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Stone et al.

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(45) **Date of Patent: Apr. 20, 2010**

(54) MICROPHONE BLEED SIMULATOR

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H03G 3/00 (2006.01)

(52) **U.S. Cl.** **381/63**; 381/17; 381/61; 381/92

(58) **Field of Classification Search** 381/1, 381/17–18, 26, 91–92, 122, 61–63, 119, 381/118; 700/94; 84/625, 630, 660, 707
See application file for complete search history.

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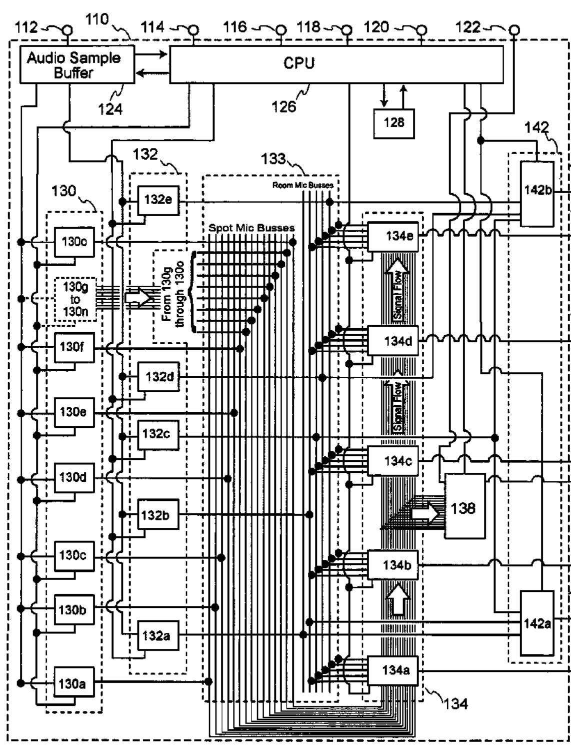
(57) **ABSTRACT**

The invention is a microphone bleed simulator which processes the audio signals recorded with one spot microphone and one room mic (or one spot mic alone) and simulates the sound that would be created in a typical studio recording environment wherein multiple spot microphones and multiple room microphones distributed throughout that environment all pick up sounds from any given sound source. Such multiple microphone contributions to the mixed sound are what imbue actual recordings with much of their characteristic sound quality and sense of realism, and what make recordings of ensemble instruments sound as though the players were performing together in one place. The system and method of the invention artificially simulates this natural microphone bleed and provides the means to create scaleable, realistic multi-channel sound mixes in formats such as stereo, 5.1 and 7.1 surround. The invention is particularly well suited to processing sampled or synthesized sound.

5 Claims, 15 Drawing Sheets



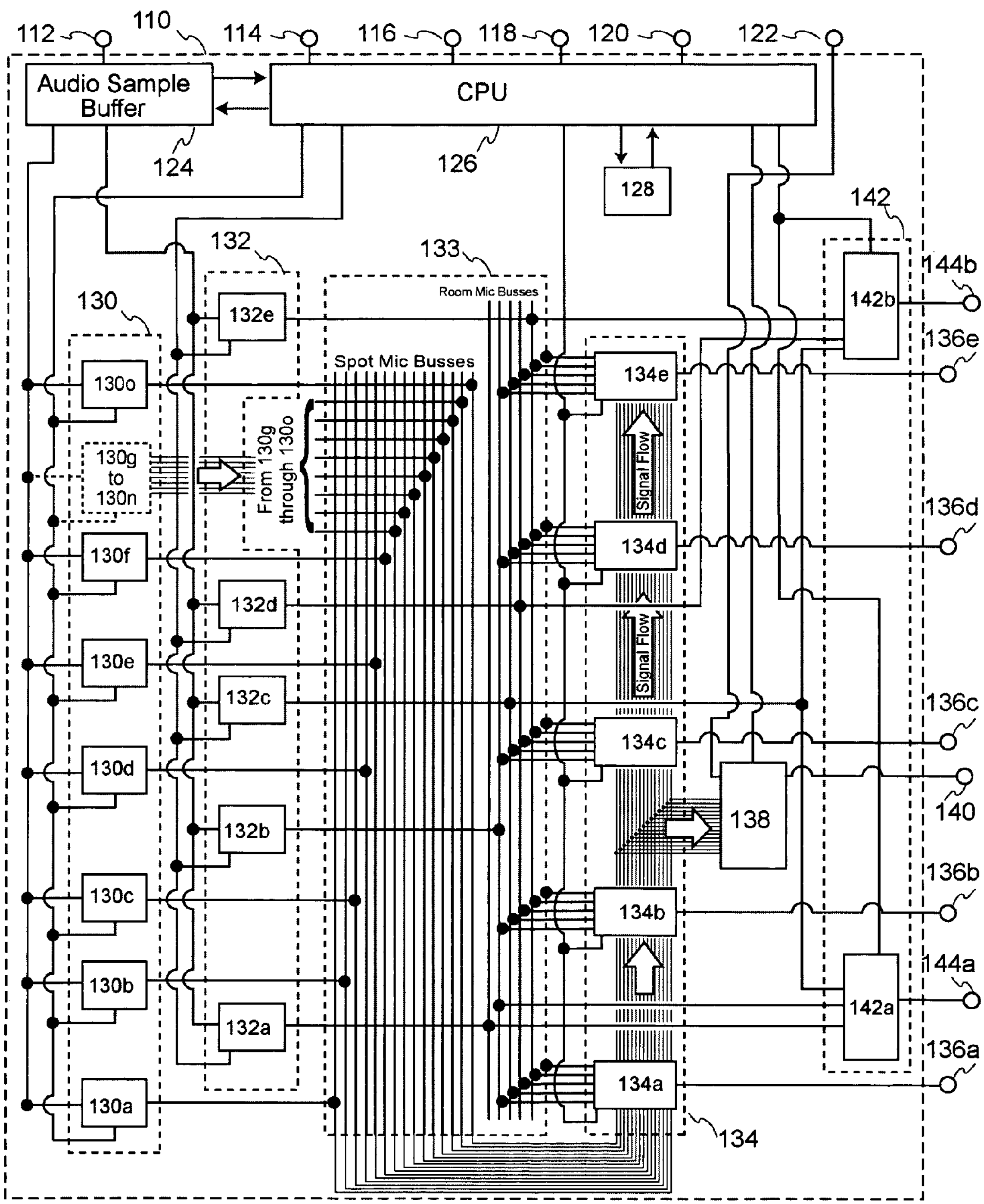

112
110
114
116
118
120
122
Audio Sample Buffer
CPU
124
126
128
142
132
133
Room Mic Busses
130
144b
132e
142b
136e
Spot Mic Busses
134e
130o
130g to 130n
From 130g through 130o
Signal Flow
136d
134d
130f
132d
Signal Flow
130e
132c
136c
134c
130d
140
138
132b
136b
134b
130c
144a
130b
142a
132a
136a
134a
130a
134

Fig. 1

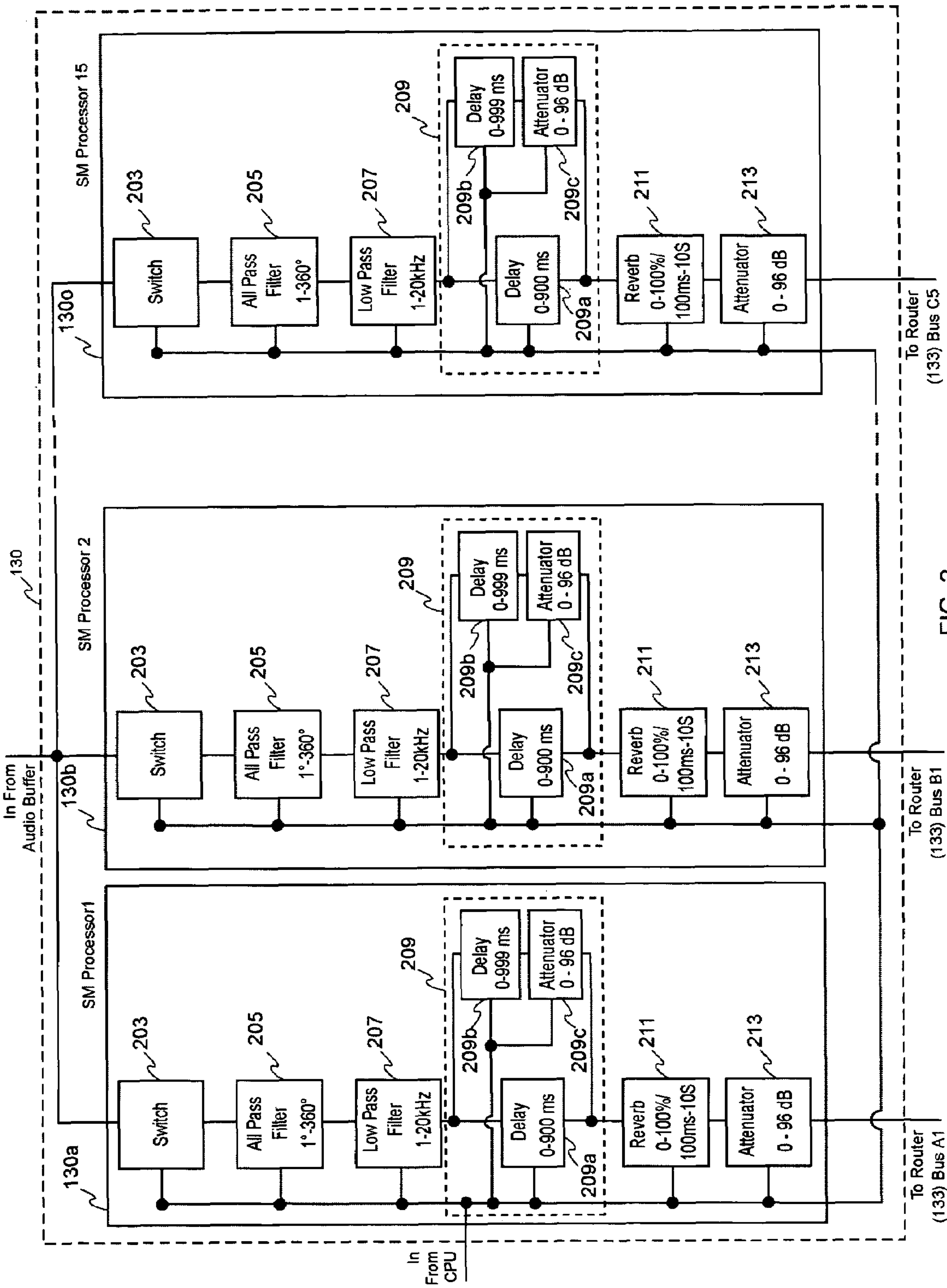

130
In From
Audio Buffer
In
From
CPU
SM Processor1
SM Processor 2
SM Processor 15
130a
130b
130o
203
Switch
205
All Pass
Filter
1°-360°
1-360°
207
Low Pass
Filter
1-20kHz
209
209a
209b
209c
Delay
0-900 ms
Delay
0-999 ms
Attenuator
0 - 96 dB
211
Reverb
0-100%/
100ms-10S
213
Attenuator
0 - 96 dB
To Router
(133) Bus A1
To Router
(133) Bus B1
To Router
(133) Bus C5

FIG. 2

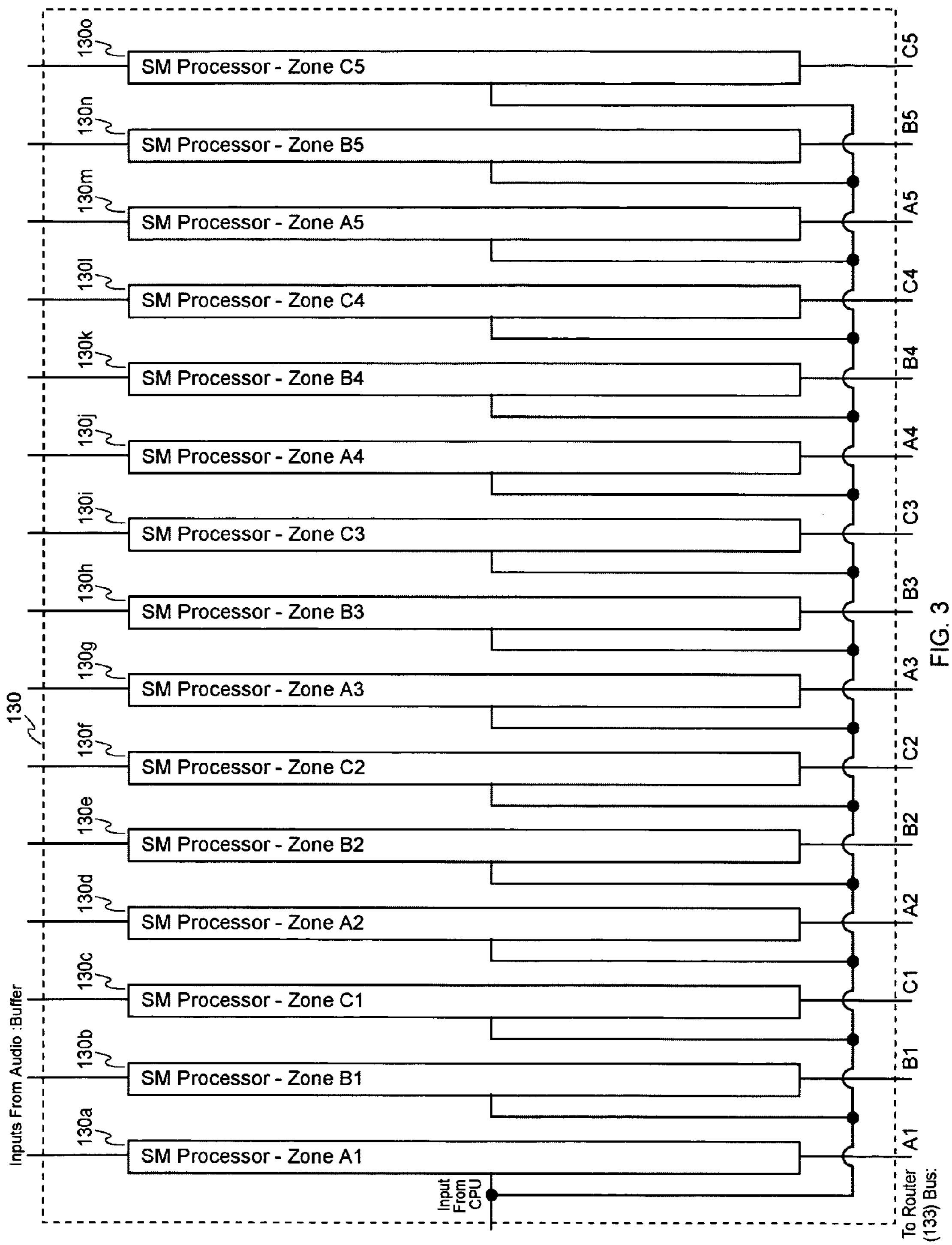
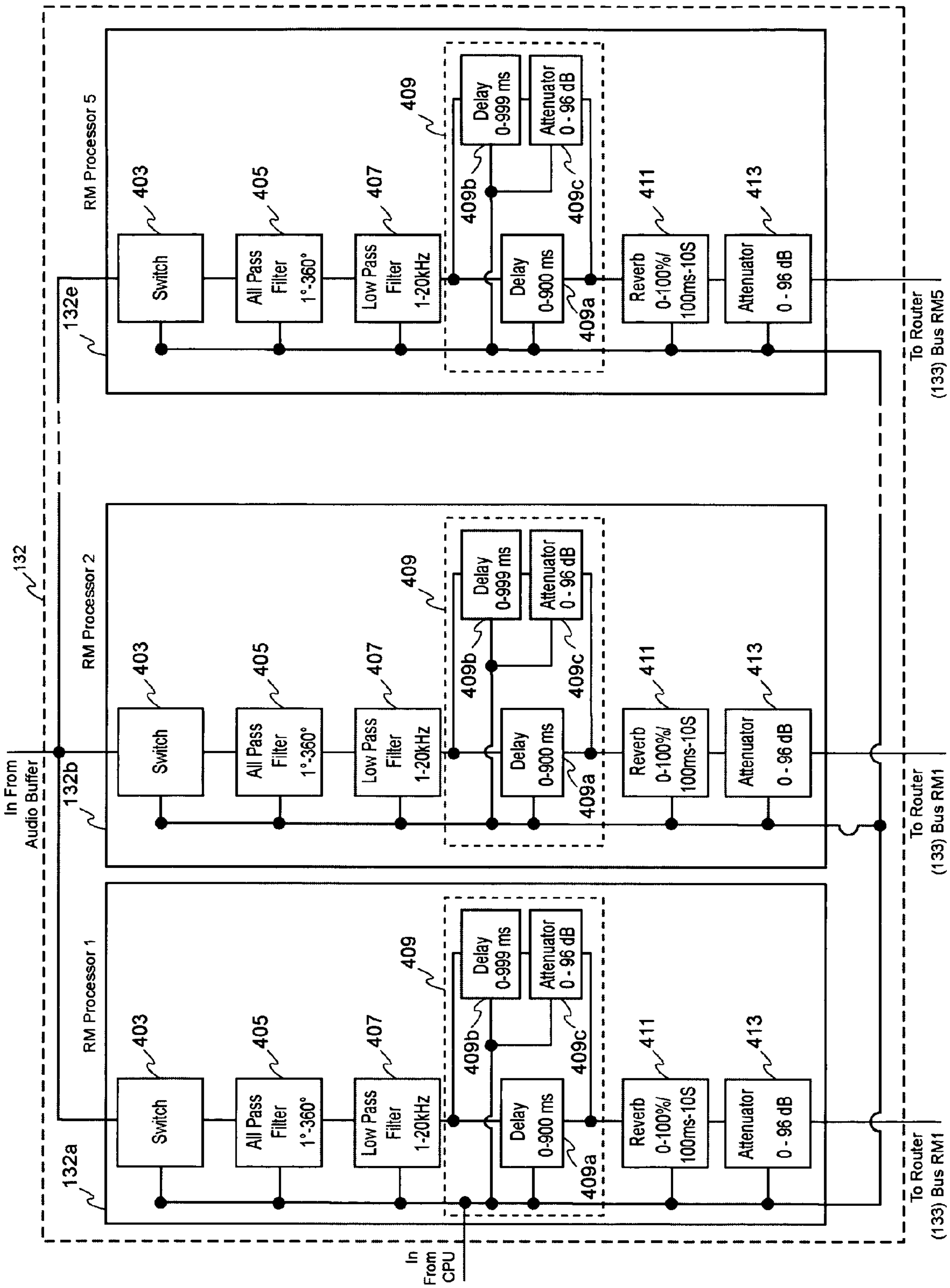

130
130o
SM Processor - Zone C5
130n
SM Processor - Zone B5
130m
SM Processor - Zone A5
130l
SM Processor - Zone C4
130k
SM Processor - Zone B4
130j
SM Processor - Zone A4
130i
SM Processor - Zone C3
130h
SM Processor - Zone B3
130g
SM Processor - Zone A3
130f
SM Processor - Zone C2
130e
SM Processor - Zone B2
130d
SM Processor - Zone A2
130c
SM Processor - Zone C1
130b
SM Processor - Zone B1
130a
SM Processor - Zone A1
Inputs From Audio :Buffer
Input From CPU
C5
B5
A5
C4
B4
A4
C3
B3
A3
C2
B2
A2
C1
B1
A1
To Router (133) Bus:

FIG. 3



132
In From
Audio Buffer
In
From
CPU
132a
132b
132e
RM Processor 1
RM Processor 2
RM Processor 5
403
405
407
409
409a
409b
409c
411
413
Switch
All Pass
Filter
1°-360°
Low Pass
Filter
1-20kHz
Delay
0-900 ms
Delay
0-999 ms
Attenuator
0 - 96 dB
Reverb
0-100%/
100ms-10S
Attenuator
0 - 96 dB
To Router
(133) Bus RM1
To Router
(133) Bus RM1
To Router
(133) Bus RM5

FIG. 4

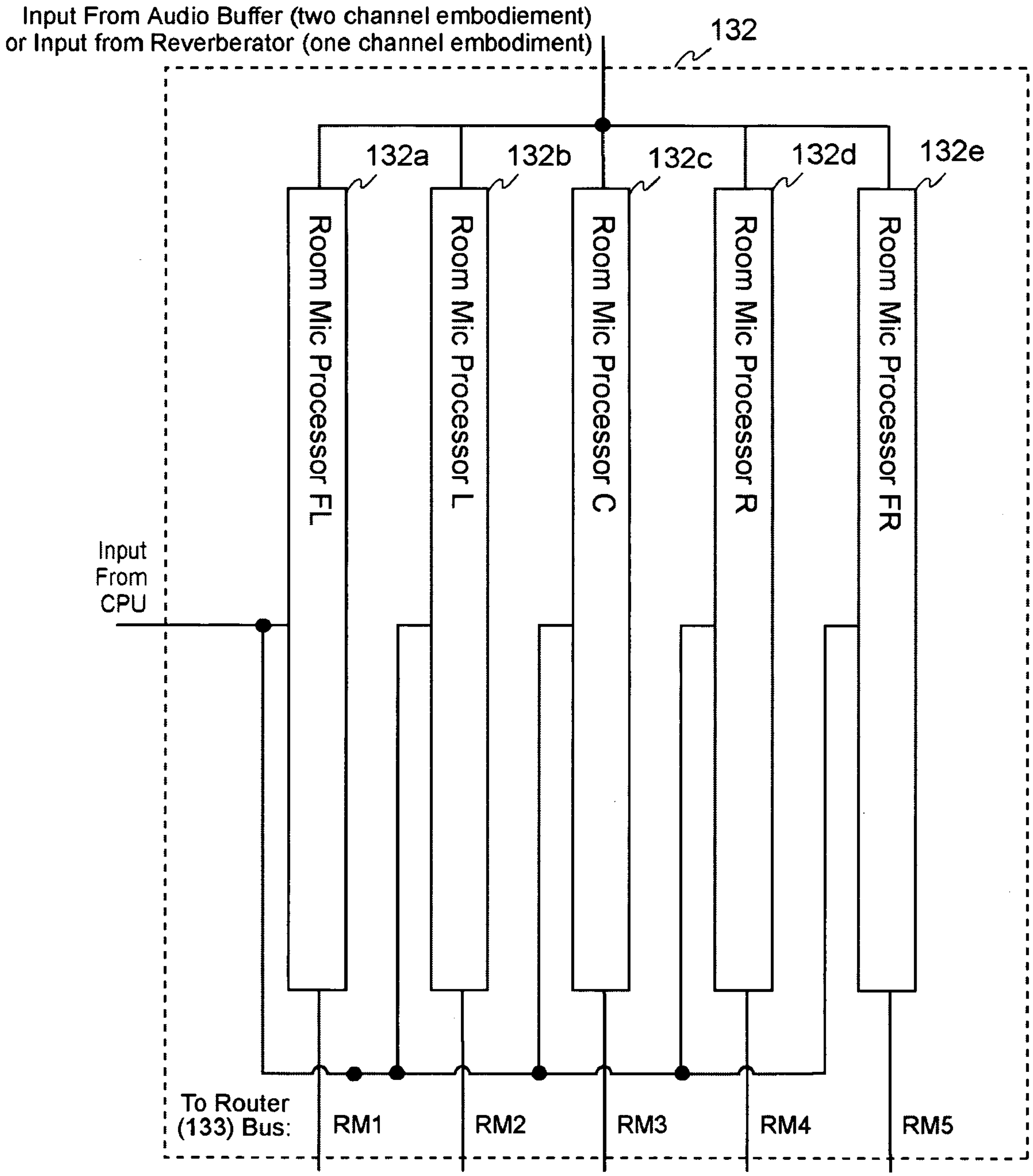

Input From Audio Buffer (two channel embodiement)
or Input from Reverberator (one channel embodiment)
132
132a
132b
132c
132d
132e
Room Mic Processor FL
Room Mic Processor L
Room Mic Processor C
Room Mic Processor R
Room Mic Processor FR
Input
From
CPU
To Router
(133) Bus:
RM1
RM2
RM3
RM4
RM5

FIG. 5

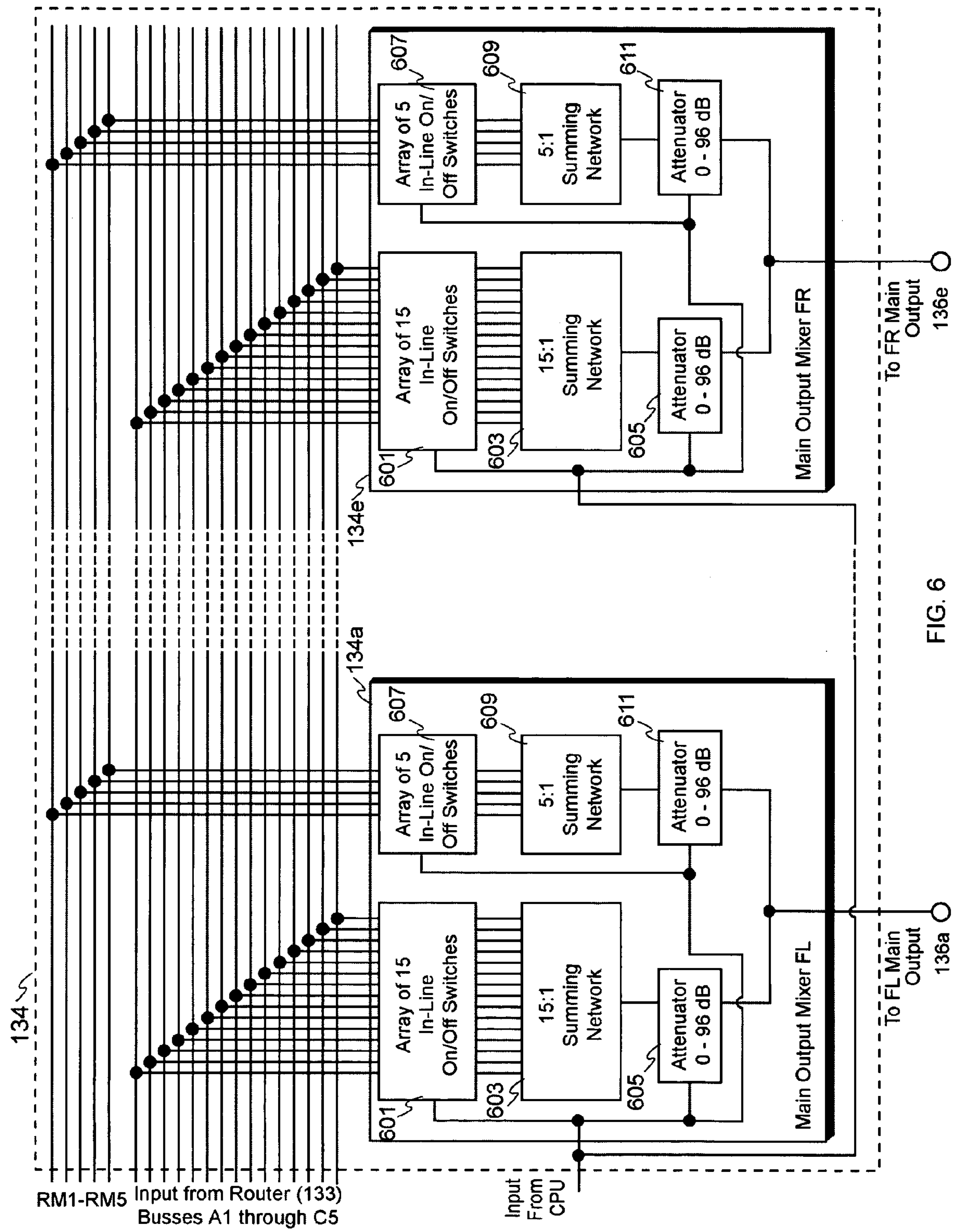

134
134a
134e
601
603
605
607
609
611
Array of 15 In-Line On/Off Switches
15:1 Summing Network
Attenuator 0 - 96 dB
Array of 5 In-Line On/ Off Switches
5:1 Summing Network
Main Output Mixer FL
Main Output Mixer FR
To FL Main Output 136a
To FR Main Output 136e
RM1-RM5
Input from Router (133) Busses A1 through C5
Input From CPU

FIG. 6

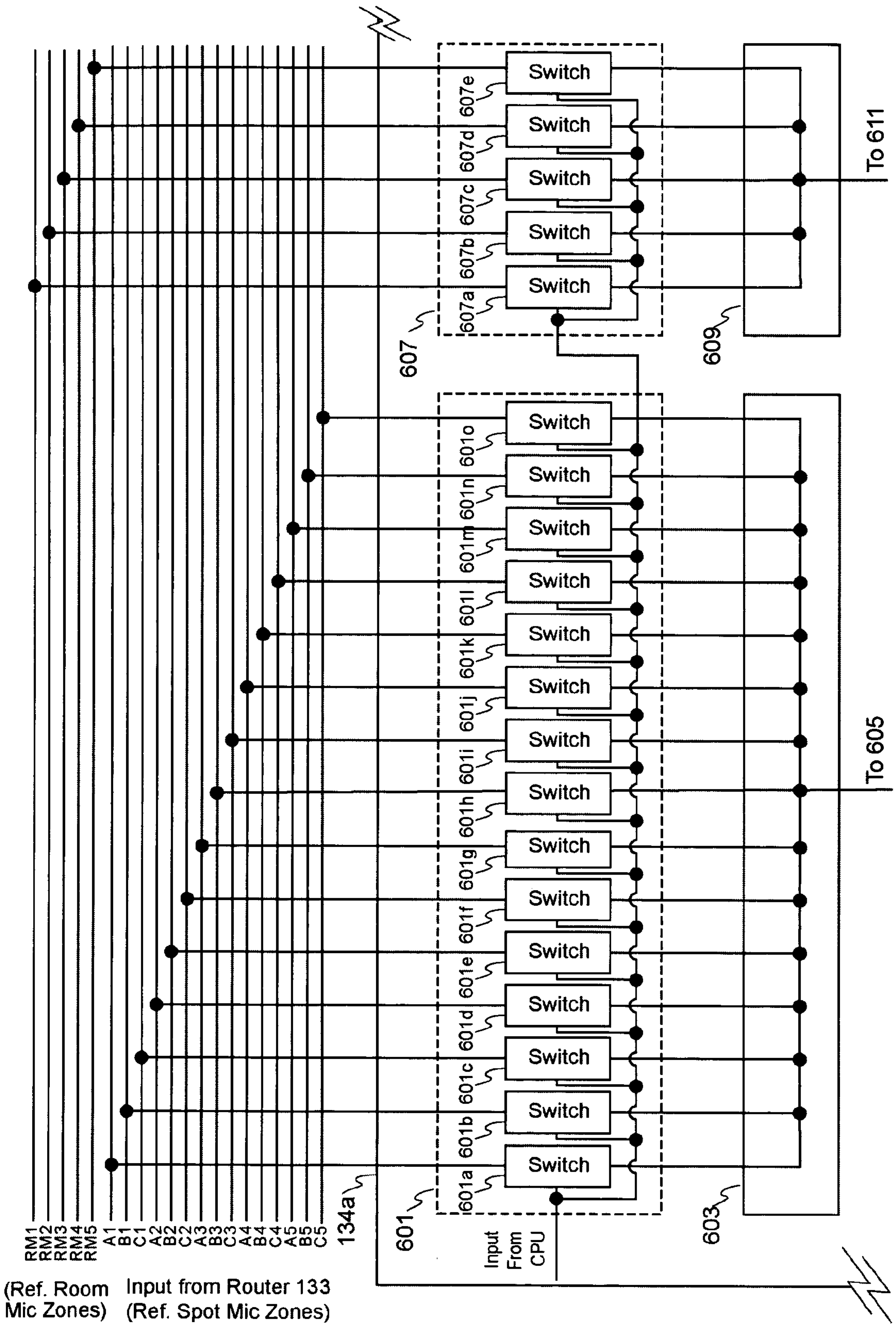
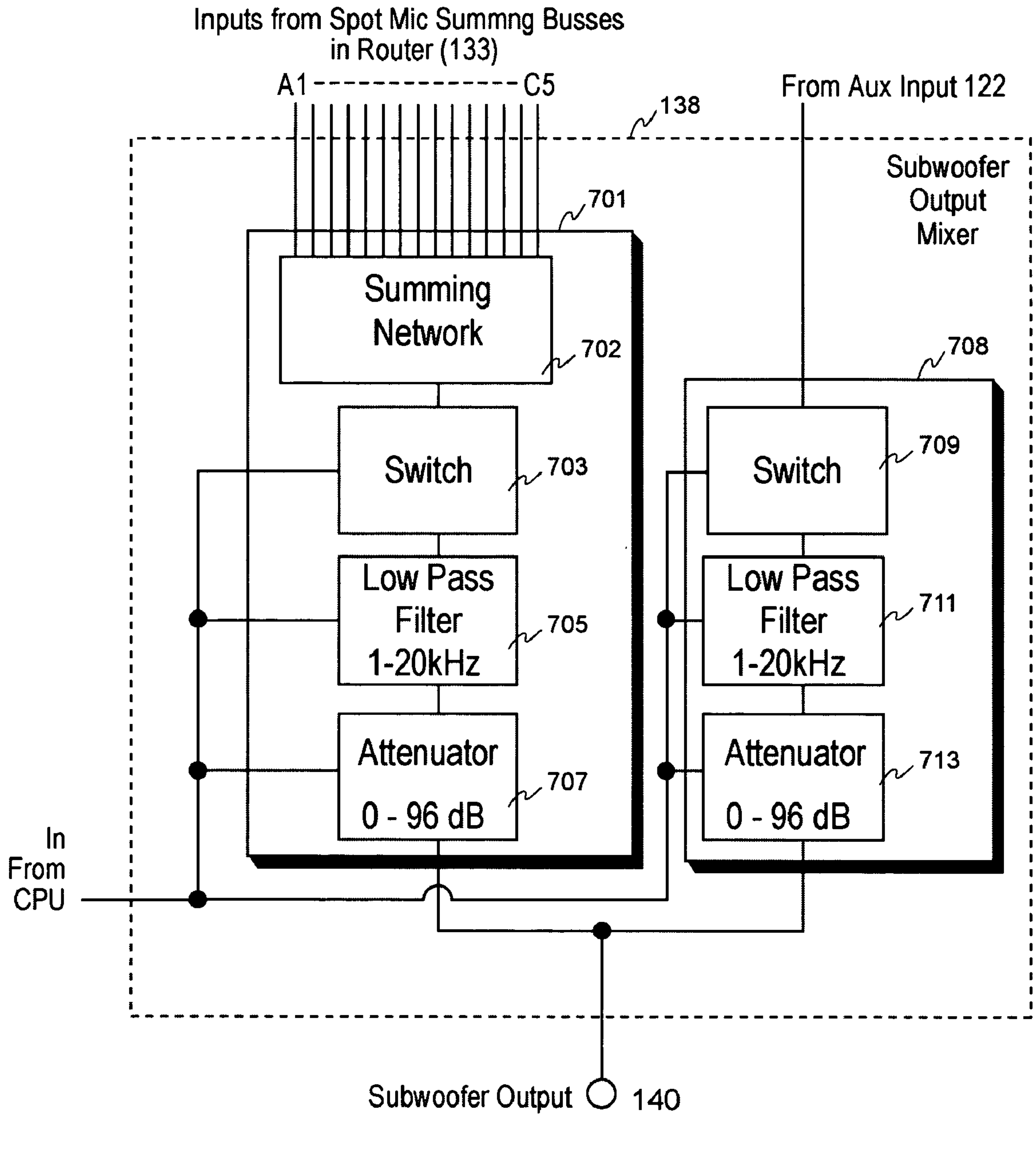
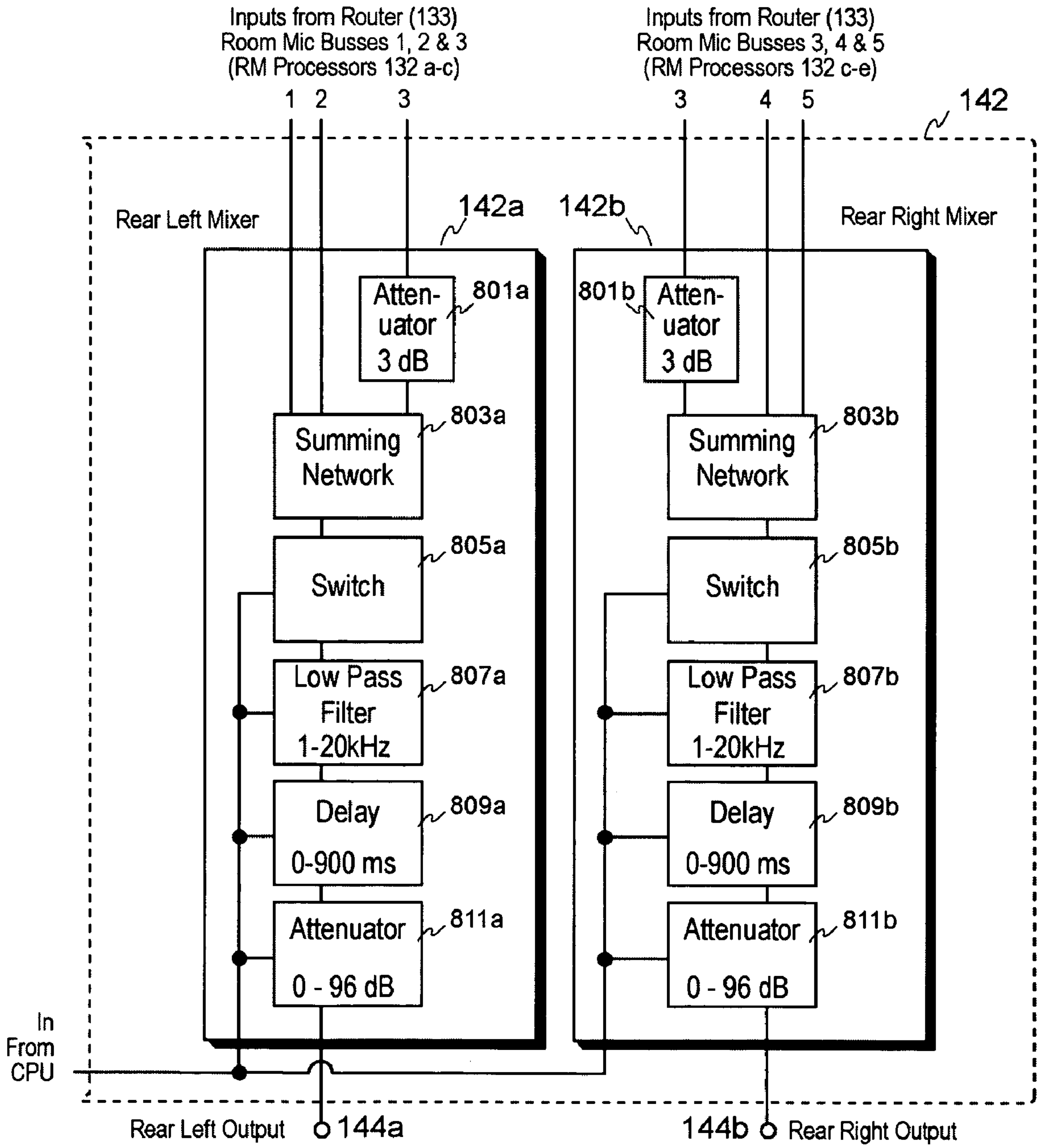

Switch
607a
607b
607c
607d
607e
607
609
To 611
601a
601b
601c
601d
601e
601f
601g
601h
601i
601j
601k
601l
601m
601n
601o
601
603
To 605
RM1
RM2
RM3
RM4
RM5
A1
B1
C1
A2
B2
C2
A3
B3
C3
A4
B4
C4
A5
B5
C5
134a
Input From CPU
(Ref. Room Mic Zones)
Input from Router 133 (Ref. Spot Mic Zones)

FIG. 7



Inputs from Spot Mic Summng Busses
in Router (133)
A1 C5
138
From Aux Input 122
Subwoofer
Output
Mixer
701
Summing
Network
702
708
Switch
703
Switch
709
Low Pass
Filter
1-20kHz
705
Low Pass
Filter
1-20kHz
711
Attenuator
0 - 96 dB
707
Attenuator
0 - 96 dB
713
In
From
CPU
Subwoofer Output
140

Fig. 8



Inputs from Router (133)
Room Mic Busses 1, 2 & 3
(RM Processors 132 a-c)
1 2 3
Inputs from Router (133)
Room Mic Busses 3, 4 & 5
(RM Processors 132 c-e)
3 4 5
142
Rear Left Mixer
142a
142b
Rear Right Mixer
Atten-uator 3 dB
801a
801b
Atten-uator 3 dB
Summing Network
803a
Summing Network
803b
Switch
805a
Switch
805b
Low Pass Filter 1-20kHz
807a
Low Pass Filter 1-20kHz
807b
Delay 0-900 ms
809a
Delay 0-900 ms
809b
Attenuator 0 - 96 dB
811a
Attenuator 0 - 96 dB
811b
In From CPU
Rear Left Output
144a
144b
Rear Right Output

Fig. 9

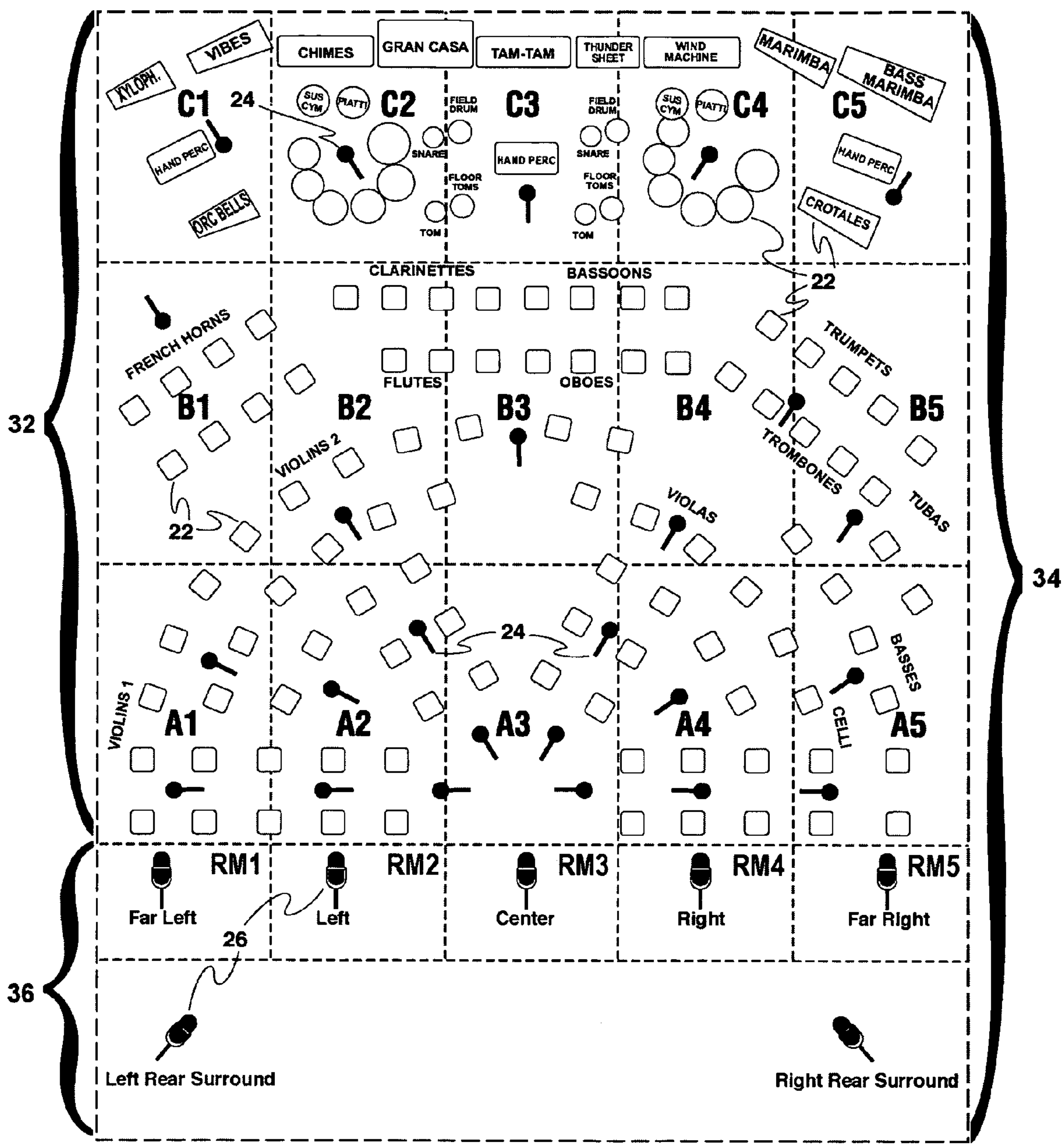
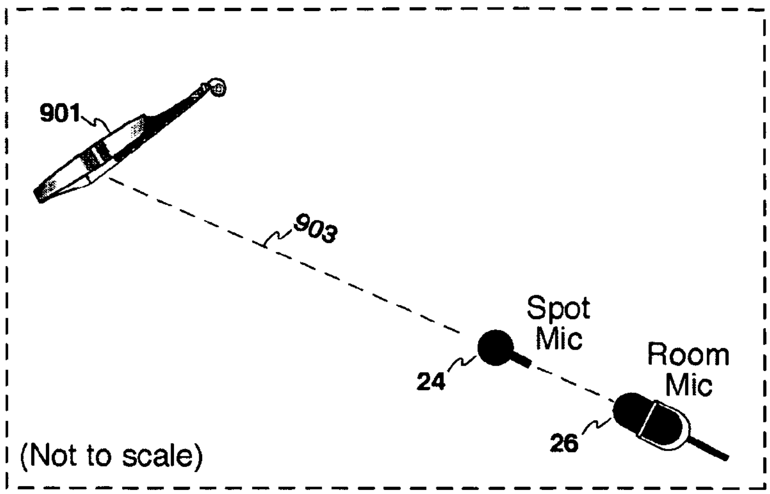
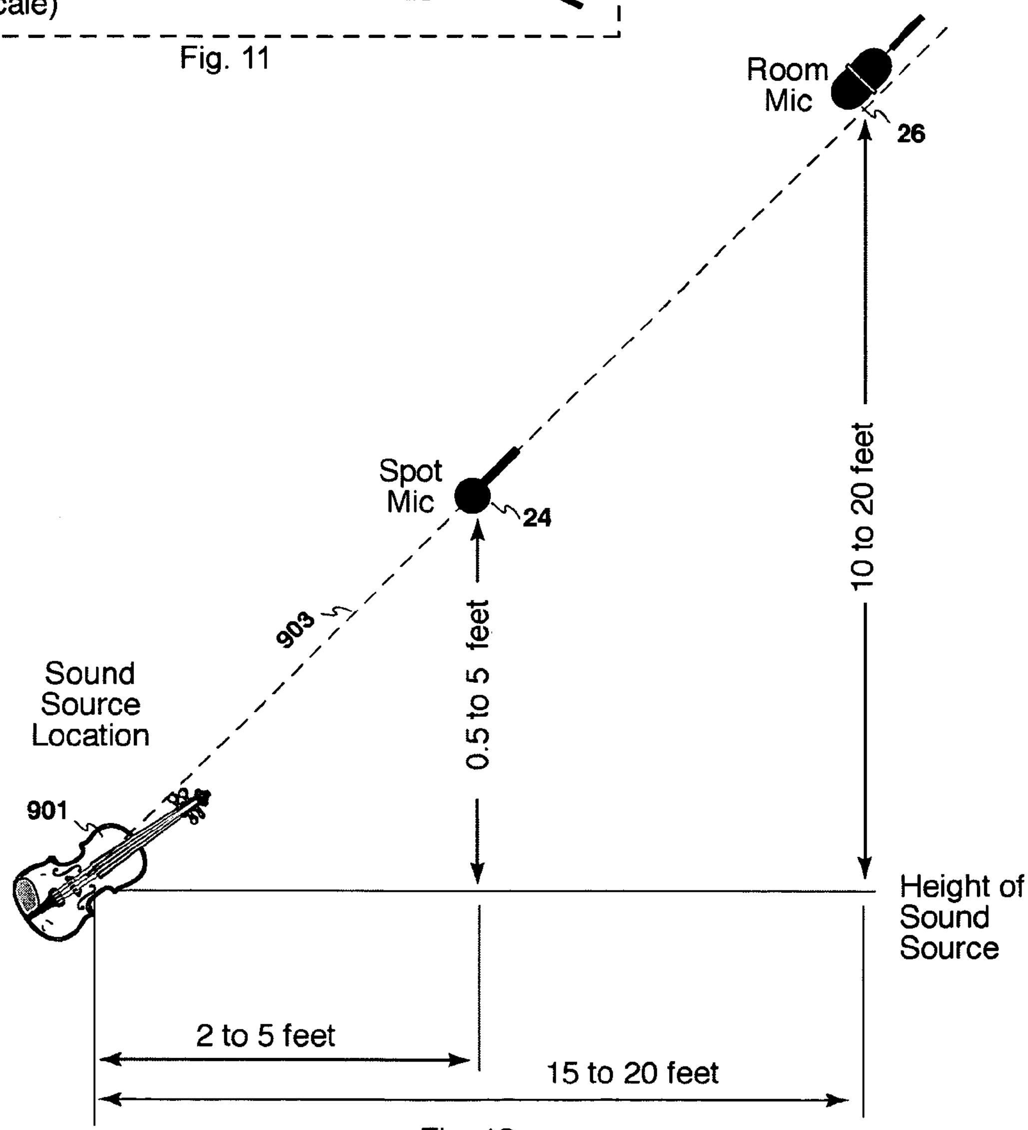

XYLOPH.
VIBES
CHIMES
GRAN CASA
TAM-TAM
THUNDER SHEET
WIND MACHINE
MARIMBA
BASS MARIMBA
C1
24
SUS CYM
PIATTI
C2
FIELD DRUM
C3
FIELD DRUM
SUS CYM
PIATTI
C4
C5
HAND PERC
SNARE
HAND PERC
SNARE
HAND PERC
FLOOR TOMS
FLOOR TOMS
ORC BELLS
TOM
TOM
CROTALES
CLARINETTES
BASSOONS
22
FRENCH HORNS
TRUMPETS
FLUTES
OBOES
B1
B2
B3
B4
B5
VIOLINS 2
TROMBONES
VIOLAS
TUBAS
22
32
34
24
VIOLINS 1
BASSES
A1
A2
A3
A4
A5
CELLI
RM1
RM2
RM3
RM4
RM5
Far Left
Left
Center
Right
Far Right
26
36
Left Rear Surround
Right Rear Surround

Fig. 10



901
903
Spot Mic
24
Room Mic
26
(Not to scale)

Fig. 11



Room Mic
26
Spot Mic
24
903
0.5 to 5 feet
10 to 20 feet
Sound Source Location
901
Height of Sound Source
2 to 5 feet
15 to 20 feet

Fig. 12

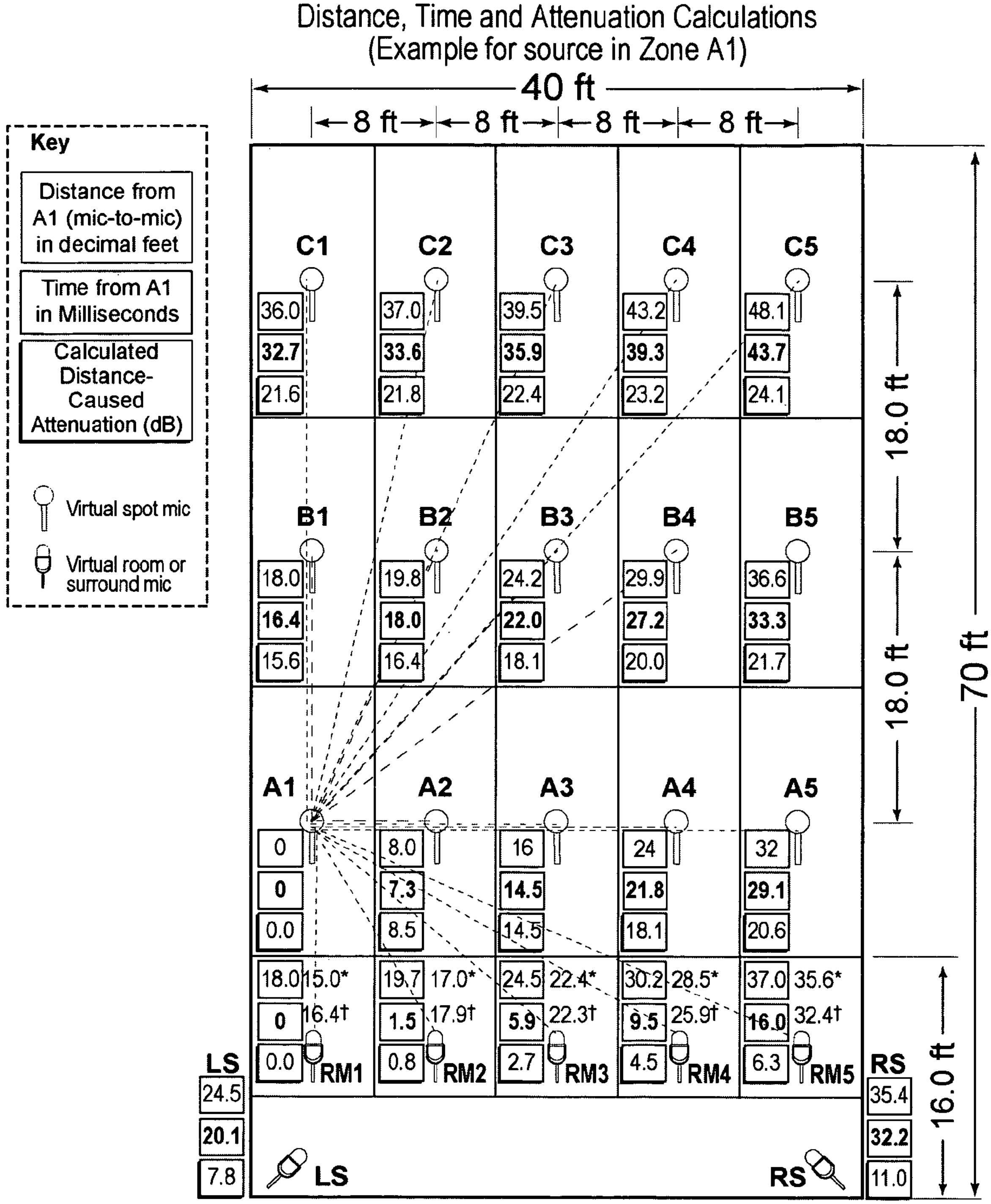

Distance, Time and Attenuation Calculations
(Example for source in Zone A1)
40 ft
8 ft
8 ft
8 ft
8 ft
Key
Distance from A1 (mic-to-mic) in decimal feet
Time from A1 in Milliseconds
Calculated Distance-Caused Attenuation (dB)
Virtual spot mic
Virtual room or surround mic
C1
36.0
32.7
21.6
C2
37.0
33.6
21.8
C3
39.5
35.9
22.4
C4
43.2
39.3
23.2
C5
48.1
43.7
24.1
B1
18.0
16.4
15.6
B2
19.8
18.0
16.4
B3
24.2
22.0
18.1
B4
29.9
27.2
20.0
B5
36.6
33.3
21.7
A1
0
0
0.0
A2
8.0
7.3
8.5
A3
16
14.5
14.5
A4
24
21.8
18.1
A5
32
29.1
20.6
18.0 15.0*
0 16.4†
0.0 RM1
19.7 17.0*
1.5 17.9†
0.8 RM2
24.5 22.4*
5.9 22.3†
2.7 RM3
30.2 28.5*
9.5 25.9†
4.5 RM4
37.0 35.6*
16.0 32.4†
6.3 RM5
LS
24.5
20.1
7.8
LS
RS
35.4
32.2
11.0
RS
18.0 ft
18.0 ft
16.0 ft
70 ft

Fig. 13

Instructions for Spot Mic Simulator (130)

ID	Element	Value	ID	Element	Value	ID	Element	Value
	203	on		203	on		203	on
	205	290		205	360		205	90
130a	207	20,000	130g	207	6,000	130m	207	4,000
	209a	0		209a	14.5		209a	29.1
	209b	0.003		209b	14.505		209b	29.104
	209c	18		209c	14		209c	12
	211	0/-		211	0/-		211	0/-
	213	0		213	14.5		213	20.6
	203	on		203	on		203	on
	205	120		205	360		205	45
130b	207	12,000	130h	207	5,000	130n	207	3,500
	209a	16.4		209a	22		209a	33.3
	209b	16.403		209b	22.002		209b	33.302
	209c	17		209c	14		209c	11
	211	0/-		211	0/-		211	0/-
	213	15.6		213	18.1		213	21.7
	203	on		203	on		203	on
	205	125		205	360		205	200
130c	207	8,000	130i	207	4,000	130o	207	2,500
	209a	32.7		209a	35.9		209a	43.7
	209b	32.703		209b	35.905		209b	43.705
	209c	16		209c	12		209c	10
	211	0/-		211	0/-		211	0/-
	213	21.6		213	22.4		213	24.1
	203	on		203	on			
	205	280		205	45			
130d	207	14,000	130j	207	4,500			
	209a	7.3		209a	21.8			
	209b	7.302		209b	21.802			
	209c	16		209c	12			
	211	0/-		211	0/-			
	213	8.5		213	18.1			
	203	on		203	on			
	205	330		205	30			
130e	207	10,000	130k	207	4,000			
	209a	18		209a	27.2			
	209b	18.004		209b	27.205			
	209c	16		209c	11			
	211	0/-		211	0/-			
	213	16.4		213	20			
	203	on		203	on			
	205	330		205	20			
130f	207	7,500	130l	207	3,000			
	209a	35.9		209a	39.3			
	209b	35.904		209b	39.304			
	209c	15		209c	11			
	211	0/-		211	0/-			
	213	21.8		213	23.2			

Instructions for Room Mic Simulator (132)

	Element	Value
	403	on
	405	360
132a	407	10,000
	409a	0
	409b	0.003
	409c	24
	411	0/-
	413	0
	403	on
	405	360
132b	407	8,000
	409a	1.5
	409b	1.510
	409c	20
	411	0/-
	413	0.8
	403	on
	405	360
132c	407	6,000
	409a	5.90
	409b	5.917
	409c	18
	411	0/-
	413	2.7
	403	on
	405	360
132d	407	4,000
	409a	9.5
	409b	9.523
	409c	16
	411	0/-
	413	4.5
	403	on
	405	360
132e	407	3,500
	409a	16.00
	409b	16.018
	409c	14
	411	0/-
	413	6.3

Instructions for Surround Simulator (142)

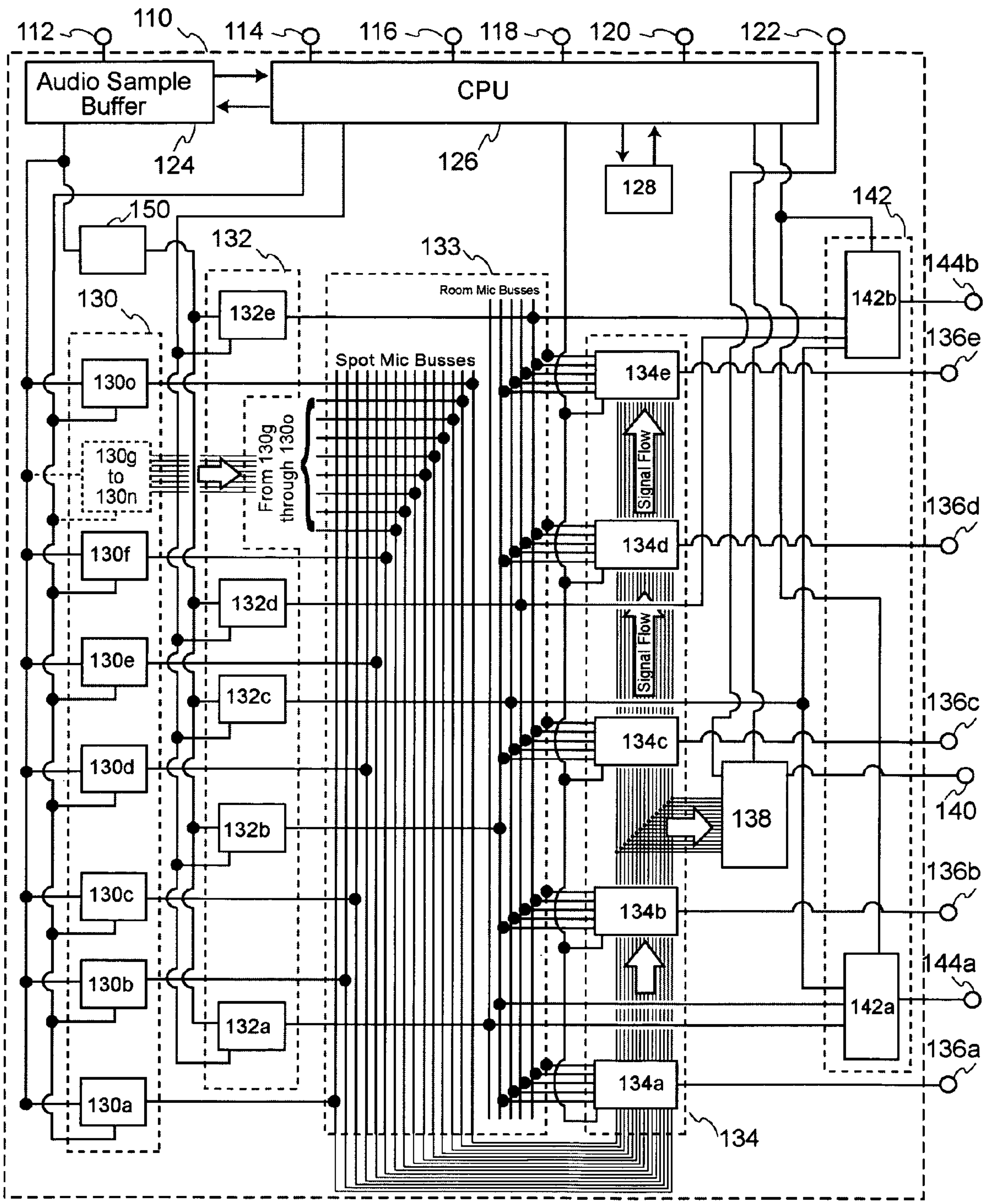
Element	142a	Element	142b
805a	on	805b	on
807a	3,000	807b	3,000
809a	20.1	809b	32.2
811a	7.8	811b	11

Fig. 14

Instructions for Main Output Mixer (134)

Element	134a	134b	134c	134d	134)
601a	on	off	off	off	off
601b	on	off	off	off	off
601c	on	off	off	off	off
601d	off	on	off	off	off
601e	off	on	off	off	off
601f	off	on	off	off	off
601g	off	off	on	off	off
601h	off	off	on	off	off
601i	off	off	on	off	off
601j	off	off	off	on	off
601k	off	off	off	on	off
601l	off	off	off	on	off
601m	off	off	off	off	on
601n	off	off	off	off	on
601o	on	off	off	off	on
605	0.0	8.5	14.5	18.1	20.6
607a	on	off	off	off	off
607b	off	on	off	off	off
607c	off	off	on	off	off
607d	off	off	off	on	off
607e	off	off	off	off	on
611	0.0	0.8	2.7	4.5	6.3

Fig. 15



112
110
114
116
118
120
122
Audio Sample Buffer
CPU
124
126
128
150
132
133
142
144b
130
132e
Room Mic Busses
142b
136e
Spot Mic Busses
130o
134e
130g to 130n
From 130g through 130o
Signal Flow
136d
130f
134d
132d
130e
Signal Flow
132c
136c
134c
130d
140
138
132b
136b
130c
134b
144a
130b
142a
132a
136a
134a
130a
134

Fig. 16

MICROPHONE BLEED SIMULATOR

TECHNICAL FIELD

This invention relates to the field of microphone bleed simulation for manipulation of apparent source position and environment.

BACKGROUND OF THE INVENTION

This invention relates to the recording of orchestral sounds, choirs, or any type of music or sound effect and to the subsequent playback in a manner that simulates a particular position on a sound stage and particular features thereof (such as size, shape and acoustics). By way of example, the invention can be used to recreate the sound of a conventional symphonic recording setup that uses numerous spot microphones and five to seven surround sound (sometimes called "Decca Tree") microphones.

As little as one microphone and one channel of actual audio for sound storage (in digital or analog format) of each individual sound source may be used. Multiple sources (e.g., multiple instruments within an orchestra) are recorded individually and stored discretely on recording media. The invention is particularly well suited for use with a multiplicity of sound sources which are distributed around a real or virtual environment, and for reproduction of the sounds made by those sources on an expanded stereo or a 7.1 stereo surround sound system (although it is suitable also for fewer or more reproduction channels).

Unlike conventional reverberation and spatial effects systems that are entirely post-production oriented, meaning they are used after sound is recorded, this invention may benefit from a specific microphone placement and recording technique used in conjunction with specialized processing upon playback. As a result, the invention may capture the sound quality of a recorded space, and it may permit continuously variable control of the size and liveliness of the apparent recording environment.

The invention enhances the quality and controllability of sampled sound libraries, and it is also applicable to many other situations where one wishes to have real-time, continuously variable control of the apparent size of the acoustic sound field upon playback of recorded or synthesized sound. The invention is applicable for both studio production use as well as for live performances of sampled or synthesized music and for computer gaming. It can also be applied to benefit broadcast and other means of sound distribution such as CD, DVD, Internet and other means, including future means, of audio distribution, storage and reproduction.

Many years ago, recordings for home listening were made in simple 2-channel (left-right) stereo. (This, of course, was subsequent to the "ancient" monaural days.) For sound reproduction in motion picture theaters, a center channel was added to the stereo channels to keep the "audio image" from appearing to drift. So, theatrical recordings were made in 3-channel (left-center-right) stereo. Then along came 5-channel stereo recording that created expanded stereo for wide-screen presentations (adding far left and far right to the left-center-right sound field). (Other channel allocations and speaker placements were used as well.) A "subwoofer" or very low bass channel was added, and this arrangement can be referred to as 5.1 (pronounced "five-dot-one") expanded stereo. Typically in 5.1 expanded stereo sound production, the subwoofer channel is artificially created from the 5-channel stereo mix, although special effects can be added to the subwoofer channel (such as rumble to simulate rocket engines,

explosions, earthquakes and so forth). Left and right rear speakers have also been added. These rear speakers are often referred to as "surround" speakers and the audio channel feeds to them are referred to as "surround" channels. In many motion picture theaters today, there are three speakers (a left speaker, a center speaker and a right speaker) behind the projection screen (i.e., at the front of the auditorium), a subwoofer (low frequency speaker) usually in front and below the screen, and two speakers in the rear of the auditorium (a left rear (or left surround) speaker and a right rear (or right surround) speaker), with a separate audio channel feeding each speaker. This system is usually referred to as a "5.1 stereo surround" system or simply a "5.1" system. Many home theater systems emulate the motion picture theater systems and 5.1 stereo surround has become a standard for high-quality motion picture soundtracks and digital video disc (DVD) soundtracks.

Some high-end productions are being released in 7.1 ("seven-dot-one"), adding far left and far right channels to the behind the screen speakers. Some are experimenting with other formats involving different numbers of channels.

In the prior art recording of orchestras, for greater realism and ease of production, symphonic recording sessions would be done in a very large room with 50 to 100 musicians all playing as though they were in a live concert. This assembly of musicians typically would be recorded using from 15 to 25 microphones ("mics"). The audio signals from the mics would be mixed down (combined) to fewer channels, which channels were ultimately preserved on discrete tracks of a professional digital or analog recording system.

This "ideal" situation is seldom available, and more frequently symphonic recordings (such as for motion picture soundtrack scoring) are made with sections of about 10 instruments at a time in multiple recording sessions. The effect of a "full orchestra" is created by mixing together the results of these multiple sessions. This compromise is made primarily to avoid the high cost of hiring a large orchestra and of renting a large enough studio or hall in which to record it. Using a process called overdubbing, a musician can listen in headphones to previously recorded portions of the program while recording additional material, thus allowing the musician to synchronize his/her playing and to control the timber and volume of the instrument similarly to what would naturally occur in a full orchestra environment. Thus, one musician can, in sequence, play several parts or different instruments. However, the overdubbing process does not provide the same sonic quality as can be achieved with all members of the orchestra actually present at once in a large space.

Despite the pressure for economy, the need for superb sonic quality has not diminished, especially for large-sounding motion picture (or television) scores. There is still a demand for the "large sound."

Primarily in response to the high cost of live musical scoring, digitally sampled music technology has been taking over the recording industry. Using sampling technology, a library of recorded instrument sounds can be played to create soundtracks without the need for any (or at least not as many) live players. The libraries are played using digital devices known as samplers, which may be embodied as computer programs, or the samplers may be embodied in dedicated hardware systems. Unfortunately, there are myriad difficulties when using today's sample libraries or even using synthesizers.

The overall systems (library and sampler) seldom deliver their promised realism, economy, or simplicity of use. Many sampled instrument libraries only go as far as 2-channel stereo, although some have been created to satisfy the 5.1 format. The 7.1 format has not yet been tackled by samplers, and even the latest releases of 5.1 libraries pose serious technical and performance issues in practical application only a handful of libraries sold in recent years are entirely new productions, with most libraries using at least some reworked recordings from older, poorer quality libraries. Since leading-edge sound quality is difficult to find in samplers, and is not uniform across the sounds offered, and since the 5.1 sampler systems are either too complex or just plain fail to function satisfactorily when pressed to the limit (i.e., recreating the sound of about one hundred different instruments at a time to simulate a full orchestra), simple 2-channel stereo libraries have continued to sell. In other words, there is a large gap between what the market would like to have at its disposal, and what is practically achievable from currently available products.

To extend the utility of 2-channel stereo libraries, end users have attempted to simulate the 5.1 surround sound environment by using 5.1 surround reverb software programs. Typical embodiments would be the plug-in software "processors" like those sold by Lexicon of Sandy, Utah or TC Electronics of Risskov, Denmark, and intended for use with ProTools™ software/hardware platforms of Digidesign, a division of Avid Technology, Inc., of Daly City, Calif. Although these are useful tools, they do not accurately simulate the sound of multiple microphones picking up all the instruments during a live recording session in a single space (such as an auditorium or studio).

Today's hardware-based and software-based 5.1 reverbs treat each signal as though it were being picked up by five room microphones, typically in a so-called "Decca Tree" configuration. A Decca Tree arrangement usually consists of five microphones many feet above and in front of the sound source (three in a frontal triangle plus added far left and far right microphones). This arrangement was first popularized in Decca Records' London studios many years ago. Other arrangements of five microphones are simply known as 5-channel sound. Such reverbs do not simulate the additional spot microphones that are present during recordings of a full complement of musicians in a larger (or even in a smaller) recording studio or hall.

The difference between typical 5.1 reverb-processed surround and live 5.1 surround recording must be understood in order to appreciate all that the invention described herein does to advance the state of the art. Reference here is made to FIG. **10**, which includes an overhead view of a traditional live orchestra arrangement for over a hundred musicians playing an even greater number of instruments **22** (including violins, flutes, brass, percussion and so forth), which are depicted in the drawing as rectangular and circular shapes, positioned in the orchestral stage portion **32** of a recording studio **34** (which could be a concert hall). Spot mics **24** are distributed throughout the area of the sound stage where the orchestra is positioned, and room mics **26** are positioned near the front of the orchestra and significantly higher than the spot mics. The spot mics are typically directional mics and primarily pick up the sonic character of the instruments near them. In FIG. **10** twenty five spot mics are shown, which would be within the range typically used for recordings of a full orchestra. The room mics are typically omni directional mics and pick up more of the sound of the environment as well as the blended sound of many instruments. In FIG. **10**, seven room mics **26** are shown. One each is positioned, respectively, at the far left, left, center, right and far right above the front of the orchestral stage (which is coincident with the front of the part of the studio representing the audience area **36**). The two remaining room mics are positioned to the far left and far right of the rear of the audience area **36**. These two remaining mics are intended to pick up the sounds which would reverberate from the rear of the audience area **36**.

During a live orchestral recording session, each microphone, including each of the spot mics **24** and room mics **26** in the studio (or on stage), picks up sounds from all the instruments present. This "mic bleed" is what gives live ensemble recordings a feeling of space and depth as well as the impression that all the musicians were playing in the same location at once. This is true whether the recording is for a full performance of a musical program or whether it is for the purpose of deriving digital samples that capture the sound of the studio or hall. Orchestral recording is discussed here because it is among the most complex and challenging, but the descriptions apply equally to almost any kind of sound recording. The five front room microphones (far left—left—center—right—far right) are located high in the air and near the front of the orchestra, and they serve to capture much of the sound of the studio (or hall) as well as of the orchestra itself (or the sound of a section of the orchestra if the recording is being done with a subgroup of musicians), but the room or surround mics are not the only mics contributing to the recording. Numerous spot mics which are placed lower and closer to specific instruments or instrument sections, are mixed into the overall recording at which time they are panned to the appropriate left-to-right location in the stereo sound field.

Many recording engineers have tried to use only or primarily the room microphones (typically in a "Decca Tree" microphone array or some variation thereof) to capture the entire performance, but they have generally found the results to be unsatisfactory. Such room-mic-only recordings can seem "muddy" or lacking in clarity and definition. The "presence" (i.e., the midrange/high frequency spectral content) of any given sound source is better captured by a nearby microphone, and that is why spot mics are almost always used in conjunction with room mics to achieve satisfactory results. However because the room and spot mics are picking up the same sound but in different locations, there often are phase-related sonic cancellations, which make it very difficult to optimize the placement and mixing of the various mics. Among its other benefits, the invention controls this phase cancellation problem.

Typically, in prior art methods, which pick up sounds that are to be similarly panned, spot mics are sub-mixed into "stems" (in which case each spot mic is not recorded discretely, but instead is recorded along with other spot mics as part of a group or stem). This means that existing master recordings are generally unsuitable for use with this invention, as will become evident once the method of this invention is understood. Although the microphone closest to one instrument will predominantly pick up the sound of that instrument, it also picks up the sound of every other instrument playing, the phenomenon previously mentioned as "mic bleed." The bleed sound from more distant instruments is usually detected at a lower volume level than the sound of the instrument(s) nearest to a given spot mic due to normal acoustic attenuation over distance. Also, because sound travels at a finite speed (about 1.1 feet per millisecond), the sound from a particular source reaches the spot mic nearest it sooner than its sound reaches a remotely spaced spot mic. Additionally, the sound arriving at the remotely spaced spot mic is diminished in high frequency content due to the differential attenuation of higher frequencies by the air itself. Thus the spectral balance of an instrument in the bleed sound picked up by each more distant spot mic is not the same as sound picked up by the spot mic nearest to the instrument. The more distant a sound source is from a given spot mic, the more reverberation is present (rapid, blended sound reflections primarily from sound bouncing off the floor, ceiling and walls). Such reverberation decays more slowly than the direct sound from the sound source. Also, the orientation of each spot mic (or room mic, for that matter) with respect to a given sound may differ, introducing additional phase shift with respect to that same sound as it is sensed by each of the various mics. Thus, the bleed in different spot mics comprises different "versions" of the sound from a given source, depending largely upon the spot mic distance from the source. With each instrument (or other sound source) bleeding into all the other spot microphones, the overall captured sound field from all these spot mics imbues a "color" and a recognizable characteristic to the recorded sound, one that gives the impression of a particular sized recording space.

The sound mix of all the spot mics provides most of the character and impression of size to a recording. The sounds recorded by the spot mics are enhanced by the elevated room microphones such as those in a Decca Tree. By way of example, if one records a single instrument in a large hall with only the nearest spot mic, and no other spot mics—that recording does not sound as spacious or appear to be as realistic as one recorded with all the typical spot mics and Decca Tree room mics contributing to the recording. In fact, it sounds almost as though it were recorded in a much smaller room. Adding conventional sound effects and reverberation processing will make the sound appear "larger" but cannot fully overcome the lack of realism and spaciousness, particularly when recording with multiple "spot-mic'd" sources. Even with a five room mic sound array added to the mix, the sound still lacks the depth and spaciousness that can be heard when all the spot mics and room mics pick up bleed from all the instruments and contribute this sound to the mix.

For prior art sampled sound, several obstacles arise for sampler playback if the samples were recorded with the typical 15 to 25 spot mics plus the 5 room mics picking up the orchestra (or any group of musicians or singers). The first and most obvious issue to anyone who has tried to use sample libraries is that the sound of the hall is "locked in" through this technique, particularly if the sample recording was done in a large space. In this instance, the long time delays and natural sound reflections in a large recorded space become part of the sampled sounds and cannot later be removed or altered appreciably. This is an insurmountable obstacle when one needs the sound of a smaller environment and the samples were recorded in a large environment. Going the other way, small studio recordings can be "stretched" somewhat through the addition of artificial reverberation and delay processing, but this does not accurately create the sound field achieved from mic bleed when spot microphones would be laid out further apart in a larger space. Because one does not have access to each spot microphone in a mixed-down multi-instrument sample recording (even if it is in 5.1 format), there is no way to alter the relationship of the spot mic contributions, nor do conventional reverbs provide means to simulate the way the spot mics function to create the impression of the live sound field.

When smaller numbers of musicians are recorded in smaller studios, as is more often the case today, the deficiency of not having the correct spot microphone scaling is especially evident. Many engineers, composers and musicians may be unaware of why the sound seems "too small" or incorrect, and so when they attempt to scale up the sound of a smaller studio using conventional 5.1 reverb embodied in software or hardware based systems, they find that no amount of conventional post-production reverberation effects can truly emulate the spaciousness of the sound which would have been achieved in the larger environment with the mic bleed. Even if they manage to come close to the "right sound" for one or a few instruments, the effect deteriorates when a whole section of instruments or an entire orchestra is processed—even using the most advanced 5.1 reverb systems. This inability to satisfactorily scale the orchestra (or other group) occurs because the individual instruments in the sample library lack the correct and unique spot mic "bleed" contributions; treating the sum of all the instruments with reverb processors as though they came from one (or even 5) locations simply cannot emulate what happens with an array of five or more room mics and a large array of many spot mics.

The inability to accurately and continuously scale the apparent size of the recorded space, particularly to make it smaller, is a drawback with conventional sample libraries and, as stated above, it is a deficiency that can barely be compensated using reverberation and effects processing. However, there is another major obstacle to the implementation of surround stereo recordings with sampler technology. That is, today's best full-orchestral library samplers with surround sound capability are difficult to set up and use. The means by which conventional samplers function to provide changes in the characteristics of notes played typically requires that several "versions" of a given note be loaded at once, multiplying the number of samples that must be processed in real time. Because the demand on computer resources rises and falls as a prior art sampler is played, the internal computer data busses and input/output ports can suddenly and unpredictably "choke" (create glitches in the sound or crash completely) if the user demands playback of "just that one more note." Consequently real-time, one-take performance is difficult to achieve from a large-scale 5.1 sample library.

Setting aside the practical aspects noted above for a moment, there are serious sonic issues even assuming one manages to get the prior art sample library playing satisfactorily and reliably. True 5.1 recordings of a sound have many open mics (that is, they have many microphones picking up and recording the sound). Not only do the mics pick up the sounds of the instruments, all these mics also pick up the sound of the room—which includes undesirable noise such as air conditioning rumble, coughs, flipping pages of music and shuffling feet. Besides room noise, for each "live" mic there is a degree of electronic noise present (even a simple carbon resistor generates electrical noise due to thermally-stimulated molecular activity). When one plays back a 5.1 recording of a typical complement of 25 spot and room microphones, one hears 25 mic's worth of electric noise, plus five or six times the ambient room noise. This noise is then multiplied by each note being played; add another note to a chord, and you get another dose of all that room and mic noise. Indeed, noise builds up rapidly, even with what seemed like a very quiet studio to the naked ear. Thus, even a genuine, well-recorded prior-art 5.1 sample library would not sound as good as would a live performance or a single-take recording of an orchestra with just one set of open mics that are heard just once. As discussed below, a sample used with the current invention has the sound picked up by only one mic (typically a spot mic) or two mics (a spot mic and a room mic).

Yet another problem occurs with conventional sampled libraries. Each individual recorded note is not simply stored in the sampler but rather it is digitally edited. In some cases this is done to make it possible to sustain a note (as if it were played by a bowed violin, a horn, or a piano with the sustain pedal depressed), however long the original recorded note might have been, for as long as a keyboard's key is held down.

Thus, instead of storing the full note, the sampler version has a truncated note, with the ends "chopped off" at precisely chosen times (or more accurately stated, precise locations in the sonic continuum of the note) selected so that the end of the note can be dove-tailed into the beginning of the note in a process known as "looping." When a key is released (i.e., when a note is no longer wanted), any looped sound ceases and the sampler now plays the "decay" or final portion of the recorded sound. With a conventionally recorded note's sound, the reverberation of the room in which the note was played is largely heard at the end of the note and is inextricably mixed with the sound of the note trailing off to silence. Sometimes the person playing these sampled notes wants the sound to end more quickly than was originally recorded, and so a sampler function can be invoked which truncates (chops off the decaying sound) at the end of such notes. Unfortunately, truncating a note causes a significant loss of the reverberant information as well so the impression of the size and depth of the recorded environment is lost. Using the subject invention to process recorded notes, even if they are truncated, overcomes the problem found with conventionally truncated samples in that the room characteristic is still present (i.e., added by the invention's processing that creates "lingering" reverberation when a note is released).

SUMMARY OF THE INVENTION

The instant invention allows the size of the apparent recording space to be scaled with continuous variability, in real time during playback, with no need to stop and retrieve alternate sound recordings. The instant invention has other benefits as well. The invention relies on a recording made either with one spot mic alone or with one spot and one room mic, and on a processor used upon playback to give broad control over apparent room size and sound source location within the apparent recording environment. However, the recorded sound itself needs no special encoding, no phase manipulation, and no additive or subtractive manipulation between the recorded tracks. This lack of manipulation of the sound being recorded is important in that sonic quality and image stability are not at all affected, issues that had been raised by certain prior art approaches to capturing multiple mics (or multiple channels such as with quadraphonic sound as described during the early 1970's in CBS' SQ™ and Sansui's QS™ matrix systems), storing them on fewer tracks (stereo), and then reconstituting the original channels upon playback; there is no complementary encoding-decoding in this invention, although there is signal processing upon playback.

Once the individual sound sources have been recorded with one or two microphones, the invention permits the real-time playback processing of multiple single-channel voices (i.e., digitally stored sounds) to place them in a 5.1 or 7.1 surround sound environment, although 9.1 surround (this is 7.1 with two additional side fill channels) or other playback formats are easily achieved using this same method. Each sound source can be placed in a realistic spatial relationship, or the apparent location can be arbitrarily moved to almost any "virtual" position regardless of where it was during the original recording, not just left-to-right but also front-to-rear.

The invention can function to some extent with any monaural or stereo recording but it works best when the sound being processed has been recorded with one spot microphone (or one spot microphone plus one room microphone in the two-channel format) as described herein. Because as little as one channel is needed to realistically produce a low-noise, high quality 5.1 or 7.1 surround playback environment, the amount of media needed to store recordings done in this fashion may be reduced, and it becomes easier to avoid exceeding the processor capability of the computers and samplers used to play back the samples. The nature of the process allows for a wide range of scaleability from a very small to a very large sonic space without compromise of quality, in real time, with smooth and continuous variability.

The spot mic bleed simulator portion of the invention processes the audio signal from a spot mic recording in order to simulate the audio bleed pickup of multiple microphones distributed among a plurality of predetermined spot mic location zones superimposed on a virtual sound stage. The spot mic bleed simulator includes a plurality of spot mic processors, preferably a spot mic processor for each one of the designated spot mic location zones, and each of these spot mic processors includes, among other elements, a series-connected spot mic low pass filter, spot mic delay and spot mic attenuator that together alter the original spot mic signal so it takes on the characteristic of a sound which would have been picked up by an actual spot mic at the location designated by the corresponding processor's zone. The output of each of these spot mic processors within the spot mic bleed simulator feeds a router and related output switching which together comprise a multiplexer that feeds summed subsets of these processed spot mic audio signals to one or more of the outputs of the mic bleed simulator system. In a similar manner the room mic bleed simulator processes the audio signal from a room mic recording in order to simulate the audio bleed pickup of multiple room microphones distributed among a plurality of predetermined room mic location zones superimposed on a virtual audience area, and feeds these signals through the invention's router-multiplexer, summers, and ultimately to one or more outputs of the mic bleed simulator system. Additionally, outputs of the spot mic bleed simulator and room mic bleed simulator may be routed and processed by the invention to derive rear surround outputs and subwoofer outputs when such outputs are designated as being active.

BRIEF DESCRIPTIONS OF THE DRAWINGS

FIG. **1** is block diagram of the bleed simulator of the instant invention as configured for use with two-channel recordings created with one spot and one room microphone wherein each channel drives its respective (spot or room) mic simulator.

FIG. **2** is a detailed block diagram of a few of the spot mic processors within the spot mic simulator of the bleed simulator.

FIG. **3** is a block diagram of all the spot mic processors for a preferred embodiment of the bleed simulator.

FIG. **4** is a detailed block diagram of a few of the room mic processors within the room mic simulator of the bleed simulator.

FIG. **5** is a block diagram of all the room mic processors for a preferred embodiment of the bleed simulator.

FIG. **6** is a detailed block diagram of two of the main output mixers of the bleed simulator.

FIG. **7** is a detailed block diagram of the in-line on/off switch array and summing networks in one of the main output mixers.

FIG. **8** is a block diagram of the subwoofer output mixer.

FIG. **9** is a block diagram of the surround simulator of the bleed simulator.

FIG. **10** is a top plan view of a typical arrangement of room and spot microphones and the instruments of a large orchestra in a sound recording studio for conventional recording, with a superimposed zone map that identifies the processing zones used in connection with the invention described herein.

FIG. **11** is a top overhead view showing the alignment and position of the room and spot microphones when two microphones are used to record one instrument.

FIG. **12** is a side elevation view showing the alignment and position of the room and spot microphones when two microphones are used to record one instrument.

FIG. **13** is a top plan view of a simplified recording environment (similar to FIG. **10**) with measured distances from a virtual instrument source in the middle of zone A**1** to all other zones (representing virtual spot and room mics), and with calculated time-to-arrive for the sound from that source as well as calculated attenuation values for the sound at each virtual spot and room mic.

FIG. **14** is a table with a particular set of CPU instructions for the spot mic simulator, the room mic simulator and the surround simulator corresponding to a particular recording environment, virtual source zone designation, and output configuration.

FIG. **15** is a table with a particular set of CPU instructions for to the main output mixer for the same recording environment, virtual source zone and output configuration as defined for FIG. **14**.

FIG. **16** is block diagram of the bleed simulator of the instant invention as configured for use with one-channel recordings.

DETAILED DESCRIPTION OF THE INVENTION

An example of a set up for making two-channel recordings for samples used with the invention is shown in FIGS. **11** and **12**. In the example, the sound source is a single violin **901** which is being recorded in a studio which is about 20 feet wide×35 feet long by 30 feet high. Spot mic **24** is positioned 2 to 5 feet in front of and ½ to 5 feet above the center of the violin. The spot mic's capsule is pointed toward the center of the violin. Room mic **26** is positioned 15 to 20 feet in front of and 10 to 20 feet above the center of the violin. When viewed from above (as in FIG. **11**), the room mic is co-linear with the violin and spot mic. The angle of the room mic's capsule with respect to imaginary horizontal and vertical axes is identical to that of the spot mic's capsule. However, when viewed from the side (as in FIG. **12**), the room mic is offset from a line which would pass through the center of the violin and the spot mic. This avoids the room mic's capsule falling in the acoustic shadow of the spot mic. While this offset results in the room mic's capsule not pointing precisely at the center of the violin (in FIG. **12** the aiming line **903** of the spot mic's capsule points to the center of the violin, showing the slight offset of room mic **26**), the phase relationship of the sounds (including reflections off of the floor) reaching the mics' capsules is preserved by the identical capsule angles.

The signals created by the spot mic and room mic are recorded on two separate channels of a high fidelity recording medium. The channel having the spot mic signal may be designated as the first channel or the spot mic channel, and the channel having the room mic signal may be designated as the second channel or the room mic channel. This two-channel recording is the basis for making digital two channel samples of the various notes played on the musical instrument during the recording session. The method for preparing the digital two-channel samples is essentially the same as used in preparing prior art stereo samples, except that instead of left and right channel signals, the two signals are spot and room mic signals (or, respectively, first and second channel audio signals).

A simplified microphone setup may be used with only the spot microphone **24** of FIG. **11** or FIG. **12**, in which case the only difference in the recording setup is that there is no room microphone and only one channel of recording media is required.

FIG. **10** does double duty in this description. On the one hand, it depicts the layout of a traditional orchestral recording studio **34**, with an illustration of where actual instruments, spot mics and room mics would be typically disposed in the stage portion **32** and the audience area **36**. FIG. **10** also illustrates a virtual recording studio (also designated with reference number **34**). Superimposed over the illustration of the stage portion **32** is a grid of **15** zones (bordered by dashed lines) which are designated as zones A**1**-A**5**, B**1**-B**5** and C**1**-C**5**. These zones are virtual spot mic location zones A**1**-C**5**. The invented bleed simulator emulates the sound that would be picked up by a single spot mic located in the center of each of the 15 virtual spot mic zones.

Another grid divides the area immediately in front of the virtual stage into five virtual room mic location zones, RM**1**-RM**5**. These represent the placement of virtual room mics. The invented bleed simulator emulates the sound that would be picked up by a single room mic located in the center of each of the five virtual room mic zones.

For purposes of indicating the virtual location of a given sound source in the stage portion of the recording studio, another grid could be used to designate virtual source zone locations. The virtual source zones need not coincide with the virtual spot mic zones (e.g., there may be fewer or greater virtual source zones than virtual spot mic zones). However, in the preferred embodiment the virtual spot mic zones and virtual source zones are coincidental with each other. (Sometimes herein the virtual spot mic zones, virtual room mic zones and virtual source zones are referred to as processing zones.) For playback of a sample prepared in accordance with the above two-channel recording method, the sample is fed into the bleed simulator **110** of the instant invention shown in FIG. **1**. (Single channel recorded samples are handled in a similar manner; the playback method variation for such samples is illustrated and discussed with FIG. **16**.)

The bleed simulator comprises,

- an audio sample input **112**, a virtual spot mic and virtual source zone designation input **114**, a sound stage definition input **116**, an output configuration designation input **118**, an auxiliary subwoofer control input **120**, and an auxiliary subwoofer input **122**;
- an audio sample buffer **124** which receives the audio sample input signal, which is comprised of the first and second audio channel signals (or the spot and room mic components), from the audio sample input **112**;
- a central processing unit (CPU) **126** which is connected to the virtual spot mic and virtual source zone designation input **114**, sound stage definition input **116**, output configuration designation input **118**, and auxiliary subwoofer control input **120**, and which is also bi-directionally connected to audio sample buffer **124** and memory location **128**;
- spot mic bleed simulator **130** which includes one spot mic processor per virtual spot mic zone location, of which there are fifteen in the embodiment of the instant invention depicted herein, designated **130***a* through **130***o*, each of which is connected to and receives a control signal from CPU **126** and is connected to and receives the first channel component of the audio sample signal (i.e., the spot mic component) from buffer **124**, with each spot mic processor having an output connected to elements described below;

room mic bleed simulator **132** which includes a plurality of room mic processors, preferably one room mic processor per virtual room mic zone, of which there are five (5) in the embodiment of the invention depicted herein, designated **132***a*-**132***e*, each of which is connected to and receives a control signal from CPU **126** and is connected to and receives the second audio channel component of the signal (i.e., the room mic component) from buffer **124**, with each room mic processor having an output connected to elements described below;

router **133** which comprises a set of physical busses (or logical switching to accomplish the same function) to deliver sample signal components from the various spot mic bleed simulator processors and room mic bleed simulator processors to the main output mixer array, subwoofer output mixer and surround simulator;

main output mixer array **134** which includes far left main output mixer **134***a*, left main output mixer **134***b*, center main output mixer **134***c*, right main output mixer **134***d*, and far right main output mixer **134***e*, each of which is connected to and receives a control signal from CPU **126** and is connected to and receives processed first and second channel components of the audio sample signals (i.e., spot and room mic components) from all spot mic and room mic processors, with the processed audio sample signals flowing through switch arrays in the mixers which pass only suitable signals for the main output channels **136***a-e* of the bleed simulator;

subwoofer output mixer **138** which is connected to and receives a control signal from CPU **126**, is connected to and receives the processed first channel component of the audio sample signal (i.e., the processed spot mic component) from all spot mic processors, and is connected to and receives an auxiliary audio signal (if one is present) from auxiliary subwoofer input **122**, and which feeds the subwoofer output **140** of the bleed simulator; and

surround output simulator **142** which includes rear left surround output mixer **142***a* and rear right surround output mixer **142***b*, each of which is connected to and receives a control signal from CPU **126**. Rear left surround output mixer **142***a* is connected to and receives the processed second channel component of the audio sample signal (i.e., room mic component) from the outputs of far left room mic processor **132***a*, left room mic processor **132***b* and center room mic processor **132***c*, via their respective busses from router **133** and feed the rear right surround output **144***a* of the bleed simulator. Rear right surround output mixer **142***b* is connected to and receives the processed second channel component of the audio sample signal (i.e., room mic component) from the outputs of center room mic processor **132***c*, right room mic processor **132***d*, and far right room mic processor **132***e*, via their respective busses from router **133** and feeds the rear right surround output **144***b* of the bleed simulator.

In viewing FIG. **1** (as well as FIG. **16**), the signal flow lines represent a bus available simultaneously to all five output mixers and are not processed sequentially by these mixers.

The spot mic bleed simulator **130** is shown in more detail in FIGS. **2** and **3**. As noted above, the spot mic bleed simulator includes fifteen (15) spot mic processors (virtual spot mic zone A**1** spot mic processor **130***a*, zone B**1** mic processor **130***b*, zone C**1** spot mic processor **130***c*, zone A**2** spot mic processor **130***d*, zone B**2** mic processor **130***e*, zone C**2** spot mic processor **130***f*, zone A**3** spot mic processor **130***g*, zone B**3** mic processor **130***h*, zone C**3** spot mic processor **130***i*, zone A**4** spot mic processor **130***j*, zone B**4** mic processor **130***k*, zone C**4** spot mic processor **130**L, zone A**5** spot mic processor **130***m*, zone B**5** mic processor **130***n*, and zone C**5** spot mic processor **130***o*), which number corresponds, in the illustrated embodiment, to the number of virtual spot mic zones into which the virtual sound stage is divided as shown in FIG. **10**. Each spot mic processor includes the same elements, which elements are configured according to suitable stored information as explained below.

The number of processing zones actually used would depend on several factors, including the number of main outputs, the amount of computing power to be made available, and the degree of bleed simulation precision desired. The number of spot mic processors actually used could be fewer or greater than the number of virtual spot mic zones. The number used would also depend on several factors, including the number of main outputs, the amount of computing power to be made available, and the degree of bleed simulation precision desired.

For the types of simulations the inventors have contemplated, particularly with respect to orchestral arrangements in typical recording studios, the number of virtual spot mic zones is designated as fifteen, a compromise between adequate spatial resolution and conservation of computing resources. Similarly the inventors contemplate using in the spot mic bleed simulator one spot mic processor per virtual spot mic zone.

Each spot mic processor includes the following elements as shown in FIG. **2**:

a switch **203** which receives the first channel component of the audio sample signal (in the example discussed here, the spot mic component of the audio sample signal, sometimes referred to as the "spot mic component") from the audio sample buffer **124**;

an all pass filter **205**, having a variable range of 1° to 360°, connected to switch **203** in series;

a low pass filter **207**, having a variable range of 1 kHz to 20 kHz, connected to all pass filter **205** in series;

a delay circuit **209** connected to low pass filter **207** in series with respect to the spot mic component, with the delay circuit including a primary delay **209***a* with a range of 0 to 900 mSec connected in parallel with series-connected delay **209***b*, having a range of 0 to 999 mSec, and attenuator **209***c*, having a range of 0 to 96 dB;

a reverb **211**, having a depth range of 0 to 100% modulation and a decay time range of 100 mSec to 10 Sec, connected in series with delay circuit **209**; and

another attenuator **213**, having a range of 0 to 96 dB, connected in series with reverb **211**.

The outputs of the attenuators **213** of each of the spot mic processors are applied to the spot mic busses of router **133**.

Each of the elements, including both delays and the attenuator of the delay circuit, of a spot mic processor includes a control input which receives the control signal from CPU **126**.

The control signal from CPU **126** sets the operating parameters of each of the various spot mic processor elements, depending upon the virtual source zone designated as the source of the sound represented by the audio sample, and with reference to memory location **128**, which stores information relating to the parameter settings for room size, source zone location and output configuration. The information may be in the form of a lookup table such as shown in FIGS. **14** and **15** or the parameters may be calculated with the use of formulas. Using the lookup table of FIG. **14** (which corresponds to, among other things, a particular virtual source zone), the CPU would issue instructions such that switch **203** in spot mic processor **130***b* may be turned on (which would enable that particular processor), all pass filter **205** may be set to 120°, low pass filter **207** may be set to 12,000 Hz, delay **209***a* may be set to 16.400 msec, delay **209***b* may be set to 16.403 msec, attenuator **209***c* may be set to 17 dB of attenuation, reverb **211** may be set to 0% depth and hence the delay time (in seconds) is of no consequence, and attenuator **213** may be set to 15.6 dB of attenuation. In this example, the first channel component of the audio sample signal (i.e., the spot mic component) passing through spot mic processor **2** would shift in phase relative to the signal at input **112**, would have reduced high frequency content, would be delayed and comb filtered, would have no added reverb, and would be reduced in amplitude.

The particular settings would correspond to control factors which are inputted to the CPU. Each spot mic processor receives its own set of instructions. When the outputs of all spot mic processors are routed and summed in accordance with the system shown and described, the desired spot mic bleed simulation is achieved.

The room mic bleed simulator **132** is shown in more detail in FIGS. **4** and **5**. As noted above, the room mic bleed simulator includes five (5) room mic processors (far left room mic processor **132***a*, left room mic processor **132***b*, center room mic processor **132***c*, right room mic processor **132***d*, and far right room mic processor **132***e*), which number corresponds, in the preferred embodiment, to the number of virtual room mic zones.

The number of room mic processors actually used could be fewer or greater than the number of virtual room mic zones. The number used would depend on several factors, including the number of main outputs, the amount of computing power to be made available, and the degree of bleed simulation precision desired. For the types of simulations the inventors have contemplated, they have found that five (5) room mic processors is a suitable quantity.

Each room mic processor includes the following elements as shown in FIG. **4**:

- a switch **403** which receives the second channel component of the audio sample signal (in the example discussed here, the room mic component of the audio sample signal, sometimes referred to as the "room mic component") from the audio sample buffer **124**;
- an all pass filter **405**, having a variable range of 1° to 360°, connected to switch **403** in series;
- a low pass filter **407**, having a variable range of 1 kHz to 20 kHz, connected to all pass filter **405** in series;
- a delay circuit **409** connected to low pass filter **407** in series with respect to the room mic component, with the delay circuit including a primary delay **409***a* with a range of 0 to 900 msec connected in parallel with series-connected delay **409***b*, having a range of 0 to 999 mSec, and attenuator **409***c*, having a range of 0 to 96 dB;
- a reverb **411**, having a depth range of 0 to 100% modulation and a decay time range of 100 mSec to 10 Sec, connected in series; and
- another attenuator **413**, having a range of 0 to 96 dB, connected in series with reverb **411**.

The outputs of the attenuators **413** of each room mic processor are applied to the room mic busses of router **133**.

Each of the elements, including both delays and the attenuator of the delay circuit, of a room mic processor includes a control input which receives the control signal from CPU **126**.

Just as with the spot mic processors, the control signal from CPU **126** sets the operating parameters of each of the various room mic processor elements. For example, with reference to the lookup table of FIG. **14**, switch **403** in center room mic processor **132***c* may be turned on (which enables this processor), all pass filter **405** may be set to 360°, low pass filter **407** may be set to 6,000 Hz, delay **409***a* may be set to 5.900 msec, delay **409***b* may be set to 5.917 msec, attenuator **409***c* may be set to 18 dB of attenuation, reverb **411** may be set to 0% depth and hence the delay time (in seconds) is of no consequence, and attenuator **413** may be set to 2.7 dB of attenuation. In this example, the second channel component of the audio sample signal (i.e., the room mic component) passing through the center room mic processor would have no shift in phase relative to the signal at input **112**, would have reduced high frequency content, would be delayed and comb filtered, would have no added reverb, and would be reduced in amplitude.

As with the spot mic processors' settings, the particular room mic processor settings would correspond to control factors which are inputted to the CPU and stored in memory location **128**, with each room mic processor receiving its own set of instructions. When the outputs of all room mic processors are routed and summed in accordance with the system shown and described, the desired room mic bleed simulation is achieved.

In the illustrated embodiment, the number of spot mic processors in the spot mic bleed simulator **130** equals the number of virtual spot mic location zones and the number of room mic processors in the room mic bleed simulator **132** equals the number of virtual room mic location zones. In the preferred embodiment (here for a 7.1 surround system) as shown in FIG. **1**, there are fifteen (15) virtual spot mic zones, namely A**1**-**5**, B**1**-**5** and C**1**-**5**, and five (5) virtual room mic zones, namely RM**1**-RM**5**. In an output configuration such as a 7.1 system, where all five main outputs are used, the output of each respective room mic processor is routed to all the main output mixers of main output mixer array **134** as shown in FIG. **1**, namely far left main output mixer **134***a*, left main output mixer **134***b*, center main output mixer **134***c*, right main output mixer **134***d*, and far right main output mixer **134***e*, which are enabled such that the signal from each room mic processor flows only to a corresponding main output. Different output configurations such as a 3.1 or 2.0 system will result in varying the use of the main output mixers, and for this reason the main output mixer array is configured with logically-controlled switches to permit activation of any of the room mic processors to feed any of the main outputs. Similarly, the outputs of all spot mic processors are routed to each of the main output mixers and logically-controlled switches permit activation of any combination of the spot mic processors to feed any of the main outputs.

The main output mixer array **134** is shown in detail in FIG. **6** and FIG. **7**. Each main output mixer includes two pairs of switch arrays and summing networks (i.e., switch array **601** connected in series to summing network **603** and switch array **607** connected in series to summing network **609**). FIG. **7** is an expanded view of these switch arrays and summing networks, the inputs of which are connected to the output mixer's spot mic component inputs and room mic component inputs, respectively. The summed output of each network in turn feeds, respectively, attenuators **605** and **611**. The attenuation range of each of the attenuators is 0-96 dB. The outputs of the attenuators of an output mixer are summed together and fed to the corresponding main output of the bleed simulator (i.e., the output of far left main output mixer **134***a* is connected to the far left main output **136***a*, the output of left main output mixer **134***b* is connected to the left main output **136***b*, the output of center main output mixer **134***c* is connected to the center main output **136***c*, the output of right main output mixer **134***d* is connected to the right main output **136***d*, and the output of far right main output mixer **134***e* is connected to the far right main output **136**_e_). Each of the elements of the main output mixers is connected to and receives a control signal from CPU **126**.

The control signal from CPU **126** sets the operating parameters of each of the various main output mixer elements with reference to information stored in memory location **128**. Based on the parameters, the relative contributions of the processed audio signal components from the spot mic and room mic processors may be adjusted to obtain a desired sound effect. Similarly, each switch would be controlled to enable or disable selected main output mixers to achieve a particular output configuration (e.g., shutting off all the on/off switches in arrays **601** and **607** of main output mixers **134**_b_ and **134**_d_ effectively reduces a 7.1 surround system to a 5.1 surround system; other switches in arrays **601** and **607** of main output mixers **134**_a_ and **134**_e_ would be turned on to avoid loss of desired sound components in this example).

Together the router **133** and main output mixer array **134** form a multiplexer.

Subwoofer mixer **138** is shown in detail in FIG. **8**. It includes a spot mic subwoofer processor **701** which is connected to and receives processed first channel (or spot mic) audio sample signal components from all the spot mic processors via feeds from the router. The spot mic subwoofer processor **701** includes a summing network **702** which sums together the processed first channel (or spot mic) audio sample signal components from all the spot mic processors, a switch **703** connected in series to the summing network, low pass filter **705**, having a range of 1-20 kHz, connected in series to the switch **703**, and attenuator **707**, having a range of 0-96 dB, which is connected in series to the low pass filter. The subwoofer mixer also includes a subwoofer aux input processor **708** which is connected to and receives an auxiliary audio signal from auxiliary subwoofer input **122**. The subwoofer aux input processor includes switch **709** which receives the auxiliary audio signal, low pass filter **711**, having a range of 1-20 kHz, connected in series to the switch, and attenuator **713**, having a range of 0-96 dB, which is connected in series to the low pass filter. The outputs of the spot mic subwoofer processor and the subwoofer aux input processor (i.e., the outputs of attenuators **707** and **713**) are summed together and are fed to subwoofer output **140** of the bleed simulator.

Except for the summing network, each of the elements of the subwoofer mixer is connected to and receives a control signal from CPU **126**. The control signal from CPU **126** sets the operating parameters of each of these elements with reference to information stored in memory location **128**. Since the subwoofer mixer is intended to deliver a signal which would drive a very low frequency speaker, each of low pass filters **705** and **711** would typically be set at or below 125 Hz. The settings of the switches **703** and **709** and attenuators **707** and **713** determine the presence and balance of the spot mic and auxiliary audio contributions.

Surround output simulator **142** is shown in detail in FIG. **9**. Rear left surround output mixer **142**_a_ includes attenuator **801**_a_ which is connected to and receives a processed second channel (or room mic) component of the audio sample signal from center room mic processor **132**_c_ via router **133**. The rear left surround output mixer also includes summing network **803**_a_, which is connected to and receives a processed second channel (or room mic) component of the audio sample signal from each of the far left room mic processor **132**_a_ and left room mic processor **132**_b_ via router **133**, and also receives from attenuator **801**_a_ the processed and attenuated signal originating from center room mic processor **132**_c_. The summed signals are fed to switch **805**_a_, which is connected in series with the summing network **803**_a_, low pass filter **807**_a_ (with a range of 1-20 kHz), delay **809**_a_ (with a range of 0-900 mSec), and attenuator **811**_a_ (having a range of 0-96 dB). The output of attenuator **811**_a_ is connected to the bleed simulator's rear left surround output **144**_a_.

Rear right surround output mixer **142**_b_, which includes attenuator **801**_b_, summing network **803**_b_, switch **805**_b_, low pass filter **807**_b_, delay **809**_b_ and attenuator **811**_b_, is essentially identical to rear left surround output mixer **142**_a_, except that where the inputs to summing network **803**_a_ are from the far left and left room mic processors, the inputs to the summing network **803**_b_ are from right and far right room mic processors **132**_d_ and **132**_e_, respectively (also via router **133**), and the output of attenuator **811**_b_ is connected to the bleed simulator's rear right surround output **144**_b_.

The two surround (i.e., rear left and rear right) outputs both derive sound from the same center room mic processor. To avoid doubling in power of this contribution, attenuators **801**_a_ and **801**_b_ would typically each be fixed at 3 dB attenuation.

Switches **805**_a_ and **805**_b_, low pass filters **807**_a_ and **807**_b_, delays **809**_a_ and **809**_b_, and attenuators **811**_a_ and **811**_b_ are each connected to and receive a control signal from CPU **126**. The control signal from CPU **126** sets the operating parameters of each of these elements with reference to information stored in memory location **128**. As long as surround sound is desired, switches **805**_a_ and **805**_b_ would be turned on, and if surround sound is not desired they would be turned off. As the simulated studio size in increased, low pass filters **807**_a_ and **807**_b_ in the preferred embodiment would be set at lower frequencies, while delays **809**_a_ and **809**_b_ would be increased. Attenuators **811**_a_ and **811**_b_ set the balance of surround to main output sound.

Before performing music with the bleed simulator, the user would provide to the bleed simulator the following setup information through inputs **114**, **116**, **118** and **120** shown in FIG. **1**.

At input **120**, the user enters information regarding whether the subwoofer aux input processor **708** should be enabled (i.e., whether switch **709** should be turned on). If the information indicates that the subwoofer aux input processor should be turned on, then the CPU-issued control signal would include an instruction for subwoofer aux input processor **708** to turn on its switch **709**. If it is turned on, then any audio signal applied to input **122** would be processed and output by the bleed simulator at output **140**.

At input **118**, the user enters information regarding which output configuration should be enabled. For example, with respect to the embodiment of the bleed simulator illustrated in FIG. **1**, if the user intends to use all of the outputs, this would be a 7.1 surround configuration. In such event, the CPU-issued control signal would include instructions for main output mixers **134**_a-e_, spot mic subwoofer processor **701**, and surround output simulator **142** such that appropriate switch elements in switch arrays **601** and **607** of the main output mixers **134**_a-e_, switch **703** in spot mic subwoofer processor **701**, and switches **805**_a_ and **805**_b_ in surround output mixer array **142** would be turned on. This would allow processed audio signals to appear at the far left, left, center, right and far right outputs, the subwoofer output and the left and right rear surround outputs. By way of another example, if the user intends to use only two channel stereo (i.e., which could be referred to as a 2.0 configuration), the CPU-issued control signal would include instructions for these elements that causes switch arrays **601** and **607** in main output mixers **134**_b_ and **134**_d_ to be turned on and switch arrays **601** and **607** in main output mixers **134**_a_, **134**_c_ and **134**_e_, switch **703** in spot mic subwoofer processor **701**, and switches **805**_a_ and **805**_b_ in surround output mixer array **142** to be turned off. This would allow processed audio signals to appear on left and right outputs **136***b* and **136***d*, respectively, of the bleed simulator, but not on any other output.

At input **116**, the user enters information about the nature of the virtual studio which the user wants to be simulated by the method and system described herein. Typically, such information would be the length, width and height of the simulated sound stage. It could also include additional information, such as information regarding the reverberance of the sound stage (e.g., whether its walls are acoustically absorptive or reflective). This information, stored in memory location **128**, would result in the CPU-issued control signal to include instructions for the elements of the bleed simulator which would be used for a particular output configuration, which instructions would enable or disable appropriate switches and adjust appropriate delays, attenuators, filters and reverbs as necessary to achieve the desired effect. (This is discussed further below.)

At input **114**, the user enters information about the number and layout of virtual spot mic zones and virtual source zones in the virtual recording studio **34** as shown for example in FIG. **10**. (In the example discussed herein, as noted before, these two sets of zones are coincident.) In addition, for each separate sample which the user intends to use, the user enters the specific source zone in which the instrument(s) on that sample are intended by the user to be located. By way of example, the user may intend to be simulating the recording of an orchestra on a sound stage such as sound stage **32** shown in FIG. **10** which the user desires to have divided into fifteen virtual spot mic location zones and identical virtual source zones A**1**-**5**, B**1**-**5** and C**1**-**5**. If for a particular sample of two violins (a particular desk as described above) the user desires the violins to be positioned at the far left front of the stage, the user would enter the source zone information for that sample as being "A**1**." This information would result in the CPU-issued control signal to include instructions for the elements of the spot mic bleed processors **130***a*-*o* and the room mic bleed processors **132***a*-*e* which would be used for a particular output configuration, which instructions would enable or disable appropriate switches and adjust appropriate delays, attenuators, filters and reverbs as necessary to achieve the desired effect. (This is discussed further below.)

By way of example, if the user desires that the instrument (or instruments) of a particular sample made pursuant to this invention (e.g., a desk of two violins) appear to be positioned in zone A**1** of a virtual recording studio which is 40 feet wide, by 70 feet deep, by 35 foot high with virtual source zones and microphones laid out as shown in FIG. **10**, and where the output configuration is a 7.1 surround sound system, the user would input the virtual spot mic zone layout information and the "A**1**" source zone designation at input **114**, the sound stage description at input **116** and the output configuration at input **118**, and when the audio sample signal is input at input **112**, CPU **126** would issue instructions in accordance with the information depicted in the tables of FIG. **14** and FIG. **15**. The CPU's associated memory **128** is used to store this data so it is available to provide instructions which control the performance of the elements of bleed simulator **110**. Various sets of data may be stored and retrieved from non-volatile memory and loaded into memory location **128** so the invention can rapidly be set to simulate a particular virtual sound source location zone, output configuration and virtual recording environment.

In the specific example discussed here (where the sound source is in virtual source zone A**1** of a virtual recording studio which is 40 feet wide, by 70 feet deep, by 35 foot high with virtual spot mic zones laid out as shown in FIG. **10**), the look up tables of FIG. **14** and FIG. **15** show that the CPU will provide instructions which, among other things, cause the following (in reading the information shown in FIG. **14** and FIG. **15**, it can be seen that units are not identified, but they are known by means of the memory location in which they are stored, and a dash ("—") indicates a null or non-applicable value):

For spot mic processor **130***a*, the CPU would issue instructions which

- turn on its switch **203**;
- set its all pass filter **205** to 290°;
- set its low pass filter **207** to 20,000 Hz;
- set its primary delay **209***a* to 0.000 mSec;
- set its side-chain delay **209***b* to 0.003 msec;
- set its side-chain attenuator **209***c* to 18 dB of attenuation;
- set its reverb **211** to 0% reflection (and its delay time is thus null); and
- set its attenuator **213** to 0 dB of attenuation.

For spot mic processor **130***i*, the CPU would issue instructions which

- turn on its switch **203**;
- set its all pass filter **205** to 360°;
- set its low pass filter **207** to 4,000 Hz;
- set its primary delay **209***a* to 35.900 mSec;
- set its side-chain delay **209***b* to 35.905 msec;
- set its side-chain attenuator **209***c* to 12.0 dB of attenuation;
- set its reverb **211** to 0% reflection (and its delay time is thus null); and
- set its attenuator **213** to 22.4 dB of attenuation.

For room mic processor **132***a*, the CPU would issue instructions which

- turn on its switch **403**;
- set its all pass filter **405** to 360°;
- set its low pass filter **407** to 10,000 Hz;
- set its primary delay **409***a* to 0.000 mSec;
- set its side-chain delay **409***b* to 0.003 mSec;
- set its side-chain attenuator **409***c* to 24 dB of attenuation;
- set its reverb **411** to 0% reflection (and it delay time is thus null);
- set its attenuator **413** to 0 dB of attenuation.

For far right main output mixer **134***e* of main output mixer array **134**, the CPU would issue instructions which

- turn on its switches **601***m*, **601***n* and **601***o* of while turning all other spot mic switches off;
- turn on its switch **607***e* while turning all other room mic switches off;
- set its spot mic component attenuator **605** to 20.6 dB of attenuation; and
- set its room mic component attenuator **611** to 6.3 dB of attenuation.

For rear left surround output mixer **142***a* of surround output simulator **142**, the CPU would issue instructions which

- turn on its switch **805***a;*
- sets its low pass filter **807***a* at 3,000 Hz;
- sets its delay **809***a* to 20.1 mSec; and
- sets its attenuator **811***a* to 7.8 dB of attenuator.

For a given sized virtual recording studio, with particular virtual acoustic characteristics, and with virtual spot mic and source zones laid out as shown in FIG. **10**, there would be fifteen sets of informational data (i.e., information per FIG. **14** and FIG. **15**), one for each virtual source zone location. This data would be stored in memory location **128** accessible to the CPU. In the preferred embodiment the inventors anticipate that there will need to be as many as one pair of spot mic bleed simulator and room mic bleed simulator per virtual source zone (for the example discussed herein, fifteen (15) pairs) so that multiple different recordings with various virtual source locations can be processed simultaneously. While it is theoretically possible to do this by means of multiplexing the input sources and rapidly changing the simulator parameters (effectively virtually multiplying the number of spot mic and room mic bleed processing pairs by time slicing), considerations of presently available processor speed and circuit latency suggest this is not necessarily achievable for a large number of simultaneous, disparately-zoned, high-bandwidth audio signals.

For a virtual recording environment having the same virtual acoustic characteristics and with virtual spot mic and virtual source zones laid out as shown in FIG. **10**, but larger in size than the example just discussed above (e.g., a virtual studio 55 feet wide, by 90 feet deep, by 40 foot high), at least some of the CPU instructions would be different from those given in the previous example. For example, the information for source zone A**1** for that size virtual recording studio may result in the CPU issuing instructions to spot mic processor **130***i* as follows (with explanations of why some values are different from the values given for the previous example):

- turn on its switch **203**;
- set its all pass filter **205** to 360°;
- set its low pass filter **207** to 3,500 Hz (because this larger room would have greater high frequency attenuation due to the greater distances between virtual spot mics);
- set its primary delay **209***a* to 37.2 mSec (because this larger room would have distances between virtual spot mics which would require a longer time for sound from a virtual source to reach a virtual spot mic in another zone);
- set its side-chain delay **209***b* to 37.204 mSec (for the same reason as for delay **209***a*);
- set its side-chain attenuator **209***c* to 11 dB of attenuation;
- set its reverb **211** to 10% reflection and 0.8 seconds decay time (because this larger room would, as a result of being larger, be more reverberant and since the ratio of reverberant to direct sound increases with distance from the sound source, the percentage of reverb would increase and the decay time would be related to the size of the room and distance from the sound source); and
- set its attenuator **213** to 23.5 dB of attenuation (because in this larger room the virtual spot mic corresponding to this processor is further from the virtual source location, the amplitude of the sound reaching this virtual spot mic would be lower).

The stored information controlling the processor would be developed by the provider of the system described herein.

The following factors are among those which may be used for establishing the information values. Some of them have already been discussed (e.g., length, width and height of the virtual recording studio, the virtual spot mic and room mic distribution in the virtual recording studio, the number and layout of the virtual source zones),

- length, width and height of the virtual recording studio;
- the reverberance of the virtual recording studio;
- the virtual spot mic and room mic distribution in the virtual recording studio;
- the directions in which the spot and room mics are aimed;
- the height of each mic above the floor;
- the sensitivity pattern of each mic;
- the frequency response of each mic;
- the output configuration (e.g., 2.0, 5.1 or 7.1 and the like); and
- the number and layout of the virtual source zones.

FIG. **13** depicts a method for determining preliminary (i.e., initial starting point) values for the time delay and attenuator information that would be associated with a particular virtual source zone and the virtual spot and room mic zones for control of the various spot and room microphone bleed processors within the spot and room mic bleed simulators. The overall virtual studio floor space measures 40 feet wide×70 feet deep, and the first 16 feet of depth is allocated for the room mics, including the surround mics. This leaves 54 feet for three front-to-back spot mic zones (A, B and C), and simple division suggests that there is therefore 18 feet depth for each of these zones.

All distances (between virtual spot mics, which are assumed to be at the center of each virtual source location zone) are expressed in feet. Since the 40-foot width of the studio is divided into five zones (B**1** to B**5** for example), each of these zones is 8 feet wide. Using simple geometric calculations (or measuring actual space if one wishes) the distance the center of source zone location A**1** to each of the remaining 14 virtual spot mic locations is calculated (and shown in the uppermost box of the three values at the lower left corner of each zone). Using the standard temperature and pressure of air at sea level as an assumed condition, sound travels approximately 1.1 feet per millisecond, and so dividing the distance from A**1** to each of the other 14 virtual spot mic locations by 1.1 yields the time in milliseconds for sound to reach each of those locations, as shown in the middle box of each zone. Because the distance from the virtual source to each of the other virtual spot mics is now known, using appropriate formulas the amount of sound attenuation can be approximated. The resulting attenuations (in dB) from the center of zone A**1** to the center of other zones are shown in the bottom of the three values at the lower left corner of each zone.

The nearest of the virtual room mics is assumed to be 18 feet from the virtual sound source (this is calculated by estimating that it is 15 feet along the floor from the center of zone A**1** to the point below virtual room mic RM**1**, but since RM**1** is about 10 feet higher than the presumed height of the virtual sound source, the slant (diagonal) distance is 18 feet. With a two-audio channel (spot and room mic) recording, the room mic is already at a greater distance from the source, though, so one does not need to apply any attenuation to RM**1** relative to A**1**. However, since the system is simulating RM**2**, RM**3**, RM**4** and RM**5** it needs to figure the additional amount of attenuation required for each of these virtual room mic locations based on their distances from A**1**. The same kind of slant range calculations are used to arrive at the upper box (feet) values in RM**2** through RM**5** (the uncorrected floor-only distances are shown adjacent to the boxes with an asterisk), and appropriate approximations are used to calculate the attenuation at each of these locations to be 0.8, 2.7, 4.5 and 6.3 dB respectively for RM**2**, RM**3**, RM**4** and RM**5**. Finally, the delay necessary for each of the locations RM**2** through RM**5** is calculated by measuring using the slant range distance divided by 1.1, and then subtracting the time delay between A**1** and RM**1** since that delay is inherent in the second channel audio signal.

In the embodiment where a first channel audio signal (spot mic) is reverbed to derive a simulated second channel audio signal (room mic), as shown in FIG. **16**, the time delay between A**1** and virtual RM**1** is actually added to the first channel signal as part of the reverberator processing.

While calculations could be used to derive the data, it is expected that empirical testing and human listening tests will more likely be used to establish the stored information values such as shown in FIG. **14** and FIG. **15**.

In the embodiment shown in FIG. **1**, the information regarding the size and reverberance could be entered at input **116**, the information regarding mic characteristics, virtual spot mic zones, and about the number and layout of the virtual source zones could be entered at input **114**, and the information about output configuration could be entered at input **118**. In other embodiments, other input facilities could be made available.

The system could be set so that a user could only input factor values which would correspond to existing stored data. On the other hand, the system could be configured to accept a wider range of input factor values. In such event, the CPU of the system could be configured to calculate intermediate values between stored data sets when the user has input factors which fall between system-provided parameters. In addition, the system can be provided with means for the user to make custom look up tables or to adjust (e.g., by scaling) the outputs of various elements to suit the user's needs.

With respect to subwoofer output mixer **138** shown in FIG. **8**, the CPU would issue an instruction to turn on switch **703** whenever a "0.1" (dot one) output configuration is designated by the user. Low pass filter **705** and attenuator **707** would have default values, which values would be user adjustable. The CPU would issue instructions to turn on switch **709** depending upon the information entered at input **120** by the user. Low pass filter **711** and attenuator **713** would have default values, which values would be user adjustable.

While the bleed simulator of the invention could use separately recorded spot and room mic components, as described above in connection with FIGS. **1**, **11** and **12**, the invention can in fact work with only a recording from a spot mic (i.e., without a corresponding recording from a room mic). Such an embodiment, depicted in FIG. **16**, depending upon circumstances (e.g., such as sample availability) may be a more appropriate choice. A simulated room mic component (or simulated second channel audio signal) can be derived from an audio sample signal originally consisting only of a spot mic signal by applying delay, reverb and possibly equalization (i.e., frequency response contouring) to the spot mic audio sample signal.

The elements comprising FIG. **16** are, with the exception of element **150**, identical to the elements comprising FIG. **1**, and so only the function and purpose of element **150** in FIG. **16** will be described. Whereas in the embodiment of FIG. **1** there were first and second channel audio signals applied to input **112**, representing respectively the spot and room microphone signals, the embodiment of FIG. **16** has only a first channel audio signal applied to input **112**, the spot microphone signal. In order to provide the necessary signal to the downstream components of the bleed simulator (i.e., room mic bleed simulator **132**, output mixer array **134** and surround simulator **142**) a second channel "room mic" audio signal is created by means of a reverberator processor **150**. The reverberator processor applies: (1) a time delay corresponding to the time it would have taken for the signal emanating from the sound source to pass the spot mic and reach the room mic of FIG. **11** and FIG. **12**, (2) a reverberation component corresponding to the additional acoustic reflections that represent the "room sound" that would have been sensed by the room mic **26** of FIG. **11** and FIG. **12**, and (3) an equalization characteristic that accounts for the difference in spectral sensitivity between a typical directional spot mic **24** and a typical omni directional room mic **26** as well as for the natural development of greater bass response at a greater distance from the sound source due to the longer wavelength of low frequency sounds. In such an embodiment additional data may be stored in memory **128** for rapid recall of differing room mic models, different room mic locations relative to the spot mic, and different acoustic environments affecting the sound that would have been sensed by an actual room mic such as used in the embodiment of FIG. **1**.

The various elements of the bleed simulator, such as the attenuators, filters, switches, reverbs and delays may implemented by various analog and/or digital means as is known by those having skill in the art (and, of course, as will become available to those skilled in the art in the future).

Many of the elements shown and discussed herein above have been described as if they were analog devices. However, those skilled in the art will recognize that with appropriate software the components may be implemented in a digital form. The use of analog or digital implementations will depend upon such factors as anticipated availability of computing power and memory, sonic quality of competing analog and digital systems, cost, and physical form factors. It is currently preferred to implement the components with digital means in a manner known to those skilled in the art of implementing digital audio circuits.

The invention described herein also includes the method of operating on audio signals described above to achieve microphone bleed simulation.

It will be understood that various changes of the details, materials, steps, arrangement of parts and uses which have been herein described and illustrated in order to explain the nature of the invention will occur to and may be made by those skilled in the art, and such changes are intended to be included within the scope of this invention.

We claim the following:

1. A bleed simulator system for playing an audio signal to simulate the audio bleed pickup of multiple microphones distributed among a plurality of predetermined virtual spot mic location zones and virtual room mic location zones associated with a virtual sound stage, said system comprising,

at least one bleed simulator output;

a spot mic bleed simulator which receives as an input said audio signal and outputs a plurality of processed audio signals, said spot mic bleed simulator including a plurality of spot mic processors, each of said spot mic processors having a series-connected spot mic low pass filter, spot mic delay and spot mic attenuator;

a reverb which receives as an input said audio signal and outputs a reverbed audio signal;

a room mic bleed simulator which receives as an input said reverbed audio signal and outputs a plurality of processed reverbed audio signals, said room mic bleed simulator including a plurality of room mic processors, each of said room mic processors having a series-connected room mic low pass filter, room mic delay and room mic attenuator; and

a multiplexer connected to said spot mic bleed simulator and to said room mic bleed simulator, which multiplexer receives said plurality of processed audio signals and said plurality of processed reverbed audio signals, and which multiplexer outputs summed subsets of said processed audio signals and said plurality of processed reverbed audio signals to at least one of the outputs of said bleed simulator system.

2. The bleed simulator system of said claim **1** wherein said spot mic bleed simulator includes a spot mic processor for each one of said plurality of virtual spot mic location zones and said room mic bleed simulator includes a room mic processor for each one of said plurality of virtual room mic location zones.

3. The bleed simulator system of claims 2 further comprising a CPU connected to each spot mic processor and each room mic processor, said CPU issuing instructions which control the settings of the sport and room mic processors.

4. The bleed simulator system of claim 3 further comprising a memory in communication with said CPU, which memory stores information corresponding to virtual sound stage definitions and virtual source zone designations, wherein said CPU issues instructions to said spot and room mic processors based on said information associated with a selected virtual sound stage definition and source zone designation.

5. The bleed simulator system of any one of claims 1-4 wherein said virtual sound stage is divided into a predetermined number of virtual source zones and wherein said bleed simulator system includes a number of spot mic bleed simulators equal to said predetermined number of virtual source zones and a number of room mic bleed simulators equal to said predetermined number of virtual source zones.

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