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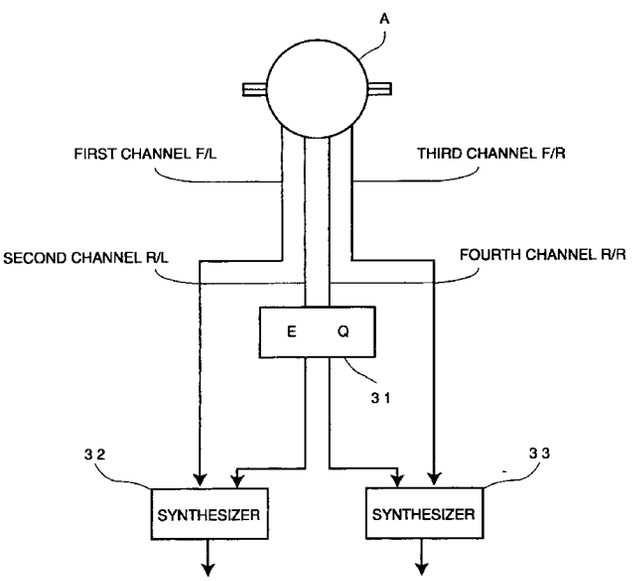
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(54) ACOUSTIC SYSTEM

(57) A microphone unit comprising a semi-spherical structure base (1), a first microphone element (2a) having at least 2 directivities which is attached to a left end of the structure (1) and a second microphone element (2b) having at least 2 directivities which is attached to a right end of the structure (1).

FIG.3



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Description

FIELD OF THE INVENTION

The present invention relates to a microphone unit for collecting sounds which can be reproduced as being substantially real and stereophonic and a sound processing method, a recording medium on which the sound processed through the microphone unit and the sound processing method is recorded and a method for broadcasting those sounds.

BACKGROUND OF THE INVENTION

Recently the sound reproduction with full reality has been increasingly demanded by the recent trend of the virtual reality environment. A stereo mode as a conventional art has been proposed for offering the sound reproduction with reality.

Referring to Fig. 11, in the above conventional stereo mode, the sound from the sound source is picked up by a microphone L and a microphone R. Each sound collected by the microphone L and the microphone R is transmitted through the respective transmission systems L and R separately. The sound is reproduced through a speaker unit L and a speaker unit R corresponding to the respective transmission systems.

As an another conventional art, a quadraphonic mode has been proposed for providing the reproduced sound with full reality.

Fig. 12 is a schematic view representing the quadraphonic mode.

As shown in Fig. 12, in the quadraphonic mode, the sound from the sound source is picked up by 4 channels, i.e., 4 microphones locating at the respective 4 corners of the area around the sound source, a front right microphone F/R, a front left microphone F/L, a rear right microphone R/R and a rear left microphone R/L, respectively.

Each sound recorded by those 4 microphones is separately transmitted through the respective 4 transmission systems to 4 speaker units corresponding to the respective 4 microphones for reproduction. As a result, the sound field provided from those speaker units can exhibit substantially a real impression.

However the conventional stereo mode fails to stabilize the sound field accurately. So the effort to reproduce the real sound field has been limited to a certain extent.

Although the quadraphonic mode is assumed to have 4-channel sound collection and 4-channel sound reproduction, stabilization of the sound field has been limited.

When the sounds picked up by 4 channels in the quadraphonic mode are grouped into 2-channel sounds and output from two speaker units (stereo mode), quality of the reproduced sound is further deteriorated in view of reality.

The conventional stereo mode and quadraphonic mode have failed to provide the sound reproduction with full reality.

Since most of the recent broadcasting system has been employing a stereo mode, the sound put on the air on such a stereo broadcasting offers no impact nor real reproduced sound. The broadcasting art for providing the real sound reproduction has never been developed so far.

It is an object of the present invention to solve the aforementioned problems.

It is another object of the present invention to provide a microphone unit for reproducing sounds with reality by stabilizing the sound field, a method for processing the collected sound and the system used in the method.

It is a still further object of the present invention to provide a recording medium for offering the realistic reproduced sound with reality through stabilized sound field and a broadcasting system.

SUMMARY OF THE INVENTION

The object of the present invention is realized by a microphone unit comprising:

a semi-spherical shape structure base;
a first microphone element having at least two directivities which is attached to a left end of said semi-spherical shape structure base; and
a second microphone element having at least two directivities which is attached to a right end of said semi-spherical shape structure base.

The definition of the term "semi-spherical" is not limited to a sphere shape in mathematical aspect but includes the shape which can be virtually recognized as being spherical as well as the one having an almost ellipsoidal shape. It is preferable that the semi-spherical structure has a diameter ranging from 10 to 50 cm. It is also preferable that the semi-spherical structure has a diameter ranging from 10 to 40 cm, more preferably, ranging from 10 to 30 cm or the most preferably, ranging from 15 to 20 cm.

If the semi-spherical structure base has the same as or slightly larger size than that of the microphone element mounted thereon, each distance between the microphone elements becomes too small to provide the reproduced sound with full reality. While if the semi-spherical structure base has an excessively large size, each distance between the microphone elements becomes so large that each microphone element cannot pick up the same sound from the sound source.

It is further preferable that the semi-spherical structure base is formed of a microporous material. The semi-spherical structure base may be formed as a solid construction. However the one formed of a porous ceramics such as a pumice stone or a foaming resin

material of a microporous structure provided more stereophonic sound reproduction compared with that of solid structure.

Preferably the semi-spherical structure base is a composite structure formed of a semi-spherical skin layer and a sound absorbing agent filled with the inside of the skin layer.

The first microphone element and the second microphone element are preferably of the stereo type.

An arrangement in the semi-spherical structure base for removably accommodating the microphone element allows for easy movement.

The object of the present invention is realized by a method of reproducing a sound using a microphone unit in which a first microphone element of 2-channel type having at least two directivities is attached to a left end of a semi-spherical structure base and a second microphone element of 2-channel type having at least two directivities is attached to a right end of said semi-spherical structure base, said method comprising steps of:

collecting sounds through a first channel as a front channel of said first microphone element, a second channel as a rear channel of said first microphone element, a third channel as a front channel of said second microphone element and a fourth channel as a rear channel of said second microphone element; and

providing an output means corresponding to said respective channels for reproducing collected sounds through 4 channels.

It is preferable that a sound is reproduced in the same place as that used for collecting the sound.

The object of the present invention is realized by a method of reproducing a sound using a microphone unit in which a first microphone element of 2-channel type having at least two directivities is attached to a left end of a semi-spherical structure base and a second microphone element of 2-channel type having at least two directivities is attached to a right end of said semi-spherical structure base, said method comprising steps of:

collecting sounds through a first channel as a front channel of said first microphone element, a second channel as a rear channel of said first microphone element, a third channel as a front channel of said second microphone element and a fourth channel as a rear channel of said second microphone element;

executing a predetermined frequency correction to said second channel signal and said fourth channel signal;

synthesizing said first channel signal and said second channel signal which has been frequency corrected; and

synthesizing said third channel signal and said fourth channel which has been frequency corrected.

Preferably the frequency correction step corrects each frequency of the second channel signal and the fourth channel signal so that winding occurs in a high frequency when plotting a sound pressure to an axis of ordinate and a frequency to an axis of abscissa.

One of the most important characteristics of the auditory sense of a human is the ability to recognize the sound which is heard from front of or behind him/her. Although each human ear is generally on the left and right side of the head, the sound heard from front or behind can also be recognized attributable to the concha of the ear which changes the sound tone. The sound coming from the front is lead to an auris external through a complex structure of the concha. While the sound behind is heard as being in a high range with a unique winding in the presence of the concha functioning as a screen.

Accordingly the frequency correction is executed by rendering the second channel signal and the fourth channel signal to have frequency characteristics exhibiting a big drop of (valley) about 9dB in the range from 500 to 1100 Hz, a sharp peak of about 3dB in the range from 2000 to 3000 Hz and a decline in the range from 4000 Hz to 16000 Hz.

The frequency correction is not limited to the one shown in Fig. 4. The similar correction is available as far as the stereophonic sound reproduction is realized.

The above-described structure preferably has a diameter ranging from 10 to 50 cm.

The object of the present invention is realized by a method of processing a collected sound using a microphone unit in which a first microphone element of 2-channel type having at least two directivities is attached to a left end of a semi-spherical structure base and a second microphone element of 2-channel type having at least two directivities is attached to a right end of said semi-spherical structure base, said method comprising:

a collecting step of collecting sounds through a first channel as a front channel of said first microphone element, a second channel as a rear channel of said first microphone element, a third channel as a front channel of said second microphone element and a fourth channel as a rear channel of said second microphone element;

a executing step of executing a predetermined frequency correction to said second channel signal and said fourth channel signal;

a first synthesizing step of synthesizing said first channel signal with said frequency corrected second channel signal;

a second synthesizing step of synthesizing said third channel signal with said frequency corrected fourth channel signal;

a first difference signal generation step of generating a difference signal by subtracting a signal generated by said second synthesizing step from a signal generated in said first synthesizing step;

a second difference signal generation step of generating a difference signal by subtracting a signal generated in said first synthesizing step from a signal generated in said second synthesizing step;

a first delay step of generating a plurality of difference signals having different delay times by delaying each difference signal generated in said first difference signal generation step for a predetermined time sequentially;

a second delay step of generating a plurality of difference signals having different delay times by delaying each difference signal generated in said second difference signal generation step for a predetermined time sequentially;

a third synthesizing step of synthesizing a signal generated in said first synthesizing step, a difference signal generated in said first difference signal generation step and a plurality of difference signals generated in said first delay step; and

a fourth synthesizing step of synthesizing a signal generated in said second synthesizing step, a difference signal generated in said second difference signal generation step and a plurality of difference signals generated in said second delay step.

This processing method is suitable for reproducing the sound collected through the above microphone unit by means of the reproducing system of stereo type, i.e., two speaker units.

It is preferable that a delay time is set in the range from 0.5 msec. to 1.0 msec. and the delay step produces two difference signals having delay times of 0.5 msec. and 1.0 msec., respectively. It is further preferable that the delay step produces two difference signals having delay times of 0.5 msec. and 1.0 msec., respectively.

Preferably the third synthesizing step synthesizes a difference signal produced in the first difference signal generation step, a plurality of difference signals produced in the first delay step and a signal produced in the first synthesizing step so that a volume level of a total of the difference signal produced in the first difference signal generation step and a plurality of difference signals produced in the first delay step is not larger than that of the signal produced in the first synthesizing step The fourth synthesizing step also synthesizes a difference signal produced in the second difference signal generation step, a plurality of difference signals produced in the second delay step and a signal produced in the second synthesizing step so that a volume level of a total of the difference signal produced in the second difference signal generation step and the plurality of difference signals produced in the second delay step is not larger than that of the signal produced in the second synthesizing step.

It is further preferable that the third synthesizing step synthesizes so that a volume level ratio of a total of the difference signal produced in the first difference signal generation step and a plurality of difference signals produced in the first delay step becomes 0.3 to 0.8 to the signal produced in the first synthesizing step as a value 1. The fourth synthesizing step further synthesizes so that a volume level ratio of a total of the difference signal produced in the second difference signal generation step and a plurality of difference signals produced in the second delay step becomes 0.3 to 0.8 to the signal produced in the second synthesizing step as a value 1.

Particularly it is the most preferable that the volume level ratio of the signal generated in the synthesizing step to the total of the difference signal and a plurality of delayed difference signals is 1 : 0.6.

The above-processed sound can be recorded on the recording medium, which is easily reproduced as full stereophonic sound through the reproducing system of the stereo type.

Being broadcasted in the stereo mode, the above-processed sounds can also exhibit full stereophonic and real impression.

The object of the present invention is realized by a channel conversion system for converting sounds collected through 4 channels, a first channel for collecting front left sound, a second channel for collecting rear left sound, a third channel for collecting front right sound and a fourth channel for collecting rear right sound, into 2-channel sounds, said channel conversion system comprising:

frequency correction means for providing a predetermined frequency correction to said second channel signal and said fourth channel signal;

first synthesizing means for synthesizing said first channel signal and said frequency corrected second channel signal; and

second synthesizing means for synthesizing said third channel signal and said frequency corrected fourth channel signal.

Preferably the frequency correction means executes frequency correction so that winding occurs in a high frequency when plotting a sound pressure to an axis of ordinate and a frequency to an axis of abscissa. It is further preferable that the frequency correction means is constructed so that each of the second channel signal and the fourth channel signal has a frequency characteristic exhibiting a big drop of about 9dB in the range from 500 to 1100 Hz, a sharp peak of about 3dB in the range from 2000 to 3000 Hz and a decline in the range from 4000 Hz to 16000 Hz.

The object of the present invention is realized by a sound signal processing system for processing sound signals comprising:

first difference signal generation means for generating a difference signal by subtracting a second channel signal from a first channel signal;

second difference signal generation means for generating a difference signal by subtracting said first channel signal from said second channel signal;

first delay means for generating a plurality of difference signals having different delay times by delaying each of a difference signal generated in said first difference signal generation means for a predetermined time sequentially;

second delay means for generating a plurality of difference signals having different delay times by delaying each of difference signal generated in said second difference signal generation means for a predetermined time sequentially;

first synthesizing means for synthesizing said first channel signal, a difference signal generated in said first difference signal generation means and a plurality of difference signals generated in said first delay means; and

second synthesizing means for synthesizing said second channel signal, a difference signal generated in said second difference signal generation means and a plurality of difference signals generated in said second delay means.

It is preferable that delay times of the first delay means and the second delay means are ranged from 0.5 msec. to 1.0 msec.

Preferably the first synthesizing means synthesizes a difference signal produced in the first difference signal generation means, a plurality of difference signals produced in the first delay means and the first channel signal so that a volume level of a total of the difference signal produced in the first difference signal generation means and a plurality of difference signals produced in the first delay means is not larger than that of the first channel signal The second synthesizing means further synthesizes a difference signal produced in the second difference signal generation means, a plurality of difference signals produced in the second delay means and the second channel signal so that a volume level of a total of the difference signal produced in the second difference signal production means and the plurality of difference signals produced in the second delay means is not larger than that of said second channel signal.

More preferably the first synthesizing means synthesizes so that a volume level ratio of a total of the difference signal produced in the first difference signal generation means and a plurality of difference signals produced in the first delay means becomes 0.3 to 0.8 to the first channel signal as a value 1. The second synthesizing means further synthesizes so that a volume level ratio of a total of the difference signal produced in the second difference signal generation means and a plurality of difference signals produced in the second delay means becomes 0.3 to 0.8 to the second channel signal as a value 1.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a perspective view of a microphone unit A of the present invention;

Fig. 2 is a plan view of the microphone unit A of the present invention;

Fig. 3 is a block diagram of a channel conversion system;

Fig. 4 is a graphical representation of frequency characteristics;

Fig. 5 is a block diagram of a sound processing system 100;

Fig. 6 is a schematic view showing function of the sound processing system 100;

Fig. 7 is a schematic view showing function of the sound processing system 100;

Fig. 8 is a schematic view showing 4-channel sound collection and 4-channel sound reproduction by means of the microphone unit;

Fig. 9 is a schematic view showing 4-channel sound collection by means of the microphone unit and sound reproduction by means of a headphone after converting 4-channel collected sounds into 2-channel sounds;

Fig. 10 is a schematic view showing sound collection by means of the microphone unit and sound processing through the channel conversion system and the sound processing system;

Fig. 11 is a schematic view showing a prior art; and

Fig. 12 is a schematic view showing another prior art.

BEST MODE FOR CARRYING OUT THE INVENTION

First a microphone unit of the present invention is described. Fig. 1 is a perspective view showing a microphone unit A of the present invention. Fig. 2 is a plan view of the microphone unit A of the present invention.

In the above drawings, a reference numeral 1 is a sphere structure having a diameter of 18 cm. The inside of the sphere structure 1 is filled with a urethane foaming resin. An FRP is used for covering a whole surface of the urethane foaming resin. That is the sphere structure 1 is formed of the urethane foaming resin and the FRP as a skin layer forming the spherical shape.

Reference numerals 2a and 2b are stereo type microphone elements. As shown in the drawings, each base section of the microphone elements 2a and 2b is removably embedded in the right and left end of the sphere structure 1. As Fig. 2 shows, the microphone element 2a is designed to pick up sounds coming from front left and rear left. The microphone element 2b is designed to pick up sounds coming from front right and rear right.

Next described is a channel conversion system for converting collected sounds through 4 channels of the

microphone unit A into those through 2 channels.

Fig. 3 is a block diagram of the channel conversion system. Among two channels (stereo) of the microphone element 2a, the one for picking up the front left sound is designated as a first channel F/L and the other for picking up the rear left sound is designated as a second channel R/L. Among two channels (stereo) of the microphone element 2b, the one for picking up the front right sound is designated as a third channel F/R and the other for picking up the rear right sound is designated as a fourth channel R/R.

A reference numeral 31 is a equalizer. The equalizer 31 receives signals of the second channel R/L and the fourth channel R/R, which will be frequency corrected by rendering characteristics shown in Fig. 4 therewith. It is intended to correct the frequency of those signals so as to have a big drop (valley) of approximately 9 dB in the range from 500 to 1100 Hz, an acute peak of approximately 3 dB in the range from 2000 to 3000 Hz, and damping in the range from 4000 Hz to 16000Hz.

A reference numeral 32 is a first synthesizer for synthesizing a sound of the first channel F/L and a sound of the second channel R/L which has been corrected into 1 channel.

A reference numeral 33 is a second synthesizer for synthesizing a sound of the third channel F/R and a sound of the fourth channel R/R which has been corrected into 1 channel.

As aforementioned, sounds collected through 4 channels are converted into 2-channels sounds.

The above channel conversion system is effective particularly for the case of listening the sound through a head phone.

Then described is a sound processing system for further stereophonic sound reproduction by processing the 2-channel mode signals.

Fig. 5 is a block diagram of a sound processing system 100.

A reference numeral 51 is a first difference signal generation circuit where a second channel signal is subtracted from a first channel signal in order to obtain a difference signal, i.e., a matrix signal.

A reference numeral 52 is a second difference signal generation circuit where the first channel signal is subtracted from the second channel signal in order to obtain a difference signal, i.e., a matrix signal.

Reference numerals 53_1 to 53_n are first delay circuits. Different delay times ranging from 0.5 msec. to 1.0 msec. are set in the respective delay circuits 53_1 to 53_n for delaying an output signal of the difference signal generation circuit 51 for a preset delay time. Supposing that 0.5 msec. is set in the delay circuit 53_1 as the delay time, the output signal of the difference signal generation circuit 51 is output from the delay circuit 53_1 delayed for 0.5 msec.

Reference numerals 54_1 to 54_n are second delay circuits. Different delay times ranging from 0.5 msec. to 1.0 msec. are set in the respective delay circuits 54_1 to 54_n for delaying an output signal of the difference signal generation circuit 52 for a preset delay time. Supposing that 0.5 msec. is set in the delay circuit 54_1 as the delay time, the output signal of the difference signal generation circuit 52 is output from the delay circuit 54_1 delayed for 0.5 msec.

Reference numeral 55 is a first synthesizer for synthesizing a first channel signal, an output signal of the difference signal generation circuit 51 and output signals of the delay circuits 53_1 to 53_n. Supposing that a volume level of the first channel signal is 1, the synthesizer 55 is designed to synthesize those signals so that the total volume level of the output signal of the difference signal generation circuit 51 and those of the delay circuits 53_1 to 53_n becomes 0.3 to 0.8.

A reference numeral 56 is a second synthesizer for synthesizing a second channel signal, an output signal of the difference signal generation circuit 52 and those of the delay circuits 54_1 to 54_n. Supposing that a volume level of the second channel signal is 1, the synthesizer 56 is designed to synthesize those signals so that the total volume level of the output signal of the difference signal generation circuit 52 and those of the delay circuits 54_1 to 54_n becomes 0.3 to 0.8.

A function of the above-constructed system is described. It is assumed that delay circuits 53_1, 53_2, 54_1 and 54_2 are used and 0.5 msec. is set in the delay circuits 53_1 and 54_1 and 1.0 msec. is set in the delay circuits 53_2 and 54_2 as delay times, respectively. Supposing that the volume level ratio of the channel signal is 1, the synthesizers 55 and 56 are designed to synthesize the output signals of the difference signals generation circuit and the delay circuit so that the total volume level ratio becomes 0.6.

Fig. 6 and Fig. 7 are schematic views showing how the system functions. Reference numeral 60 is a person who listens to the reproduced sounds, reference numeral 61 is a speaker unit left to the listener 60 and reference numeral 62 is a speaker unit right to the listener 60. The speaker unit 61 outputs signals output from the synthesizer 55. The speaker unit 62 outputs signals output from the synthesizer 56.

The sound field may become full stereophonic if the reproduced sound output from the speaker unit 61 is caught by a left ear of the listener 60 and the reproduced sound output from the speaker unit 62 is caught by his/her right ear.

However the reproduced sound output from the speaker unit 61 is also picked up by the right ear of the listener 60, which is slightly over hie/her head as the unnecessary sound. Similarly the reproduced sound output from the speaker unit 62 is also picked up by the left ear of the listener 60, which is slightly delayed over his/her head as the unnecessary sound.

The system of the present invention superimposes the sound for suppressing the unnecessary sound with the intrinsically necessary sound, which is output from

the right and left speaker units. The explanation of the system is hereinafter described.

The speaker unit 61 outputs an intrinsically necessary sound 61a (the first channel signal) and a sound for suppressing the unnecessary sound 61b (difference signal). A reference numeral 61c is the unnecessary sound input to the right ear of the listener 60.

The speaker unit 62 outputs an intrinsically necessary sound 62a (the second channel signal) and a sound for suppressing the unnecessary sound 62b (difference signal). A reference numeral 62c is the unnecessary sound input to the left ear of the listener 60.

It is assumed that the listener 60 sits before the speaker units 61 and 62 at a center of the distance therebetween as shown in Fig. 6.

The left ear of the listener 60 catches the sound 62c as well as the sound 61a. The sound 62c is intended to be suppressed by the sound 61b (delayed time: 0.5 msec).

Similarly the right ear of the listener 60 catches the sound 61c as well as the sound 62a. The sound 61c is intended to be suppressed by the sound 62b (delayed time: 0.5 msec).

It is assumed that the listener 60 sits relatively closer to the speaker unit 61 as shown in Fig. 7.

The left ear of the listener 60 catches the sound 62c as well as the sound 61a. The sound 62c is intended to be suppressed by the sound 61b (delayed time: 1.0 msec). The right ear of the listener 60 similarly catches the sound 61c as well as the sound 62a. The sound 61c is suppressed by the sound 62b (difference signal; delay time: 0) which has not been delayed.

While in case the listener 60 sits relatively closer to the speaker unit 62, the similar results as described above will be obtained.

The above-described microphone unit was used for sound collection. The collected sound was processed through the channel conversion system and the sound processing system. The result is described hereinafter.

The microphone unit is used for 4-channel sound collection. The sound is then reproduced through 4 channels.

Fig. 8 is a schematic view showing 4-channel sound collection with the microphone unit and 4-channel sound reproduction.

Right in the center of a room 81, the microphone unit A was placed, a drum set, piano, guitar and a base guitar were placed at a front left, front right, rear left and rear right, respectively for sound collection.

In the center of a room 82 similar to the room 81, the speaker units 83, 84, 85 and 86 were placed at a front left, front right, rear left and rear right, respectively. A listener 87 was expected to listen to the sound in the center of the room 82.

The listener 87 heard the drum sound from the front left, piano sound from the front right, guitar sound from the rear left and the base guitar sound from the rear right. The sounds derived from reproducing those collected sounds through 4 channels were found to be substantially stereophonic owing to stable sound field.

Described hereinafter is with respect to the sound reproduction with a headphone by converting 4-channel sound collected with the microphone unit into the 2-channel sound.

Fig. 9 is a schematic view showing a mechanism in which the sound is collected with the microphone unit A via 4 channels and the collected sound is converted into 2-channel sound with the channel conversion system for sound reproduction with the headphone.

Among two channels (stereo) of the microphone element 2a, one channel for picking up the front left sound was designated as a first channel F/L. The other channel for picking up the rear left sound was designated as a second channel R/L. Among two channels (stereo) of the microphone element 2b, one channel for picking up the front right sound was designated as a third channel F/R. The other channel for picking up the rear right sound was designated as a fourth channel R/R. Referring to Fig. 9, a drum set, piano, guitar and base guitar were arranged so that the microphone unit A was located central to them. Then sound collection was started.

Among sounds collected with the microphone unit A, the sounds received by the second channel R/L and the fourth channel R/R were input to an equalizer 31.

The equalizer 31 corrected the frequency of each input sound, R/L sound and R/R sound, by rendering those sounds to have the characteristic as shown in Fig. 4.

The second channel R/L sound was synthesized with the first channel F/L sound in a synthesizer 32. The fourth channel R/R sound was synthesized with the third channel F/R sound in a synthesizer 33.

The 2-channel sound converted from the 4-channel sound was heard through the headphone 91.

The reproduced sound had far more excellent quality in stereophonic aspect than those reproduced through a conventional stereo mode. The listener 92 listened to the sound of each instrument separately, i.e., the drum sound from the front left, the piano sound from the front right, the guitar sound from the rear left and the base guitar sound from the rear right, respectively.

Unlike with the speaker units, unnecessary sounds were rarely heard with the headphone. So the sound field can be reproduced with substantially an excellent stereophonic feel through the frequency correction.

Described is a process for collecting the sound by the microphone unit and processing the collected sound through a channel conversion system and a sound processing system.

Fig. 10 is a schematic view showing a mechanism where the sound is collected by the microphone unit and the collected sound is processed through the channel conversion system and the sound processing system.

The microphone unit A collected sounds through 4

channels. Then the collected sound was converted into 2-channel sound through the channel conversion system, which was further processed through the sound processing system 100. In this case, delay circuits 53_1, 53_2, 54_1 and 54_2 were used. In each delay circuit 53_1 and 54_1, 0.5 msec. was set as the delay time. In each delay circuit 53_2 and 54_2, 1.0 msec. was set as the delay time. Supposing that the volume level ratio of the channel signal was 1, the synthesizers 55 and 56 were designed to synthesize the output signals of the difference signal generation circuit and the delay circuit so that the total volume level ratio became 0.6.

Finally the sound was reproduced with the speaker units 101 and 102. The reproduced sound was found to have the sound field with the similar stereophonic sound to that heard through the headphone. A listener 103 heard the sound of each instrument distinctly, that was the drum sound from the front left, piano sound from the front right, guitar sound from the rear left and base guitar sound from the rear right.

As aforementioned, the microphone unit using 4 channels gave the most excellent reproduced sound. In case the headphone was supposed to be used, frequency correction through converting the sound into 2-channel also allowed for an excellent reproduced sound field which never has been offered by the conventional stereo mode. Additionally the above frequency correction did not require a great deal of labor.

Even if the conventional stereo mode was used for the reproduction using two speaker units, the listener could hear stereophonic sounds both front and behind.

The thus processed sound was recorded on the tape and then reproduced. The resultant sound field presented excellent quality in view of reality equivalent to the aforementioned effect.

In this example, the sound is recorded by a 2-channel (track) tape recorder after being collected and processed. However, it is also allowable that the collected sound is recorded by a 4-channel (track) tape recorder, processed and then recorded by the 2-channel (track) tape recorder.

The recording medium is not limited to a magnetic tape. A record, CD or the like are available.

When the thus processed sound was put on the air of FM radio broadcast, the resultant sound on the radio (stereo) gives an excellent stereophonic quality and reality.

Employing a stereo mode (2-channel) enables the AM radio broadcasting to offer a reproduction of the real sound field as excellent as the FM radio broadcast.

INDUSTRIAL APPLICABILITY

The present invention allows both 2-channel reproduction and 4-channel reproduction to provide excellent stereophonic and real sounds, which is an advantageous art for an acoustic field.

Claims

1. A microphone unit comprising:

   a semi-spherical shape structure base;
   a first microphone element having at least two directivities which is attached to a left end of said semi-spherical shape structure base; and
   a second microphone element having at least two directivities which is attached to a right end of said semi-spherical shape structure base.

2. The microphone unit of claim 1, wherein said semi-spherical structure base has a diameter ranging from 10 to 50 cm.

3. The microphone unit of claim 1, wherein said semi-spherical structure base is formed of a microporous material.

4. The microphone unit of claim 1, wherein said semi-spherical structure base is a composite structure formed of a semi-spherical skin layer and a sound absorbing agent filled with the inside of said skin layer.

5. The microphone unit of claim 1, wherein said first microphone element and said second microphone element are microphone elements of a stereo type.

6. A microphone unit comprising:

   a semi-spherical shape structure base; and
   each section of a left end and a right end of said spherical structure for connecting a first microphone element having at least two directivities and a second microphone element having at least two directivities, respectively.

7. The microphone unit of claim 6, wherein said semi-spherical structure base has a diameter ranging from 10 to 50 cm.

8. The microphone unit of claim 6, wherein said semi-spherical structure base is formed of a microporous material.

9. The microphone unit of claim 6, wherein said semi-spherical structure base is a composite structure formed of a semi-spherical skin layer and a sound absorbing agent filled with the inside of said skin layer.

10. A method of reproducing a sound using a microphone unit in which a first microphone element of 2-channel type having at least two directivities is attached to a left end of a semi-spherical structure base and a second microphone element of 2-chan-

nel type having at least two directivities is attached to a right end of said semi-spherical structure base, said method comprising steps of:

collecting sounds through a first channel as a front channel of said first microphone element, a second channel as a rear channel of said first microphone element, a third channel as a front channel of said second microphone element and a fourth channel as a rear channel of said second microphone element; and
providing an output means corresponding to said respective channels for reproducing collected sounds through 4 channels.

11. The method of reproducing a sound of claim 10, wherein said reproducing step reproduces a sound in the same place as that used for collecting said sound.

12. A method of reproducing a sound using a microphone unit in which a first microphone element of 2-channel type having at least two directivities is attached to a left end of a semi-spherical structure base and a second microphone element of 2-channel type having at least two directivities is attached to a right end of said semi-spherical structure base, said method comprising steps of:

collecting sounds through a first channel as a front channel of said first microphone element, a second channel as a rear channel of said first microphone element, a third channel as a front channel of said second microphone element and a fourth channel as a rear channel of said second microphone element;
executing a predetermined frequency correction to said second channel signal and said fourth channel signal;
synthesizing said first channel signal and said second channel signal which has been frequency corrected; and
synthesizing said third channel signal and said fourth channel which has been frequency corrected.

13. The method of processing a collected sound of claim 12, wherein said semi-spherical structure base has a diameter ranging from 10 to 50 cm.

14. The method of processing a collected sound of claim 12, wherein said frequency correction step comprises a step of correcting each frequency of said second channel signal and said fourth channel signal so that winding occurs in a high frequency when plotting a sound pressure to an axis of ordinate and a frequency to an axis of abscissa.

15. The method of processing a collected sound of claim 14, wherein said frequency correction step comprises a step of executing frequency correction by rendering said second channel signal and said fourth channel signal to have a frequency characteristic exhibiting a big drop of about 9dB in the range from 500 to 1100 Hz, a sharp peak of about 3dB in the range from 2000 to 3000 Hz and a damping in the range from 4000 Hz to 16000 Hz.

16. A method of processing a collected sound using a microphone unit in which a first microphone element of 2-channel type having at least two directivities is attached to a left end of a semi-spherical structure base and a second microphone element of 2-channel type having at least two directivities is attached to a right end of said semi-spherical structure base, said method comprising:

a collecting step of collecting sounds through a first channel as a front channel of said first microphone element, a second channel as a rear channel of said first microphone element, a third channel as a front channel of said second microphone element and a fourth channel as a rear channel of said second microphone element;
a executing step of executing a predetermined frequency correction to said second channel signal and said fourth channel signal;
a first synthesizing step of synthesizing said first channel signal with said frequency corrected second channel signal;
a second synthesizing step of synthesizing said third channel signal with said frequency corrected fourth channel signal;
a first difference signal generation step of generating a difference signal by subtracting a signal generated by said second synthesizing step from a signal generated in said first synthesizing step;
a second difference signal generation step of generating a difference signal by subtracting a signal generated in said first synthesizing step from a signal generated in said second synthesizing step;
a first delay step of generating a plurality of difference signals having different delay times by delaying each difference signal generated in said first difference signal generation step for a predetermined time sequentially;
a second delay step of generating a plurality of difference signals having different delay times by delaying each difference signal generated in said second difference signal generation step for a predetermined time sequentially;
a third synthesizing step of synthesizing a signal generated in said first synthesizing step, a

difference signal generated in said first difference signal generation step and a plurality of difference signals generated in said first delay step; and

a fourth synthesizing step of synthesizing a signal generated in said second synthesizing step, a difference signal generated in said second difference signal generation step and a plurality of difference signals generated in said second delay step.

17. The method of processing a collected sound of claim 16, wherein said frequency correction step comprises a step of correcting each frequency of said second channel signal and said fourth channel signal so that winding occurs in a high frequency when plotting a sound pressure to an axis of ordinate and a frequency to an axis of abscissa.

18. The method of processing a collected sound of claim 17, wherein said frequency correction step comprises a step of executing frequency correction by rendering said second channel signal and said fourth channel signal to have a frequency characteristic exhibiting a big drop of about 9dB in the range from 500 to 1100 Hz, a sharp peak of about 3dB in the range from 2000 to 3000 Hz and a damping in the range from 4000 Hz to 16000 Hz.

19. The method of processing a collected sound of claim 16, wherein a semi-spherical structure base has a diameter ranging from 10 cm to 50 cm.

20. The method of processing a collected sound of claim 16, wherein a delay time is set in the range from 0.5 msec. to 1.0 msec.

21. The method of processing a collected sound of claim 20, wherein said first delay step comprises a step of producing two difference signals having delay times of 0.5 msec. and 1.0 msec., respectively and said second delay step is a step of producing two difference signals having delay times of 0.5 msec. and 1.0 msec., respectively.

22. The method of processing a collected sound of claim 16, wherein said third synthesizing step comprises a step of synthesizing a difference signal generated in said first difference signal generation step, a plurality of difference signals generated in said first delay step and a signal generated in said first synthesizing step so that a volume level of a total of said difference signal generated in said first difference signal generation step and a plurality of difference signals generated in said first delay step is not larger than a volume level of said signal produced in said first synthesizing step; and said fourth synthesizing step comprises a step of synthesizing a difference signal generated in said second difference signal generation step, a plurality of difference signals generated in said second delay step and a signal produced in said second synthesizing step so that a volume level of a total of said difference signal generated in said second difference signal generation step and said plurality of difference signals generated in said second delay step is not larger than a volume level of said signal produced in said second synthesizing step.

23. The method of processing a collected sound of claim 22, wherein said third synthesizing step comprises a step of synthesizing a difference signal generated in said first difference signal generation step, a plurality of difference signals generated in said first delay step and a signal generated in said first synthesizing step so that a volume level ratio of a total of said difference signal generated in said first difference signal generation step and a plurality of difference signals generated in said first delay step becomes 0.3 to .08 to said signal generated in said first synthesizing step as a value ratio 1; and said fourth synthesizing step comprises a step of synthesizing a difference signal generated in said second difference signal generation step, a plurality of difference signals generated in said second delay step and a signal generated in said second synthesizing step so that a volume level ratio of a total of said difference signal generated in said second difference signal generation step and said plurality of difference signals generated in said second delay step becomes 0.3 to 0.8 to said signal generated in said second synthesizing step as a value ratio 1.

24. A recording medium in which sounds processed through said microphone unit or said sound processing method as in any of claims 1 to 23 are recorded.

25. A broadcasting method of broadcasting sounds processed through said microphone unit or said sound processing method as in any of claims 1 to 23.

26. A channel conversion system for converting sounds collected through 4 channels, a first channel for collecting front left sound, a second channel for collecting rear left sound, a third channel for collecting front right sound and a fourth channel for collecting rear right sound, into 2-channel sounds, said channel conversion system comprising:

frequency correction means for providing a predetermined frequency correction to said second channel signal and said fourth channel signal;

first synthesizing means for synthesizing said first channel signal and said frequency corrected second channel signal; and

second synthesizing means for synthesizing said third channel signal and said frequency corrected fourth channel signal.

27. The channel conversion system of claim 26, wherein said frequency correction means comprises means for equalizing said second channel signal and said fourth channel signal so that winding occurs in a high frequency when plotting a sound pressure to an axis of ordinate and a frequency to an axis of abscissa.

28. The channel conversion system of claim 27, wherein said frequency correction means comprises means for equalizing said second channel signal and said fourth channel signal so that each of said second channel signal and said fourth channel signal has a frequency characteristic exhibiting a big drop of about 9dB in the range from 500 to 1100 Hz, a sharp peak of about 3dB in the range from 2000 to 3000 Hz and a decline in the range from 4000 Hz to 16000 Hz.

29. A sound signal processing system for processing sound signals comprising:

first difference signal generation means for generating a difference signal by subtracting a second channel signal from a first channel signal;

second difference signal generation means for generating a difference signal by subtracting said first channel signal from said second channel signal;

first delay means for generating a plurality of difference signals having different delay times by delaying each of a difference signal generated in said first difference signal generation means for a predetermined time sequentially;

second delay means for generating a plurality of difference signals having different delay times by delaying each of difference signal generated in said second difference signal generation means for a predetermined time sequentially;

first synthesizing means for synthesizing said first channel signal, a difference signal generated in said first difference signal generation means and a plurality of difference signals generated in said first delay means; and

second synthesizing means for synthesizing said second channel signal, a difference signal generated in said second difference signal generation means and a plurality of difference signals generated in said second delay means.

30. The sound signal processing system of claim 29, wherein delay times of said first delay means and said second delay means are in a range from 0.5 msec. to 1.0 msec.

31. The sound signal processing system of claim 29, wherein said first synthesizing means comprises means for synthesizing a difference signal generated in said first difference signal generation means, a plurality of difference signals generated in said first delay means and said first channel signal so that a volume level of a total of said difference signal generated in said first difference signal generation means and a plurality of difference signals generated in said first delay means is not larger than that of said first channel signal; and said second synthesizing means comprises means for synthesizing a difference signal generated in said second difference signal generation means, a plurality of difference signals generated in said second delay means and said second channel signal so that a volume level of a total of said difference signal generated in said second difference signal production means and said plurality of difference signals generated in said second delay means is not larger than that of said second channel signal.

32. The sound signal processing system of claim 31, wherein said first synthesizing means comprises means for synthesizing a difference signal generated in said first difference signal generation means, a plurality of difference signals generated in said first delay means and said first channel signal so that a volume level ratio of a total of said difference signal generated in said first difference signal generation means and a plurality of difference signals generated in said first delay means becomes 0.3 to 0.8 to said first channel signal as a value ratio 1; and said second synthesizing means comprises means for synthesizing a difference signal generated in said second difference signal generation means, a plurality of difference signals generated in said second delay means and said second channel signal so that a volume level ratio of a total of said difference signal generated in said second difference signal generation means and said a plurality of difference signals generated in said second delay means becomes 0.3 to 0.8 to said second channel signal as a value ratio 1.

FIG.1

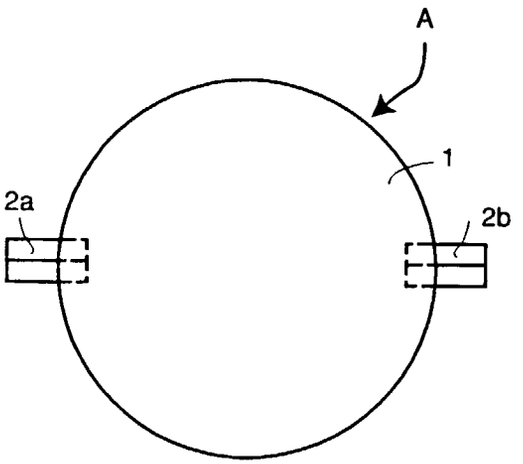


FIG.2

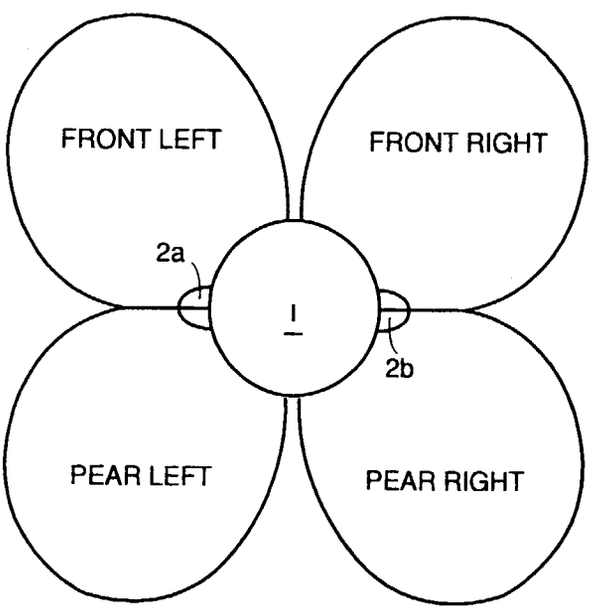


FIG.3

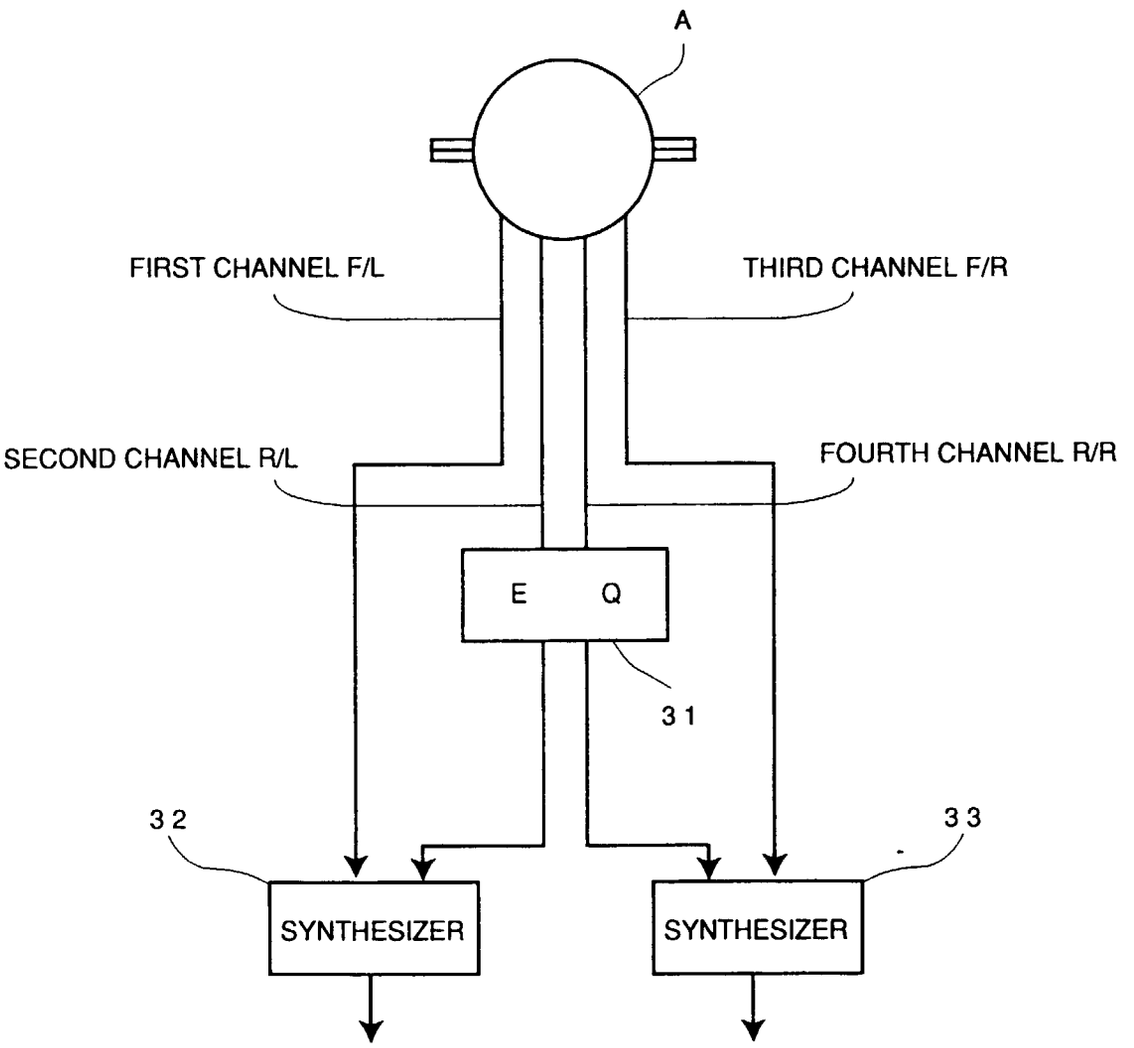

A
FIRST CHANNEL F/L
THIRD CHANNEL F/R
SECOND CHANNEL R/L
FOURTH CHANNEL R/R
E
Q
31
32
33
SYNTHESIZER
SYNTHESIZER

FIG.4

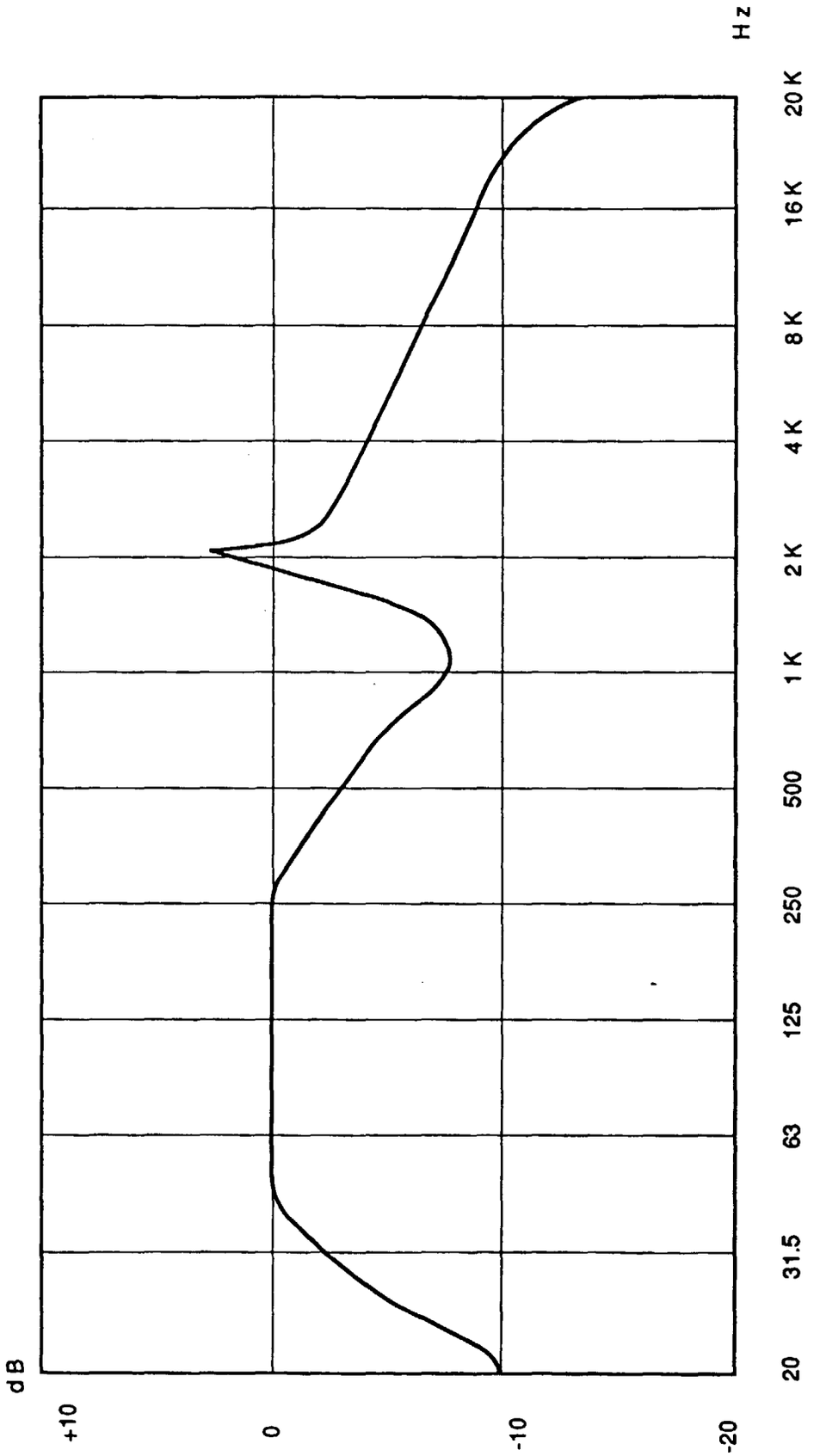


FIG.5

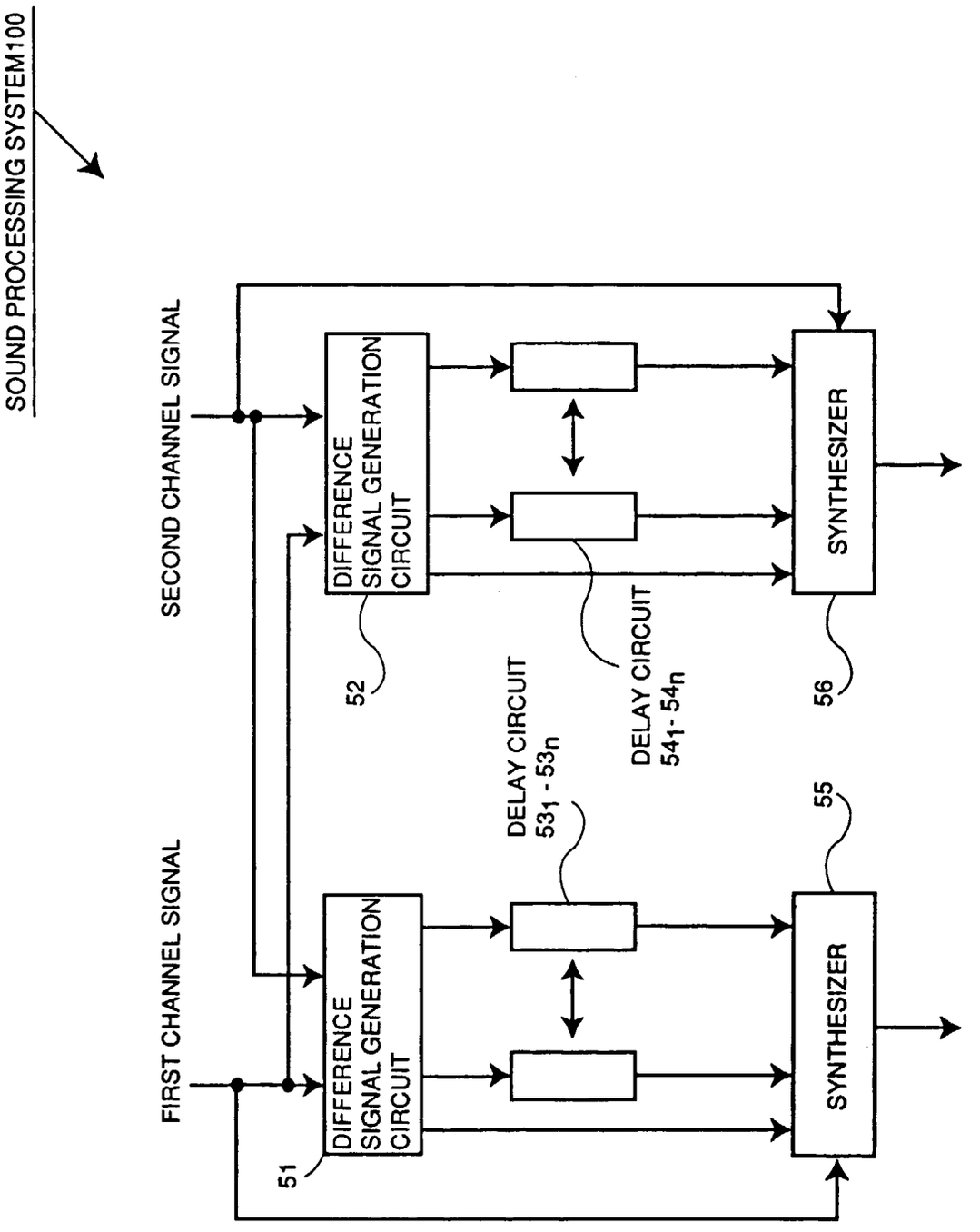

SOUND PROCESSING SYSTEM100
FIRST CHANNEL SIGNAL
SECOND CHANNEL SIGNAL
DIFFERENCE SIGNAL GENERATION CIRCUIT
DIFFERENCE SIGNAL GENERATION CIRCUIT
DELAY CIRCUIT 53₁ - 53n
DELAY CIRCUIT 54₁ - 54n
SYNTHESIZER
SYNTHESIZER
51
52
55
56

FIG.6

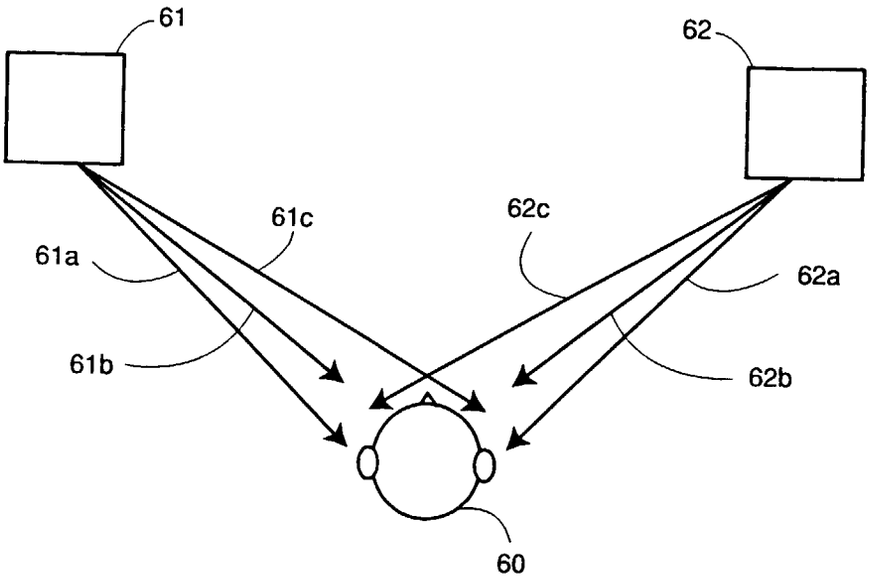


FIG.7

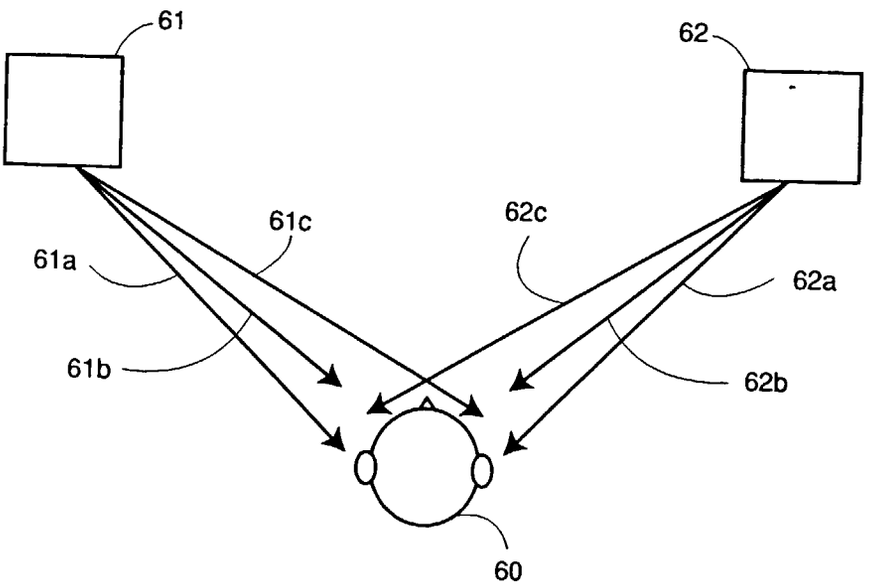


FIG.8

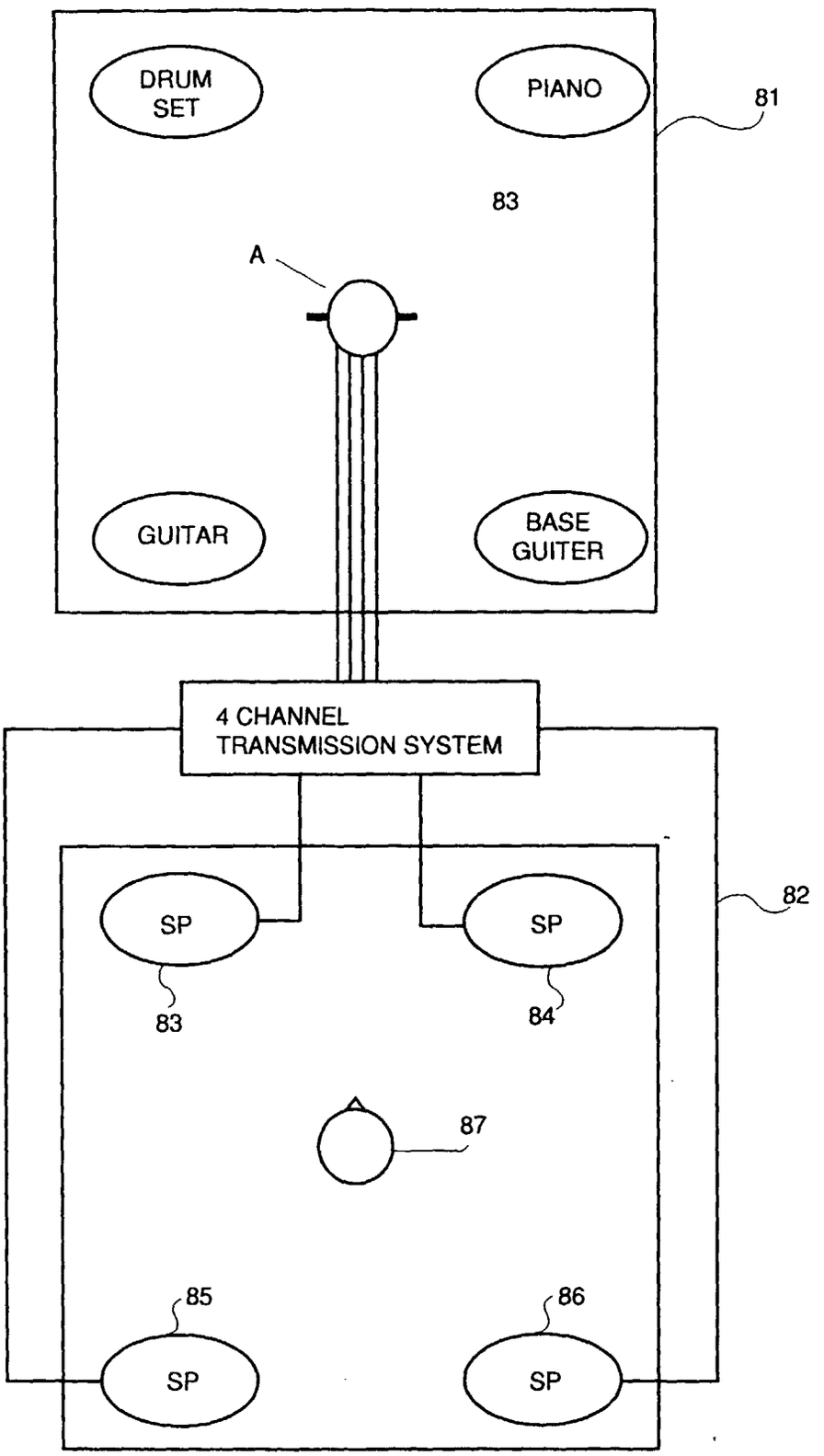

DRUM SET
PIANO
81
83
A
GUITAR
BASE GUITER
4 CHANNEL TRANSMISSION SYSTEM
82
SP
SP
83
84
87
85
86
SP
SP

FIG.9

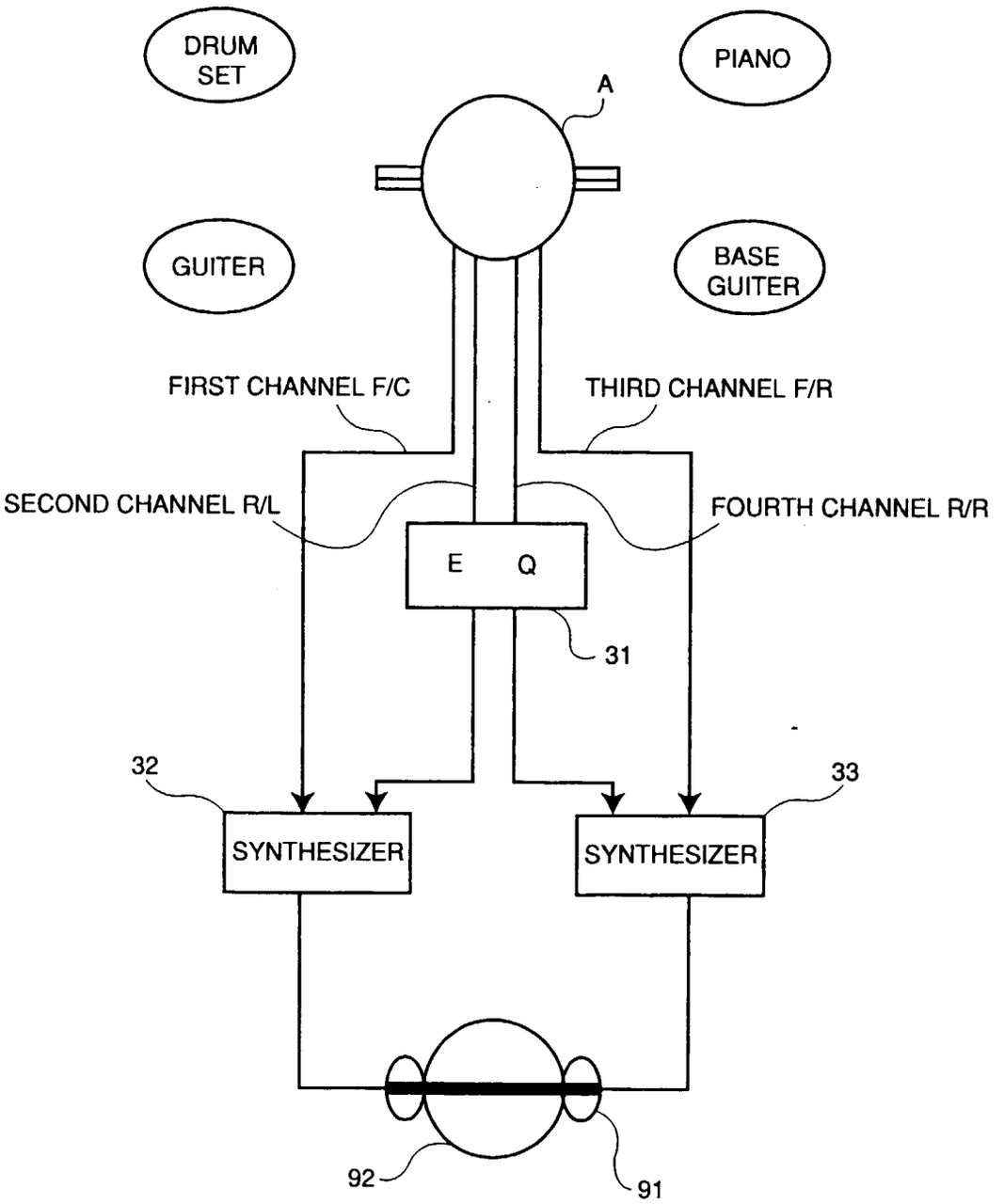

DRUM SET
PIANO
A
GUITER
BASE GUITER
FIRST CHANNEL F/C
THIRD CHANNEL F/R
SECOND CHANNEL R/L
FOURTH CHANNEL R/R
E Q
31
32
33
SYNTHESIZER
SYNTHESIZER
92
91

FIG.10

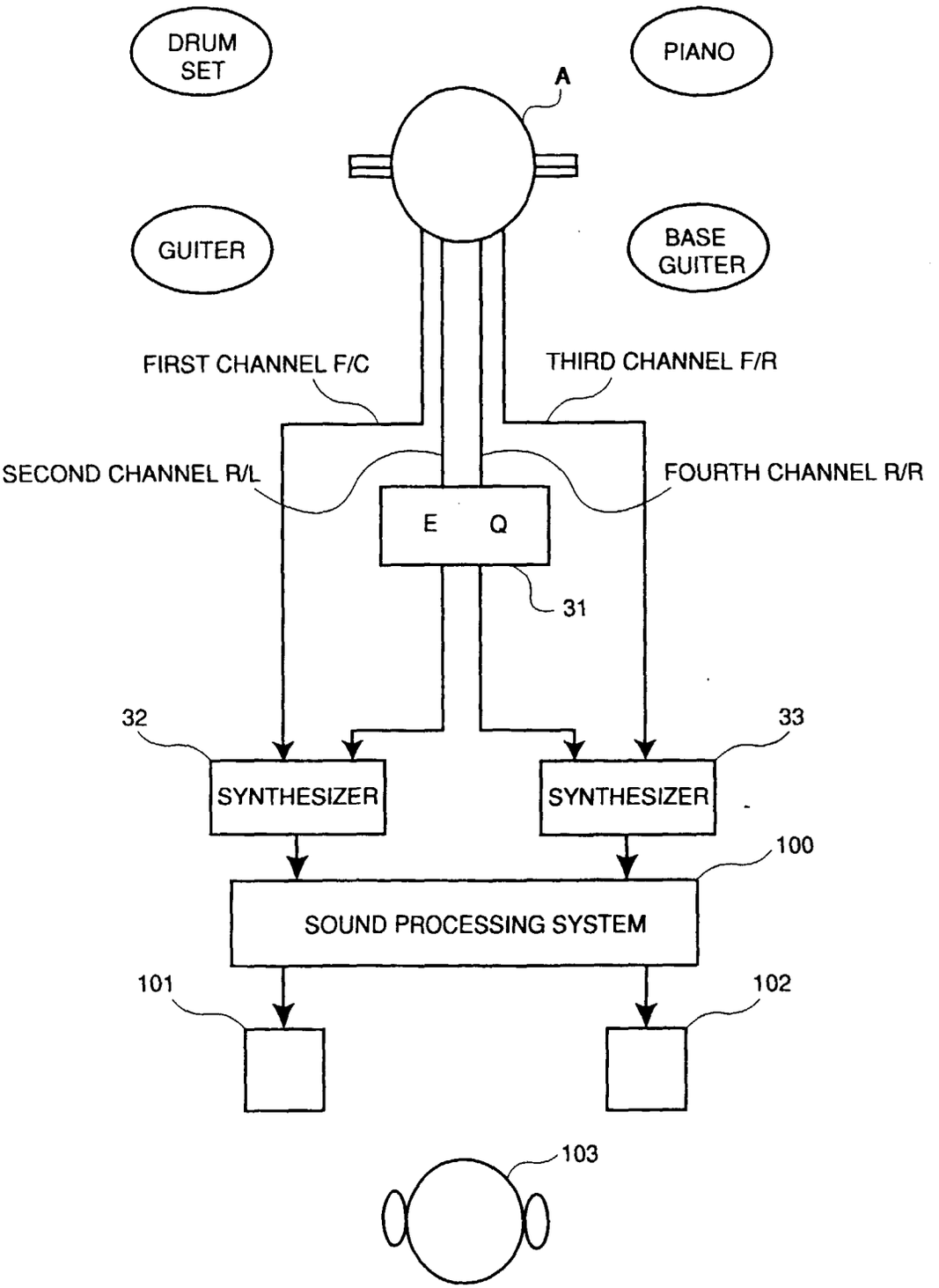

DRUM SET
PIANO
A
GUITER
BASE GUITER
FIRST CHANNEL F/C
THIRD CHANNEL F/R
SECOND CHANNEL R/L
FOURTH CHANNEL R/R
E Q
31
32
33
SYNTHESIZER
SYNTHESIZER
100
SOUND PROCESSING SYSTEM
101
102
103

FIG.11

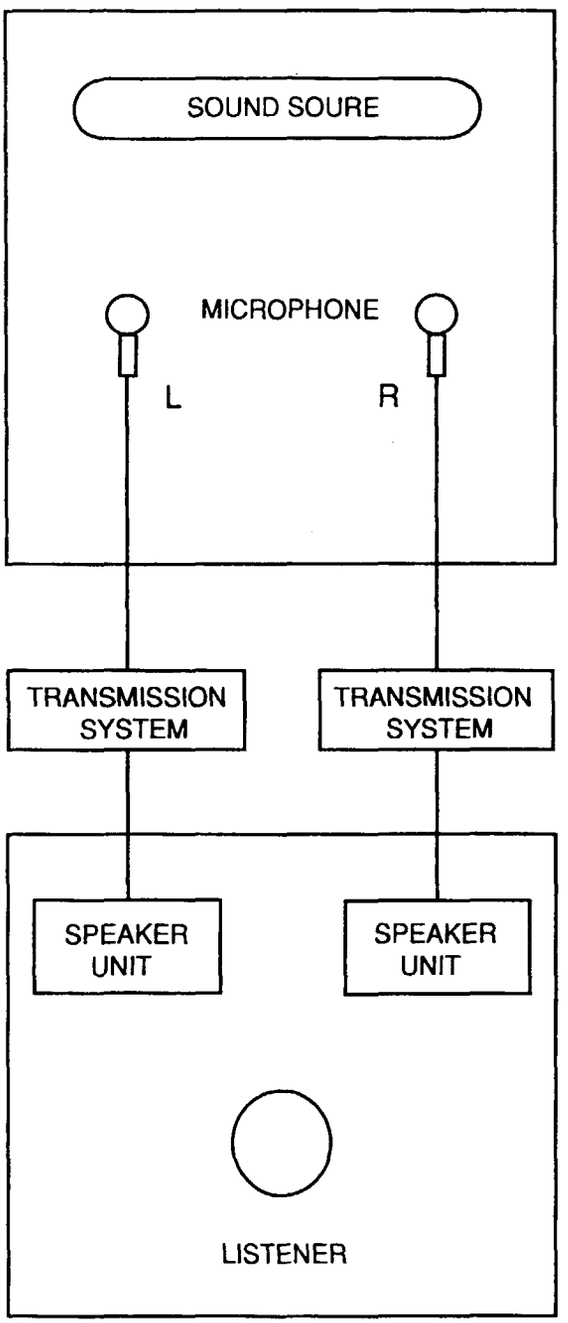
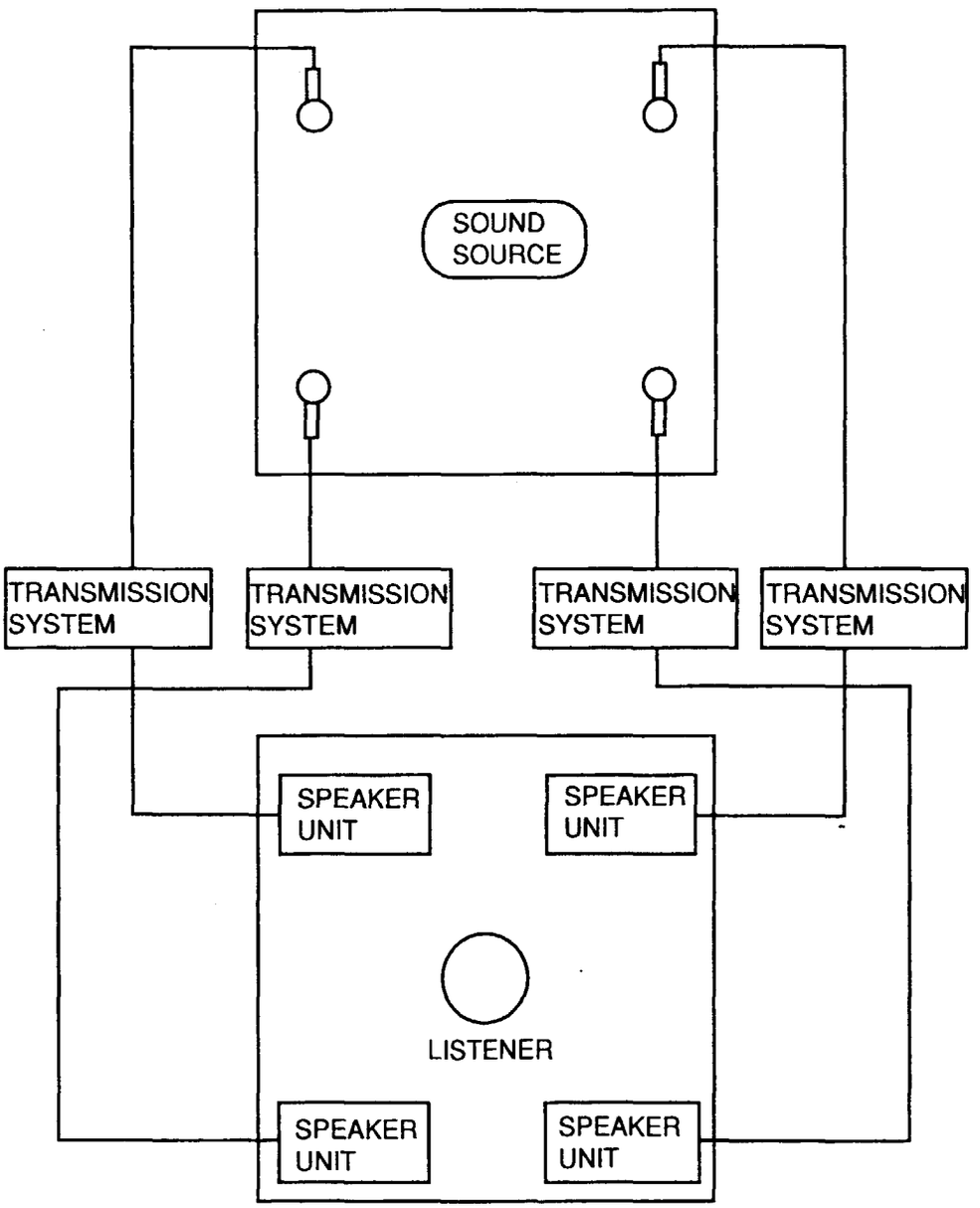

SOUND SOURE
MICROPHONE
L
R
TRANSMISSION SYSTEM
TRANSMISSION SYSTEM
SPEAKER UNIT
SPEAKER UNIT
LISTENER

FIG.12


SOUND SOURCE
TRANSMISSION SYSTEM
TRANSMISSION SYSTEM
TRANSMISSION SYSTEM
TRANSMISSION SYSTEM
SPEAKER UNIT
SPEAKER UNIT
LISTENER
SPEAKER UNIT
SPEAKER UNIT

INTERNATIONAL SEARCH REPORT

International application No.
PCT/JP96/00939

A. CLASSIFICATION OF SUBJECT MATTER

Int. Cl^6 H04R5/027

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

Int. Cl^6 H04R5/027

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

Jitsuyo Shinan Koho 1926 - 1996
Kokai Jitsuyo Shinan Koho 1971 - 1996
Toroku Jitsuyo Shinan Koho 1994 - 1996

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

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	JP, 58-43959, B2 (Victor Company of Japan, Ltd.), September 30, 1983 (30. 09. 83)(Family: none)	1, 2, 4-7, 9-11
Y	Microfilm of the specification and drawings annexed to the written application of Japanese Utility Model Application No. 180185/1975 (Laid-open No. 89001/1977)(Matsushita Electric Industrial Co., Ltd.), July 2, 1977 (02. 07. 77)	4, 9
Y	JP, 53-140002, A (Victor Company of Japan, Ltd.), December 6, 1978 (06. 12. 78)(Family: none)	29, 30

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

* Special categories of cited documents:
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"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed
"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
June 21, 1996 (21. 06. 96)	July 2, 1996 (02. 07. 96)
Name and mailing address of the ISA/ Japanese Patent Office Facsimile No.	Authorized officer Telephone No.

Form PCT/ISA/210 (second sheet) (July 1992)