

DIGITAL AUDIO SIGNAL PROCESSING METHOD AND SYSTEM FOR GENERATING AND CONTROLLING DIGITAL REVERBERATIONS FOR AUDIO SIGNALS

Field of the Invention

[0001] The present invention generally relates to digital audio signal processing technologies, and more particularly to a digital audio processing method and system with a reverberation device for generating and controlling artificial reverberations for audio signals, wherein the reverberation device has a uniform structure and the generated artificial reverberations have the characteristics extracted from real environments.

Background of the Invention

[0002] Artificial reverberations are often used for dry audio contents to simulate effects of real environments. In many applications such as headphone and speaker playbacks, artificial reverberations are added to give the listeners a sense of being in real environments.

[0003] In nature, reverberations are echoes from various reflections in real environments, such as a room. The ideal way of generating reverberations will be convolving the audio signal with the impulse response of the desired environment. Such a method in practice is computationally costly. In a digital signal processing application, it takes huge computational and storage resources to implement this method. To reduce the cost, U.S. Pat. 5,317,104 discloses an electronic sound processor for creating reverberation effect by convolving random white noise with dry audio signals to simulate the late part of the reverberation.

[0004] A number of methods have been proposed to approximate the exact reverberation or to create only the salient signals. Most of the algorithms use feedback loops with delay lines, sometimes combined with allpass filters. U.S. Pat. 4,181,820 discloses an electric reverberation apparatus that comprises a plurality of loops having different delay times and adapted to form sound repetitions of diminishing intensity, wherein the loops are provided with tappings each of which has a particular delay time

associated with it. U.S. Pat. 5,621,801 discloses a reverberation effect imparting system that includes plural comb filters, each of which has a signal delay line and a feedback loop for filtering a delayed output signal from the delay line and feeding the filtered signal back to the input side with a variable loop gain. The drawback of such feedback systems is that they will create resonates thus colorizes the sound. The problems are overcome by phase-shifting or time-variant delay lines in some algorithms, which introduce certain undesired pitch shifting effects. See, U.S. Pats. 4,955,057; 5,740,716. Some use only delay lines and feed forward loops, tapping at different locations of the delay lines. See, U.S. Pat. 5,555,306. Some other algorithms separate the reverberations to early and later parts and generate them separately. See, U.S. Pats. 5,040,219; 5,146,507. This will lead to a sudden increase of echo density at the boundary, which is not true in a nature environment.

Summary of the Invention

[0005] Therefore, one embodiment of the present invention provides a reverberation generator for generating reverberations of a digital audio signal input in a digital audio signal processing system to simulate a real environment. In the embodiment, the reverberation generator comprises an input means for receiving the digital audio signal input; a summing means for generating a digital audio signal output containing the digital audio signal input and reverberations; a digital audio signal direct path electronically connected to the input means and the summing means for transmitting the digital audio signal input directly; and a plurality of feed forward loops configured in a cascade manner for generating reverberations; wherein the output of each loop is provided to the summing means and at the same time to the next loop; thereby the plurality of feed forward loops generate the reverberations simulating the real environment that are combined with the digital audio signal input to produce the digital audio signal output. In another embodiment of the reverberation generator, each of the feed forward loops comprises a gain stage, a delay line, an allpass filter, and a lowpass filter. In a further embodiment of the reverberation generator, the allpass filter comprises an input adder for summing up the input to the allpass filter and a feedback from a delay line, wherein the delay line is electronically downstream of the input adder; a feedback loop with a feedback amplifier (-a) for using the output of the delay line as the feedback to the input adder; a feed forward

loop with an amplifier (a) electronically connected to the input adder; and an output adder for summing up the outputs from the delay line and the feed forward loop. In yet another embodiment of the reverberation generator, the values of both -a and a are between about 0.6 and about 0.7. In yet another embodiment of the reverberation generator, the length of the delay line in the first allpass filter is approximately equal to the delay time between the first echo and the second echo.

[0006] In another embodiment of the reverberation generator, the lengths of all the delay lines and allpass filters are preferably prime numbers. In another embodiment of the reverberation generator, the length of the delay lines in the allpass filters except for the first allpass filter is calculated by the following equations:

$$AP_{n+1} \approx AP_n * y$$

where AP_n is the length of the delay line in the nth allpass filter; y is the environment coefficient; and the value of y is from about 1.1 to about 1.5.

[0007] In another embodiment of the reverberation generator, the delay line in the first loop is approximately equal to the delay time between a direct signal and its first echo. In another embodiment of the reverberation generator, the length of the delay line in any loop except for the first loop is calculated by the following equation:

$$DL_{n+1} \approx DL_n * x$$

where DL_n is the length of delay line in the nth loop; x is the environment coefficient; and the value of x is from about 1.1 to about 1.5.

[0008] In another embodiment of the reverberation generator, the delay lines used in the feed forward loops and allpass filters are preferably realized by circular buffers in digital signal processing.

[0009] In another embodiment of the reverberation generator, the gain is calculated by following equations:

$$G_1 = \sqrt{\frac{dE}{dt} * DL_1}$$

$$G_{n+1} = G_n \sqrt{\frac{dE}{dt} * DL_{n+1}}$$

where G_n is the gain for the nth loop and DL_n is the length of the delay line in the nth loop.

[0010] In another embodiment of the reverberation generator, the gain value in the first loop varies from about 0.2 to about 0.5; and the gain values in subsequent loops vary

from about 1 to about 2. In another embodiment of the reverberation generator, the lowpass filters are preferably FIR and IIR filters, and more preferably first order IIR filters.

[0011] Another embodiment of the present invention provides a digital audio signal processing system for generating and controlling reverberations simulating real environments for a digital audio signal input. In the embodiment, the digital audio signal processing system comprises a digital I/O interface for inputting and outputting digital audio signals; a controlling unit electronically connected to the digital I/O interface for receiving the input digital audio signals; and a reverberation generator electronically connected to the controlling unit; wherein the controlling unit receives reverberation characteristics of a digital audio signal of the real environment; and wherein the received reverberation characteristics will control the configuration of the reverberation generator so as to generate the reverberations for the digital audio signal input to simulate the real environment.

[0012] In another embodiment, the digital audio signal processing system further comprises a controlling mechanism that extracts reverberation characteristics of the digital audio signal when the digital audio signal contains reverberations from a real environment; wherein the reverberation characteristics include one or more of the following ones: final echo density, rate of the echo density to be built up, decay rate of overall energy level of the echoes, and differential decay rates of high-frequency signals and low-frequency signals.

[0013] In another embodiment of the digital audio signal processing system, the controlling unit receives one or more of the following reverberation characteristics: final echo density, rate of the echo density to be built up, decay rate of overall energy level of the echoes, and differential decay rates of high-frequency signals and low-frequency signals.

[0014] In another embodiment of the digital audio signal processing system, the reverberation generator comprises an input means for receiving the digital audio signal input; a summing means for generating a digital audio signal output containing the digital audio signal input and reverberations; a digital audio signal direct path electronically connected to the input means and the summing means for transmitting the digital audio signal input directly; and a plurality of feed forward loops configured in a cascade manner for generating reverberations; wherein the output of each loop is provided to the summing

means and at the same time to the next loop; thereby the plurality of feed forward loops generate the reverberations simulating the real environment that are combined with the digital audio signal input to produce the digital audio signal output.

[0015] In another embodiment of the digital audio signal processing system, the reverberation characteristics control the reverberation generator in the following manner: the final echo density determines the number of feed forward loops; the rate of echo density to be built up determines the delay lengths of delay lines; the decay rate of the overall energy of echoes determines the gains in each loop; and the differential decay rates of high-frequency signals and low-frequency signals determines the cutoff frequencies and roll off rate of lowpass filters.

[0016] Another embodiment of the present invention provides an article of manufacture having the capacity of generating reverberations for digital audio signals to simulate real environments. In the embodiment, the article of manufacture comprises a computer-readable medium with a memory domain for storing files and programs, and a computer-executable domain for enabling the article to perform the computer programs; and a digital audio signal processing system for generating and controlling reverberations simulating real environments for a digital audio signal input, wherein the digital audio signal processing system is embedded in the computer-readable medium; wherein the digital audio signal processing system comprises: a digital I/O interface for inputting and outputting digital audio signals; a controlling unit electronically connected to the digital I/O interface for receiving the input digital audio signals; and a reverberation generator electronically connected to the controlling unit; wherein the controlling unit receives reverberation characteristics of a digital audio signal of the real environment; and wherein the received reverberation characteristics will control the configuration of the reverberation generator so as to generate the reverberations for the digital audio signal input to simulate the real environment.

[0017] In another embodiment, the article of manufacture further comprises a controlling mechanism that extracts reverberation characteristics of the digital audio signal when the digital audio signal contains reverberations from a real environment; wherein the reverberation characteristics include one or more of the following ones: final echo density, rate of the echo density to be built up, decay rate of overall energy level of the echoes, and differential decay rates of high-frequency signals and low-frequency signals.

[0018] In another embodiment of the article of manufacture, the article includes MP3 players, handphones, portable players, TVs, DVD players, and the like.

[0019] Another embodiment of the present invention provides a method for generating reverberations for a digital audio signal to simulate real environments. In the embodiment, the method comprises the following operations: receiving the reverberation characteristics of the digital audio signal for a real environment; translating the extracted reverberation characteristics into controlling parameters for a reverberation generator with a plurality of feed forward loops configured in a cascade manner; and generating the reverberations using the controlling parameters to control the reverberation generator.

[0020] In another embodiment of the method, it further optionally comprises the following operation: extracting the reverberation characteristics of the digital audio signal when the digital audio signal contains reverberations from a real environment; wherein the reverberation characteristics include one or more of the following: final echo density, rate of the echo density to be built up, decay rate of overall energy level of the echoes, and differential decay rates of high-frequency signals and low-frequency signals.

[0021] In another embodiment of the method, the reverberation characteristics include one or more of the following: final echo density, rate of the echo density to be built up, decay rate of overall energy level of the echoes, and differential decay rates of high-frequency signals and low-frequency signals.

[0022] In another embodiment of the method, each of the feed forward loops comprises a gain stage, a delay line, an allpass filter, and a lowpass filter. In another embodiment of the method, the allpass filter comprises: an input adder for summing up the input to the allpass filter and a feedback from a delay line, wherein the delay line is electronically downstream of the input adder; a feedback loop with a feedback amplifier for using the output of the delay line as the feedback to the input adder; a feed forward loop with an amplifier electronically connected to the input adder; and an output adder for summing up the outputs from the delay line and the feed forward loop.

[0023] In another embodiment of the method, the delay line in the first loop is approximately equal to the delay time between a direct signal and its first echo.

[0024] In another embodiment of the method, controlling parameters of the reverberation generator include number of feed forward loops, lengths of the delay lines,

gain used in the feed forward loops, and cutoff frequencies and roll off rates of the lowpass filters.

[0025] In another embodiment of the method, the reverberation generator generates reverberations in the following controlled manner: controlling the final echo density by the number of feed forward loops; controlling the rate of the echo density to be built up by the lengths of the delay lines used in the feed forward loops and allpass filters; controlling the decay rate of the overall energy level of the echoes by the gain used in the feed forward loops; and controlling the decay of high frequency signals with respect to low frequency signals with the cutoff frequencies and roll off rates of the lowpass filters.

[0026] These and additional objectives and advantages of the invention will become apparent from the following detailed description of preferred embodiments thereof in connection with the accompanying drawings.

Brief Description of the Drawings

[0027] Preferred embodiments according to the present invention will now be described with reference to the Figures, in which like reference numerals denote like elements.

[0028] FIG 1 is a schematic block diagram illustrating components of a typical digital audio signal processor.

[0029] FIG 2 shows a typical amplitude response of an audio signal in a real environment.

[0030] FIG 3 is a schematic function block diagram of the controlling mechanism of the reverberation-generating process of a digital audio signal processing system in accordance with one embodiment of the present invention.

[0031] FIG 4 is a schematic block circuit diagram illustrating the allpass filter used in the digital audio signal processor for the generation of reverberation in accordance with one embodiment of the present invention.

[0032] FIG 5 is a schematic block circuit diagram of a reverberation generator used in the digital audio signal processing system in accordance with one embodiment of the present invention.

[0033] FIG 6 is a schematic functional diagram of an electronic audio device illustrating the applications of the digital audio signal processor in accordance with one embodiment of the present invention.

[0034] FIG 7 is a flowchart of generating reverberations for a digital audio signal in accordance with one embodiment of the present invention.

Detailed Description of the Invention

[0035] The present invention may be understood more readily by reference to the following detailed description of certain embodiments of the invention.

[0036] Throughout this application, where publications are referenced, the disclosures of these publications are hereby incorporated by reference, in their entireties, into this application in order to more fully describe the state of art to which this invention pertains.

[0037] Most of the modern reverberation generation methods use digital signal processors (DSP), which have limited computational and memory resources. FIG 1 is a schematic block diagram illustrating components of a typical digital audio signal processor. The digital audio signal processor 1 comprises a digital I/O interface 2 for inputting and outputting the audio data, a data bus 3 for transporting audio data within the processor and interconnecting with peripherals, a memory unit 4 for storing the input audio data and intermediate data from the executions of the processor, a computational unit 5 for loading the audio data and program data to host registers 6 and performing the processing then storing the processed audio data back to the I/O interface 2 for output. The memory unit 4 comprises RAM, ROM, DMA, and I2C where the computational unit executes its programs and stores all the data. The computation unit 5 comprises ALU, MAC and Shift for performing additions, subtractions, multiplications, and other operations. It is well known that multiplications usually need more resources, and short filter lengths and fewer multiplications will save the load of the processor. The digital audio signal processor 1 further comprises a controller 7 that is usually present to control the processor through host registers which are interfaced with the computational unit through data bus. In addition, the controller 7 is connected to a User Interface so that the user of the processor could input its instructions to the processor. Furthermore, the digital signal processor comprises

a peripheral interface **8** through which the processor can interact with other components of an audio processing system. The peripheral can be devices including, but not limited to, keyboards and mice.

[0038] Now referring to FIG 2, there are provided the illustrative amplitudes of a direct signal and its reverberations in a time domain in a real environment such as a room. It is apparent that the direct signal reaches a listener's ears first and is followed by the echoes caused by reflections of floor, walls, ceiling and other surfaces. The characteristics of the echoes will be discussed in detail hereinafter. It is to be noted that the echoes do not change their pitches.

[0039] As illustrated in FIG 2, the reverberation shows certain general characteristics including the followings: that the early echoes are quite sparse after the direct sound; that the density of the echoes increases in the time domain; and that in the late part of the reverberation in the time domain, the echoes become increasingly diffused and dense. However, to simulate the reverberations, a reverberation model has to be established by extracting certain peculiar characteristics of the reverberations in each type of real environments. The peculiar characteristics considered in the present invention include final echo density, rate of echo density to be built up, decay rate of the overall energy of echoes, and differential decay rates of high frequency signals and low frequency signals. For example, in a room, the final echo density and the rate of echo density to be built up depend on the size of the room. The smaller the room is, the faster the density of the echoes will be built up. Furthermore, the rate of decay of the overall energy level of the echoes depends on the absorption of the surfaces. In addition, the reflection surfaces generally absorb more high-frequency signals than low-frequency signals. As a result, the high-frequency signals decay faster than do the low-frequency signals. How fast the high-frequency signals decay with respect to the low-frequency signals depends on the surfaces of reflections.

[0040] Now referring to FIG 3, there is provided a schematic function block diagram of the controlling mechanism of the reverberation-generating process of a digital audio signal processing system in accordance with one embodiment of the present invention. As shown in FIG 3, the digital audio signal processing system **10** comprises a digital I/O interface **11**, a core processor **12**, and a controlling unit **13**. The digital I/O interface **11** and the core processor **12** are very similar or identical to the ones shown in

FIG 1, thus no detail description herein. The controlling unit **13** may be electronically connected to the controller **7** of FIG 1 to control the reverberation generating process.

[0041] Still referring to FIG 3, there is provided a more detailed description of the operation of the controlling unit **13**. First, extract the peculiar reverberation characteristics of an audio signal from the audio signal reverberations of one real environment to be simulated. The peculiar reverberation characteristics include final echo density **14a**, rate of the echo density to be built up **14b**, decay rate of overall energy level of the echoes **14c**, and differential decay rates of high-frequency signals and low-frequency signals **14d**.

[0042] Then, these reverberation characteristics are translated into controlling parameters. More specifically, the final echo density **14a** will be translated into the number of feed forward loops **15a**. The final echo density is the number of echoes of a given time duration at the tail of the response. The number of feed forward loops to be used is determined in the following manner: the denser the echoes to be built up, the more loops should the structure have. Generally, 3 or more loops are required to have the desired effects. Because of the diffusive nature of the late reverberation and the way human auditory system works, a reasonable close approximation for the final echo density will give sufficient sensation of the real environment when other controlling parameters are correctly set. Generally, an open space such as a square will have lower echo density and experiment shows 3 to 4 loops are sufficient for the simulation. An enclosed massy environment such as a market will have a high echo density and a minimum of 4 loops is necessary.

[0043] The rate of echo density to be built up **14b** will be translated into the delay lengths of delay lines **15b**. As discussed hereinafter, the delay lines used in the digital signal processing device include the delay lines used in the loops and the delay lines used in the allpass filters. The rate of the echo density to be built up is defined as the distance between the echoes. It is important for the simulation of the reverberation to have the first few echoes well generated because the human auditory system judges the environment depending very much on the first few echoes. As the echoes become more and more diffused in the later part of the reverberation, the distances between the consecutive echoes are of less importance to the human auditory system. The delay lengths of the delay lines used in the loops and the delay lines used in the allpass filters can be determined in the following manner: the longer the delay lengths, the slower the echo density will be built

up. The delay length of the delay line in the first loop (delay line 1) will be equal to the delay between the direct sound and the first echo. The delay length of the delay line in the first allpass filter (AP1) will be equal to the delay between the first echo and second echo. To simulate a large room like a church, the delay lengths in each delay line and each allpass filter will be relatively large. After the first loop, the delay lengths in the delay lines and allpass filters can be approximately calculated with the following equations respectively:

$$[0044] \quad DL_{n+1} \approx DL_n * x \quad \text{Equation 1}$$

$$[0045] \quad AP_{n+1} \approx AP_n * y \quad \text{Equation 2}$$

[0046] where DL_n is the length of delay line in the n th loop; AP_n the length of the delay line in the n th allpass filter; x and y are the environment coefficients. The values of x and y vary from about 1.1 to about 1.5. The lengths of the delay lines DL_n and AP_n are preferable to be prime numbers, which will ensure a smooth decay of the reflection sound without significant burst signals.

[0047] The decay rate of the overall energy of echoes **14c** will be translated into the gains in each loop **15c**. The decay rate of the overall energy level of the echo is defined by the reduction of the energy of the echoes given a time period, which can be expressed by $\frac{dE}{dt}$, where E represents the energy of the echo and t represents the time. For example, a room with carpet floor absorbs sound much better than wooden floor. This characteristic can be translated into the gains in each loop: the smaller the gains are, the faster the over energy level of the echoes decays. The gain can be approximately calculated by the following equations:

$$[0048] \quad G_1 = \sqrt{\frac{dE}{dt} * DL_1} \quad \text{Equation 3}$$

$$[0049] \quad G_{n+1} = G_n \sqrt{\frac{dE}{dt} * DL_{n+1}} \quad \text{Equation 4}$$

[0050] where G_n is the gain for the n th loop and DL_n is the length of the delay line in the n th loop. To simulate a room with higher absorption of sound, the gains in each loop will be small. Typically, the gain value in the first loop varies from about 0.2 to about 0.5. The gain values in subsequent loops vary from about 1 to about 2.

[0051] The differential decay rates of high-frequency signals and low-frequency signals **14d** will be translated into the cutoff frequencies and roll off rate of lowpass filters **15d**; the cutoff frequencies and roll off rates of the filters will determine how fast high-frequency signals decay with respect to low-frequency signals. For each environment, the decay rates of different frequencies vary. Generally, high frequency signals will be more absorbed by the reflection surfaces. The characteristics can be quantified as the relative difference in the change of energy of different frequencies. The mathematical expression for this characteristic is $\frac{dE_f}{df * dt}$, where E_f represents the energy for a certain frequency f .

This characteristic will be a very complex scenario to model. But in most of the cases, some lowpass filters can be used to have a reasonably close approximation due to the fact that high frequencies decay faster than low frequencies most of the time. The lowpass filters in each loop are used to simulate this characteristic. The lowpass filters can be realized by finite response filters (FIR) or infinite response filters (IIR). The cutoff frequencies and roll off rates of the filters will determine how fast high-frequency signals decay with respect to low-frequency signals. A simple implementation of such filter can be a first order lowpass filter as in the form of

$$\mathbf{[0052] \quad y_n = b * x_n - a * y_{n-1} \quad \text{Equation 5}}$$

[0053] where $a = 1 - b$. It should be understood by those who are skilled in the art that the lowpass filters can be implemented with different structures and methods, without being limited to the one this patent provides. The cutoff frequencies of the lowpass filter will be very specific environment dependent. The cutoff frequency for a typical room environment is recommended to be between 5000 Hz and 15000 Hz with the first order lowpass filter implementation provided.

[0054] Then, these parameters will be passed to a control unit controlling the core processor, which loads the input digital audio data from the I/O interface, performs the reverberation generation. The output signal including the reverberation generated is sent out through the I/O interface.

[0055] The method of the present invention for generating reverberations is unique because it gradually builds up the density of the reverberations and at the same time decays different frequency components discriminately. At the same time, other characteristics including the final echo density and the decay rate of the overall energy level will also be

controlled depending on the real environment characteristics. Therefore, the reverberations generated will closely match the characteristics of the real environments. Coloration of the sound is also minimized through the use of allpass filters and delay lines.

[0056] Now referring to FIG 4, there is provided a schematic block circuit diagram illustrating the allpass filter used in the digital audio signal processor for the generation of reverberation in accordance with one embodiment of the present invention. The allpass filter **20** comprises an input adder **21**, a delay line **22**, an output adder **23**, a feedback loop **24** with an amplifier (-a), and a feed forward loop **25** with an amplifier (a). The allpass filter **20** has a flat frequency response, thus introducing little coloration to the sounds. The values of both -a and a can be between about 0.6 and about 0.7.

[0057] Now referring to FIG 5, there is provided a schematic block circuit diagram of a reverberation generator used in the digital audio signal processing system in accordance with one embodiment of the present invention. The reverberation generator **30** comprises a plurality of feed forward loops **31, 32, 33, 34** configured in a cascade manner, and a summer **35**. Each of the feed forward loops comprises a gain stage, a delay line, an allpass filter shown in FIG 4 and a lowpass filter. The reverberation generator **30** uses the controlling parameters passed by the control unit to perform the generation process of reverberations for an input signal. The input signal is sent without manipulation to the summer **35** to simulate the direct signal in the output. The input signal is also to be sent to a first feed forward loop. The output of the first feed forward loop is sent to the summer **35** to simulate early reverberations in the output, and at the same time is used as the input of a second feed forward loop. The output of the second feed forward loop is sent to the summer **35** to simulate later-than-early reverberations in the output, and is used as the input of a third feed forward loop and so on. The output of the reverberation generator is the sum of the direct signal and all the outputs of the feed forward loops. The diagram only shows 4 feed forward loops, but the number of loops is not limited to 4 and can be changed when necessary. The delay line in the first loop is recommended to be equal to the delay time between the direct signal and the first echo. The delay lines used in the feed forward loops and allpass filters can be realized by circular buffers in digital signal processing. The lowpass filters can be realized by FIR and IIR filters, generally, first order IIR filters will be sufficient for most of the environments.

[0058] With this circuit, the direct and reverberation signals are generated. The gain in each loop controls the rate of decay of the overall energy level of the reverberation signals. The cascaded allpass filters will create dense echoes. With the delay lines used in each loop, the structure will create reverberations with increasing density of the echoes. The lowpass filters used in each loop will create the effect of faster decay of high-frequency signals.

[0059] Moreover, the computational cost of generating reverberations using the digital signal processing device of the present invention is reasonably low for the following reasons: the design involves very few multiplications; all the delay lines can be realized by circular buffers; and the lowpass filters can be as simple as first order IIR filters.

[0060] Now referring to FIG 6, there is provided a schematic functional diagram of an electronic audio device illustrating the applications of the digital audio signal processor in accordance with one embodiment of the present invention. The electronic audio device **40** comprises a memory domain **41** for storing all databases and enabling all computational executions, an audio media file database **42**, a decoder **43** for decoding all audio media files before each file is output, a controlling unit **44** for performing the controlling process of the reverberation generation, and a reverberation generator **45** for generating the reverberations according to the characteristics controlled by the controlling unit. The memory domain **41**, file database **42**, and decoder **43** are well known in the art. The electronics that can employ the digital audio signal processing system of the present invention further include handphones, portable players, TV, DVD player, and the like.

[0061] Now referring to FIG 7, there is provided a flowchart of generating reverberations for a digital audio signal in accordance with one embodiment of the present invention. The generation of reverberation **50** of an input digital audio signal **51** starts by choosing one real environment to be simulated and extracting the reverberation characteristics for the chosen environment **52**; then the reverberation generator is configured with the control of the reverberation characteristics (i.e., setting up the parameters of the reverberation generator including the number of feed forward loops, and the gains, delay lines, allpass filters, and low pass filters for each loop) **53**; then the simulated reverberation is generated **54** and output **55**. In the step of extracting reverberation characteristics, the extracted reverberation characteristics include the final echo density **14a**, the rate of the echo density to be built up **14b**, the decay rate of overall

energy level of the echoes **14c**, and the differential decay rates of high-frequency signals and low-frequency signals **14d**, as shown in FIG 3. The translation of the characteristics into controlling parameters of the reverberation generator has been discussed above.

[0062] While the present invention has been described with reference to particular embodiments, it will be understood that the embodiments are illustrative and that the invention scope is not so limited. Alternative embodiments of the present invention will become apparent to those having ordinary skill in the art to which the present invention pertains. Such alternate embodiments are considered to be encompassed within the spirit and scope of the present invention. Accordingly, the scope of the present invention is described by the appended claims and is supported by the foregoing description.

CLAIMS

What is claimed is:

1. A reverberation generator for generating reverberations of a digital audio signal input in a digital audio signal processing system to simulate a real environment, the reverberation generator comprising:
 - an input means for receiving the digital audio signal input;
 - a summing means for generating a digital audio signal output containing the digital audio signal input and reverberations;
 - a digital audio signal direct path electronically connected to the input means and the summing means for transmitting the digital audio signal input directly; and
 - a plurality of feed forward loops configured in a cascade manner for generating reverberations; wherein the output of each loop is provided to the summing means and at the same time to the next loop; thereby the plurality of feed forward loops generate the reverberations simulating the real environment that are combined with the digital audio signal input to produce the digital audio signal output.
2. The reverberation generator of claim 1, wherein each of the feed forward loops comprises a gain stage, a delay line, an allpass filter, and a lowpass filter.
3. The reverberation generator of claim 2, wherein the allpass filter comprises:
 - an input adder for summing up the input to the allpass filter and a feedback from a delay line, wherein the delay line is electronically downstream of the input adder;
 - a feedback loop with a feedback amplifier (-a) for using the output of the delay line as the feedback to the input adder;
 - a feed forward loop with an amplifier (a) electronically connected to the input adder; and
 - an output adder for summing up the outputs from the delay line and the feed forward loop.

4. The reverberation generator of claim 3, wherein the values of both $-a$ and a are between about 0.6 and about 0.7.

5. The reverberation generator of claim 3, wherein the length of the delay line in the first allpass filter is approximately equal to the delay time between the first echo and the second echo.

6. The reverberation generator of claim 3, wherein the lengths of all the delay lines and allpass filters are preferably prime numbers.

7. The reverberation generator of claim 3, wherein the length of the delay lines in the allpass filters except for the first allpass filter is calculated by the following equations:

$$AP_{n+1} \approx AP_n * y$$

where AP_n is the length of the delay line in the n th allpass filter; y is the environment coefficient; and the value of y is from about 1.1 to about 1.5.

8. The reverberation generator of claim 2, wherein the delay line in the first loop is approximately equal to the delay time between a direct signal and its first echo.

9. The reverberation generator of claim 8, wherein the length of the delay line in any loop except for the first loop is calculated by the following equation:

$$DL_{n+1} \approx DL_n * x$$

where DL_n is the length of delay line in the n th loop; x is the environment coefficient; and the value of x is from about 1.1 to about 1.5.

10. The reverberation generator of claim 3, wherein the delay lines used in the feed forward loops and allpass filters are preferably realized by circular buffers in digital signal processing.

11. The reverberation generator of claim 2, wherein the gain is calculated by following equations:

$$G_1 = \sqrt{\frac{dE}{dt} * DL_1}$$

$$G_{n+1} = G_n \sqrt{\frac{dE}{dt} * DL_{n+1}}$$

where G_n is the gain for the n th loop and DL_n is the length of the delay line in the n th loop.

12. The reverberation generator of claim 11, wherein the gain value in the first loop varies from about 0.2 to about 0.5; and the gain values in subsequent loops vary from about 1 to about 2.

13. The reverberation generator of claim 3, wherein the lowpass filters are preferably FIR and IIR filters, and more preferably first order IIR filters.

14. A digital audio signal processing system for generating and controlling reverberations simulating real environments for a digital audio signal input, comprising:

a digital I/O interface for inputting and outputting digital audio signals;

a controlling unit electronically connected to the digital I/O interface for receiving the input digital audio signals; and

a reverberation generator electronically connected to the controlling unit; wherein the controlling unit receives reverberation characteristics of a digital audio signal of the real environment; and wherein the received reverberation characteristics will control the configuration of the reverberation generator so as to generate the reverberations for the digital audio signal input to simulate the real environment.

15. The digital audio signal processing system of claim 14, further comprises a controlling mechanism that extracts reverberation characteristics of the digital audio signal when the digital audio signal contains reverberations from a real environment; wherein the reverberation characteristics include one or more of the following ones: final echo density, rate of the echo density to be built up, decay rate of overall energy level of the echoes, and differential decay rates of high-frequency signals and low-frequency signals.

16. The digital audio signal processing system of claim 14, wherein the controlling unit receives one or more of the following reverberation characteristics: final echo density, rate of the echo density to be built up, decay rate of overall energy level of the echoes, and differential decay rates of high-frequency signals and low-frequency signals.

17. The digital audio signal processing system of claim 16, wherein the reverberation generator comprises:

- an input means for receiving the digital audio signal input;

- a summing means for generating a digital audio signal output containing the digital audio signal input and reverberations;

- a digital audio signal direct path electronically connected to the input means and the summing means for transmitting the digital audio signal input directly; and

- a plurality of feed forward loops configured in a cascade manner for generating reverberations; wherein the output of each loop is provided to the summing means and at the same time to the next loop; thereby the plurality of feed forward loops generate the reverberations simulating the real environment that are combined with the digital audio signal input to produce the digital audio signal output.

18. The digital audio signal processing system of claim 17, wherein each of the feed forward loops comprises a gain stage, a delay line, an allpass filter, and a lowpass filter.

19. The digital audio signal processing system of claim 18, wherein the allpass filter comprises:

- an input adder for summing up the input to the allpass filter and a feedback from a delay line, wherein the delay line is electronically downstream of the input adder;

- a feedback loop with an feedback amplifier for using the output of the delay line as the feedback to the input adder;

- a feed forward loop with an amplifier electronically connected to the input adder;

and

- an output adder for summing up the outputs from the delay line and the feed forward loop.

20. The digital audio signal processing system of claim 19, wherein the values of both α and β are between about 0.6 and about 0.7.

21. The digital audio signal processing system of claim 19, wherein the length of the delay line in the first allpass filter is approximately equal to the delay time between the first echo and the second echo.

22. The digital audio signal processing system of claim 19, wherein the lengths of all the delay lines and allpass filters are preferably prime numbers.

23. The digital audio signal processing system of claim 19, wherein the length of the delay lines in the allpass filters except for the first allpass filter is calculated by the following equations:

$$AP_{n+1} \approx AP_n * y$$

where AP_n is the length of the delay line in the n th allpass filter; y is the environment coefficient; and the value of y is from about 1.1 to about 1.5.

24. The digital audio signal processing system of claim 18, wherein the delay line in the first loop is approximately equal to the delay time between a direct signal and its first echo.

25. The digital audio signal processing system of claim 18, wherein the length of the delay line in any loop except for the first loop is calculated by the following equation:

$$DL_{n+1} \approx DL_n * x$$

where DL_n is the length of delay line in the n th loop; x is the environment coefficient; and the value of x is from about 1.1 to about 1.5.

26. The digital audio signal processing system of claim 18, wherein the delay lines used in the feed forward loops and allpass filters are preferably realized by circular buffers in digital signal processing.

27. The digital audio signal processing system of claim 18, wherein the gain is calculated by following equations:

$$G_1 = \sqrt{\frac{dE}{dt} * DL_1}$$

$$G_{n+1} = G_n \sqrt{\frac{dE}{dt} * DL_{n+1}}$$

where G_n is the gain for the n th loop and DL_n is the length of the delay line in the n th loop.

28. The digital audio signal processing system of claim 18, wherein the gain value in the first loop varies from about 0.2 to about 0.5; and the gain values in subsequent loops vary from about 1 to about 2.

29. The digital audio signal processing system of claim 18, wherein the lowpass filters are preferably FIR and IIR filters, and more preferably first order IIR filters.

30. The digital audio signal processing system of claim 18, wherein the reverberation characteristics control the reverberation generator in the following manner: the final echo density determines the number of feed forward loops; the rate of echo density to be built up determines the delay lengths of delay lines; the decay rate of the overall energy of echoes determines the gains in each loop; and the differential decay rates of high-frequency signals and low-frequency signals determines the cutoff frequencies and roll off rate of lowpass filters.

31. An article of manufacture having the capacity of generating reverberations for digital audio signals to simulate real environments, comprising:

a computer-readable medium with a memory domain for storing files and programs, and a computer-executable domain for enabling the article to perform the computer programs; and

a digital audio signal processing system for generating and controlling reverberations simulating real environments for a digital audio signal input, wherein the digital audio signal processing system is embedded in the computer-readable medium;

wherein the digital audio signal processing system comprises:

a digital I/O interface for inputting and outputting digital audio signals;
 a controlling unit electronically connected to the digital I/O interface for receiving the input digital audio signals; and

a reverberation generator electronically connected to the controlling unit; wherein the controlling unit receives reverberation characteristics of a digital audio signal of the real environment; and wherein the received reverberation characteristics will control the configuration of the reverberation generator so as to generate the reverberations for the digital audio signal input to simulate the real environment.

32. The article of manufacture of claim 31, further comprises a controlling mechanism that extracts reverberation characteristics of the digital audio signal when the digital audio signal contains reverberations from a real environment; wherein the reverberation characteristics include one or more of the following ones: final echo density, rate of the echo density to be built up, decay rate of overall energy level of the echoes, and differential decay rates of high-frequency signals and low-frequency signals.

33. The article of manufacture of claim 31, wherein the controlling unit receives one or more of the following reverberation characteristics: final echo density, rate of the echo density to be built up, decay rate of overall energy level of the echoes, and differential decay rates of high-frequency signals and low-frequency signals.

34. The article of manufacture of claim 33, wherein the reverberation generator comprises:

- an input means for receiving the digital audio signal input;
- a summing means for generating a digital audio signal output containing the digital audio signal input and reverberations;
- a digital audio signal direct path electronically connected to the input means and the summing means for transmitting the digital audio signal input directly; and
- a plurality of feed forward loops configured in a cascade manner for generating reverberations; wherein the output of each loop is provided to the summing means and at the same time to the next loop; thereby the plurality of feed forward loops generate the

reverberations simulating the real environment that are combined with the digital audio signal input to produce the digital audio signal output.

35. The article of manufacture of claim 34, wherein each of the feed forward loops comprises a gain stage, a delay line, an allpass filter, and a lowpass filter.

36. The article of manufacture of claim 35, wherein the allpass filter comprises:
an input adder for summing up the input to the allpass filter and a feedback from a delay line, wherein the delay line is electronically downstream of the input adder;

a feedback loop with an feedback amplifier for using the output of the delay line as the feedback to the input adder;

a feed forward loop with an amplifier electronically connected to the input adder;
and

an output adder for summing up the outputs from the delay line and the feed forward loop.

37. The article of manufacture of claim 36, wherein the values of both -a and a are between about 0.6 and about 0.7.

38. The article of manufacture of claim 36, wherein the length of the delay line in the first allpass filter is approximately equal to the delay time between the first echo and the second echo.

39. The article of manufacture of claim 36, wherein the lengths of all the delay lines and allpass filters are preferably prime numbers.

40. The article of manufacture of claim 36, wherein the length of the delay lines in the allpass filters except for the first allpass filter is calculated by the following equations:

$$AP_{n+1} \approx AP_n * y$$

where AP_n is the length of the delay line in the nth allpass filter; y is the environment coefficient; and the value of y is from about 1.1 to about 1.5.

41. The article of manufacture of claim 35, wherein the delay line in the first loop is approximately equal to the delay time between a direct signal and its first echo.

42. The article of manufacture of claim 35, wherein the length of the delay line in any loop except for the first loop is calculated by the following equation:

$$DL_{n+1} \approx DL_n * x$$

where DL_n is the length of delay line in the nth loop; x is the environment coefficient; and the value of x is from about 1.1 to about 1.5.

43. The article of manufacture of claim 35, wherein the delay lines used in the feed forward loops and allpass filters are preferably realized by circular buffers in digital signal processing.

44. The article of manufacture of claim 35, wherein the gain is calculated by following equations:

$$G_1 = \sqrt{\frac{dE}{dt} * DL_1}$$

$$G_{n+1} = G_n \sqrt{\frac{dE}{dt} * DL_{n+1}}$$

where G_n is the gain for the nth loop and DL_n is the length of the delay line in the nth loop.

45. The article of manufacture of claim 35, wherein the gain value in the first loop varies from about 0.2 to about 0.5; and the gain values in subsequent loops vary from about 1 to about 2.

46. The article of manufacture of claim 35, wherein the lowpass filters are preferably FIR and IIR filters, and more preferably first order IIR filters.

47. The article of manufacture of claim 35, wherein the reverberation characteristics control the reverberation generator in the following manner: the final echo density determines the number of feed forward loops; the rate of echo density to be built up determines the delay lengths of delay lines; the decay rate of the overall energy of echoes

determines the gains in each loop; and the differential decay rates of high-frequency signals and low-frequency signals determines the cutoff frequencies and roll off rate of lowpass filters.

48. The article of manufacture of claim 31, wherein the article includes MP3 players, handphones, portable players, TVs, DVD players, and the like.

49. A method for generating reverberations for a digital audio signal to simulate real environments, comprising the following operations:

receiving the reverberation characteristics of the digital audio signal for a real environment;

translating the extracted reverberation characteristics into controlling parameters for a reverberation generator with a plurality of feed forward loops configured in a cascade manner; and

generating the reverberations using the controlling parameters to control the reverberation generator.

50. The method of claim 49, further optionally comprising the following operation:

extracting the reverberation characteristics of the digital audio signal when the digital audio signal contains reverberations from a real environment; wherein the reverberation characteristics include one or more of the following: final echo density, rate of the echo density to be built up, decay rate of overall energy level of the echoes, and differential decay rates of high-frequency signals and low-frequency signals.

51. The method of claim 49, wherein the reverberation characteristics include one or more of the following: final echo density, rate of the echo density to be built up, decay rate of overall energy level of the echoes, and differential decay rates of high-frequency signals and low-frequency signals.

52. The method of claim 49, wherein each of the feed forward loops comprises a gain stage, a delay line, an allpass filter, and a lowpass filter.

53. The method of claim 52, wherein wherein the allpass filter comprises:
an input adder for summing up the input to the allpass filter and a feedback from a delay line, wherein the delay line is electronically downstream of the input adder;
a feedback loop with an feedback amplifier for using the output of the delay line as the feedback to the input adder;
a feed forward loop with an amplifier electronically connected to the input adder;
and
an output adder for summing up the outputs from the delay line and the feed forward loop.
54. The method of claim 51, wherein the delay line in the first loop is approximately equal to the delay time between a direct signal and its first echo.
55. The method of claim 51, wherein the controlling parameters of the reverberation generator include number of feed forward loops, lengths of the delay lines, gain used in the feed forward loops, and cutoff frequencies and roll off rates of the lowpass filters.
56. The method of claim 55, wherein the reverberation generator generates reverberations in the following controlled manner:
controlling the final echo density by the number of feed forward loops;
controlling the rate of the echo density to be built up by the lengths of the delay lines used in the feed forward loops and allpass filters;
controlling the decay rate of the overall energy level of the echoes by the gain used in the feed forward loops; and
controlling the decay of high frequency signals with respect to low frequency signals with the cutoff frequencies and roll off rates of the lowpass filters.
57. The method of claim 56, wherein the values of both $-a$ and a are between about 0.6 and about 0.7.
58. The method of claim 56, wherein the length of the delay line in the first allpass filter is approximately equal to the delay time between the first echo and the second echo.

59. The method of claim 56, wherein the lengths of all the delay lines and allpass filters are preferably prime numbers.

60. The method of claim 56, wherein the length of the delay lines in the allpass filters except for the first allpass filter is calculated by the following equations:

$$AP_{n+1} \approx AP_n * y$$

where AP_n is the length of the delay line in the n th allpass filter; y is the environment coefficient; and the value of y is from about 1.1 to about 1.5.

61. The method of claim 56, wherein the delay line in the first loop is approximately equal to the delay time between a direct signal and its first echo.

62. The method of claim 56, wherein the length of the delay line in any loop except for the first loop is calculated by the following equation:

$$DL_{n+1} \approx DL_n * x$$

where DL_n is the length of delay line in the n th loop; x is the environment coefficient; and the value of x is from about 1.1 to about 1.5.

63. The method of claim 56, wherein the delay lines used in the feed forward loops and allpass filters are preferably realized by circular buffers in digital signal processing.

64. The method of claim 56, wherein the gain is calculated by following equations:

$$G_1 = \sqrt{\frac{dE}{dt} * DL_1}$$

$$G_{n+1} = G_n \sqrt{\frac{dE}{dt} * DL_{n+1}}$$

where G_n is the gain for the n th loop and DL_n is the length of the delay line in the n th loop.

65. The method of claim 56, wherein the gain value in the first loop varies from about 0.2 to about 0.5; and the gain values in subsequent loops vary from about 1 to about 2.