

METHOD AND APPARATUS FOR SIMULATING A SOUND IN VIRTUAL SPACE TO
HAVE A LISTENER ENJOY ARTIFICIAL EXPERIENCE OF THE SOUND

BACKGROUND OF THE INVENTION

5 Field of the Invention:

The present invention relates to a method and apparatus for determining the acoustic characteristics of a sound in an arbitrary space to have a listener enjoy his acoustic experience of the sound of the arbitrary space in a real space.

10 More particularly, the present invention relates to a method and apparatus for reproducing the sound of the arbitrary space in the real space on the basis of the acoustic characteristics of the sound.

Description of the Prior Art:

15 Some types of methods for calculating the acoustic characteristics of a sound in an existing real space and also in a virtual space have been previously researched. For example, there are many conventional wide-use calculuses such as the sound-ray method, virtual-image method and the like based on the orthodox
20 geometrical acoustics. However, these calculuses are disadvantageous in that they ignore wave characteristics of a sound and phase information in the calculation results to cause a considerably large calculation error. On the other hand, there are some calculuses such as the finite element method, boundary
25 element method and like calculuses, which consider wave

characteristics of a sound and therefore are excellent in precision in determining the actual acoustic characteristics of the sound. However, these calculuses such as the finite element method and the boundary element method are also disadvantageous in that a very large volume of calculation is required to obtain the calculation result as to a so-called audiofrequency band with a high frequency such as 16 KHz and the like. These facts make it very difficult even for a super computer now under development to obtain the calculation result.

Thus, it is apparent that the need exists for a method and apparatus for determining with high accuracy the acoustic characteristics of a sound in a relatively short time, the acoustic characteristics having little error in comparison with real acoustic characteristics the sound.

The acoustic characteristics to be determined are defined as a set of sounds reaching a predetermined observation point, for example such as the audience seats, which sounds are derived from a basic sound based on a single-pulse transient signal with a sharp leading edge as is shown in the delta approximate function, the transient signal serving as the basic sound being produced at a predetermined position, for example, on a stage and performing three-dimensional propagation in a space and being reflected by walls defining the space to reach the predetermined observation point such as the audience seats. In general, the sounds in this set are called the transient response signals. Consequently, the

transient response signals represent a result of overlapping of the sounds which are derived from the transient signal and reach the same observation point with various time delays from various directions in the space. By the use of this transient response signals, it is possible to reproduce voices and sounds such as music and like general sounds. Consequently, it is possible for a listener to enjoy music discoursed on the stage as if he were in the audience seat in the space.

The acoustic characteristics of a sound, which are determined according to a first basic method of the present invention and will be described later in detail, are transient response signals capable of realizing great reality. Consequently, such transient response signals form a set of numberless pulses representing with high accuracy the modification of frequencies and delays in phases and the like. Therefore, as shown in the description below, when a sound is actually reproduced by the use of the transient response signals, a sound reproducing apparatus capable of high-speed processing is required.

In other words, in order to store such accurate transient response signals in a digital mode, it is necessary to digitally represent, in size, the response pulses of the transient response signals in each of so-called sampling time intervals, which pulses vary with respect to time and have been divided by a very short time interval (i.e., the sampling time interval, the reciprocal of which is the sampling frequency) in a period of from issuance of

the transient signal to disappearance of the transient response signals. Then, by the use of such transient response signals, sounds appearing in an actual musical performance and the like are reproduced. In this reproduction of the sounds, as shown in Fig.

5 9, a central processing unit 76 (hereinafter referred to as the CPU 76) for generating a discrete digital signal 86 is used. The CPU 76 is provided with: a memory portion for setting and storing the transient response signals 83 in which a plurality of the response pulses are digitally allocated on the time axis; and, a
10 sum-of-products calculator 84, in which the sounds appearing in the actual musical performance and being digitally represented by a discrete signal 80 are subjected in real time to a convolution transformation together with its transient response signals 83 thus set to generate the discrete digital signal 86.

15 Based on the thus stored transient response signals, sounds appearing in the musical performance are reproduced. For this purpose, it is necessary for the CPU 76 to continuously sum up all the data of the transient response signals in real time in each of the sampling time intervals. For example, in order to
20 reproduce a frequency of 20 KHz, it is necessary to perform a digital sampling of 48 KHz. In case that the reverberation time, i.e., the time taken for the transient response signals to die away to the point of being substantially neglected, is 1.3 seconds, the number of required samplings is 65,536. Consequently,
25 in order to reproduce the sounds, it is necessary to sum up 65,536

pieces of pulse data per 20.8 microseconds (which is the reciprocal of 48 K) and output its summed-up result. Such high-speed convolution transformation function is provided only in a very expensive CPU 76. Due to this high cost, it is not practical to use such expensive CPU 76.

Thus, it is apparent that the need exists for a simple method and apparatus for reproducing a sound without impairing its reality.

On the other hand, in these days of wide-use of video instruments enabling us to enjoy various movies and like pictures through video cassette players at our homes or in our own private rooms, in order to enhance our enjoyment, it is important to reproduce with reality the sounds accompanied with such pictures. For this purpose, television sets are provided with conventional loud speaker output terminals for the surround phonic sound system. Of these television sets, some ones are further provided with simulated sound reproduction systems such as the Dolby surround system and the like. However, in an actual movie theater, in order to have the audience enjoy a two- or three-dimensional sound field, a large number of loud speakers, for example, 10 or more speakers are disposed on the side walls of the theater or on appropriate positions in the vicinity of a movie screen of the theater. Consequently, when such sounds of the pictures are reproduced through the loud speaker terminals for the surround phonic sound system in the television set at our home, it is not

possible to enjoy a feeling of three-dimensional sound field that we can experience at the movie theater.

On the other hand, recently, the need for a so-called virtual reality such as virtual images and the like is increased. Since the virtual reality enables us to experience a feeling of a virtual environment as if we were in an actual environment, it is expected that the virtual reality is applied in various fields. However, up to this time, the virtual reality is considered essentially as to picture images, and, therefore considered little as to reproduction of sounds accompanied with such picture images. Thus, in development of the virtual reality, it is apparent that the need exists for a method of reproduction of sounds enabling us to feel as if we were actually in a virtual environment which varies with respect to time.

As for picture scenes, they vary with respect to time, and, therefore often require their sounds to vary also with respect to time. For example, there is a scene of a room, followed by a subsequent scene of a prairie. Or, there are a series of scenes in which the situation becomes gradually more serious. In these cases, up to this time, according to the situation in each of the scenes, reverberations are added or removed by a human feeling of an operator. Consequently, the sounds actually reproduced are not necessarily adequate to the scenes, and, therefore often give the audience a feeling of physical disorder, or often impair the effect of the virtual reality. In these days,

reality is more required than ever. Consequently, the need exists for reproduction of sounds which are adequate to a space of each of the movie scenes, and, therefore give reality to the space to a maximum. As described above, according to such space used in each
5 of the movie scenes, the sounds are automatically processed and accurately reproduced so as to be adequate ones appearing in the space. This may improve the sounds in amusement, and, therefore enable us to enjoy a sound field improved in reality, which is different from that obtained in the movie theater using the Dolby
10 surround phonic sound system.

As described above, it is apparent that the need exists: for a sound simulation method and apparatus for determining actual and effective acoustic characteristics of a sound; and, further for a method and apparatus for actually utilizing the results of
15 the above simulation to generate a more real sound.

SUMMARY OF THE INVENTION

It is an object of the present invention to provide a sound simulation method and apparatus for generating the acoustic characteristics of a sound with high accuracy in an arbitrarily
20 set three-dimensional space (hereinafter referred to as the virtual space) in a relatively short time by the use of a relatively low-speed processing unit.

It is another object of the present invention to provide a method and apparatus for generating a sound by using the above
25 acoustic characteristics of the sound appearing in the virtual

space to enable a listener to have an actual feeling as if he were in the virtual space.

It is further another object of the present invention to provide a method and apparatus for simply generating a sound by
5 using transient response signals (which are the above precise acoustic characteristics of the sound appearing in the virtual space), while essential acoustic characteristics of the sound in the space are kept.

It is a still further another object of the present
10 invention to provide a method and apparatus for generating a sound for enabling a user enjoying a movie and like pictures at his home or in his own private room to have an actual feeling as if he were the audience in a hall such as movie theaters and like halls.

It is another object of the present invention to provide a
15 method and apparatus for generating a sound for enabling the user to have an actual feeling as if he were in a space of each of the scenes of the movie and like pictures as the scenes vary with respect to time.

It is further another object of the present invention to
20 provide a method and apparatus for generating a sound adequate to a picture scene transmitted to the user's home or his private room through electric cables, optical fibers, or wireless communication.

According to a first aspect of the present invention, the
25 above objects of the present invention are accomplished by

providing:

In a sound simulation method for determining transient response signals which affect a predetermined observation point when a transient signal, which is issued from a sound source
5 disposed in an arbitrary position in an arbitrarily set virtual space, performs a wave motion propagation through the virtual space, the improvement wherein:

the transient signal is represented by a plurality of sound ray vectors radiating from the sound source in the virtual
10 space; and

the sound simulation method further comprises at least a step for determining a potential reaching the observation point from each of portions of a boundary defining the virtual space, in each of which portions each of the plurality of sound ray vectors
15 is reflected.

By the use of this sound simulation method of the present invention, it is possible to determine in a simple manner the acoustic characteristics of the sound in a relatively short time without performing a large volume of calculations increasing in a
20 manner of infinite series as is in a conventional method in which: sound waves are issued from a sound source and reflected by a boundary so that the acoustic characteristics thereof perform wave motions to radiate in all directions in a space; all of these acoustic characteristics are then determined, each of the thus
25 determined all of the acoustic characteristics being followed by

subsequent acoustic characteristics further reflected by a subsequent boundary; the subsequent acoustic characteristics are then determined; and so on.

5 The above method of the present invention is improved in efficiency of its processing as follows.

In the above method of the present invention:

as for each of the plurality of the sound ray vectors, each time the sound ray vector is incident on and reflected by a portion of the boundary defining the virtual space, a propagation
10 history data comprising data of the sound ray vector having been incident on and reflected by the portion of the boundary and the position data of such portion of the boundary is stored; and

based on the propagation history data and an area data of the boundary occupied by the incident/reflected sound ray vector
15 in the position of the portion of the boundary in which the sound ray vector is incident on and reflected by the boundary, a potential reaching the observation point from the portion of the boundary defining the virtual space is determined.

Further, in the above method of the present invention:

20 a plurality of memory portions are arranged in a time-series array;

of the memory portions, one, which corresponds to a time delay produced until the potential reaches the observation point, adds and stores the value of the potential to determine a
25 time-series transient response signal, which makes it possible to

digitally represent the transient response signal (which comprises response signals varying with respect to time) each time a sampling operation is conducted.

5 In order to have a loud speaker in an existing real space reproduce a sound with so reality as to have a listener feel as if he were in the virtual space, it is necessary to perform the following steps when the transient response signal are generated in the method of the present invention.

Namely, the steps are:

- 10 (a) a step for setting an arbitrary space which is a virtual space;
- (b) a step for setting a predetermined real space in which a plurality of loud speakers are disposed;
- (c) a step for superimposing a listening point disposed in a
15 real space over an observation point disposed in the virtual space;
- (d) a step for determining the potential affecting the observation point after the elapse of the time delay of the transient signal which is issued from the sound source in the
20 virtual space and propagates in the virtual space to produce the time delay thereof;
- (e) a step for dividing the potential into a plurality of pieces allocated to the plurality of the loud speakers arranged in the superimposed real space; and
- 25 (f) a step for determining a transient response signal of each

of the loud speakers by adding each of the pieces of the potential thus allocated to each of the loud speakers according to its time delay.

As described above, in the above method of the present invention, it is possible to reproduce the sound excellent in directivity and in depth by dividing the potential (which is reflected from the boundary defining a wall surface of the virtual space to reach the observation point) into the plurality of pieces each of which is allocated to each of the loud speakers arranged in the real space.

In most cases, the sound source used in the above method of the present invention is generally constructed of a plurality of sound sources. Consequently, in case that the plurality of sound sources are used in the method of the present invention, it is necessary to perform the above step for generating the transient response signal as to each of the plurality of sound sources. For example, in case that a plurality of loud speakers installed in a hall are used as these sound sources and the hall is constructed of the virtual space, it is possible, by determining the transient response signal as to each of the sound sources, to realize an excellent sound simulation which enables a listener to enjoy the sound as if he were in a real hall. This is particularly effective in case that the listener wants to enjoy the sound as if he were in a movie theater provided with a plurality of loud speakers.

In the above method of the present invention, in order to have the listener enjoy his experience as if he were in the scenes of a movie and like pictures, the virtual space is varied according to the scenes. Further, the positions of the sound sources, which correspond the positions of the people and animals serving as sound sources, are also varied according to the scenes.

Since the above-mentioned transient response signal is excellent in accuracy and determined by performing a large number of digital sampling operations, generally it is necessary to use a high-speed sum-of-products calculator in order to actually reproduce the sound based on such transient response signal. Consequently, in order to reduce the number of data processing operations while the reproduced sound is kept high in reality, it is necessary to determine a new transient response signal by extracting only a predetermined characteristic signal pulse from the transient response signals representing the acoustic characteristics of the space. In this extraction, particularly, of the pulses of the transient response signals varying on the delay time base, ones are selected in ascending order of the absolute value of their amplitude peak. Further, of the pulses of the transient response signals varying on the delay time base, positive ones are selected in ascending order of the positive value of their amplitude peak, while the other negative ones following the above-mentioned positive ones are also selected in the same manner. By extracting such negative pulses with which the

positive pulses are accompanied, it is possible to reproduce the sound with a considerable depth. Further, it is also possible: to extract the transient response signals by the use of a predetermined thresh value; or, to divide the transient response signals into a plurality of groups, in each of which groups the signals are packed on the time base and extracted therefrom.

On the other hand, the present invention realizes an actual sound reproduction by the use of the thus determined transient response signals. More particularly, as for the thus extracted transient response signals, each of the pulses thereof are set in its delay time and its relative amplitude value. Then, each of the sound signals sequentially inputted is outputted on each of the delay time according to each of the above-mentioned relative amplitude values, so that these signals thus outputted at the same time are summed up to produce a new sound signal being outputted. Further, of frequencies of the thus outputted sound signal, it is possible to remove predetermined ones so as to remove high-frequency components in the pulse of the sound signal. It is also possible to apply reverberation to the gap between the thus extracted pulses so as to reproduce the sound in a continuous manner.

The method of the present invention comprises the steps of:

generating a plurality of transient response signals which are different from each other according to a sound being

reproduced; and

setting the transient response signals in a
sum-of-products calculator when the sound is reproduced, the
transient response signals having been stored according to the
5 sound;

whereby a new sound is reproduced, the new sound having
acoustic components properly varying according to a space in which
an original sound to be reproduced is generated.

The plurality of the transient response signals may be
10 previously stored, and synchronized with a synchronizing signal
accompanied with the sound signal. The thus obtained corresponding
transient response signals may be set in the sum-of-products
calculator, or, it is also possible to receive the transient
response signals before the sound signal thus transmitted is
15 received, whereby the sound is reproduced based on such transient
response signals.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a flowchart illustrating a processing procedure
for determining the transient response signals;

20 Fig. 2 is a flowchart illustrating a processing procedure
for determining and storing the propagation history;

Fig. 3 is a flowchart illustrating a processing procedure
for calculating the transient response based on the propagation
history of each of the sound ray vectors by the use of a so-called
25 approximate boundary integral;

Fig. 4 is a flowchart of a processing procedure for calculating, adding and storing the transient response by the use of the approximate boundary integral;

Fig. 5(a) is a perspective view of the boundary, illustrating a micro-area element and its area size defined by the sound ray vector;

Fig. 5(b) is a plan view of the boundary, illustrating the micro-area element and its area size defined by the sound ray vector;

Fig. 6 is a graph of a series of the sound ray vectors and the transient responses of the direct and the reflected sound both to be determined, illustrating their total potentials at the observation point versus time;

Fig. 7(a) is a graph of the response obtained by the use of a conventional orthodox calculation called the virtual-image method;

Fig. 7(b) is a graph of the impulse response obtained in the virtual space by the use of the calculating procedure of the present invention;

Fig. 8 is a block diagram of the apparatus for reproducing the sound by the use of the transient response signals;

Fig. 9 is a schematic diagram of the concept for reproducing the sound by the use of the transient response signals;

Fig. 10(a) is a graph of the transient response signal

obtained in the same method for generating the transient response signal as that shown in Fig. 7(b), illustrating the amplitude of the transient response signal varying with respect to time in logarithmic scale;

5 Fig. 10(b) is a graph of the transient response signal obtained in the same method for generating the transient response signal as that shown in Fig. 7(b), illustrating the energy of the transient response signal varying with respect to time in logarithmic scale;

10 Fig. 11(a) is an enlarged view of a wave shape of the transient response signal shown in Fig. 10(a), the wave shape being enlarged on the time base and taken from the samplings ranging from Nos. 0 (zero) to 4,096;

 Fig. 11(b) is an enlarged view of a wave shape of another
15 new or characteristic transient response signal sampled from that shown in Fig. 10(a), the wave shape being enlarged on the time base;

 Fig. 12(a) is a graph of signal groups of the transient response signals, particularly illustrating the signal group 90
20 different from the other transient response signals;

 Fig. 12(b) is an enlarged portion of the graph shown in Fig. 12(a), illustrating the signal group 90 enlarged on the time base;

 Fig. 12(c) is an enlarged portion of the graph shown in
25 Fig. 12(a), illustrating only the sampled pulses of the signal

group 90 enlarged on the time base;

Fig. 13 a block diagram of the concept of a simple apparatus for reproducing the sound by the use of the sampled ones of the transient response signals;

5 Fig. 14 is a schematic diagram illustrating the potential affecting the observation point in the space;

Fig. 15 is a plan view of a movie theater, illustrating one of the sound ray vectors and its potential relative to the audience;

10 Fig. 16(a) is a plan view of a space in which the virtual space shown in Fig. 15 is superimposed over the real space, illustrating the potential of the sound ray vector which is divided into four pieces each allocated to each of loud speakers;

15 Fig. 16(b) is a schematic diagram illustrating a method for dividing the potential of the sound ray vector in the space shown in Fig. 16(a);

Fig. 17 is a block diagram of an apparatus for reproducing the sound in the movie theater shown in Fig. 15;

20 Fig. 18(a) is a plan view of the real space superimposed over the virtual space provided with the loud speakers serving as the sound sources, illustrating these loud speakers for sound reproduction and the listener;

25 Fig. 18(b) is a plan view of the real space superimposed over the virtual space provided with the loud speakers serving as the sound sources, illustrating these loud speakers for sound

reproduction and the listener, the virtual space containing a few movable sound sources such as persons and animals;

Fig. 19 is a schematic diagram illustrating the concept of superimposition of the real space over the virtual space varying
5 with respect to time;

Fig. 20 is a block diagram of a first apparatus for reproducing the sound in the real space in a condition in which the virtual space varies with respect to time;

Fig. 21 is a block diagram of a second apparatus for
10 reproducing the sound in the real space in a condition in which the virtual space varies with respect to time; and

Fig. 22 is a schematic diagram illustrating the concept of picture/sound signal transmission conforming to the MPEG standards.

15 DESCRIPTION OF THE PREFERRED EMBODIMENTS

Hereinbelow, the present invention will be described in detail with reference to the accompanying drawings.

First of all, in order to facilitate the description of the present invention, Kirchhoff integral and a so-called
20 approximate-boundary integral for determining wave motion characteristics of a sound, which are essential parts of the present invention, will be described with reference to Fig. 14. Then, the present invention will be clarified.

Kirchhoff integral equation (i.e., equation 2) is derived
25 from Kirchhoff's three-dimensional wave equation (i.e., equation

1). A modification of the equation 2 is called the approximate-boundary integral equation (i.e., equation 3).

The Kirchhoff's three-dimensional wave equation (i.e., equation 1) is represented as follows:

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$$\nabla^2 \phi - \frac{1}{C^2} \frac{\partial^2 \phi}{\partial t^2} = -F(x, y, z, t)$$

where: " ϕ " is the velocity potential; C is the sound velocity; t is the time; F is a wave source; and, x , y and z are three-dimensional coordinates. Further, in Fig. 14: S_2 is the surface defining an arbitrary space; Q is an arbitrary point on the S_2 ; P is the observation point for determining the transient response; V is an area except an area V_1 in the vicinity of the observation point P ; and, dS_2 is a micro-area vector on the S_2 .

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The equation 2 is the integral of the Kirchhoff's three-dimensional wave equation (i.e., equation 1) to represent the velocity potential at the observation point P , and is represented as follows:

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$$\phi_P = \frac{1}{4\pi} \int_V \frac{1}{r} [F]_{td} dv + \frac{1}{4\pi} \int_{S_2} \left\{ \frac{1}{r} [\nabla_Q \phi]_{td} - [\phi]_{td} \nabla_Q \left(\frac{1}{r} \right) + \frac{1}{cr} \left[\frac{\partial \phi}{\partial t} \right]_{td} \nabla r \right\} \cdot d\vec{S}_2$$

where: ϕ_P is the velocity potential at the observation point P ; $(\)_{td}$ is the delay time $t-r/c$; V is an arbitrary space surrounding

the observation point P; S2 is the surface of the V; Q is an arbitrary point on the S2; and, r_0 is the distance between the Q and the observation point P.

5 The modification of the integral of the Kirchhoff's equation (i.e., the approximate-boundary integral equation 3) is represented as follows:

$$\phi_p = \frac{1}{d} f\left(t - \frac{d}{c}\right) + \frac{1}{4\pi} \int_{S_2} \left[\frac{1}{cr_0} \frac{\partial}{\partial t} f\left(t - \frac{r_0}{c}\right) (\cos(\vec{r}, \vec{n}) - \cos(\vec{r}_0, \vec{n})) + \frac{1}{r_0} f\left(t - \frac{r_0}{c}\right) \left\{ \frac{1}{r} \cos(\vec{r}, \vec{n}) - \frac{1}{r_0} \cos(\vec{r}_0, \vec{n}) \right\} \right] ds_2$$

10

where: ϕ_p is the velocity potential at the observation point P; $(t - r/c)$ is the delay time $t - r/c$; V is an arbitrary space surrounding the observation point P; S2 is the surface of the V; Q is an arbitrary point on the S2; r is the distance between the Q and the observation point P; and, r_0 is a total distance between the wave source and the Q.

15

Further, $f(t)$ represents a transient signal of a sound produced at the wave source.

Now, a means for generating the transient response signal on the basis of the above-mentioned approximate-boundary integral equation 3 will be described. Although the means mentioned above is described by the use of the approximate-boundary integral equation 3, it is also possible to describe the above means by the use of any other suitable equation, provided that such other equation may determine the wave characteristics of a sound as is

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in the Kirchhoff integral equation and the like.

Figs. 1 to 4 show flowcharts of processing procedures of the present invention for determining the transient response signal, i.e., impulse response signal.

5 Fig. 1 shows the entire of the processing procedure ranging from the start to the stop of the procedure, and is divided into two essential parts:

"processing procedure A" and "processing procedure B".

10 A step 7, i.e., "processing procedure C" shown in Fig. 1 corresponds to a processing procedure shown in Fig. 2. On the other hand, a step 10, i.e., "processing procedure D" shown in Fig. 1 corresponds to a processing procedure shown in Fig. 3. Further, "processing procedure E" shown in Fig. 3 corresponds to a processing procedure shown in Fig. 4.

15 As shown in Fig. 1, in a first step 1 "initialize", performed as prerequisite for executing the processing procedure are "setting of calculation conditions", "setting of wave-source conditions", "setting of a boundary", and "setting of wave-source radiating sound ray vectors".

20 Set in the "setting of calculation conditions" are: three-dimensional coordinates of the wave source; the number of observation points; three-dimensional coordinates of each of the observation points; the temperature and moisture of the air in the space; an analytic frequency (i.e., the highest frequency in the
25 transient response to be calculated); the duration time, T of the

transient response to be calculated; and, the like. Such setting is performed by means of an input unit through which a user inputs necessary data, or by means of an external memory unit from which the necessary data is retrieved.

5 In the "setting of wave-source conditions", an initial value of the sound issued from the sound source is represented to be a transient signal by the use of a delta approximate function and its derivative, which resembles an impulse in shape. Such initial value may be modified if necessary in application.

10 Further, propagation velocity of the sound is calculated based on the temperature and moisture of the air set in step 1. Based on the analytic frequency, a discrete-time interval, which corresponds to a frequency of equal to or over twice as high as the analytic frequency, is set according to Shanon's sampling
15 theorem. A size of memory area for storing therein the transient response to be calculated, which is used in the "processing procedure B", is determined on the basis of both the duration time of the transient response and the discrete-time interval.

 In the "setting of a boundary", information as to the
20 boundary defining a sound field is set. This represents a numerical space for simulating wave-motion propagation of the sound therein, and constructed of a plurality of polygons in plane. In this "setting of a boundary", each of the polygons is called the boundary. Set in the "setting of a boundary" are: the
25 number of the boundaries; a normal, i.e., perpendicular line of

each of the boundaries; coordinates of each vertex in each of the boundaries; reflection and absorption at the boundaries; and, the like. Incidentally, in the following description of the embodiment of the present invention, in order to facilitate the description, it is assumed that each of the boundaries only performs total reflection and total absorption of the sound over the entire frequency band.

In the "setting of wave-source radiating sound ray vectors", the number of sound ray vectors, N , radiating from the wave source are calculated. In this setting, the wave motion of the sound propagated from the wave source is simulated to calculate both a position and a time at which the wave motion is reflected at the boundary, so that acoustic characteristics (i.e., velocity potential " ϕ_p " of the sound at the observation point are calculated. Such simulation of the wave motion propagation of the sound is performed by the use of numerous vectors, each of which has the same solid angle and radiates from the wave source in the space. Such vectors are defined as sound ray vectors in the present invention. The number N of the radiating sound ray vectors is set so as to let the distance between adjacent sound ray vectors be equal to or smaller than $1/2$, preferably $1/4$ of a wave length " λ " of the analytic frequency. The length between adjacent sound ray vectors depends on the capacity of processing units and the degree of approximation in the reproduction of the sound, and, therefore may be more than $1/2$ of the " λ " in some cases.

Incidentally, vectors representing the wave surface of the sound propagated in the space, are generally called the sound ray vectors. Of these sound ray vectors, ones radiating from the wave source are especially called the radiating sound ray vectors.

5 In step 3, the wave motion propagated from the wave source is simulated with the use of the sound ray vectors. Namely, first of all, a direction vector D_n of the n 'th radiating sound ray vector is calculated so as to have directions of N pieces of the radiating sound ray vectors form the same solid angles
10 therebetween with respect to the wave source.

 Selected in step 4 following step 3 is the boundary B having an intersection at which it intersects the sound ray vectors propagated from the sound source. In step 5 following step 4, when there is no boundary intersecting with the sound ray
15 vectors, calculation is performed as to a subsequent one of the radiating sound ray vectors.

 On the other hand, when there is some boundary having the intersection, the sound ray vector is reflected at the boundary B . In this case, it is necessary to judge whether the boundary showed
20 its front or its back to the sound ray vector.

 Such judgment may be accomplished by determining whether an angle formed between a normal, i.e., perpendicular line of the boundary and the sound ray vector is within an angle range of from 0 to 180 degrees. In other words, in case that the normal vector
25 of the boundary is so defined as to extend outward in a virtual

space, when the angle formed between the normal boundary and the radiating sound ray vector is within the angle range of from 0 to 180 degrees (i.e., when an inner product thereof is positive), the boundary is defined to be the front. Otherwise (i.e., when the angle is within an angle range of from 180 to 360 degrees), the boundary is defined to be the back. The reason why the angle is defined in a manner described above is that it is necessary for the computer to judge in operation: whether the sound ray vector is issued from the inside of the boundary toward the front side thereof (i.e., whether or not the sound ray vector is reflected at the boundary); or, whether the sound ray vector is issued from the outside of the boundary toward the back side thereof (i.e., whether or not the sound ray vector merely passes through the boundary without being reflected thereby).

According to the above definitions, in step 6 following step 5, when the boundary B is to the back of the sound ray vector, it is necessary to select any other boundary having the above-described intersection. On the other hand, when the boundary B is, in front of to the sound ray vector, step 6 is followed by step 7, i.e., processing procedure "C".

In step 7, i.e., processing procedure "C", when the boundary B is set so as to completely absorb the wave motion of the sound in step 11, the processing procedure goes to "End", as shown in Fig. 2. Otherwise, step 11 is followed by a subsequent step 12 in which the total propagation distance d between the wave

source and the intersection Q of the boundary B is calculated.

Step 12 is followed by a subsequent step 13. In step 13, when the thus calculated propagation distance d is one permitting the elapsed time of the wave motion for reaching the point Q to exceed the duration time of transient response, step 13 is followed by "End", as shown in Fig. 2. Although these steps 11, 12 and 13 may be interchangeable in processing order, the processing order shown in Fig. 2 is the best in efficiency to minimize the processing time.

Otherwise, i.e., in step 13, when the propagation distance d is not such one, step 13 is followed by a subsequent step 14 in which an incident angle " α " of the sound ray vector incident upon the boundary B is calculated to determine the incident angle formed between the incident sound ray vector and the normal (i.e., perpendicular line) of the boundary plane in the above-described equation 3. The incident sound ray vector is defined as one which is the n'th radiating sound ray vector reaching the boundary B. On the other hand, the incident sound ray vector incident upon the boundaries "r" times is defined as "En, r". Consequently, the "En, r" is the radiating sound ray vector which is issued from the wave source to reach the r'th boundary B.

Then, step 14 is followed by a subsequent step 15. Calculated in the step 15 is the direction vector of the reflected one of this incident sound ray vector, the reflected one being reflected at the intersection Q. The reflected sound ray vector is

defined to be one which is the n 'th radiating sound ray vector D_n having been reflected at the boundary B and is found there. Of the reflected sound ray vectors, one having been reflected at the boundaries " r " times is defined as " F_n, r ". Consequently, the " $F_n, 1$ " is the n 'th sound ray vector having been reflected at the boundary for the first time.

Step 15 is followed by a subsequent step 16. Stored in the main memory or an external memory unit in step 16 are: arrangement No. B of the boundary; total propagation distance " d " between the wave source and the intersection Q ; three-dimensional coordinates of the intersection Q ; the direction vector " E_n, r " of the incident sound ray vector; and, the direction vector " F_n, r " of the reflected sound ray vector. A series of data calculated each time the radiating sound ray vector D_n is reflected at the boundary is defined as the propagation history of the radiating sound ray vector D_n . The reason why the propagation history is stored in the memory is because it is necessary to calculate the velocity potential of the sound ray vector, which potential affects the observation point P each time the sound ray vector is reflected at the arbitrary point of the boundary surface S_2 (shown in the above equation 3). The total propagation distance " d " above described corresponds to the " r ." of equation 3.

Step 16 is followed by a subsequent step 17. Selected in step 17 is another or subsequent one of the boundaries B having a new intersection Q at which such another one intersects with the

reflected sound ray vector having the direction " F_n, r ". Step 17 is followed by a subsequent step 18. When there is no boundary B having the intersection Q, the processing procedure goes to "End", as shown in Fig. 2. When there is any other boundary having such intersection Q, the step 18 is followed by a subsequent step 19. In step 19, when the incident sound ray vector having the intersection Q is incident on the back side of the boundary B, it is necessary to find another boundary B having the intersection Q. When the incident sound ray vector having the intersection Q is incident on the front side of the thus found other boundary B, step 19 is followed by step 11 to repeat the above-described processing procedure.

After completion of the processing procedure "C" with respect to the n'th radiating sound ray vector, as is clear from Fig. 1, the same processing procedure is repeated for each of the subsequent radiating sound ray vectors. Namely, after completion of the processing procedures as to all the radiating sound ray vectors, the processing procedure "C" is followed by a subsequent processing procedure "B".

In summary, the processing procedure "A" comprising of the series of steps 1 to 9 is performed to calculate and store the propagation histories of all the sound ray vectors.

As is clear from the above description, an apparatus for performing the processing procedure "A" comprises: a means for storing the initialization data; a means for calculating or

determining the propagation record data; and, a means for storing the propagation record data.

On the other hand, as is clear from Fig. 3, calculation performed in the subsequent processing procedure "D" is based on
5 the propagation histories of n pieces of the radiating sound ray vectors, as follows:

Namely, the processing procedure "D" starts with step 21. In step 21, coordinates of the first observation point P for calculating the transient response are retrieved from memory and
10 the like. First, data as to the propagation history of the radiating sound ray vector of " $n = 0$ " is retrieved in step 22 following step 21. As for a first one of the boundaries B in the thus retrieved propagation history, when this one has its back side faced to the observation point P , subsequent propagation
15 histories retrieved in step 32 to continue the processing procedure. On the other hand, when such boundary B has its front side faced to the observation point P , the processing procedure goes to step 25 in which a direction vector R extending from the observation point P toward the intersection Q recorded in the
20 propagation history is calculated. Step 25 is followed by a subsequent step 26 in which the distance RD between the observation point P and the intersection Q is calculated. Step 26 is followed by a subsequent step 27 in which when a straight line connecting the observation point P with the intersection Q
25 intersects with the remaining one of the boundaries, it is judged

that the velocity potential from the intersection Q does not affect the observation point P. As a result, the processing procedure goes to step 32. On the other hand, in step 27, when the straight line connecting the observation point P with the intersection Q does not intersect with any other boundaries, step 27 is followed by a subsequent step 28, and therein it is judged whether the wave motion reaches the observation point P within a period of time T. When it is judged that the wave motion reaches the observation point P within the period of time T, then the velocity potential on the boundary B affects the observation point P, and therefore step 28 is followed by the processing procedure "E". Incidentally, the above judgment is formed depending on whether the wave motion travels a predetermined distance within the duration time T of the initialized transient response, the predetermined distance being the sum of the total propagation distance d (i.e., that between the wave source and the intersection Q) and the distance RD.

As shown in Fig. 4, in the processing procedure "E": the velocity potential, which is formed by a so-called micro-area element of the boundary represented by the intersection Q and affects the observation point P, is calculated by the use of the equation 3; and, based on a time at which the velocity potential affects the observation point P, the transient response is stored in array. When some data has been already stored in the same location of array, the transient response is added to the data.

More particularly, the micro-area element is so defined as to be an area formed in the intersection Q by a solid angle used in the definition of the sound ray vector. In Fig. 5(A), a plurality of the sound ray vectors are issued toward the boundary, and one
5 (shown in solid line) of such sound ray vectors forms the micro-area element on the boundary. In Fig. 5(B), is a plan view of the micro-area element formed on the boundary. As is clear from these drawings, the micro-area element on the boundary varies in size and determined by an angle formed between the sound ray
10 vector and the boundary plane. More particularly, in data processing, the area size of the micro-area element on the boundary may be equal to that of a bottom area of a cone defined by both the total distance d from the wave source to the intersection Q and the solid angle described above. In this case,
15 although approximation slightly becomes poor in accuracy, it is sufficient in practice. Further, in this case, since the area size of the micro-area element can be determined on the basis of only both the distance d and the solid angle, it is possible to increase the operation speed in data processing. Incidentally,
20 this area size of the micro-area element corresponds to the term " ds^2 " of the equation 3.

As is clear from Fig. 4, the processing procedure starts with step 41 in which the first term of the integration term of equation 3 is calculated. Step 41 is followed by a subsequent step
25 42 in which the second term of the integration term of equation 3

is calculated. In calculation in the embodiment of the present invention, since the reflection and the absorption of a sound in the boundary are defined to be the total reflection and the total absorption only, the function $f(t)$ used in the calculation may be defined as to be equal to the initialized transient function of the wave source. When each of the reflection and the absorption are a partial one, the function $f(t)$ corresponding to the properties of the boundary may be determined as to the propagation history " F_n, r " each time the sound ray vector is reflected at the boundary.

Step 42 is followed by a subsequent step 43 in which the area size of the micro-area on the boundary is calculated to determine the integral approximation.

Step 43 is followed by a subsequent step 44 in which: a product of the calculation result obtained in step 41 and the area size of the micro-area element on the boundary obtained in step 43 is determined; and, the thus determined product and the initial value of the wave source are processed by the use of a calculus of convolution transformation.

Step 44 is followed by a subsequent step 45 in which: a product of the calculation result obtained in step 42 and the area size of the micro-area element on the boundary obtained in step 43 is determined; and, the thus determined product and the derivative value of the wave source's initial value are processed by the use of the calculus of convolution transformation.

In steps 46 to 48 following the step 45, the transient response base on the integral result of the approximate boundary is stored in the memory and the like. In order to reproduce the acoustic characteristics of the sound at the observation point P, there is provided a numerical array in which the transient response to be calculated is stored. Such array may be provided in the initialization performed in the processing procedure "A". In the array, a location j in the array in which the transient response is stored is determined depending on a time Dt at which the wave motion reaches the observation point P. Consequently, the result of the transient response corresponding to the time Dt at the observation point P is added and stored in the corresponding location j in the numerical array. More particularly, in the step 46, the time Dt at which the wave motion reaches the observation point P is determined based on the sum of the total distance d between the wave source and the intersection Q of the sound ray vector and the distance RD.

Step 46 is followed by a step 47 in which the location j in the numerical array corresponding to the time Dt is determined. Step 47 is followed by step 48 in which the data in time series obtained in steps 44, 45 is added and stored in the corresponding location j in the numerical array. In the above addition and storing, when some other data is already stored in location j , the data is added to such some other data and the resultant is stored in the location j , whereby the same effect as that in the

approximate integration is obtained. Further, by arranging the array location in a manner of time series, it is possible to retrieve the sound-field characteristics at the observation point P in time series (i.e., in the order of the array location), which enables the sound to be reproduced, whereby the method and apparatus for efficiently reproducing the sound are realized.

After completion of the processing procedure "E", data processing continues for the propagation history of a subsequent radiating sound ray vector. When all the data of the propagation history as to the n'th radiating sound ray vector is calculated, the same operation is conducted as to the (n+1)'th radiating sound ray vector. Then, after completion of calculation as to all the radiating sound ray vectors, data processing for a subsequent observation point P is conducted in a manner as shown in Fig. 3.

In summary, in processing procedure "D": the velocity potential of the sound ray vector affecting the observation point P is calculated based on the propagation history of each of the sound ray vectors by the use of integration, the sound ray vector being reflected at the boundary; and, the transient response at the observation point P is determined and stored.

As is clear from the above, a means for accomplishing the processing procedure "D" may be any means, provided that the means processes the stored propagation history by the use of approximate integration. In other words, the means may be: a processing unit operated by the use of software comprising a series of steps; or a

computer; or some other unit provided with hardware for executing an appropriate processing procedure by the use of the approximate-boundary integration.

After completion of the processing procedure "D", step 49 shown in Fig. 1 follows it. In step 49, in order to determine the acoustic characteristics of a direct sound issued from the sound source and affecting the observation point P, transient characteristics are added and stored in the location j of the array, the location j corresponding to a time at which the direct sound reaches the observation point P.

Fig. 6 shows a graph of a series of the sound ray vectors and the transient responses of the direct and the reflected sound both to be determined, illustrating their total potentials at the observation point P versus time. More particularly, in the processing procedure "D", the potential affecting the observation point P is calculated each time each of the sound ray vectors is reflected at the boundary and stored in time series so that the transient response at the observation point P is determined as a whole. The processing procedure "D" and adding of the direct sound are easily understood with reference to the graph shown in Fig. 6.

Now, a method for generating the transient response signal will be described with reference to Figs. 7(a) and 7(b). Fig. 7(a) shows a response signal obtained by using a conventional orthodox calculus called the virtual-image method. On the other hand, Fig. 7(b) shows an amplitude wave shape of the transient response

signal actually obtained through the above processing procedures. In comparison with a conventional wave shape shown in Fig. 7(a), it is clear that the amplitude wave shape shown in Fig. 7(b) comprises negative wave shapes in addition to the positive wave shapes. As is clear from this fact, the present invention realizes reproduction of the amplitude wave shape of the transient response which is hitherto hard to determine or requires a large amount of operation time in determination thereof.

Now, actual sound reproduction according to the above wave shapes will be described.

As shown in Fig. 8, an analog sound signal is reproduced by a sound reproduction equipment 70 such as CD players, record players, cassette players and like equipments, and inputted to a sum-of-products calculator 72 through a line 71. The sound signal thus processed in the calculator 72 is then issued to an amplifier 80 in an analog mode through a line 79 and amplified therein. The thus amplified sound signal is finally supplied to a loud speaker 82 through a line 81, so that the sound is reproduced. The sum-of-products calculator 72 comprises: a central processing unit (hereinafter referred to as the CPU) 76 for actually performing a convolution calculus; an A/D converter 73 which is connected with an input terminal of the CPU 76 through a line 75; and a D/A converter 78 which is connected with an output terminal of the CPU 76 through a line 77. However, in case that the sound reproduction equipment 70 issues its output signal in a digital mode, it is

natural that the A/D converter 73 is removed. As an example of the sum-of-products calculator 72, there is "Digital Audio Convolution Processor FDP 1 plus", which is a product supplied from a company called the Lake DSP Pty. Ltd.

5 Now, with reference to Fig. 9, a concept of signal processing performed in the sum-of-products calculator 72 will be described, in which a digital signal issued from the A/D converter 73 is inputted to the CPU 76 and processed therein, the thus processed signal being issued to the line 77.

10 As shown in Fig. 9, a discrete signal 85 is a discrete representation of the analog signal on the time base and appears on the line 71. Data as to timing and amplitude in wave shape of this discrete signal 85 is inputted as digital signals to the CPU 76, stored in which is the transient response signal determined
15 through the above processing procedure for generating the transient response signal or through any other proper processing procedures. In order to determine an actual discrete digital signal 86, all the data of the discrete signal 85 (which is inputted in real time to the CPU 76) for its every divided
20 discrete time are processed in a convolution calculus portion 84 of the CPU 76 so that the convolution calculus portion 84 produces a response signal corresponding to the transient response signal 83. Since all the sound with its delay time caused by the transient response signal must be reproduced, all the above data of the
25 discrete signal 85 for its every discrete time interval is summed

up in the convolution calculus portion 84 by a plurality of times the number of which corresponds to the number of divided pieces of time in the transient response signal 83. For example, in order to reproduce the sound in frequency up to 20 KHz, it is necessary to
5 use 65,536 pieces of taps when the transient response signal (which is divided into 48,000 pieces per second) with the duration time of 1.3 second is used. Consequently, in order to reproduce the sound, it is necessary to sum up the data as to 65,536 pieces of the taps for every 20.8 microseconds (i.e., $1/48,000$ second) so
10 as to produce an analog output.

In the above processing, it is preferable to use the above-mentioned "Digital Audio Convolution Processor FDP 1 plus", which is a product supplied from a company called the Lake DSP Pty. Ltd.

15 On the other hand, the need also exists for a simple and inexpensive method and apparatus for reproducing the sound, which will be now described.

Fig. 10(a) is a logarithmic representation of the amplitude of the transient response signal on the time base, which
20 signal is generated through the same method as that for generating the transient response signal shown in Fig. 7(b). Fig. 10(b) is a logarithmic representation of the energy of the transient response signal on the time base, which signal is generated through the same method as that for generating the transient response signal
25 shown in Fig. 7(b). In the following description, a discrete time

interval of the transient response signal is set at 20.8
microseconds corresponding to 48 KHz sampling times, and the
duration time in reproduction is set at 1.3 second, so that the
total number of sampling times of the transient response signal on
5 the time base is set at 65,536 times. Here, each of 65,536
sampling times is identified by its sampling number (i.e.,
sampling No.).

In Fig. 10(a), a time when the transient signal is
produced at the sound source is defined as a sampling No. zero
10 (0), and the wave form of the transient response signal subsequent
to the sampling No. 0 is illustrated. As is clear from Fig. 10(a),
essential pulses are concentrated in a front half portion of the
graph up to approximately the sampling No. 4,000. On the other
hand, as is clear from Fig. 10(b), attenuation in the transient
15 response in the graph on the time base is marked at about 60 dB at
a sampling No. 65,536. Consequently, although such attenuation in
the transient response depends on the size of the space, materials
in the wall defining the space and like factors, it is apparent
that the duration time of 1.3 second is sufficient for the purpose
20 of the present invention.

Fig. 11(a) shows an enlarged view of the wave shape
ranging of from the sampling Nos. zero (0) to 4,096 shown in Fig.
10(a). As is clear from Fig. 11(a), essential pulses exist in a
range of the sampling numbers up to about No. 4,000. Of these
25 essential pulses, ones with particularly large amplitudes exist in

a further limited range. Therefore, only characteristic ones of the pulses shown in Fig. 10(a) are extracted so that another new transient response signal is produced as shown in Fig. 11(b). Then, the thus extracted transient response signal is used as the transient response signal 83 shown in Fig. 9 in sound reproduction, which proves the resultant or reproduced sound to be a three-dimensional sound.

From the above fact, it is apparent that the pulses with sufficiently large amplitudes capable of reaching a listener's ear after the elapse of a predetermined time can stimulate the listener's sense of hearing. In contrast with reverberations such as echoes which merely attenuate in an exponential mode and have been widely used to produce a three-dimensional reverberation effect, the above pulses enable the listener to enjoy the three-dimensional characteristics of the space and the spacious feeling inherent therein. Consequently, by reproducing such characteristic parts, it is possible to reproduce the characteristics of the sound detected at a predetermined observation point (i.e., listening point) in the space being reproduced, the sound being detected as if it were generated in the space. On the other hand, the remaining pulses other than the above characteristic pulses serve as supplementary elements to the characteristic pulses in the listener's sense of hearing, and, therefore are not necessarily reproduced with high accuracy in comparison with the characteristic pulses.

Further, it is possible to recognize an actual image of the space by using the transient response signal comprising both the negative and the positive pulses, the positive pulses being followed by the negative pulses. In effect, by using the negative
5 pulses, it is possible to take out the components in frequency of the transient response, and also possible to have the listener recognize the three-dimensional shape of the space and especially its intermittent boundaries, because, as is clearly derived from the above equations 2 and 3, the negative pulses represent
10 discontinuous portions of the boundaries. In other words, these negative pulses are produced by predetermined portions of the boundaries or wall surfaces defining the three-dimensional space, which portions comprise convex intersectional area formed between the boundaries, and some areas of the boundaries where the
15 boundaries change in material. These negative pulses are considered to give an real image of the space in the sense of hearing. Consequently, it is important to reproduce the negative pulses together with the corresponding positive pulses in reproduction of the three-dimensional sound.

20 As described above, it is possible to recognize essential parts of the acoustic characteristics of the sound in the space by: extracting the characteristic pulses of the transient response signal; and, reproducing the characteristic portions of the sound with the use of the thus extracted characteristic pulses of the
25 transient response signal. Further, it is also possible to

remarkably reduce a load in data processing (i.e., the number of the transient response signal data to be summed up) by: extracting the data for preparing a new transient response signal from a large number of the transient response signal data.

5 Of the portions of the transient response signal, the characteristic ones are taken out or sampled preferably by the following processes (1) to (6):

PROCESS (1):

10 The characteristic portions of the transient response signal are sampled in the order of maximum (i.e., absolute) values in amplitude of the pulses. This process (1) is easy to perform. In the process (1), the amount of acoustic stimulation applied to the listener's sense of hearing is represented by magnitudes in amplitude of the pluses. In this sampling, in order to prevent the
15 number of the pulses to be sampled from infinitely increasing, it is possible to limit the number of the pulses to be sampled to a predetermined number. Alternatively, it is also possible to set a predetermined threshold value, which enables only the pulses exceeding in amplitude such threshold value to be sampled.

20 PROCESS (2):

 In a graph such as shown in Fig. 11(b), a predetermined threshold value is previously set. Then, only the pluses exceeding this threshold value in absolute value of their amplitudes are sampled in the order of time in a condition in which a sound is
25 produced at a time corresponding to the origin of the time base of

the graph. The process (2) can be effectively applied to cases where a starting area of sound occurrence is especially important, that is, the pulses not reaching the threshold value are not useful.

5 PROCESS (3):

In the process (3), when the number of samplings performed in the process (2) reaches to a predetermined number, the sampling operation of the pulses is finished. Consequently, the process (3) is especially effective in case that the capacity of a central
10 processing unit (i.e., CPU) summing up the numbers is limited.

PROCESS (4):

Of the pulses in each of groups of signals, one with the largest amplitude is selected and defined as a first pulse. In an example shown in Fig. 12(a), signals 90, different from the
15 remaining signals, form a signal group 90. Fig. 12(b) shows a graph in which this transient response signals 90 are enlarged on the time base. As is clear from Fig. 12(b), there are discrete peak points of the pulses in an area ranging from the sampling Nos. about 1024 to about 2000. Depending on the sampling timing,
20 the time intervals between these pulses may be often below a predetermined value which the listener's sense of hearing can not detect. Consequently, it is not necessary to reproduce or sample all the peak points of the pulses. In other words, it is
25 sufficient to sample one of these peak points of the pulses, which one is sufficient to enable the listener's sense of hearing to

detect it. For example, as is clear from the well-known "Haas" effect, it is not possible for the listener to distinguish individual signals when these signals are spaced apart from each other at intervals of 10 milliseconds and like very short time on the time-based graph. In this case, a selected one of the pulses may be a first peak pulse in each of the pulse groups, or a maximum peak pulses therein. In case of Fig. 12(c), only a first pulse in each of the pulse groups 90 is sampled.

PROCESS (5):

10 The duration of each of the above processes (1) to (4) may be limited to a predetermined length of time. As for the transient response signal in the space, depending on superimposition of the direct waves, which are issued from the wave source in various directions, over the reflected waves, the amplitude of the sound
15 affecting the observation point is determined. Under such circumstances, after the elapse of a predetermined period of time, the energy of the wave source damps, which makes it impossible to obtain a sufficient peak value in the pulses. For example, as is clear from Figs. 10(a) and 10(b), energy attenuation at the
20 observation point reaches approximately 60 dB as to the sampling No. 65,530. Consequently, it is not necessary to sample the pulses subsequent to this sampling No. 65,530. In the above, based on the energy attenuation, the duration of each of the processes (1) to (4) is determined. It is also possible to determine the duration
25 of each of the processes (1) to (4) on the basis of the magnitudes

in amplitude of the pulses. As shown in Figs. 10(a) and 11(a), a first range on the time base of the pulses with amplitudes equal to or more than 0.005 extends substantially up to the sampling No. 4,096. In other words, there is no pulse equal to or more than
5 0.005 in amplitude in a second range beyond the sampling No. 4,096. Consequently, it is possible to sample the peak pulses only in the above first range.

PROCESS (6):

In each of the above processes (1) to (5) for sampling the
10 pulses, it is preferable in effect to sample the negative peak pulses together with the corresponding positive peak pulses. As described above, the negative peak pulses cooperate with the corresponding positive peak pulses, which makes it possible to reproduce their frequency components and also to enable the
15 listener to enjoy the three-dimensional real image of the space.

As described above, there are various processes for sampling the pulses.

Now, with reference to Fig. 13, an apparatus of the present invention for simply reproduce, without using the
20 high-performance sum-of-products calculator, the sound on the basis of the transient response signal after completion of the above sampling of the pulses will be described.

As shown in Fig. 3, in place of the CPU 76 of the sum-of-products calculator 72 shown in Fig. 8, a simple-type
25 sum-of-products calculator 100 is used in the present invention.

The simple-type sum-of-products calculator 100 comprises a multi-tap delay circuit 110, a reverberation portion 120, and an equalizer 130.

5 The multi-tap delay circuit 110 comprises a delay line 112, a plurality of weighting portions 114, and a plurality of adding portions 118. A plurality of output lines 115 for a plurality of tap signals are provided in the adding portions 118. The signal is transmitted from the delay line 112 to the output lines 115 of the adding portions 118 through the weighting
10 portions 114.

The multi-tap delay circuit 110 receives a sound signal from an input line 75 and processes the signal on the basis of predetermined delay times and amplitudes to issue an output signal. Now, with the use of the new transient response signal
15 prepared by sampling the transient response signal shown in Figs. 11(a) and 11(b), the function of the multi-tap delay circuit 110 will be now described.

Signals to be sampled are denoted by the reference characters P1 to P30 (hereinafter referred to as the pulses P1 to
20 P30). Consequently, the total number of the pulses is 30.

Set in the delay line 112 is a delay time corresponding to a position of the pulse P1 issued from a first tap of the plurality of the output lines (i.e., taps) 115. The pulse P1 represents a direct sound reaching the observation point, and
25 corresponds to the sampling No. 793. The duration of the pulse P1

is, therefore, $793 \times 1 / 48,000 = 16.52$ milliseconds. The delay line 112 thus set receives a signal from the input line 75, and issues the signal to a first one of the weighting portions 114 after the elapse of the delay time thus set.

5 Further, the magnitude of the signal to be reproduced is set, and controlled in the weighting portions 114. In this case, the amplitude of the pulses in the transient response signal to be reproduced are so set as to be proportional in magnitude to each other. More particularly, in case of Fig. 11(b), the pulse P29 is
10 maximum in amplitude. Consequently, the amplitude of the pulse P1 is so set as to be approximately 70 percent of the pulse P29. As a result, in the first one of the weighting portions 114 corresponding to the pulse P1, the signal supplied from the delay line 112 is weighted in amplitude by a weighting ratio of 70
15 percent, and issued to the first one of the output lines 115.

As a result of the above setting, the inputted signal supplied from the input line 75 is issued from the first one of the output lines 115 after the elapse of 16.52 milliseconds. The thus outputted signal has its amplitude so weighted as to be 70
20 percent of the amplitude of the inputted signal, and is used for reproduction of the pulse P1, which makes it possible for the listener to catch a sound as a direct sound from its sound source in 16.52 milliseconds from the occurrence of the sound.

Then, setting for the pulse P2 is conducted as to a second
25 one of the output lines 115. A delay time to be set is 22.16

(i.e., 1,064/48,000) milliseconds. In this case, the weighting ratio is 63 percent. Due to such setting, it is possible for the listener to catch a first characteristic sound in 22.16 milliseconds from the occurrence of the sound at its sound source.

5 In the same manner as that described above, setting is conducted as to each of the remaining pulses P3 to P30. As for the negative pulses, in the above setting of amplitude, it is necessary to additionally set negative indications with respect to these negative pulses. After completion of all these setting, in
10 the multi-tap delay circuit 110, the signals, which correspond to the individual delay times having been set for the individual output lines 115, are sequentially issued from the output lines 115.

 In general, the input signal on the line 75 is a
15 continuous sound signal such as musical ones and the like. Consequently, the multi-tap delay circuit 110 is provided with the adding portions 118 in which all the signals issued at the same time from the output lines 115 are summed up, because, for example: a first one of the inputted sound signals is issued from
20 a first output line 115 (after the elapse of a first delay time) at the same time when a second inputted sound signal subsequent to the first one is issued from a second output line 115 after the elapse of a second delay time. Consequently, all the simultaneous signals for the pulses P1 to P30 are summed up every sampling
25 interval which is, for example, 1/48,000 second when the sampling

frequency of the inputted signals is 48 KHz.

On the other hand, in order to use the thus outputted signals on the line 116 so as to have the listener enjoy a more realistic sound field, it is preferable to use the reverberation portion 120 to add reverberations to the sound. The reverberation portion 120 may be a commercially available product. In the reverberation portion 120, reverberations are added to the sound in a manner such that the signal processed every sampling frequency in the multi-tap delay circuit 110 exponentially damps with respect to time. In this case, the reverberation time and the damping characteristic of reverberations may vary according to the actually used reverberation portion 120. The thus added reverberations may easily smooth the gaps between the pulses of the new transient response signal thus sampled. Further, there is no fear that the thus sampled characteristic pulses impair the sound in image. The addition of reverberations is particularly effective in the above process (5) in which one of the pulses in the groups of signals is selected.

The signal with reverberation is inputted to the equalizer 130 through the line 117. The signal appearing on the lines 116, 117 comprises the sampled pulse-like characteristic signals only, and, therefore comprises extremely many high-frequency components.. The equalizer 130 serves as a low-pass filter to remove these high-frequency components, and controls the frequency characteristics of the output signal issued to the line 77.

As described above, of a large number of the pulses in the transient response signal, only the characteristic ones are selected and sampled to prepare the new simple transient response signal, which realizes a so-called three-dimensional sound reproduction in the space through a convolution calculus of only the limited number of the pulses in the transient response signal. Consequently, in the present invention, it is possible to use a very simple sum-of-products calculator or a popular tap/delay system with the limited number of taps in sound reproduction in a three-dimensional virtual space.

Now, a method for generating the transient response signal used in sound reproduction in the virtual space by the use of a plurality of loud speakers disposed in a real space will be described. The method for generating the transient response signal is adapted for the real space for reproducing the sound, and enables the listener staying in the real space to enjoy the sound reproduced in the virtual space.

Fig. 15 shows a plan view of a movie theater, illustrating a plurality of the audience seats and a plurality of loud speaker arranged therein for reproducing the sound in a so-called Dolby surround system mode. The number of the loud speakers arranged in the theater shown in Fig. 15 is fourteen. These loud speakers are denoted by the reference characters SP1 to SP14, and so disposed as to surround the audience, as shown in Fig. 15. Sounds issued from the loud speakers SP1 to SP14 form direct sounds and

reflected sounds from walls. These direct and reflected sounds reach the audience, which enables the audience to enjoy the sound inherent in the movie theater.

In order to actually reproduce such sound which the audience may enjoy, it is necessary to generate a potential serving as a component of the transient response signal, which potential reaches the audience in sound reproduction. One of methods for generating such potential is the same as that for generating the transient response signal which is described in the above with reference to Figs. 7(a) and 7(b). In this method, a wall surrounding the audience in the movie theater is defined as a boundary. A space defined by such boundary forms a virtual space. The loud speakers forms a plurality of sound sources so that sounds from these loud speakers propagate in the thus defined virtual space to affect the audience. Consequently, by calculating the above potential to determine the transient response signal thus affecting the audience, it is possible to obtain the same potential as that of the transient response signal actually affecting the audience in the movie theater.

For example, in Fig. 15: a sound ray vector of a direct sound from the loud speaker SP1 affecting a listener P is denoted by the reference character D_0 ; an incident sound ray vector incident on the boundary corresponding to a wall W is denoted by the reference character D_1 ; and, a velocity potential incident on the listener P is denoted by the reference character ϕ . The

velocity potential ϕ forms the component of the entire transient response signal, and is derived from the incident sound ray vector D1 reflected from a micro area such as shown in Fig. 5(b) on the boundary.

5 Now, reproduction of the sound of the movie theater in the real space will be described. It is possible to realize such sound reproduction by synthetically producing a three-dimensional virtual sound field of the movie theater in a sound field formed in a real space. Incidentally, the sound field thus synthetically
10 produced in the real space is hereinafter referred to as the synthetic sound field.

 The virtual space is superimposed over the real space in a manner such that a position of the listener P in the real space is coincident with a position of the audience P in the virtual space.
15 Shown in Fig. 16(a) is a real space 144. A listener 142 in the real space 144 is superimposed in position over the audience P shown in Fig. 15. Disposed in the real space 144 are four pieces of loud speakers denoted by the reference numerals 140a, 140b, 140c and 140d. As shown in Fig. 16(a), the listener 142 is
20 surrounded by these loud speakers. Suppose that the sound based on the potential affecting the audience P shown in Fig. 15 can be reproduced by the use of these four loud speakers 140a, 140b, 140c, 140d. The listener would be able to enjoy the sound of the virtual space. Incidentally, only required in setting of the real
25 space are the relative positions of the listener 142 and the loud

speakers 140a, 140b, 140c, 140d relative to each other. Based on information of these relative positions, the synthetic sound field is produced with respect to the virtual space.

5 In Figs. 16(a) and 16(b), the potential " $\phi(1, 1)$ " shown in Fig. 15 is illustrated in detail. This potential " $\phi(1, 1)$ " is divided into a plurality of pieces allocated to the individual loud speakers 140a, 140b, 140c, 140d. As shown in Fig. 16(a), since the potential " $\phi(1, 1)$ " reaches the listener 142 by passing through a space between the loud speakers 140a and 140b, the
10 potential " $\phi(1, 1)$ " is divided into two pieces allocated to these loud speakers 140a, 140b.

Now, with reference to Fig. 16(b), such method for dividing the potential " $\phi(1, 1)$ " will be described.

In Fig. 16(b): the line a-a' passes through a center of
15 the loud speaker 140a and a center of the listener 142; and, the line b-b' passes through a center of the loud speaker 140b and a center of the listener 142. The line a-a' intersects the line b-b' at a terminal point " ϕe " of the potential " $\phi(1, 1)$ " which has a starting point " ϕb ". One of the lines a-a', b-b' is moved
20 parallel to its original position to form the line c-c' which intersects the other of the lines a-a', b-b' at an intersection point G. In this case, the line " ϕb "-G forms a vector having a potential parallel to the line a-a', which potential is used at the loud speaker 140a. On the other hand, the line G-" ϕe " forms a
25 vector having a potential parallel to the line b-b', which

potential is used at the loud speaker 140b.

Such dividing operation described above is conducted with respect to all the potentials, which enables the transient response signal affecting the audience into the plurality of pieces allocated to the four loud speakers 140a, 140b, 140c, 140d. Incidentally, although the potentials used here are determined through the approximate sound integral, it is also possible to use the potentials determined through another suitable calculus. Further, the above dividing operation is a conventional mathematical dividing operation in vector. Consequently, even in case that the loud speakers in the real space is three-dimensionally disposed, it is possible to determine the potentials for the individual loud speakers by conducting the three-dimensional mathematical dividing operation in vector when each of the potentials pass through a space defined by three pieces of the loud speakers.

Then, a method for reproducing an actual sound by the use of the transient response signal, which is prepared by summing up the potentials allocated to the individual loud speakers, will be described with reference to Fig. 17.

Shown in Fig. 17 are: a sound output unit 150 with fourteen Dolby surround output channels SP1 to SP14; fifty-six sum-of-products calculators 152 with convolution-calculus function comprising A/D converters and D/A converters; a mixer 154 for mixing the outputs from the sum-of-products calculators 152 to

prepare signals supplied to the individual loud speakers; four
amplifiers 156 for the individual loud speakers; and, four loud
speakers 140a, 140b, 140c, 140d. Incidentally, in the real space
144, there is also provided a television set disposed in front of
5 the listener 142, the television set being used to reproduce the
picture of a movie.

The sound issued from the loud speakers for the channel
SP1 of the sound output unit 150 shown in Fig. 15 propagates in
various modes in the virtual space to affect the audience P
10 through its various velocity potentials. In this case, each of
such potentials are allocated to each of the four loud speakers,
as described with reference to Fig. 16(a). Consequently, such
potentials allocated to the four loud speakers are summed up with
respect to each of the loud speakers, so that four pieces of the
15 transient response signals for the individual loud speakers are
prepared and stored in a first to a fourth one of the
sum-of-products calculators 152, respectively. As is in the above
channel SP1, in each of the remaining channels SP2 to SP14 of the
sound output unit 150, there are provided four pieces of the
20 sum-of-products calculators 152 for the individual loud speakers.
Consequently, the necessary number of the sum-of-products
calculators 152 for reproducing all the transient response signals
is fifty-six (56), because $14 \times 4 = 56$, where 14 is the number of
the sound sources, i.e., loud speakers in the virtual space; and,
25 4 is the number of the actual loud speakers for sound

reproduction.

The output signal issued from the sound output unit 150 to the sum-of-products calculators 152 is processed in these calculators 152 and issued therefrom as a virtual sound signal, according to the process having been already described with reference to Figs. 8 and 9. The virtual sound signal thus issued from the sum-of-products calculators 152 is then inputted to the mixer 154 and subjected therein to mixing operations for the individual loud speakers. After that, the sound signal thus subjected to the mixing operations is issued from the mixer 154 to the amplifiers 156, and amplified therein. The thus amplified sound signal is supplied from the amplifiers 156 to the individual loud speakers 140a, 140b, 140c, 140d, as shown in Fig. 17, so that the sound is reproduced by these loud speakers.

As described above, according to the present invention, it is possible for the listener 142 to enjoy the sound in the Dolby surround mode as if he were in the movie theater. In the above embodiment of the present invention, there are used four pieces of the loud speakers for reproducing the sound. However, it is also possible for the present invention to use any other desired number of the loud speakers such as two, eight channels or more. Further, in the above embodiment of the present invention, although the loud speakers are disposed in a plane, i.e., are two-dimensionally arranged, it is also possible for the present invention to three-dimensionally dispose the loud speakers, allocated to each

of which speakers thus three-dimensionally disposed is a piece of the velocity potential affecting the observation point, the piece being prepared by dividing the velocity potential. Such division of the velocity potential may be performed through the
5 mathematical division in vector which is publicly well known as described above.

Incidentally, in place of the central processing units which actually perform the convolution calculuses being performed in the sum-of-products calculators 152, it is possible to use the
10 multi-tap delay circuit 110 which is already shown in Fig. 13 and more inexpensive. By the use of such inexpensive multi-tap delay circuit 110, it is possible for the listener to enjoy the sound at his home in the Dolby surround mode for the movie theater.

As described above, in order to reproduce the sound
15 existing in the movie theater and like halls by means of the loud speakers of the real space (hereinafter referred to as the reproducing loud speakers), it is necessary to: define a desired movie theater and like halls as a virtual space; determine the transient response signal for each of the reproducing loud
20 speakers of the real space, the transient response signal being derived from the transient signal which is generated at the sound source by the loud speakers of the movie theater (hereinafter referred to as the installation loud speakers so as to be distinguished from the reproducing loud speakers) to affect the
25 listener as the transient response signal; and, perform the

convolution calculuses with respect to the stereo sound or
Dolby-surround sound signals allocated to the individual
installation loud speakers. The above is adequate for sound
reproduction by means of the reproducing loud speakers of the real
5 space so as to have the listener enjoy at his home the sound
existing in the movie theater and like halls in the virtual space.

Further, the present invention is effective also in
preparing a new sound source. For example, in a scene of a movie,
in order to reproducing sounds affecting the listener who enjoys
10 the movie in an acoustic environment in which the sounds caused by
people, animals and other objects affect the listener, it is
necessary to: define such scene of the movie as a virtual space;
define the positions of the people, animals and other objects in
the acoustic environment as the positions of sound sources in the
15 virtual space; and, apply the same synthetic process as that
described in the above to the virtual space with respect to the
real space. Fig. 18(a) shows a position relationship between the
reproducing loud speakers 140a to 140d and the listener in the
real space when the loud speakers SP disposed in the virtual space
20 of the movie theater and like halls described above serve as the
sound sources. Fig. 18(b) shows a position relationship between
the reproducing loud speakers 140a to 140d and the listener in the
real space when a scene of a movie is formed in the virtual space
with an acoustic environment in which people and animals serve as
25 the sound source.

A difference between Fig. 18(a) and Fig. 18(b) is that the sound sources of the virtual space in Fig. 18(a) are the loud speakers thus fixedly installed, while the sound sources of the virtual space in Fig. 18(b) are the people and the animals both movable. However, since any of the loud speakers thus fixedly installed and the people and animals both movable forms the sound source in the virtual space, it is possible to carry out the present invention in both cases shown in Figs. 18(a) and 18(b). As for each of the movable sound sources such as the people and the animals in the virtual space, it is necessary to calculate the position of each of such movable sound sources in the virtual space every desired time interval so as to determine the transient response signals every desired time interval, which transient response signals are derived from the sounds generated in the thus calculated positions to affect the listener. Consequently, in the present invention: their potentials are determined every desired time interval; and, then, based on the thus determined potentials, the transient response signals are generated every desired time interval according to the process described above. The important thing here is that application of the movable sound sources in the present invention may realize a means for reproducing a realistic sound in computer graphics that is now already in the limelight, which makes it possible to automatically produce and edit the sounds in future movie production processes.

Also provided in the present invention is a dynamic sound

which makes it more effective to change a plurality of the virtual spaces. Now, a method of sound reproduction in the plurality of the virtual spaces, which are changed according to the scenes of the movie and not limited to the virtual space of the movie theater, will be described.

Fig. 19 shows the relationship between an acoustic virtual space 158 varying with respect to time and a real space 144 in such acoustic virtual space 158. More particularly, at a time T1, the listener 142 in the real space locates himself in a substantially central portion of an elongated virtual space in a condition in which the real space 144 is combined with the virtual space 158 in a manner such that: the four loud speakers 140a, 140b, 140c, 140d installed in the real space 144 in which the listener 142 stays reproduce the sounds issued from the virtual space 158 surrounding the real space 144. On the other hand, at a time T2, the real space 144 is combined with a corner portion of the virtual space surrounded by a wall assuming a substantially square shape in plan view. Further, at a time T3, the real space is combined with an upper portion of the virtual space with a considerable height. The virtual space described above is set according to the scenes of the movie. As described above, it is possible for the listener staying in the real space to enjoy the optimum sound adapted to every scene of the movie when the real space is combined with a desired portion of the virtual space, the desired portion varying with respect to time. Consequently, for

example, in a wide scene of the movie such as a prairie scene and the like, it is possible for the listener to enjoy such a wide feeling through the sound thus reproduced together with the wide scene of the movie. In another case, for example, it is also possible to produce a movie in a manner such that: the more the scene of the movie becomes serious, the more the virtual space is enlarged to increase its reverberation effect so as to increase the seriousness in the scene.

Fig. 20 shows a block diagram of an apparatus for reproducing a sound according to the scene of the movie. In the apparatus shown in Fig. 20, an input signal 172 comprising information of both a sound signal and a synchronizing signal synchronized with the sound signal is inputted to an input portion 160 of the apparatus. The information as to the synchronizing signal is used to synchronize the reproduced sound with the scene of the movie. In the input portion 160, the input signal 172 is demodulated or sampled in frequency within a predetermined band to obtain a sound signal 180 and a synchronizing signal 174 each of which is divided and inputted to both a D/A converter 168 and a control portion 162. The reason why the D/A converter 168 is shown in dotted line in Fig. 20 is that it is necessary to show that the D/A converter 168 is not required when the input signal 172 is of a digital data. On the other hand, according to the synchronizing signal 174 thus inputted in the control portion 162, the control portion 162 issues a control signal 176 to both a memory portion

164 and a central processing unit portion 166. Previously stored in the memory portion 164 are a plurality of the transient response signals being used by the central processing unit portion 166. Of these transient response signals, one specified by a control signal issued from the control portion 162 is retrieved from the memory portion 164, so that a predetermined one of the transient response signals 178 depending on the scene of the movie is issued from the memory portion 164 in real time. Incidentally, in the method for specifying the predetermined one of the transient response signals in the above, for example, it is also possible to sequentially retrieve the transient response signals upon occurrence of the synchronizing signal. Alternatively, it is also possible that the synchronizing signal comprises address information for identifying a predetermined one of the transient response signals, through which information the predetermined transient response signal is specified. In the central processing unit portion 166, according to the control signal 176 issued from the control portion 162, an additionally issued one of the transient response signal 178 is set. The central processing unit portion 166 has two functions which are: the substantially same convolution calculus function as that of the central processing unit portion 76 shown in Fig. 8; and, a setting function in setting an arbitrary one of the transient response signals 178 issued from the memory portion 164, according to the control signal 176. It is easy for those skilled in the art to realize

these setting function and the convolution calculus function by means of a commercially available DSP (i.e., digital signal processor). In the central processing unit portion 166 in which the predetermined transient response signal is thus already set, a
5 digital sound signal 182 is subjected to a convolution calculus based on the transient response signal according to the processes which have been already described in the above with reference to Figs. 8 and 9. The resultant signal thus obtained through the convolution calculus is then issued from the central processing
10 unit portion 166 to the A/D converter 170, as shown in Fig. 20. Consequently, the sound reproduction signal 186 issued from the A/D converter 170 represents a sound corresponding to a virtual space which is set according to each scene of the movie. This sound reproduction signal 186 (hereinafter also referred to as the
15 virtual space sound) is then issued to the loud speakers through the amplifiers so that the sound is reproduced, which enables the listener to enjoy the sounds corresponding to the individual scenes of the movie.

Although described in the above is a setting of the
20 transient response signal corresponding to the virtual space which varies with respect to time, it is apparent that the present invention is also applicable to another case in which the transient response signal itself may vary together with the virtual space, as is in case that the sound sources are moved with
25 respect to time.

Incidentally, in the today's widely used movie video tape LD and the like, there is no synchronizing signal. Consequently, in picture reproduction in this case, time is counted from a start of such picture reproduction, so that the control portion 162

5 controls the previously stored transient response signals so as to be issued from the memory portion 164 to the central processing unit portion 166 according to the scene of the movie varying with respect to the time thus counted. Consequently, in this case, the input portion 160 is negligible. In other words, a trigger signal

10 generated at the start of picture reproduction is inputted to the control portion 162, so that the control portion 162 starts time counting according to the trigger signal. After the elapse of a predetermined period of time, the control portion 162 issues a control signal to both the memory portion 164 and the central

15 processing unit portion 166.

As for the various transient response signals stored in the memory portion or means 164 in the above, in case that a video signal is reproduced through signal transmission from a remote location, it is apparent that the transient response signals are

20 transmitted to the memory portion 164 prior to transmission of the video signal.

According to the development of multimedia in future, appearance of digital television broadcasting, and the development in transmitting and receiving techniques of video/sound signals

25 through data compression/encode between computers, the input

portion 160 may be properly modified in design. For example, it is possible to modify the input portion 160 as follows with respect to MPEG2 standards in the ISO standards.

The concept of signal transmission using the MPEG2 standards will be now described with reference to Fig. 22. A video signal and a sound signal are separately inputted to different encoders which issue encode signals separately. Each of the encode signal is packed for its packet transmission to both a transport/stream (TS) multiplexing unit of the MPEG2 and a program/stream (PS) multiplexing unit of the MPEG2, issued from each of which units is an actual stream signal (TS or PS). TS packet comprises in construction: a packet header; an adaptation field; and/or, a bay load (i.e., data portion). The adaptation field comprises time information, i.e., a standard time of a program called PCR (i.e., Program Clock Reference). On the other hand, the bay load is used to issue various data therefrom. Consequently, by using such adaptation field and the bay load (i.e., data portion), it is possible to use a signal for synchronization, which makes it possible to perform the above-described synchronized control. Incidentally, in this case, since the input signal is a packed stream signal, it is naturally necessary to use a function for decoding the actual video signal and also necessary to separate the signal for synchronization, which may be realized by properly modifying the decoding/sampling control according to a transmitted protocol. In other words, the

present invention is characterized in that the present invention provides a more dynamic acoustic piece of information by arbitrarily changing the virtual space with respect to time according to each scene of a movie, and, therefore is not limited
5 to the block diagram shown in Fig. 20.

The block diagram shown in Fig. 21 may be also used, provided that it achieves the objects of the present invention. A difference between the block diagram shown in Fig. 20 and that of Fig. 21 is that the latter lacks the memory portion so that the
10 transient response signal 178 is issued from the control portion 162 directly to the central processing unit portion 166. In this case, the transient response signal for each scene of a movie is received by the input portion 160 through an input signal 192 comprising a video signal and a sound signal both issued from a
15 broadcasting station, a Karaoke master station, a picture CD, or the like. In the input portion 160, the input signal 192 thus received therein is then divided into a sound signal 180 and another signal 194 comprising the transient response signal 178, so that the transient response signal 178 is supplied to and set
20 in the central processing unit portion 166 through the control portion 162 under control performed by the control signal 176 issued from the control portion 162 to the central processing unit portion 166.

Incidentally, the central processing unit portion 166
25 shown in Figs. 20 and 21 may be replaced with a proper means

having the same convolution calculus function as that of the central processing unit portion 100 with a simplified function shown in Fig. 13, which may realize an inexpensive three-dimensional sound reproduction in an easy manner.

5 Consequently, it is possible for the present invention to provide a consumer products for home use. Particularly, since a sound output accompanied with a moving picture as is in a home-use video instrument and computer is of a two-channel mode, the sound may be reproduced by four loud speakers using the central processing unit
10 portion 166 provided with eight pieces of the sum-of-products calculators, the number of which calculators is considerably less than that (i.e., fifty-six pieces) of the sum-of-products calculators shown in Fig. 13, and, therefore makes it possible to apply the present invention to the consumer products.

**CLAIMS :**

1. A method of reproducing sound by using transient response signals each representing acoustic characteristic of the sound generated at a sound source in virtual space on an observation point,
5 wherein:

each said transient response signal is produced as to each phase of the virtual space, the sound source or the observation point which are variable in size of the space, in location of the sound source or in location of the observation point.
10

2. A method of reproducing sound according to claim 1, further comprising the steps of:

storing a plurality of the transient response signals; and
setting corresponding one of the transient response signals
15 to a convolution processor according to each phase of the variable virtual space, the sound source location or the observation point location, when the sound is reproduced.

3. A method of reproducing sound according to claim 1, further comprising the steps of:

20 storing a plurality of the transient response signals;
adding a synchronizing signal to sound signal for identifying each transient response signal to be used; and

setting corresponding one of said stored transient response signals to a convolution processor according to said added
25 synchronizing signal when the sound is reproduced.

4. An apparatus of reproducing sound by using transient response signals each representing acoustic characteristic of sound generated at a sound source in virtual space on an observation point, which comprises;

30 convolution means having input portion and output portion for performing convolution calculus of sound signal inputted through the input portion according to a transient response signal set to the convolution means, and for outputting the calculated sound signal through the output portion; and

35 sound output means connected to the output portion of the



convolution means for receiving the calculated sound signal, and outputting sound to be reproduced.

5. An apparatus of reproducing sound according to claim 4, wherein:

5 said input portion is adapted to input a synchronizing signal transmitted together with the inputted sound signal, said synchronizing signal being adapted to be used for identifying one of a plurality of the stored transient response signals; further comprising:

10 means for storing a plurality of transient response signals; and

 control means for retrieving a predetermined one of said plurality of transient response signals from said storing means according to the inputted synchronizing signal, and setting said
15 predetermined one to said convolution means for reproducing sound varying according to said predetermined transient response signal.

6. An apparatus of reproducing sound according to claim 4, wherein:

 said input portion is adapted to input a transient response
20 signal transmitted together with the inputted sound signal, the transient response signal representing an acoustic characteristic of the sound to be reproduced; and

 said transient response signal is set to said convolution means for performing convolution calculus of the sound signal
25 inputted through the input portion according to the transient response signal.