Method for acquiring audio signals is described, wherein a microphone probe (11) equipped with a plurality (Y) of microphone capsules (B) detects a plurality of audio signals, and wherein said detected audio signals are combined together in order to obtain a virtual microphone signal. The latter is generated as a function of characteristic probe data (IRs) measured during a probe characterization step, wherein the signals detected by each microphone capsule (B) are measured following a corresponding predetermined test signal.

An audio acquisition system is also described which allows to implement the method.

15 Claims, 8 Drawing Sheets
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Test signals

Microphone

Y x M x K signals

Probe characterisation

Y x M x K IRs

Filter generation

Y x N FIRs

M azimuth
K elevation

N virtual microphones
N azimuth values
N elevation values

Fig. 2
Signals picked up by the real microphone captures B

Convolution and generation of virtual microphone signals
Fig. 5
Fig. 6

Microphone (32 capsules)

Audio device
EMIB

Controller

DSP

RECORDER

Analog OUTPUT
Digital OUTPUT

N = number of microphones
An = azimuth of n-microphone
En = elevation of n-microphone
Pn = polar pattern of n-microphone

300
Fig. 7
Fig. 8

- Microphone (32 capsules)
- Audio device EMIB
- Controller
- GUI
- Polar pattern (type and order)
  - Azimuth
  - Elevation
- Target function
- Kirkeby Inversion algorithm
- FIR Matrix
- CONVOLVER
- Jack server
- Matrix of measured IIs

32 Audio channels

Internal connections:
- 32 Audio channels to Audio device EMIB
- Jack server to Audio device EMIB
- FIR Matrix to Audio device EMIB

External connections:
- GUI to Controller
- Microphone to Controller

Inversion algorithm

CONVOLVER

Virtual Microphone Audio Output
METODO PARA ObtENER SEÑALES DE ÁUDIO, ENSAYOS DE ACQUISICIÓN DE SEÑALES DE ÁUDIO, Y SISTEMA DE ACQUISICIÓN DE SEÑALES DE ÁUDIO DEL MISMO

FONDO DE LA INVENCIÓN

1. Campo del Invenção

El presente invento se refiere a un método para obtener señales de áudio y un sistema de obtención de señales de áudio capaz de implementar dicho método.

En la televisión y los medios cinematográficos, en general, hay un aumento de las necesidades de grabar sonidos en un entorno tridimensional en el que la grabación está en curso, para que puedan ser reproducidos fielmente en el punto del oyente.

Registrar sonidos en un entorno tridimensional implica la necesidad de conocer la presión y la velocidad de los partículas de aire en un cierto punto espacial.

Para ello, se conoce el uso de microfonos probadores que compilan micrófonos de cápsulas múltiples dispuestos en una superficie, e.g. una superficie esférica.

Un ejemplo de tales microfonos es un microfono probador disponible en el mercado bajo el nombre "EigenMike32" fabricado por la compañía americana "mhAcoustics".

La Fig. 1 muestra un ejemplo de un probador 11 que permite obtener señales de áudio para ser adquiridas de múltiples direcciones espaciales. Dicho probador 11 consta de un número Y (en este caso treinta y dos) de cápsulas de micrófono B dispuestas en un rígido y estrechamente esférico C.

Cada cápsula B detecta una señal de audio que llega desde una dirección espacial diferente.

Al combinarse apropiadamente estas señales es posible obtener una señal correspondiente a la señal que sería registrada por un micrófono con ciertas características deseadas.

Gracias a estos probadores, la persona puede usar “virtual” microfonos con las características deseadas de directividad (cardióide, supercardióide o similar) y posición (azimuth, elevación, etc.).

2. Estadío del Arte

Estos probadores se utilizan generalmente en combinación con sistemas gráficos para mostrar para el usuario cualquier ruido detectado y identificar cualquier defecto mecánico (por ejemplo, un diente roto de una rueda dentada) o cualquier fuente de ruido no deseados.

Para este propósito, se asigna mucha importancia a los probadores de micrófonos en la directividad de micrófonos, y el esfuerzo es canalizado para definir filtros óptimos que puedan asegurar la mejor posible directividad.

Una vez que los óptimos teóricos de estos filtros han sido identificados, la señal de audio del micrófono virtual requerida por el usuario se genera de manera apropiada, medidos por los filtros de salida y aplicando alteraciones de demora y ganancias que son adecuados calculados y luego combinados juntos para obtener ciertas formas de funcionamiento de directividad de micrófono.

El primer límite de estos probadores se relaciona con el hecho de que el uso de filtros teóricos predeterminados, a pesar de que proporcionan buena directividad, a menudo no aseguran una buena calidad de señales de audio.

Además, otro límite de estos probadores es el hecho de que se pueden proporcionar buena directividad hasta ciertas frecuencias, generalmente a 4 kHz, mientras que más allá de las cuales la directividad tiende a deteriorarse.

Estos probadores, por tanto, no son adecuados para su uso en el mundo de la televisión o cinematográfico, en el que, además, en la adquisición de señales de audio, es también muy importante ser capaz de obtener señales de audio de alta calidad.

RESUMEN DEL INVENCIÓN

Es el objeto del presente invento proporcionar un método para obtener señales de audio y un sistema de adquisición de señales de audio que puede superar los defectos de las anteriores.

Este objeto se logra a través de un método y un sistema que incorpora las características definidas en los derechos de autor presentes, y que son considerados como una parte integral del presente invento.

El presente invento se basa en el concepto de que las señales adquiridas por las cápsulas del probador se detectan al principio de forma empírica a partir de un test señal, y se utilizan para obtener un señal correspondiente a la señal que se obtendría al micrófono virtual incluso a frecuencias superiores a 4 kHz.

Asimismo, está posible llevar en cuenta el efecto de la función en el test señal, que de hecho rompe la simetría perfecta del test señal.

Además, el probador puede mantener una buena directividad del micrófono virtual, especialmente a altas frecuencias superiores a 4 kHz, en el que el test señal del micrófono virtual no está basado en un proceso de filtrado teórico, pero en un proceso de filtrado que depende de las características reales del test señal, y en particular en las respuestas impulsivas de las cápsulas, calculadas con el fin de obtener resultados que son más precisos durante el proceso de caracterización del test señal.

DESCRIPTIVO DETALLADO DE LOS DIBUJOS

A continuación se indican las ventajas y los beneficios del presente invento que se verán evidentes a partir de la descripción siguiente de una de las realizaciones del presente invento como se muestra en los dibujos, que se encuentran en el anexo de los dibujos, en los que se presentan, de manera no restringida, por ejemplo:

La Fig. 1 muestra un probador de micrófonos virtual como se describió anteriormente;

La Fig. 2 esquematicamente muestra los pasos del método correspondiente al presente invento;

La Fig. 3 sintéticamente ilustra una operación de convolución ejecutado por el método correspondiente al presente invento;

La Fig. 4 es un diagrama de bloque de un paso del método correspondiente al presente invento;

La Fig. 5 es un diagrama de bloque de un paso del método correspondiente al presente invento en el que se cambia las características de un micrófono virtual;

La Fig. 6 muestra un sistema de adquisición de señales de audio del mismo tipo que se mencionó anteriormente, para ejecutar el método correspondiente al presente invento;

La Fig. 7 muestra el primer variante del sistema de adquisición de señales de audio correspondiente al presente invento;

La Fig. 8 muestra el segundo variante del sistema de adquisición de señales de audio correspondiente al presente invento.

DESCRIPTIVO DETALADO DE LA REALIZACIÓN DE LA INVENCIÓN

Refiriendo al Fig. 2, el método correspondiente al presente invento proporciona la ejecución preliminar de un
In order to choose which source must be listened to and recorded by the probe, it is necessary to synthesize a virtual microphone by starting from the signals detected by the Y microphone capsules.

In other words, the audio signals picked up by the real capsules B of the microphone probe 11 are processed in a manner such as to obtain a signal which ideally corresponds to the one that would be acquired by a microphone whose parameters could be chosen at will by an operator, more specifically pointing direction and directivity.

By “microphone directivity” it is meant the way in which the sensitivity of the microphone varies as the sound incidence angle changes: it may be, for example, cardioid, supercardioid, cardioid of the 3rd order or the like.

The other parameters of a microphone are, more in general, sensitivity, response curve, noise, distortion, dynamic range, impedance, and transient response; in the present text, however, only pointing direction and directivity will be taken into account as parameters of the virtual microphone, leaving out the remaining parameters listed above.

The operator thus chooses the parameters of one or more virtual microphones to be used in the environment where the sound field is to be picked up, e.g. to concentrate on certain areas of the environment to be detected with (virtual) microphones having a certain directivity.

The definition of the parameters of the virtual microphones is schematized in Fig. 2 by block 202.

In accordance with the teachings of the present invention, the virtual microphones are generated in the method step designated in Fig. 2 as "FILTER GENERATION" (reference numeral 203), and involves the generation of a matrix of FIRs filters which is used (as will be explained more in detail hereafter) for filtering the signal picked up by the real microphone capsules B of the probe 11.

As will be better explained below, the operator interacting with the audio acquisition system defines the parameters of the virtual microphone(s) by giving inputs to the system, e.g. by moving a joystick and selecting in real time an area of the environment to be listened to.

Based on the operator inputs, the system generates (step 204 in Fig. 4) a matrix called “target function” A, of size (MxK), which depends on the characteristics of the virtual microphone(s) corresponding to the inputs received from the operator.

The matrix A is thus that matrix which represents the directivity model of the virtual microphone, i.e. that spatial figure which the virtual microphone must tend to.

The elements a_{ik} generally have a value, preferably between 0 and 1, which depends on the spatial coordinates (azimuth and elevation) and directivity of the desired virtual microphone.

The mathematical expression of directivity (e.g. cardioid, supercardioid, cardioid of the 3rd order, etc.) is per se known and is described by functions known in the literature; therefore, the man skilled in the art can create the matrix A corresponding to the desired microphone(s).

The matrix H of FIRs filters is then generated (step 203 in Figs. 2 and 4) by using the known Kirkeby algorithm (in "matlab" notation):

\[ H = A \cdot \frac{\text{Con}(|R_s(w)|)}{\text{Con}(|R_s(\omega)|) \cdot |R_s(\omega)| + e(\omega)} \]  \hspace{1cm} (1)

that is (in standard notation):

\[ H = \frac{\text{Con}(|R_s(\omega)|) \cdot |R_s(\omega)| + e(\omega)}{\text{Con}(|R_s(\omega)|) \cdot |R_s(\omega)| \cdot |R_s(\omega)| + e(\omega)} \]  \hspace{1cm} (2)
Once the matrix H has been thus determined, the virtual microphones are synthesized by filtering the signals picked up by the capsules through the filters determined in accordance with the above-described method.

The signal coming from each capsule is combined (step 207), by means of a convolution operation, with a suitable filter and is then added to the other signals in order to obtain the signal of the desired virtual microphone:

\[
\text{Virtual}_\text{Mic}_1 = \sum_{i=1}^{N} \text{FIR}_i \odot C_i \\
\vdots \\
\text{Virtual}_\text{Mic}_N = \sum_{i=1}^{N} \text{FIR}_N \odot C_i
\]

where:
\[
\text{Virtual}_\text{Mic}_1, \ldots, \text{N} \text{ indicates the audio signal detected by each virtual microphone,} \\
\text{FIR}_1, \ldots, \text{N} \text{ indicates the element } i \ldots, \text{N} \text{ of the matrix H,} \\
\text{C}_i \text{ indicates the signal picked up by the i-th microphone capsule of the probe.}
\]

A graphic diagram of said convolution is also shown in FIG. 3, whereas the second step of the method, called FILTER GENERATION, is also shown in the information flow of FIG. 4.

The above-described method advantageously allows the virtual microphone parameters to be changed in real time.

The operator can change the parameters of the virtual microphone in use (e.g. in order to follow an actor in a cinematographic scene or the action taking place in a certain point of the environment) by acting upon a dedicated control console.

Upon receiving an input corresponding to a change in the parameters of one of the virtual microphones or a request to add or eliminate a virtual microphone, the system will recalculate the filter matrix H.

The flow chart of this operation is shown in FIG. 5.

After turning on a virtual microphone (step 500), it is checked whether an input has arrived which requires a change to the azimuth (step 501); if not, it is checked whether an input has arrived which requires a change in elevation (step 502) and, if also this check gives a negative result, it is checked whether an input has arrived which requires a change in directivity (step 503).

If this last check is also negative, the method goes back to step 501.

Otherwise, if any one of the checks made in the steps 501 to 503 gives a positive result, then the coefficients of the target functions A are recalculated based on the new input (step 504).

After the coefficients have been changed, they can be used by the processor to generate the filter matrix H.

The algorithm schematized in FIG. 5 provides for checking whether the microphone is still active or not (step 505) after the coefficients of the matrix A have been updated. If the microphone is still active, then the process goes back to step 501 and the parameters of the virtual microphone are checked again; if the microphone is not active anymore, then the algorithm is ended (step 506).

In short, therefore, when the operator varies the azimuth and/or elevation and/or directivity of the virtual microphone (and thus the parameters thereof), the coefficients of the target function matrix A are changed accordingly and the matrix H is recalculated.
According to a further improvement, it is also possible to change a virtual microphone without generating a sensation of "jerky" motion affected by disturbances or ground noise: this can be done by executing a dynamic "crossfade" between the audio coming from the virtual microphone in use and that coming from the virtual microphone to which the operator wants to move.

In substance, when the operator changes the virtual microphone in use and chooses a second one, the switch between a first matrix \( \mathbf{T}_1 \) corresponding to a first microphone (the microphone in use) and a second matrix \( \mathbf{H} \) corresponding to a second microphone (the microphone to which the operator wants to move) is carried out gradually by means of an ordered set of transaction matrices (i.e. transaction filters). The sound picked up by the capsules \( \mathbf{B} \) is filtered with the transaction matrices according to their order. More in detail, the ordered set of transaction matrices \( \mathbf{T}_1, \mathbf{T}_2, \mathbf{T}_3 \ldots \mathbf{T}_n \) allows to switch between the first matrix and the second matrix as follows: at the beginning the sound is filtered by the first matrix, then it is filtered by transaction matrix \( \mathbf{T}_1 \), then by transaction matrix \( \mathbf{T}_2 \), then by transaction matrix \( \mathbf{T}_3 \) and so on till to arrive at the second matrix.

Each of the transaction matrices \( \mathbf{T}_1, \mathbf{T}_2, \mathbf{T}_3 \ldots \mathbf{T}_n \) comprises submatrices corresponding to submatrices belonging to either the first matrix or the second matrix. In particular, transaction matrix \( \mathbf{T}_k \) (corresponding the \( k \)-th matrix of the ordered set of transaction matrices, with \( k=2,3,\ldots, n \)) comprises a number of submatrices corresponding to submatrices of the second matrix greater than a previous transaction matrix \( \mathbf{T}_{k-1} \) comprises. Moreover, transaction matrix \( \mathbf{T}_k \) comprises a number of submatrices corresponding to submatrices of the first matrix lower than the previous transaction matrix \( \mathbf{T}_{k-1} \) comprises.

Then, using a mathematical syntax, the transaction matrices comprise submatrices so that:

\[
#S_{2k} \geq #S_{2k+1} \quad \text{and} \quad #S_{1k} \geq #S_{1k+1}, \quad k=2,3,\ldots, n
\]

where:
- \#S2k indicates the number of submatrices of the transaction matrix \( \mathbf{T}_k \) that correspond to submatrices of the second matrix,
- \#S2k+1 indicates the number of submatrices of the transaction matrix \( \mathbf{T}_{k+1} \) that correspond to submatrices of the second matrix,
- \#S1k indicates the number of submatrices of the transaction matrix \( \mathbf{T}_k \) that correspond to submatrices of the first matrix,
- \#S1k+1 indicates the number of submatrices of the transaction matrix \( \mathbf{T}_{k+1} \) that correspond to submatrices of the first matrix,

index \( k \) is any integer value between 2 and \( n \), where \( n \) is the number of the transaction matrices.

As a result, the transaction matrix \( \mathbf{T}_1 \) is the most similar to the first matrix, whereas the transaction matrix \( \mathbf{T}_n \) is the most similar to the second matrix.

In a preferred embodiment, all submatrices have the same sizes and in particular a size (row or column) equal to \( N \). The switch between different filters (i.e. the different matrices) can be done by a standard "crossfade" (i.e. a decrease in the level of an audio signal corresponding to a filter while the audio signal corresponding to another filter increases) between the audio coming from a filter in use and that coming from a following filter: the signal of the filter in use and the one of the following filter are then mixed so as to progressively fade the volume of the former to zero and progressively increase the volume of the latter to the maximum value, thus giving the user a sensation of great smoothness.

Referring now to FIG. 6, there is shown an audio acquisition system \( \mathbf{1} \) for implementing the above-described method. The system \( \mathbf{1} \) allows to pick up audio signals coming from an environment.

The system \( \mathbf{1} \) comprises a microphone probe \( \mathbf{11} \) comprising a plurality of capsules (e.g. a 32-channel microphone probe called “em32.Eigenmike”, sold by company mAhAcoustics), whose signals are pre-amplified and converted into digital form.

The probe \( \mathbf{11} \) is connected to an electronic computer \( \mathbf{3} \) equipped with an audio interface \( \mathbf{2} \) (e.g. an EM1B firewire audio interface), which receives the signals from the probe and transmits them, after having possibly processed them, to a processor \( \mathbf{300} \), e.g. a DSP (Digital Signal Processor), programmed for executing the above-described audio acquisition method.

The system \( \mathbf{1} \) further comprises a data or command input unit \( \mathbf{4} \), also connected to the computer \( \mathbf{3} \), e.g. through a USB (Universal Serial Bus) port, by means of which an operator can supply information about the area where sound must be acquired or directly enter the parameters of one or more virtual microphones (e.g. by selecting predefined forms of directivity by means of buttons).

The data or command input unit \( \mathbf{4} \) may be, for example, a control console equipped with a joystick for controlling the pointing of the virtual microphones.

The system \( \mathbf{1} \) further comprises a recorder \( \mathbf{5} \) and/or an analog output \( \mathbf{6} \) and/or a digital output \( \mathbf{7} \) through which it can record or transmit the signal picked up by the virtual microphones.

In the example of FIG. 6, the recorder \( \mathbf{5} \), the analog output \( \mathbf{6} \) and the digital output \( \mathbf{7} \) are all installed inside the computer \( \mathbf{3} \); alternatively, the recorder \( \mathbf{5} \) may be external to the computer \( \mathbf{3} \) and connected thereto.

FIG. 7 shows an enhanced version of the system \( \mathbf{1} \), designated \( \mathbf{1'} \); this enhanced system allows audio signals to be acquired from an environment and synchronized with video images of that same environment.

In addition to the parts designated by the same reference numerals in FIG. 6 and having the same functions, the system \( \mathbf{1} \) also comprises a video camera \( \mathbf{8} \) that films the environment whose audio signals are to be detected by the probe \( \mathbf{11} \), graphic interface means \( \mathbf{9} \), and a timer \( \mathbf{10} \) (preferably internal to the computer \( \mathbf{3} \) and connected to the processor \( \mathbf{300} \) for synchronizing the audio picked up by the probe \( \mathbf{11} \) with the video captured by the video camera \( \mathbf{8} \).

The video camera \( \mathbf{8} \) frames the environment where the scene whose audio is to be acquired is taking place; for this purpose, the video camera \( \mathbf{8} \) is a wide angle video camera, e.g. of the "dome" type typically used for surveillance purposes or the like.

The video camera \( \mathbf{8} \) transmits the acquired video signal to the graphic interface means \( \mathbf{9} \), which comprise a monitor for displaying the images taken by the video camera \( \mathbf{8} \).

The same graphic interface means \( \mathbf{9} \) are operationally connected to the data or command input unit \( \mathbf{4} \), and therefore receive information about the virtual microphone(s) selected by the operator.

The graphic interface means \( \mathbf{9} \) process this information and translate it graphically; in particular, they display, superimposed on the images taken by the video camera \( \mathbf{8} \), a mobile pointer which indicates the area being listened to by the virtual microphone chosen by the operator.

Preferably, the shape and size of the pointer are related to the microphone’s directivity and orientation, so as to reflect the parameters of the microphone in use and allow it to be controlled more intuitively by the operator.
The data or command input unit 4 may advantageously be fitted with a control lever or a slider or the like to allow an operator to zoom in or out the sound field of the virtual microphone in a quick and intuitive manner.

Through the data or command input unit 4, the operator thus moves the microphone within the filmed scene and can listen separately to different sound sources included in the taken image.

By moving the joystick, the operator moves the virtual microphone and can follow the movement thereof thanks to the images displayed by the graphic interface means 9. By acting upon the slider the operator can control directivity, and the pointer’s size changes accordingly.

In a further alternative embodiment, the pointer may be replaced with coloured areas corresponding to the regions being listened to by the microphone; for example, the best received area may be displayed in red, the other areas being displayed with colder colours according to their reception quality. When the virtual microphone is moved or its directivity is changed, the colour of the images will change as well.

Fig. 8 shows a variant of the system of Fig. 7.

In this example, the operator has the possibility of setting the parameters of the virtual microphone through the data or command input unit 4 or the graphic interface 90, thereby pointing the virtual microphone (in terms of azimuth and elevation) and selecting its directivity (cardioid, supercardioid, cardioid of the 3rd order, etc.).

The graphic interface means 90 of Fig. 8 comprise for this purpose a touch screen which displays the images coming from the video camera 8 and the microphone pointer, as previously explained with reference to Fig. 7.

By interacting with the touch screen, the operator can move the microphone or change the extent of the space to be listened to, i.e. change the microphone’s orientation and directivity.

The virtual microphone data thus set by the user is sent to the processor 300, where the execution of some code portions allows for the generation of the above-mentioned target function A and the calculation of the Kirkeby algorithm, which is made by using the IRs matrix of impulse responses (measured in the aforementioned PROBE CHARACTERIZATION step) pre-loaded into the memory and relating to the microphone probe 11.

The filter matrix H is then generated as previously described.

The file containing the FIRs filter coefficients is then used in order to carry out the filtering process with the audio data coming from the microphone probe 11.

The virtual microphone signal synthesized by said filtering process is returned to a Jack interface 15, which may then deliver it to digital outputs (ADAT) provided on the EMIB card or divert it towards a memory card.

Every time the virtual microphone’s parameters are changed (e.g. when directivity is changed), the Kirkeby algorithm is executed again and a new matrix H is calculated, so that a change is made in real time.

In this respect, the processor 3 or the processor 300 preferably comprises a memory area (e.g. a flash memory) which stores the matrix

\[ \Gamma = \frac{\text{Conf}[	ext{IRs}]}{\text{Conf}[	ext{IRs}] \cdot \text{IRs} + \epsilon} \]

calculated during the probe characterization step and therefore dependent on the capsules’ impulse responses calculated by using the predetermined and known test signals.

This solution allows to reduce the computational cost required by the above-described audio acquisition method;

when the matrix \( H \) is to be re-calculated, it is not necessary to recalculate \( \Gamma \), but only the product of the matrices \( A \) and \( \Gamma \).

Although the present invention has been described herein with reference to some preferred embodiments, it is apparent that those skilled in the art may make several changes to the above-described audio acquisition system and audio acquisition method.

In particular, the various elements and logic blocks of the audio acquisition system may be composed and distributed in many different ways while still carrying out, as a whole, the same functions or functions being equivalent to those described herein.

The invention claimed is:

1. Method for acquiring audio signals, wherein a microphone probe equipped with a plurality of microphone capsules detects a plurality of audio signals and wherein said detected audio signals are combined in order to obtain a signal of a virtual microphone, wherein said signal of the virtual microphone is generated as a function of characteristic probe data measured during a probe characterization step, wherein the signals detected by each microphone capsule are measured following a corresponding predetermined test signal, wherein said signal of a virtual microphone is calculated on the basis of desired virtual microphone parameters of the virtual microphone, and wherein said signal of a virtual microphone is generated by filtering the signals received by said plurality of capsules through a filter \( H \) calculated according to the following formula:

\[ H = \frac{\text{Conf}[	ext{IRs}]}{\text{Conf}[	ext{IRs}] \cdot \text{IRs} + \epsilon} \]

where:

- IRs is the matrix of the impulse responses of each microphone capsule in response to said predetermined test signal,
- \( \text{Conf} \) is a so-called “target function” matrix generated on the basis of said virtual microphone parameters,
- \( \epsilon \) is a predefined adjustment parameter.

2. Method according to claim 1, wherein the probe characterization step comprises:

- subjecting said probe to multiple test signals whose emission coordinates are relative to the probe are known, detecting the signals picked up by each microphone capsule of said probe at said test signals,
- generating a matrix of the impulse responses of said capsules.

3. Method according to claim 1, wherein every change in the virtual microphone parameters of said virtual microphone is followed by a new generation of filters which can be used for filtering the signals received by said plurality of capsules and generating a new audio signal of said virtual microphone.

4. Method according to claim 3, wherein the following occurs when the virtual microphone parameters of said virtual microphone are changed in order to switch from a first virtual microphone, corresponding to a first filter, to a second virtual microphone:

- a second filter corresponding to the second virtual microphone is calculated;
- an ordered set of transaction filters is calculated, wherein each of said transaction filters comprises submatrices corresponding to submatrices of either said first filter or said second filter,
Audio acquisition system, comprising at least one microphone probe equipped with a plurality of microphone capsules for detecting a plurality of audio signals, and at least one processor adapted to combine the signals received by said plurality of capsules according to the order of said set of transaction filters; and wherein the number of second filter submatrices of said transaction filter is greater than the number of second filter submatrices of a previous transaction filter, and wherein the number of first filter submatrices of said transaction filter is lower than the number of first filter submatrices of a previous transaction filter; the signal picked up by said capsules is filtered through said transaction filters according to the order of said set of transaction filters; after the last transaction filter of said set, the signal picked up by said capsules is filtered through said second filter.

Method according to claim 4, wherein the following occurs in order to switch from a filter in use to a filter following said filter in use:
- said filter following said filter in use is calculated;
- the signal picked up by said capsules (B) is filtered through said filter following said filter in use;
- signals of said filter in use and of said filter following said filter in use are mixed together;
- the level of the signal of said filter in use is decreased proportionally to the increase in the level of the signal of said filter following said filter in use.

Method according to claim 1, wherein an operator sets orientation and/or directivity characteristics of said virtual microphone.

Method according to claim 1, wherein said virtual microphone parameters comprise orientation and/or directivity characteristics of said virtual microphone.

Audio acquisition system, comprising at least one microphone probe equipped with a plurality of microphone capsules for detecting a plurality of audio signals, and at least one processor adapted to combine the signals received by said plurality of capsules in order to obtain a signal of a virtual microphone.

wherein said processor comprises code portions which, when executed, allow said signal of a virtual microphone to be generated by filtering the signals received by said plurality of capsules through a filter H calculated according to the following formula:

\[
H = A \cdot \frac{\text{Conf}[\text{IR}(s(o))]}{\text{Conf}[\text{IR}(s(o))] \cdot C - \text{IR}(s(o)) + e(s(o))}
\]

where:
- \(\text{IR}(s(o))\) is the matrix of the impulse responses of each microphone capsule in response to said predetermined test signal,
- A is a so-called “target function” matrix generated on the basis of said virtual microphone parameters,
- \(e(s(o))\) is a predefined adjustment parameter.

System according to claim 9, further comprising means feasible to an operator of said system for setting the virtual microphone parameters of at least one virtual microphone.

System according to claim 10, wherein said said virtual microphone to said graphic interface means, so that said virtual microphone parameters can be set on the basis of desired virtual microphone parameters of the virtual microphone.

System according to claim 9, wherein said processor comprises code portions which, when executed, allow said signal of a virtual microphone to be generated by filtering the signals received by said plurality of capsules through a filter H calculated according to the following formula:

\[
H = A \cdot \frac{\text{Conf}[\text{IR}(s(o))]}{\text{Conf}[\text{IR}(s(o))] \cdot C - \text{IR}(s(o)) + e(s(o))}
\]

where:
- \(\text{IR}(s(o))\) is the matrix of the impulse responses of each microphone capsule in response to said predetermined test signal,
- A is a so-called “target function” matrix generated on the basis of said virtual microphone parameters,
- \(e(s(o))\) is a predefined adjustment parameter.

System according to claim 9, further comprising means feasible to an operator comprise a touch screen.

System according to claim 9, further comprising a recorder and/or an analog output and/or a digital output for recording and/or transmitting the signal picked up by the at least one virtual microphone.

System according to claim 9, wherein said system comprises a video camera operationally connected to graphic interface means adapted to display on a monitor the images taken by said video camera, and wherein said processor is adapted to transmit information about characteristics of said virtual microphone to said graphic interface means, so that said graphic interface means can generate a graphic element adapted to be superimposed on said images displayed on said monitor.

System according to claim 9, wherein said system comprises a video camera operationally connected to graphic interface means adapted to display on a monitor the images taken by said video camera, and wherein said system comprises a timer for synchronizing the audio picked up by the probe with the video picked up by the video camera.

System according to claim 9, wherein said virtual microphone parameters comprise orientation and directivity of the virtual microphone.