_h(\sin\phi_1 - \sin\phi_2)}\text{; and}$$

said second ratio (R_L) of step h is generated in accordance with the mathematical formula

$$R_L = \frac{\Delta T}{\Delta t + t_h(\sin\phi_1 - \sin\phi_2)}.$$

12. The method of Claim 7 wherein:

said first ratio (R_R) of step f is generated in accordance with the mathematical formula

$$R_R = \frac{\Delta T + t_h(\sin\phi_1 - \sin\phi_2)}{\Delta t}\text{; and}$$

said second ratio (R_L) of step h is generated in accordance with the mathematical formula

$$R_L = \frac{\Delta T - t_h(\sin\phi_1 - \sin\phi_2)}{\Delta t}.$$

13. The method of Claim 7 wherein step j comprises the steps of:

j1) sequentially fetching said monaural sound samples from said storage where the number of said monaural sound samples fetched is a function of said present first summation ratio;

j2) storing said fetched monaural sound samples;

j3) interpolating between the values of the last two stored monaural sound samples, the interpolation factor for said interpolation being a function of said fractional portion of said first summation ratio, for generating said first sound sample; and

wherein step k comprises the steps of:

k1) sequentially fetching said monaural sound samples from said storage where the number of said monaural sound samples fetched is a function of said integer portion of said second summation ratio;

k2) storing said fetched monaural sound samples;

k3) interpolating between the values of the last two stored monaural sound samples, the interpolation factor for said interpolation being a function of said fractional portion of said second summation ratio, for generating said second sound sample; and

said first and second sound samples being generated at the same rate as said sample rate for said monaural sound samples.

14. The method of Claim 7 comprising the additional steps of:

l) receiving and storing reverberation data;

m) generating a first reverberation signal and a second reverberation signal as a function of said reverberation data;

n) adding said first reverberation signal to said first sound sample to form a first reverberized sound sample; and

o) adding said second reverberation signal to said second sound sample to form a second reverberized sound sample.

15. The method of Claim 13 comprising the additional steps of:

l) receiving and storing reverberation data;

m) generating a first reverberation signal and a second reverberation signal as a function of said reverberation data;

n) adding said first digital reverberation signal to said first digital sound sample to form a first reverberized sound sample; and

o) adding said second reverberation signal to said second sound sample to form a second reverberized sound sample.

16. The method of Claim 7 comprising the additional steps of:

p) generating a set of first control values for a first digital notch filter and a first digital low pass filter as a function of said segment;

q) setting said first digital notch filter and said first digital low pass filter by said first set of control values;

r) filtering said first sound sample by said first digital notch filter and said first digital low pass filter for forming a first filtered sound sample;

s) generating a set of second controlled values for a second digital notch filter and a second digital low pass filter as a function of said segment;

t) setting said second digital notch filter and said second digital low pass filter by said second set of control values;

u) filtering said second digital sound sample by said second digital notch filter and said second digital low pass filter for forming a second filtered sound sample.

17. The method of Claim 13 comprising the additional steps of:

p) generating a set of first control values for a first digital notch filter and a first digital low pass filter as a function of said segment;

q) setting said first digital notch filter and said first digital low pass filter by said first set of control values;

r) filtering said first sound sample by said first digital notch filter and said first digital low pass filter for forming a first filtered sound sample;

s) generating a set of second controlled values for a second digital notch filter and a second digital low pass filter as a function of said segment;

t) setting said second digital notch filter and said second digital low pass filter by said second set of control values;

u) filtering said second sound sample by said second digital notch filter and said second digital low pass filter for forming a second filtered sound sample.

18. The method of Claim 15 comprising the additional steps of:

p) generating a set of first control values for a first digital notch filter and a first digital low pass filter as a function of said segment;

q) setting said first digital notch filter and said first digital low pass filter by said first set of control values;

r) filtering said first reverberized sound sample by said first digital notch filter and said first digital low pass filter for forming a first filtered sound sample;

s) generating a set of second control values for a second digital notch filter and a second digital low pass filter as a function of said segment;

t) setting said second digital notch filter and said second digital low pass filter by said second set of control values;

u) filtering said second reverberized sound sample by said second digital notch filter and said second digital low pass filter for forming a second filtered sound sample.

19. The method of Claim 16 comprising the additional steps of:

v) generating a volume control value as a first volume multiplier and a second volume multiplier as a function of said segment;

w) multiplying said first sound sample by said first volume multiplier for forming a first volume adjust sound sample;

x) multiplying said second sound sample by said second volume multiplier for forming a second volume adjust sound sample;

y) converting said first volume adjust sound sample into a first analog signal for the right ear of said listener; and

z) converting said second volume adjusted sound samples into a second analog signal for the left ear of said listener.

20. The method of Claim 17 comprising the additional steps of:

v) generating a volume control value as a first volume multiplier and a second volume multiplier as a function of said segment;

w) multiplying said sound sample by said first volume multiplier for forming a first volume adjust sound sample;

x) multiplying said second sound sample by said second volume multiplier for forming a second volume adjust sound sample;

y) converting said first volume adjust sound sample into a first analog signal for the right ear of said listener; and

z) converting said second volume adjusted sound sample into a second analog signal for the left ear of said listener.

21. The method of Claim 18 comprising the additional steps of:

v) generating a volume control value as a first volume multiplier and a second volume multiplier as a function of said input data;

w) multiplying said first reverberized sound sample by said first volume multiplier for forming a first volume adjust sound sample;

x) multiplying said second reverberized sound sample by said second volume multiplier for forming a second volume adjust sound sample;

y) converting said first volume adjust sound sample into a first analog signal for the right ear of said listener; and

z) converting said second volume adjusted sound samples into a second analog signal for the left ear of said listener.

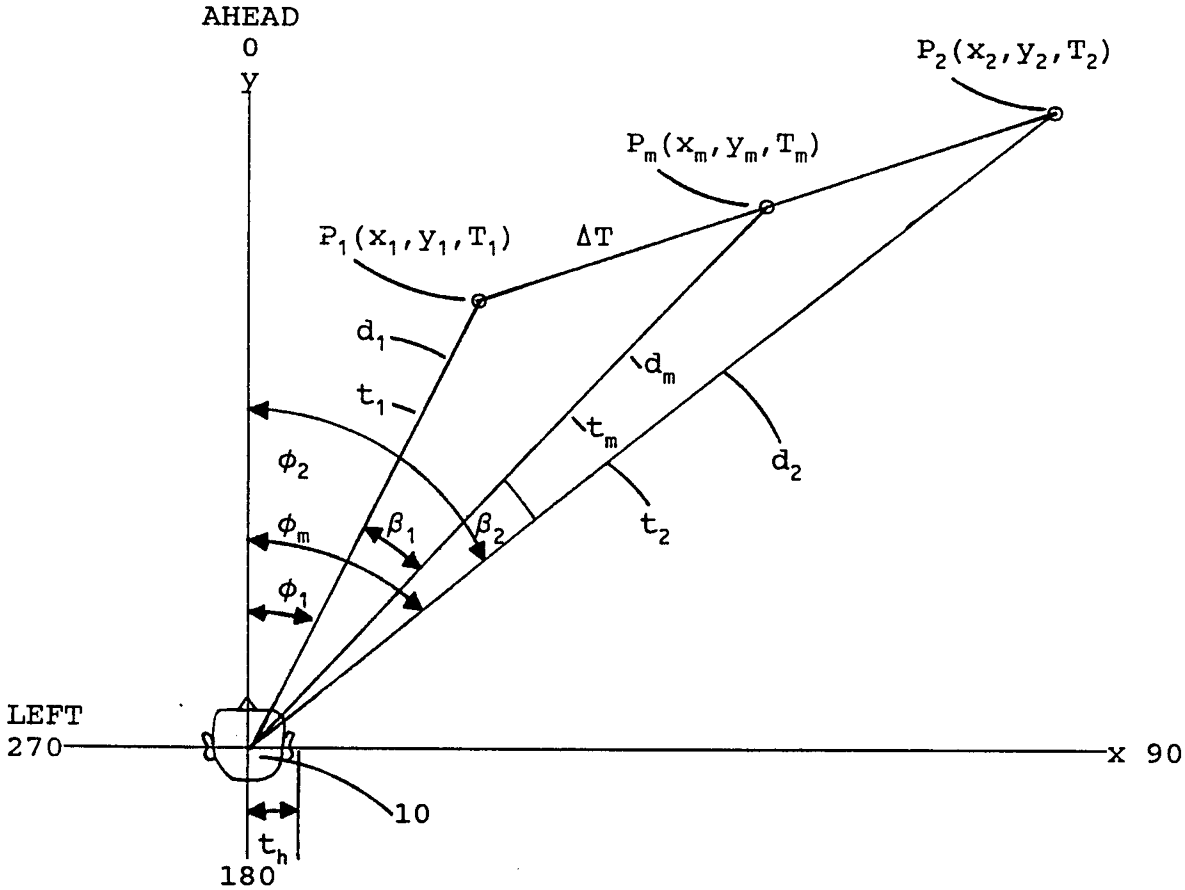

AHEAD
0
y
$P_2(x_2, y_2, T_2)$
$P_m(x_m, y_m, T_m)$
$P_1(x_1, y_1, T_1)$
ΔT
d_1
d_m
t_1
t_m
d_2
ϕ_2
β_1
β_2
t_2
ϕ_m
ϕ_1
LEFT
270
x 90
10
t_h
180

FIG. 1

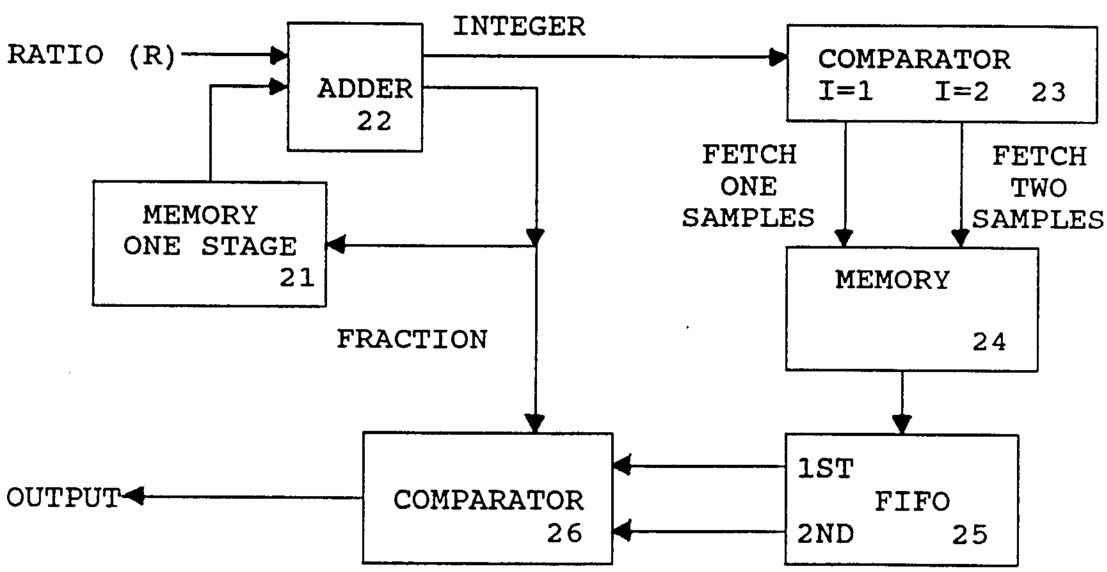

RATIO (R)
ADDER
22
INTEGER
COMPARATOR
I=1 I=2 23
MEMORY
ONE STAGE
21
FETCH
ONE
SAMPLES
FETCH
TWO
SAMPLES
MEMORY
24
FRACTION
OUTPUT
COMPARATOR
26
1ST
FIFO
2ND 25

FIG. 2

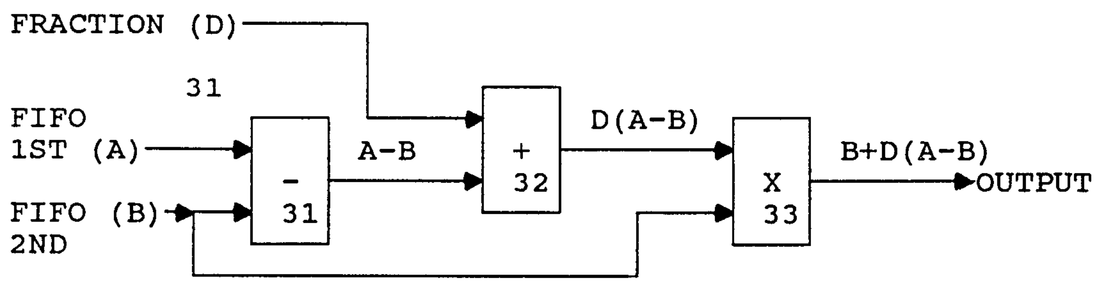


FIG. 3

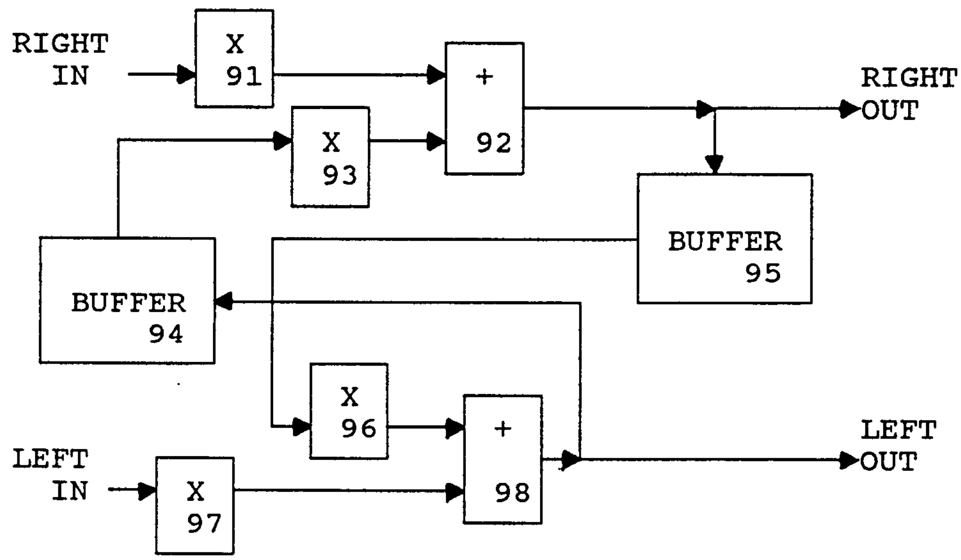


FIG. 9

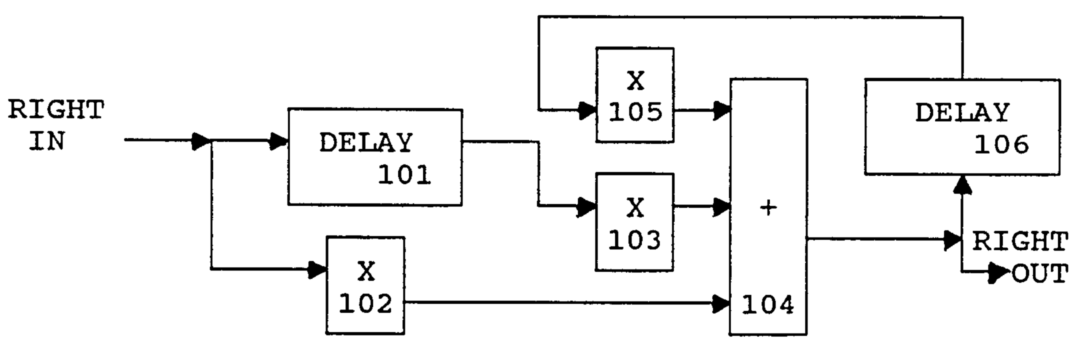


FIG. 10

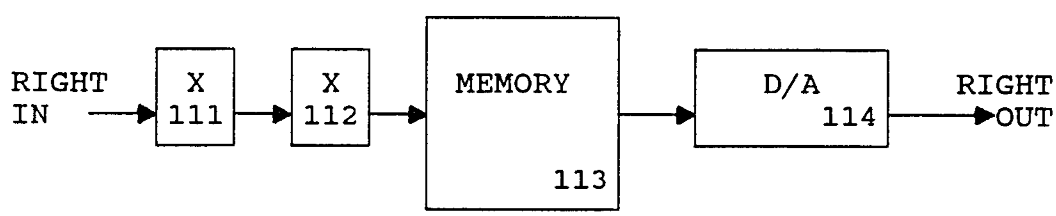


FIG. 11

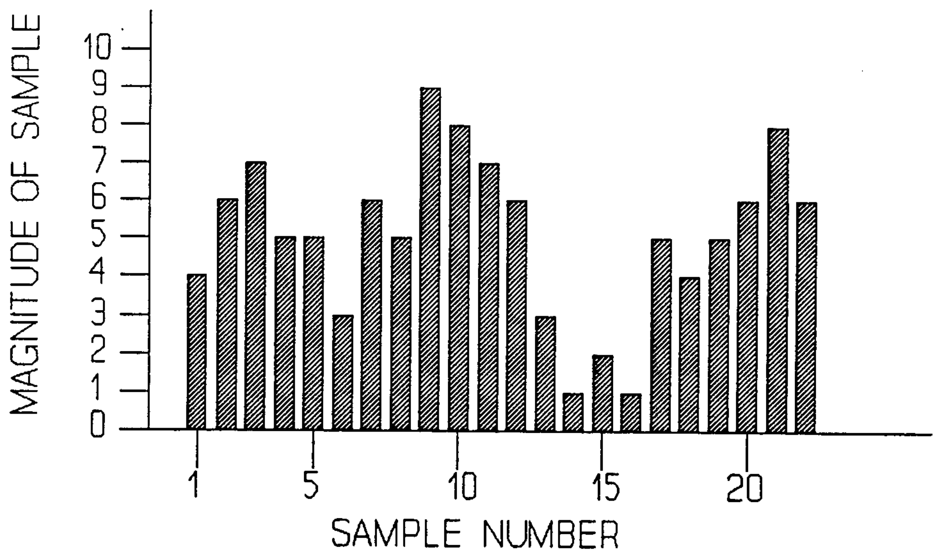


FIG. 4

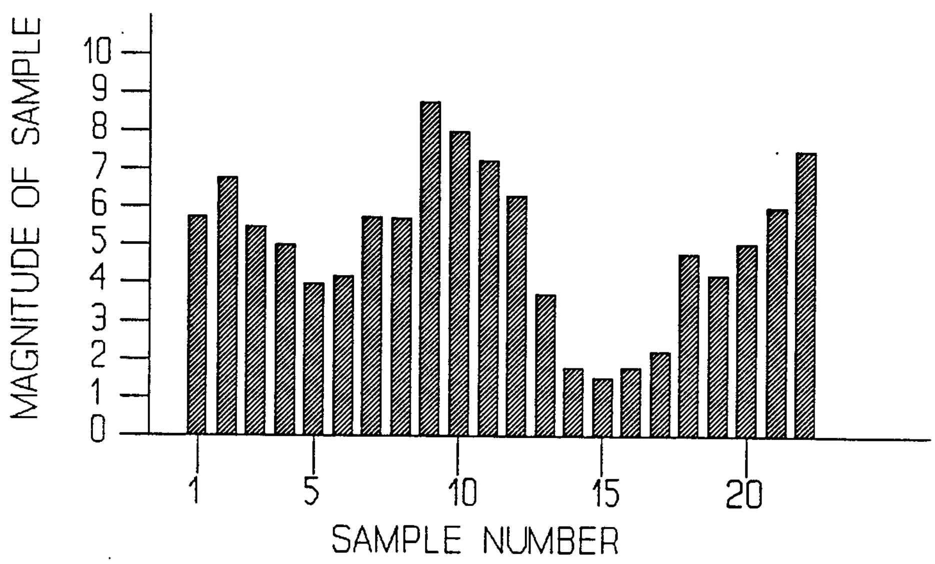


FIG. 5

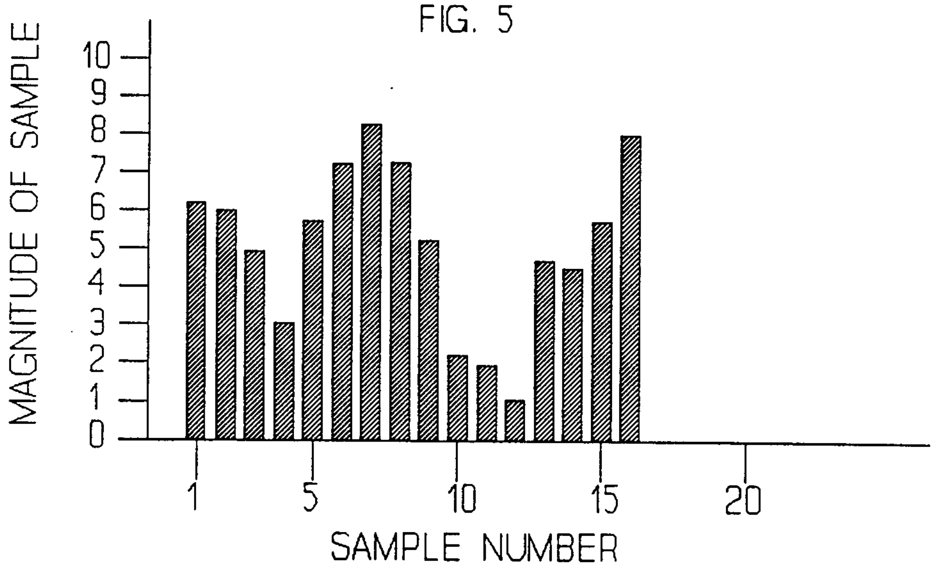


FIG. 6

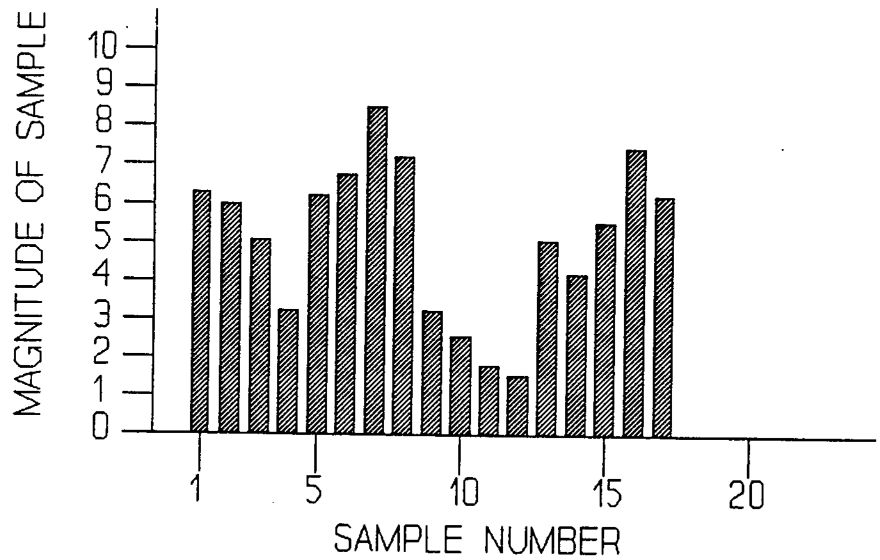

MAGNITUDE OF SAMPLE
0
1
2
3
4
5
6
7
8
9
10
1
5
10
15
20
SAMPLE NUMBER

FIG. 7

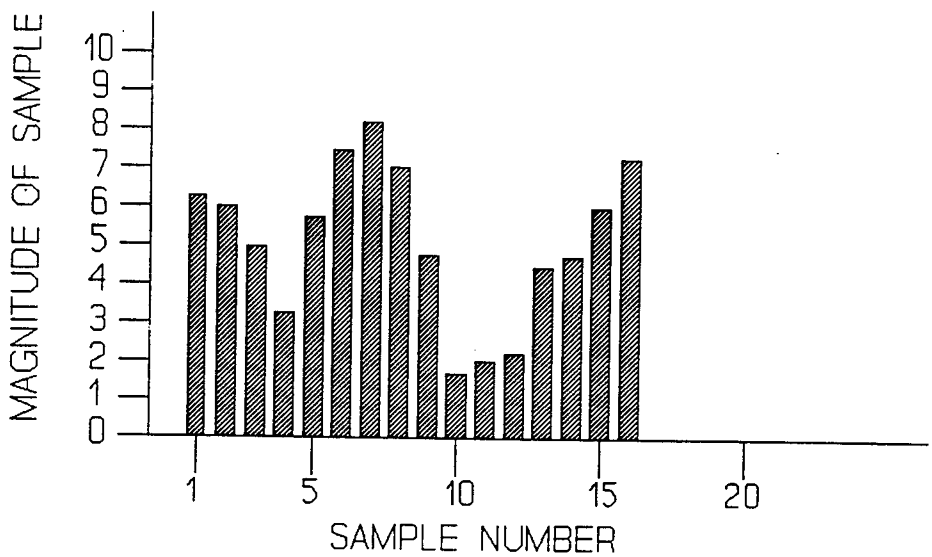
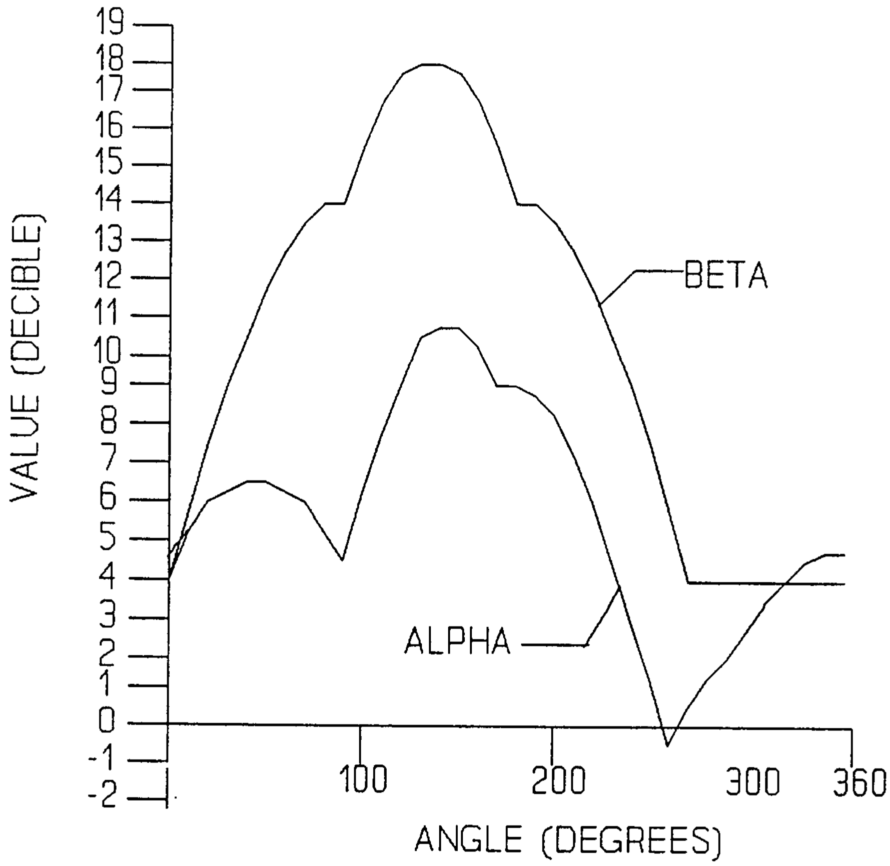

MAGNITUDE OF SAMPLE
0
1
2
3
4
5
6
7
8
9
10
1
5
10
15
20
SAMPLE NUMBER

FIG. 8


VALUE (DECIBLE)
19
18
17
16
15
14
13
12
11
10
9
8
7
6
5
4
3
2
1
0
-1
-2
BETA
ALPHA
100
200
300
360
ANGLE (DEGREES)

FRG. 12

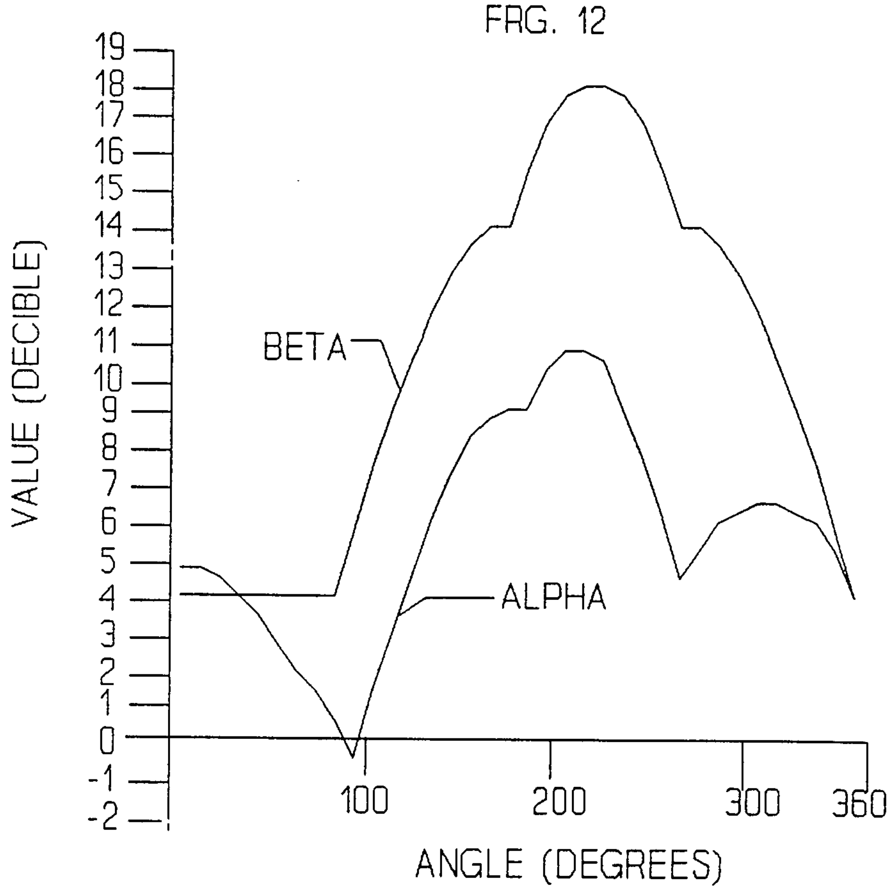

VALUE (DECIBLE)
19
18
17
16
15
14
13
12
11
10
9
8
7
6
5
4
3
2
1
0
-1
-2
BETA
ALPHA
100
200
300
360
ANGLE (DEGREES)

FIG. 13

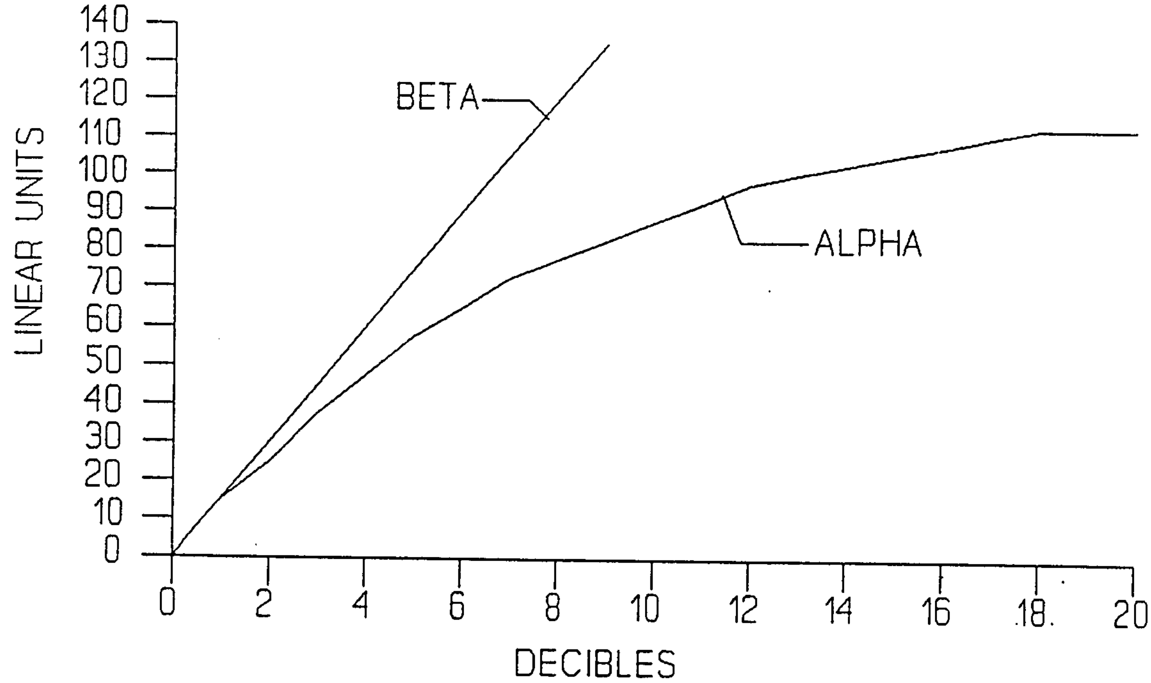

140
130
120
110
100
90
80
70
60
50
40
30
20
10
0
LINEAR UNITS
BETA
ALPHA
0
2
4
6
8
10
12
14
16
18
20
DECIBLES

FIG. 14

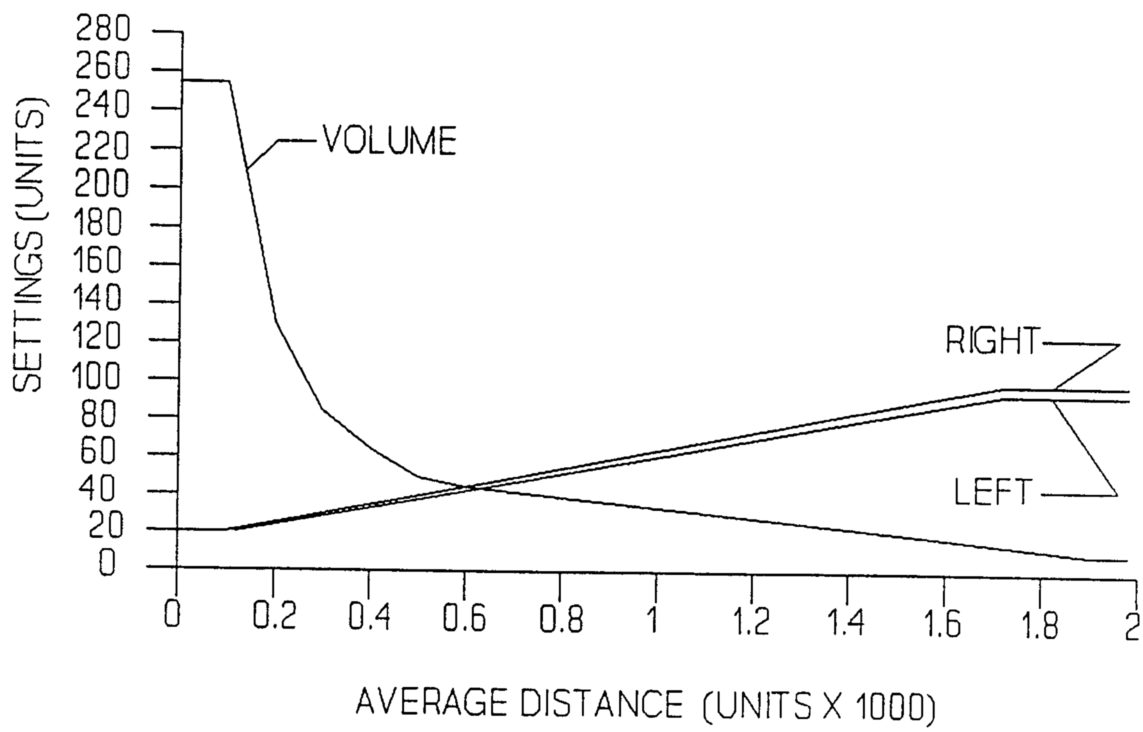

280
260
240
220
200
180
160
140
120
100
80
60
40
20
0
SETTINGS (UNITS)
VOLUME
RIGHT
LEFT
0
0.2
0.4
0.6
0.8
1
1.2
1.4
1.6
1.8
2
AVERAGE DISTANCE (UNITS X 1000)

FIG. 15

INTERNATIONAL SEARCH REPORT

PCT/US92/09348

A. CLASSIFICATION OF SUBJECT MATTER

IPC(5) :H04R 5/00 G06K 9/00
US CL :381/17,63
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 381/17,63 381/1,18,61

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

(Three Dimensional(5A)Sound) and (Mov?(5A)Listener?)
USPTO APS

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	JP,A, 62-140600 (Obata) 24 June 1987 Fig. 3.	19,20,21
A	US,A, 5,046,097 (Lowe et al.) 03 September 1991 Figs. 1,3 and 8, Col. 8, line 32-42, Col. 12, line 54, Col. 13, line 11, Col. 17, line 44-52, Col. 20, line 62-68.	14-21
A	US,A, 4,817,149 (Myers) 28 March 1989 Figs. 1,6,7,13,14 and 25, Col. 6, lines 27-45, Col. 7, line 19-25, Col. 15, line 59-68, Col. 8, line 59-66.	14-21
A	US,A, 4,731,848 (Kendall et al.) 15 March 1988 Fig. 2, Col. 7, line 16-35.	14,15

[X] Further documents are listed in the continuation of Box C. [ ] See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be part of particular relevance

"E" earlier document published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
01 JANUARY 1993	05 FEB 1993

Name and mailing address of the ISA/US	Authorized officer
Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231	PING WONG
Facsimile No. NOT APPLICABLE	Telephone No. (703) 305-4865

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US92/09348

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	JP,A, 53-137101 (Mori) 30 November 1978 Fig. 1 and constitution.	14-21
A	JP,A, 58-68400 (Kikuchi) 23 April 1983 constitution.	14-21

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US92/09348

Box I Observations where certain claims were found unsearchable (Continuation of item 1 of first sheet)

This international report has not been established in respect of certain claims under Article 17(2)(a) for the following reasons:

1. [X] Claims Nos.: 1-13
because they relate to subject matter not required to be searched by this Authority, namely:

Mathematical algorithm for data gathering.

2. [ ] Claims Nos.:
because they relate to parts of the international application that do not comply with the prescribed requirements to such an extent that no meaningful international search can be carried out, specifically:

3. [ ] Claims Nos.:
because they are dependent claims and are not drafted in accordance with the second and third sentences of Rule 6.4(a).

Box II Observations where unity of invention is lacking (Continuation of item 2 of first sheet)

This International Searching Authority found multiple inventions in this international application, as follows:

1. [ ] As all required additional search fees were timely paid by the applicant, this international search report covers all searchable claims.

2. [ ] As all searchable claims could be searched without effort justifying an additional fee, this Authority did not invite payment of any additional fee.

3. [ ] As only some of the required additional search fees were timely paid by the applicant, this international search report covers only those claims for which fees were paid, specifically claims Nos.:

4. [ ] No required additional search fees were timely paid by the applicant. Consequently, this international search report is restricted to the invention first mentioned in the claims; it is covered by claims Nos.:

Remark on Protest

[ ] The additional search fees were accompanied by the applicant's protest.

[ ] No protest accompanied the payment of additional search fees.