TIMBRAL CORRECTION OF AUDIO REPRODUCTION SYSTEMS BASED ON MEASURED DECAY TIME OR REVERBERATION TIME

ABSTRACT

The invention relates to a method and system for use in directly adjusting the timbre of a reproduced audio signal in any closed or partially enclosed space according to the measured reverberation time or other function describing the decay of sound within the space. The measurement of the reverberation time and the correction of the timbre are performed by a system that can be incorporated within the installed audio reproduction system, although a separate measuring system could alternatively be used. The measurement of decay time or reverberation time for the space is by known methods. The invention centres around the calculation and application of a correction filter determined directly from the measured decay time or reverberation time for the space.

When a loudspeaker is placed within an enclosed space, the timbre of the loudspeaker as perceived by a listener is affected by the acoustical properties of the space or room. Consequently, the same reproduction system placed in different rooms with differing acoustical properties will sound differently. This coloration of the timbral balance by differing room acoustics often has a detrimental effect on the timbral balance and the perceived sound quality of the audio reproduction system.

The method and system according to the invention avoid the detrimental effect that differing reverberation time between different listening spaces has on the timbral balance of an audio reproduction system by insertion of a correction filter in the signal path, the filter characteristic of which filter is determined based on the decay of sound in the room.

TECHNICAL FIELD

The invention relates generally to the use of decay time or reverberation time of a room or other at least partially enclosed spaces for directly adjusting or correcting the timbre of sound reproduced by an audio reproduction system in this room or space and to methods and systems for use in directly adjusting or correcting the timbre of a reproduced audio signal in any at least partially enclosed room or space based on the decay time or reverberation time within the room or space.

BACKGROUND OF THE INVENTION

When a loudspeaker is placed within an enclosed space, the timbre of the loudspeaker as perceived by a listener or listeners is affected by the acoustical properties of the space. Consequently, the timbre of a given reproduction system or loudspeaker(s) placed in different rooms with differing acoustical properties will be perceived differently - they will sound different in different rooms.

When listening to a loudspeaker in a closed space or room, the listener hears both the direct sound from the loudspeaker and also reflected sound from surfaces within the space or room. The combination of direct and reflected energy colours the timbral balance of the audio reproduction system. This coloration of the timbral balance often has a detrimental effect on the timbral balance and the perceived sound quality of the audio reproduction system.

The designer of a sound reproduction system usually wishes to give the listener the same intended listening experience regardless of the acoustical properties of the listening space. In order to compensate for differing acoustical properties of different listening spaces, knowledge of the reverberation time or another function describing the sound decay within the space is necessary.

Reverberation time is a known acoustical parameter and is a measure of the time taken for sound to decay in a space or room. Reverberation time RT, which is a function of frequency, is per definition the time required for the sound energy density to decay 60 dB. Decay time is also a measure of the time taken for sound to decay in a space or room and is a fraction of the reverberation time according to the available measuring conditions. For example, the influence of background noise may limit the available measurable decay of sound in a space or room.

SUMMARY OF THE INVENTION

Based upon the above background, it is an objective of the present invention to provide a method and corresponding devices and systems that compensate for and reduce the detrimental effect the acoustic properties of a listening space or room have on the perceived acoustic performance of an audio reproduction system. According to a specific embodiment of the invention, the audio reproduction system itself measures the reverberation time or other function describing the sound decay within the space or room and thereafter applies appropriate correction.

Specifically – but not exclusively – the determination of the decay time or reverberation time and the adjustment or correction of the timbre are according to the invention performed by a system that is incorporated within the installed audio reproduction system and not by a separate system, but implementation of the method according to the invention could also be accomplished by a separate system. A basic feature of the invention is the calculation and application of a correction filter determined directly from the measured decay time or reverberation time for the space.

As mentioned above, reverberation time RT, which is a function of frequency, is per definition the time required for the sound energy density to decay 60 dB. In practice it is often not possible to measure sound decay over the full 60 dB dynamic range and sound decay may be measured over any other dynamic range according to for instance the signal to noise ratio obtainable in the particular situation. Thus, the measurement of reverberation time according to the above definition is not a prerequisite for the present invention and sound decay may be determined in other manners, as exemplified below in the detailed description of the invention.

The method and corresponding devices and systems according to the present invention could find use within all fields of audio reproduction in domestic and professional listening environments, where listening is performed within a closed space or room and where an audio reproduction system may be placed in spaces or rooms with differing acoustic properties.

The above and other objectives and advantages are according to a first aspect of the invention as defined by claim 1 attained by the use of a function describing the decay of acoustical energy in a room or other at least partially enclosed space as a function of time. This function will also typically be a function of frequency. Specifically a pre-determined decay time or reverberation time RT of a room or other at least partially enclosed space as a function of frequency is according to the invention used for directly adjusting or correcting the timbre of sound reproduced by a sound reproduction system in said room or other at least partially enclosed space.

The above and other objectives and advantages are according to a second aspect of the invention as defined by claim 2 attained by a method for adjusting or correcting the timbre of sound reproduced by at least one transducer, such as a loudspeaker in a room or other at least partially enclosed space, the method comprising the steps of:

- determining a function describing the decay of acoustical energy such as the decay time or reverberation time RT of said room or space as a function of frequency;

- based on said function, such as the decay time or reverberation time RT, determining a correction curve (filter characteristic) C as a function of frequency, where said correction curve C is a function of said function that describes the decay of acoustical energy, such as the decay time or reverberation time RT;

- implementing said correction curve (filter characteristic) as an electronic filter;

- processing an electrical signal via said electronic filter and providing the processed signal to one or more of said transducers and/or additional transducers.

The above and other objectives and advantages are according to a third aspect of the invention as defined by claim 8 attained by a system for adjusting or correcting the timbre of an audio signal reproduced by at least one loudspeaker in a room, the system comprising:

- at least one sound source, such as a loudspeaker for emitting sound energy to said room, thereby creating a sound field in said room;

- at least one sound sensitive means, such as a microphone for converting acoustical energy from said sound field in the room to electrical energy;

- means for generating a test signal for emission by said at least one sound source into said room;

- means for determining a function describing the decay of acoustical energy such as the decay time or reverberation time RT as a function of frequency based on said test signal and on a signal provided by said at least one sound sensitive means;

- means for determining a correction curve (filter characteristic) C as a function of frequency, where said correction curve C is a function of said function that describes the decay of acoustical energy, such as the reverberation time RT or decay time;

- correction filter means, the frequency response of which is determined based on said correction curve C;

whereby said correction filter means can be used for processing an electrical signal and where the processed electrical signal is provided to one or more of said sound sources and/or additional sound sources.

According to a preferred embodiment of the present invention, a number of loudspeaker-microphone combinations are used within a space or room. The loudspeaker-microphone combinations are designed in such a way that the microphone is an integrated part of the loudspeaker's design. Furthermore, according to this embodiment active loudspeaker systems are used, where the internal signal conditioning for the loudspeaker and microphone, within the loudspeaker systems, is performed digitally. The loudspeaker systems are connected to a network enabling 2-way data communication. A master unit provides control of the system. This master unit may be a separate master unit or one of the loudspeaker-microphone combinations on the network that has been designated as the master unit.

The total number of loudspeakers can exceed the number of microphones in the audio reproduction system, in other words, loudspeakers without microphones can be included in the system but they cannot themselves provide a microphone measurement for the calculation of the decay time or reverberation time. They can however, be used to reproduce a test signal for measurement by the microphones in the audio reproduction system. A calculated correction or corrections can then be applied to some or all loudspeakers connected to the audio reproduction system.

For loudspeaker systems where the internal signal conditioning for the loudspeaker is not performed digitally, for example in the case of an analogue active loudspeaker or a passive loudspeaker system, or the loudspeaker system does not have network capabilities, the loudspeaker system can be connected to the network and thus to the said audio reproduction system by an interface that can communicate with the audio reproduction system. The interface can then initiate a test signal that can be reproduced by the said loudspeaker(s) and also apply the necessary correction or corrections.

In certain specially designed loudspeaker systems, the microphones in the loudspeaker-microphone combinations mentioned can be replaced by using the loudspeaker diaphragm(s) within the loudspeaker system as the microphone.

Upon installation of an audio reproduction system incorporating the present invention, the system itself, or a user, initiates a measurement sequence that automatically measures the decay time or reverberation time within the space or room using the installed audio reproduction system. The measured decay time or reverberation time is then used to calculate one or more correction filters that are then applied to the audio reproduction system. According to the invention, a single calculated correction filter can be used for all loudspeakers in the system, but it is also possible to apply different calculated filters to each individual loudspeaker or to groups of loudspeakers in the system. The measurement sequence can be initiated at any time should the user wish such as when the acoustical properties of the space or room are changed.

The then calibrated audio reproduction system should give the same intended listening experience regardless of the measured decay time or reverberation time within the listening space or room.

According to further aspects the present invention also relates to an audio reproduction system comprising correction filter means receiving an audio signal and providing adjusted or corrected output signals to one or more loudspeakers, where said correction filter means has a filter curve C determined by the method according to the present invention or by the system according to the present invention.

The method and system according to the invention for adjusting or correcting timbre of an audio reproduction system can also be applied in connection with combinations of loudspeaker drivers for instance mounted in a single cabinet, where all of said drivers or chosen drivers are provided with signals that are adjusted or corrected according to the invention This is in the detailed description of the invention illustrated by combinations of a traditional loudspeaker driver and gradient loudspeakers.

BRIEF DESCRIPTION OF THE FIGURES

The present invention will be more fully understood with reference to the following detailed description of embodiments of the invention in conjunction with the figures, where

figure 1 shows an enclosed space or room with a number of loudspeaker-microphone combinations and a number of loudspeakers without microphones placed throughout the space at any position;

figure 2 shows an example of a control set-up for the system, where a master unit controls the loudspeaker-microphone combinations and loudspeakers without microphones via a 2-way serial link;

figure 3 shows an example of a control set-up for the system, where one of the loudspeaker-microphone combinations is designated as the master unit and controls the other loudspeaker-microphone combinations and loudspeakers without microphones via a 2-way serial link;

figure 4 shows an example of a control set-up for the system, where a master unit controls the loudspeaker-microphone combinations and loudspeakers without microphones via 2-way parallel links;

figure 5 shows an example of a control set-up for the system, where one of the loudspeaker-microphone combinations is designated as the master unit and controls the other loudspeaker-microphone combinations and loudspeakers without microphones via 2-way parallel links;

figure 6 shows a block diagram of the measurement source, where a trigger from the master unit initiates a test signal that is reproduced by the loudspeaker in question and where the level of the test signal can be controlled;

figure 7 shows a block diagram of the measurement receiver, where a trigger from the master unit initiates the measuring sequence and where the microphone in question measures the impinging sound at its position, the microphone signal is amplified, the signal is then processed and final data is then available for this microphone position;

figure 8 shows that data from the microphone(s) in question can be combined, where this data is then weighted resulting in a new data set, which can be sent to the loudspeaker(s) in question and where some loudspeakers may have an interface that receives the data;

figure 9 shows a schematical representation of two different loudspeaker systems, i.e. a traditional system (to the left) with a loudspeaker unit or units typically mounted only on the front of an enclosure and a traditional loudspeaker provided with another electrically and acoustically separate combination of loudspeaker units mounted such that they face in another direction, a so called gradient loudspeaker with, preferably, a bidirectional response;

figure 10 shows the audio signal path for correction of the timbre described by embodiment 1 with a traditional loudspeaker system and also the typical directivity of such a loudspeaker system;

figure 11 shows the audio signal path described by embodiment 2 with a traditional loudspeaker system in combination with a gradient loudspeaker system and also the typical directivity for these two loudspeaker systems;

figure 12 shows the audio signal path described by embodiment 3 with a traditional loudspeaker system in combination with a gradient loudspeaker system and also the typical directivity for these two loudspeaker systems;

figure 13 shows an average reverberation time curve Y (frequency in Hertz versus time in seconds) for a typical medium-sized listening space;

figure 14 shows a typical weighting function W;

figure 15 shows the reverberation time curve Y weighted with the function W to give a new weighted reverberation time curve C_1;

figure 16 shows the weighted reverberation time curve C_1 (solid curve) and the same curve forced to zero at the upper and lower ends of the frequency range C_2 (dashed curve);

figure 17 shows a smoothed version C_3 of the curve C_2;

figure 18 shows the smoothed curve C_3 (dashed curve) and the equalisation curve C_4 (solid curve) based upon the measured reverberation time Y shown in figure 13;

figure 19 shows the correction or equalisation curve C_4 shown as gain in decibels C_5;

figure 20 shows a reverberation time curve for an atypical listening space; and

figure 21 shows the correction or equalisation curve in decibels for the reverberation time curve shown in figure 20.

DETAILED DESCRIPTION OF EMBODIMENT 1 OF THE INVENTION

Referring to figure 1, a number of loudspeaker-microphone combinations 2, 3, 4, 5 and 6 are installed in a listening space or room 1. They are connected together with a 2-way network (not shown in the figures) and one of the loudspeaker-microphone combinations, or a separate part of the sound reproduction system such as an audio unit (CD/radio player or Hard Disc system or server), is designated as the master unit. The number of loudspeaker-microphone combinations can be supplemented with a number of loudspeakers 7 and 8 without microphones, potentially via an interface(s).

With reference to figures 2 through 5 various loudspeaker/microphone/control unit combinations are illustrated, but other configurations would also fall within the scope of the present invention. Thus, figure 2 shows an example of a control set-up for a system according to the invention, where a master unit 9 controls the loudspeaker-microphone combinations 11, 12 and loudspeakers 13 without microphones via a 2-way serial link.

Figure 3 shows an example of a control set-up for the system, where one of the loudspeaker-microphone combinations 16 is designated as the master unit and controls the other loudspeaker-microphone combinations and loudspeakers without microphones via a 2-way serial link;

Figure 4 shows an example of a control set-up for the system, where a separate master unit 18 controls the loudspeaker-microphone combinations and loudspeakers without microphones via 2-way parallel links;

Figure 5 shows an example of a control set-up for the system, where one of the loudspeaker-microphone combinations 25 is designated as the master unit and controls the other loudspeaker-microphone combinations and loudspeakers without microphones via 2-way parallel links;

Once the system according to the invention is connected, the measurement process can be initiated as schematically illustrated in figures 6-7. Referring to figure 6 the designated master unit triggers 27 the first loudspeaker LS_1 (reference numeral 2 in figure 1 and reference numeral 31 in figure 6) to reproduce the test signal 28. The test signal is a band-limited signal that can excite the sound field in the listening space or room 1. The level of the test signal is controlled as schematically indicated by the amplifier 30, the gain of which can be controlled as indicated by the level control 29, such that a sufficient sound pressure level is obtained within the listening space or room and at the measuring microphones in the audio reproduction system when the test signal is active. The test signal is preferably an interrupted signal. When the test signal is reproduced by loudspeaker LS_1 (2 or 31) the measurement receivers in the system (see figure 7), i.e. the microphones (32 in figure 8) and associated signal processing means 34, 35, 36, 37 are triggered 33 to measure the impinging sound at the microphones. It is the decay of sound within the listening space or room when the test signal is interrupted that is relevant for the measurement of decay time or reverberation time. The relevant period of time can be divided into three intervals:

a) a period where the test signal is at its maximum or steady-state level,

b) a period of decay immediately after the test signal is interrupted,

c) a period of background noise.

The individual microphone signals are amplified by suitable amplifier means 34 and subsequently processed as indicated by reference numerals 35 and 36. At least two methods could be used:

1) The processing can involve Fast Fourier Transforms (FFTs) of the microphone signal at a certain frequency resolution and at discrete time intervals for a period of time as schematically indicated by block 35 in figure 7. The FFT information is grouped into frequency bands f_f and a slice of data for each frequency band is calculated for the period of time in question as indicated by reference numeral 36.

2) Alternatively, the microphone signal can be filtered with filter banks (digital or analogue) into the desired frequency bands f_f and a slice of data for each frequency band is calculated for the period of time in question 36.

Within each frequency band f_f the level of time intervals (a) and (c) is calculated and a suitable interval for the measurement of the sound decay is selected. The steady state level (a) is determined from an average of the initial levels within the measurement slice. The end of this steady state period (the start of the decay) is determined when the average level of a number of following points in the slice falls below the first average level less a limit value. The level of the background noise (c) is determined in a similar manner by calculating an average level at the end of the measurement slice and by finding the end of the decay (when the average level of a preceding number of points rises above the calculated average by a limit value). The rate of decay -X dB/s is then determined by linear regression from the data points within the period of decay (b) within each frequency band f_f. The result is a data set

$X(M_m)$ for each microphone position M_m which is a function of frequency. The data set $X(M_m)$, reference numeral 37, consists of decay time versus frequency band.

The measurement process described is repeated for each loudspeaker LS_n such that each loudspeaker in turn reproduces the test signal to be measured by the microphones in the audio reproduction system.

All the data sets are collected by the designated master unit and are processed as schematically illustrated by the block diagram in figure 8. The number of individual data sets will usually be M(N-1), where N is the total number of loudspeakers in the audio reproduction system and M is the total number of microphones in the audio reproduction system. This indicates that according to a specific embodiment of the invention, the microphone in a loudspeaker-microphone combination is not included in the measurement when the loudspeaker in the said combination reproduces the test signal. However, the invention also relates to the specific case where the calculations may comprise the microphone signal from the loudspeaker-microphone combination actually emitting the sound.

The data sets $X(M_m)$ can now be used to calculate a correction or corrections for the audio reproduction system. In the simplest case, all of the data sets can be combined (reference numeral 41) using a simple average of the individual data sets $X(M_M)$ for each frequency f_f, as follows:

$$Y = \frac{\sum X(M_m)}{M(N-1)}$$

The resulting combined data set Y is a function of frequency. A typical data set is shown in figure 13 that illustrates an average reverberation time curve Y (frequency in Hertz versus time in seconds) for a typical medium-sized listening space.

As previously mentioned, the present invention is according to a specific embodiment also applicable in cases, where the test signal is emitted from a given loudspeaker and the resulting sound decay, after interruption of the test signal, is recorded by means of a microphone provided in the same loudspeaker as the loudspeaker emitting the test signal. Instead of using a microphone, the loudspeaker itself may even be used to record the sound decay by using the loudspeaker as a microphone. In this case the above expression should be replaced by:

$$Y = \frac{\sum X(M_m)}{MN}$$

Furthermore, in the case where only a single loudspeaker/microphone is present in the system, this expression reduces to:

$$Y = X(M_m)$$

In a more complicated case, the individual data sets can be combined as described above, but in groups that have similar data, or combined in areas within the listening space or room, should the listening space have significantly differing acoustic properties from one area within the space to another area within the space, for example if there are ‘live’ and ‘dead’ areas of the listening space or room.

Data points within a data set that differ significantly from the average value can be automatically excluded from the calculation of the final combined data set Y.

The combined data set or sets can be transposed with a weighting curve or curves 42 (see also figure 14) into a correction curve or curves 43 (see also figure 15) that can be applied to some or all of the loudspeakers in the audio reproduction system 44. For loudspeakers that do not have network or internal digital signal processor capabilities, the correction can be applied by an interface 45 or by the master unit.

The weighting curve W (figure 14) typically describes, but is not limited to, the decay time in a reference listening space or room where the values have been shifted such that the weighting curve has a nominal value of zero between two predefined frequencies. In the example of the curve described in figure 14, the two frequencies are 10 kHz and 20 kHz.

The data set Y (figure 13) is at least according to the shown embodiment weighted by the function W (figure 14) which itself is offset by a factor O, where O is typically the average value of the curve Y between two frequencies as follows:

$$C_1 = \frac{Y}{W + O}$$

The resulting weighted curve is shown in figure 15, where O is the average value of the data Y between two predefined frequencies. In the example of the curve described in figure 15, the two frequencies are 10 kHz and 20 kHz.

The resulting data C_1 is then typically forced to unity at low and high frequencies as shown in figure 16. The low frequencies where the data is forced to unity are typically below the Schroeder Frequency for the listening space or room.

The new data C_2 (see figure 16) is then typically smoothed with a simple smoothing function to give a new curve C_3 as shown in figure 17. The equalisation curve C_4 as shown in figure 18 is derived from the data set C_3 as follows:

$$C_4 = G\left(\frac{1}{C_3}\right)$$

The function G can be, but is not limited to, a simple square-root function such that for a doubling of the decay time a correction of √0.5 or 0.707 is made, however the function G is typically more non-linear in a fashion that compresses high gains if a limit is desired due to system limitations such as headroom.

This correction or equalisation curve (figure 18), which is a function of gain versus frequency, can now be applied to the sound reproduction system. Figure 19 shows the correction or equalisation curve C_4 in decibels.

In this embodiment of the invention, this correction filter or equalisation curve C_4 is applied to the audio signal path as shown in figure 10 for a traditional loudspeaker system 51 which is shown schematically in figure 9 designated by reference numeral 47. This loudspeaker system is preferably a multi-way active design, but may be full-range and/or passive. Changes to the signal from the signal source 49 by the correction filter 50 will directly affect the loudspeaker system's frequency response 53 and power response thus changing the response within the listening space or room according to the measured decay time Y. 52 represents a typical directivity pattern for a traditional loudspeaker system.

Figure 20 shows an atypical reverberation time curve (higher values of reverberation time at mid frequencies (around 1 kHz) than in the upper bass region around 100 Hz) for another listening space or room.

Figure 21 shows a correction or equalisation curve in decibels for this space.

Once applied, the correction remains as an active part of the audio reproduction system until it is disabled or until the system is re-calibrated, for example, if the system is moved to another listening space or room, or more loudspeaker-microphone combinations or loudspeakers are added to the system, or the acoustic properties of the listening space or room are changed.

DESCRIPTION OF EMBODIMENT 2 OF THE INVENTION

Reverting to figure 9 there is shown a simple representation of two different loudspeaker systems. 47 represents a traditional system with a loudspeaker unit or units typically mounted only on the front of an enclosure. 48 represents a traditional loudspeaker with a unit or units typically mounted only on the front of an enclosure and another electrically and acoustically separate combination of loudspeaker units mounted such that they face in another direction, a so called gradient loudspeaker with, preferably, a bidirectional response.

According to the second embodiment of the invention, the correction filter or equalisation curve 55 is applied as shown in figure 11 for a traditional loudspeaker system 57 combined with a gradient loudspeaker system 56 which is shown schematically in figure 9. The system is designated by reference numeral 48 and represents a traditional loudspeaker with a unit or units typically mounted on the front of an enclosure and another electrically and acoustically separate combination of loudspeaker units mounted such that they face in another direction. Referring to figure 11 this additional combination of drive units is designed and driven in such a way to achieve a certain directivity response 58 with a null on the axis 60 of the traditional forward-facing drive unit or units by means of a so called gradient loudspeaker 56 with, preferably, a bidirectional response. 59 represents a typical directivity pattern for the traditional loudspeaker system, i.e. for the loudspeaker system 57 itself. Each of these two loudspeaker systems is preferably a multi-way active design but may be full-range and/or passive. The signal from a signal source 54 is fed through the correction filter 55 and thereafter to the gradient loudspeaker system 56. The original signal is also fed to the traditional loudspeaker system 57. Therefore, the correction filter will affect the loudspeaker system’s power response, thereby correcting the non-direct sound field in a space or room according to the measured decay time Y. The loudspeaker system’s free-field on-axis frequency response will be unchanged thus preserving the direct sound on the axis 60 of the loudspeaker system. The correction filter 55 is given by:

$$CGL1 = H\left(\frac{1}{C_3}\right)$$

H is a function depending on the actual power response of the gradient loudspeaker system in question.

Once applied, the correction remains as an active part of the audio reproduction system until it is disabled or until the system is re-calibrated, for example, if the system is moved to another listening space or room, or more loudspeaker-microphone combinations or loudspeakers are added to the system, or the acoustic properties of the listening space or room are changed.

DESCRIPTION OF EMBODIMENT 3 OF THE INVENTION

In this embodiment of the invention, the correction filter or filters or equalisation curve or curves 62 and 63 are applied as shown in figure 12 for a traditional loudspeaker system 65 combined with a gradient loudspeaker system 64 which is shown schematically in figure 9. This loudspeaker system is designated by reference numeral 48 in figure 9 and represents a traditional loudspeaker with a unit or units typically mounted on the front of an enclosure and another electrically and acoustically separate combination of loudspeaker units mounted such that they face in another direction. This additional combination of drive units are designed and driven in such a way to achieve a certain directivity response 66 with a null on the axis 68 of the traditional forward-facing drive unit or units, a so called gradient loudspeaker with, preferably, a bidirectional response. The gradient loudspeaker in itself is a known acoustical method. 67 represents a typical directivity pattern for the traditional loudspeaker system 65. Each of these two loudspeaker systems is preferably a multi-way active design but may be full-range and/or passive. The signal from a signal source 61 is fed through the correction filters 62 and 63 and thereafter to a gradient loudspeaker system 64 and a traditional loudspeaker system 65, respectively. Therefore the correction filters will affect the loudspeaker system’s power response and frequency response thereby correcting the non-direct and direct sound field in a space or room according to the measured decay time Y. The correction filters 62 and 63 are given by:

$$CGL2 = I\left(\frac{1}{C_3}\right) \qquad CTL = J\left(\frac{1}{C_3}\right)$$

I and J are functions depending on the actual power response of the gradient loudspeaker system and traditional loudspeaker system in question.

Once applied, the correction remains as an active part of the audio reproduction system until it is disabled or until the system is re-calibrated, for example, if the system is moved to another listening space or room, or more loudspeaker-microphone combinations or loudspeakers are added to the system, or the acoustic properties of the listening space or room are changed.

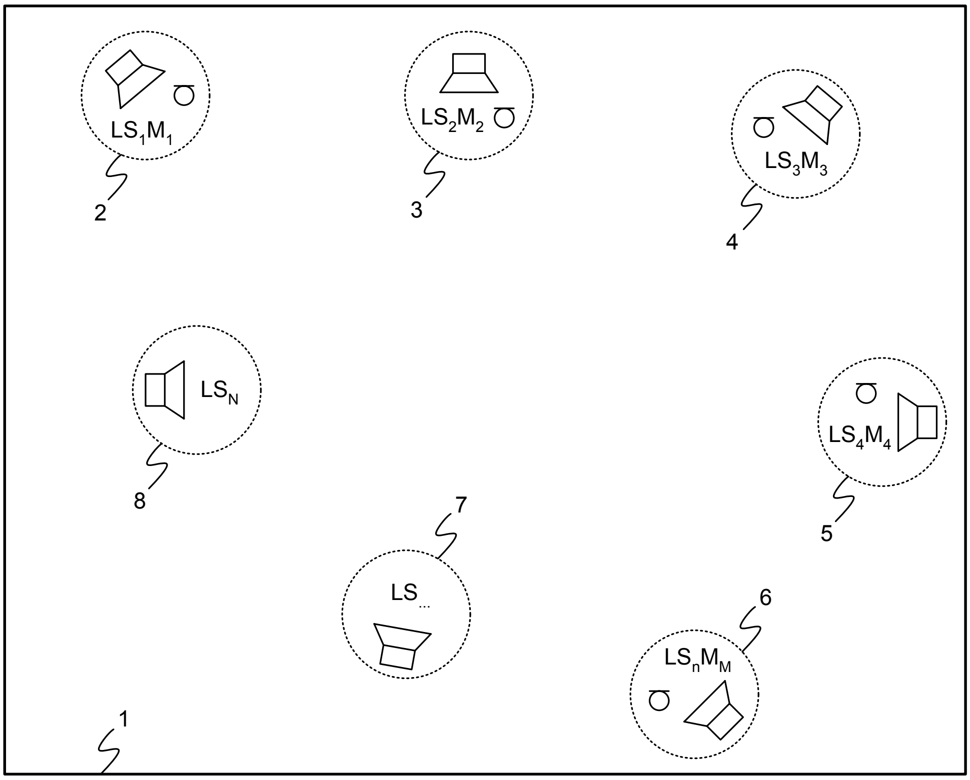


Figure 1

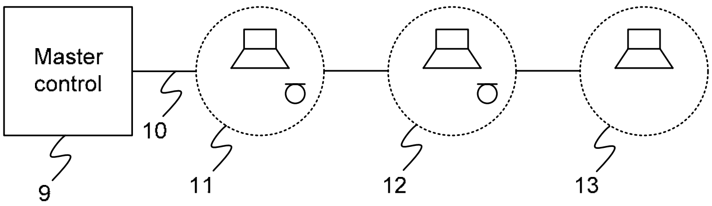


Figure 2

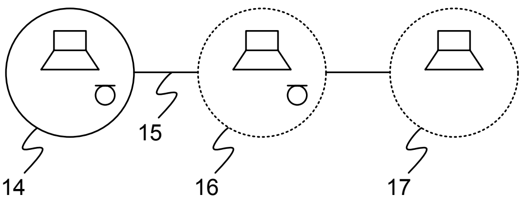


Figure 3

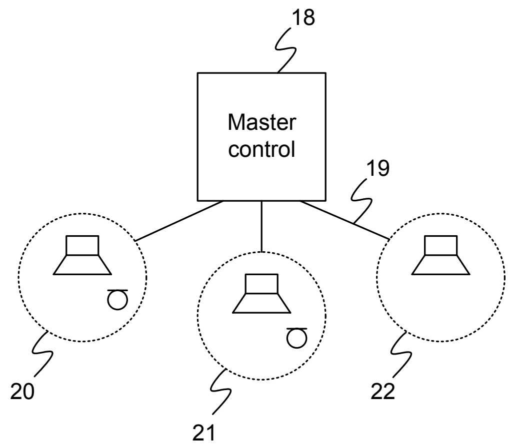


Figure 4

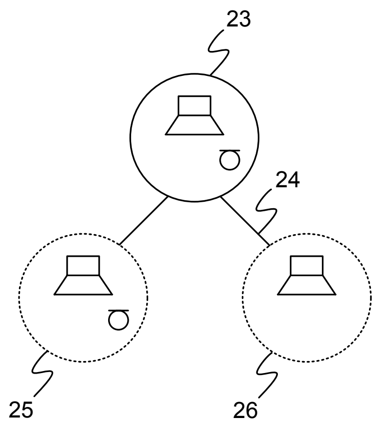


Figure 5

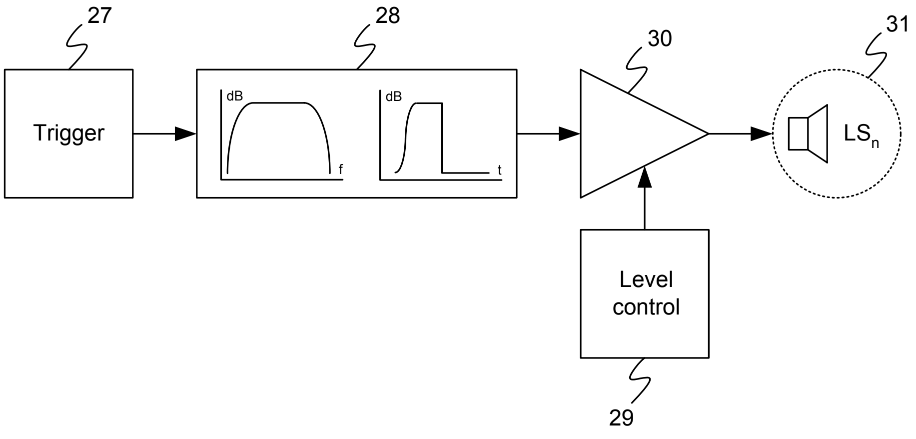


Figure 6

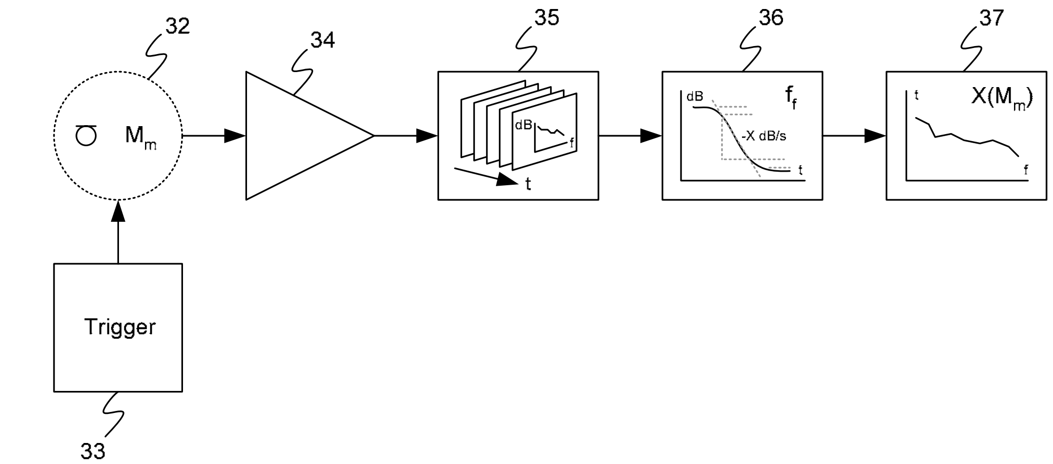


Figure 7

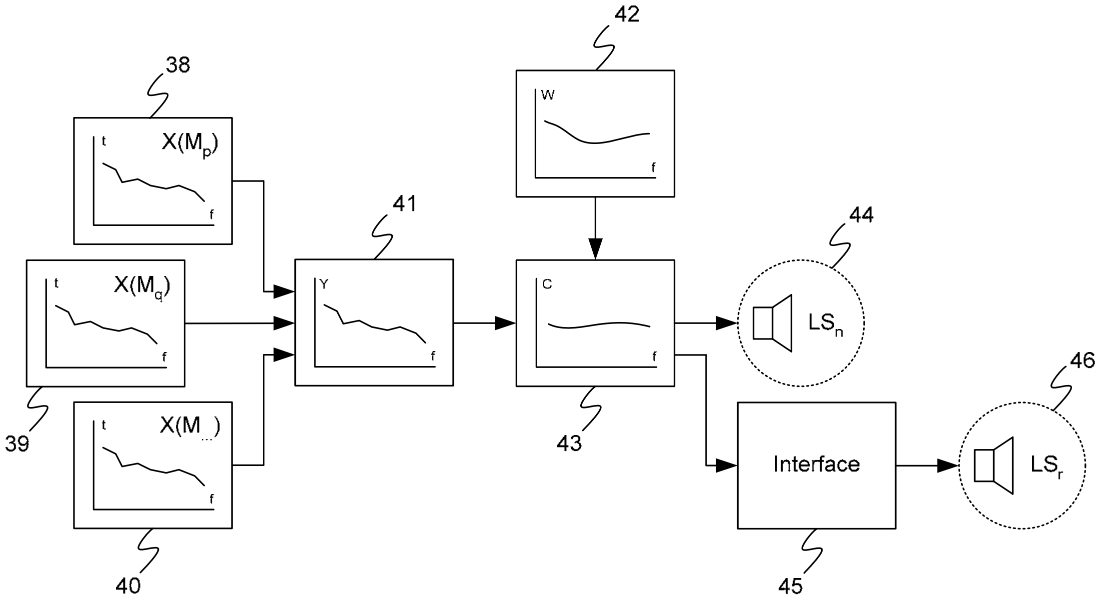
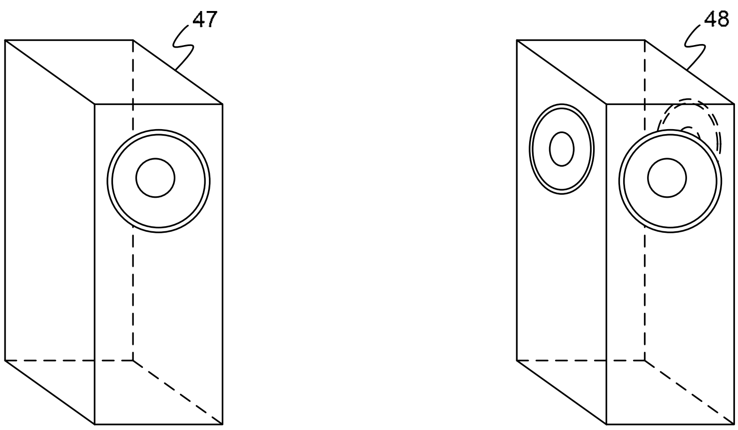


Figure 8



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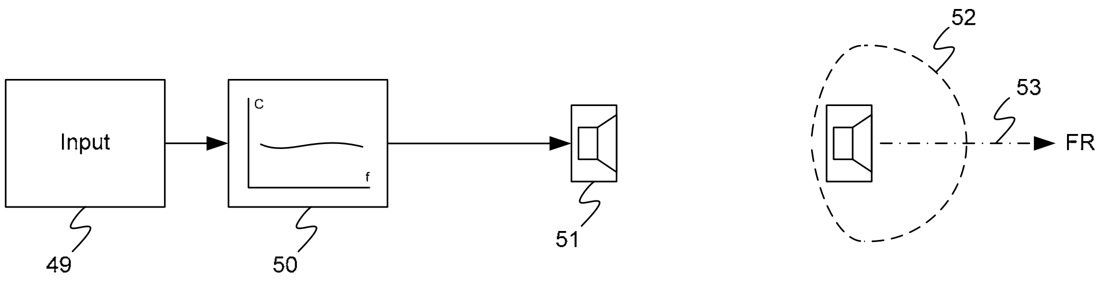

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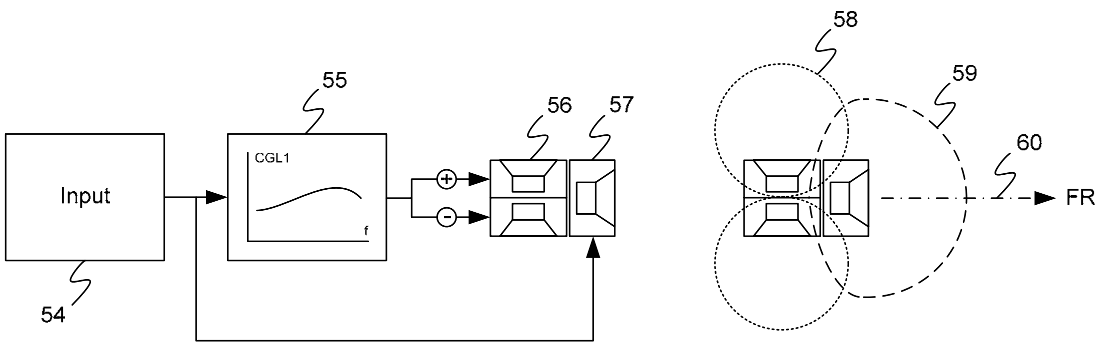

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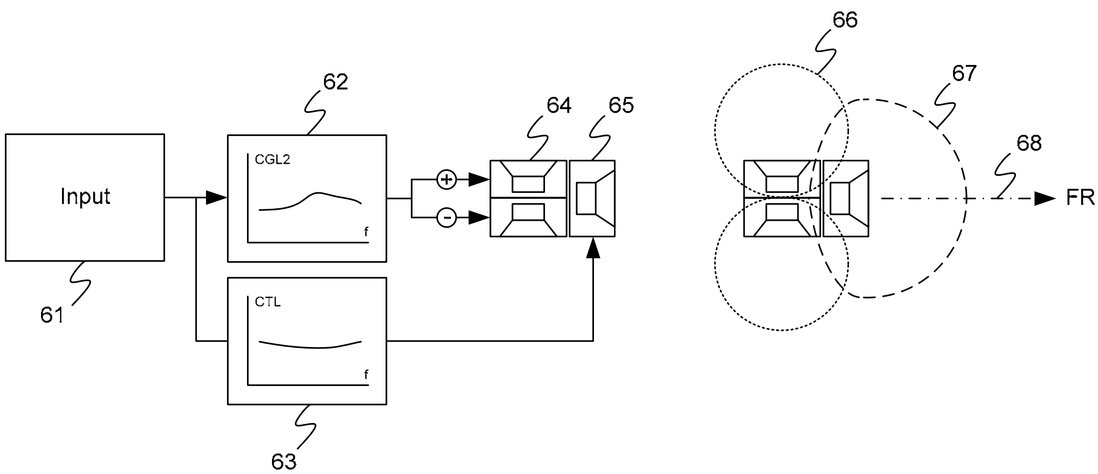

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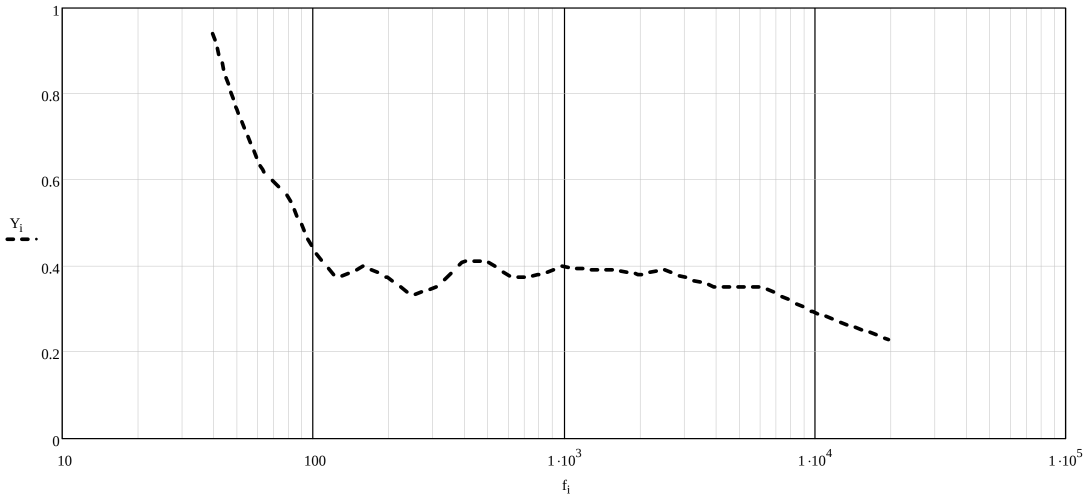

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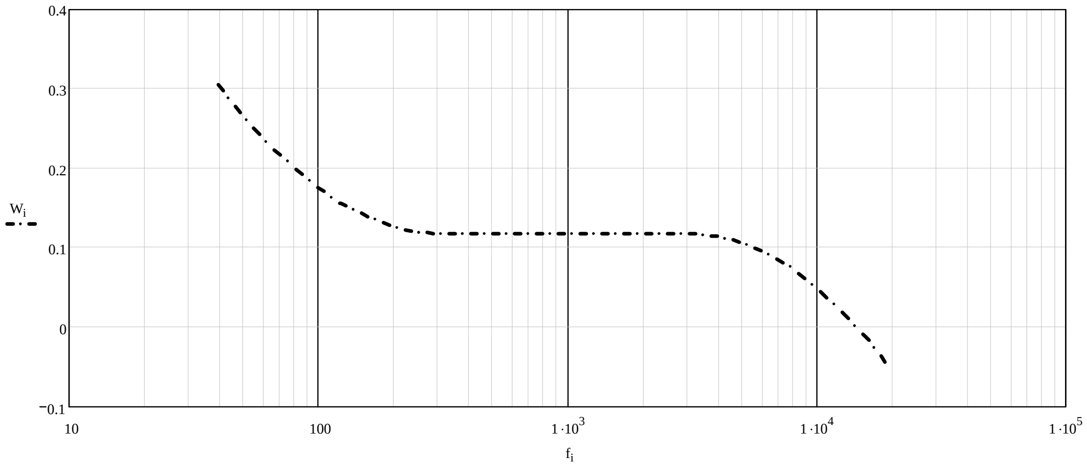

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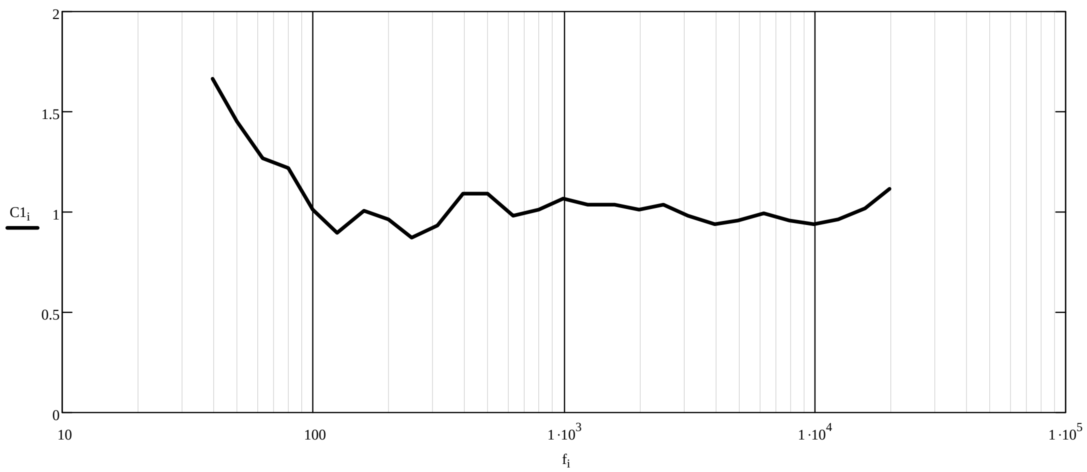

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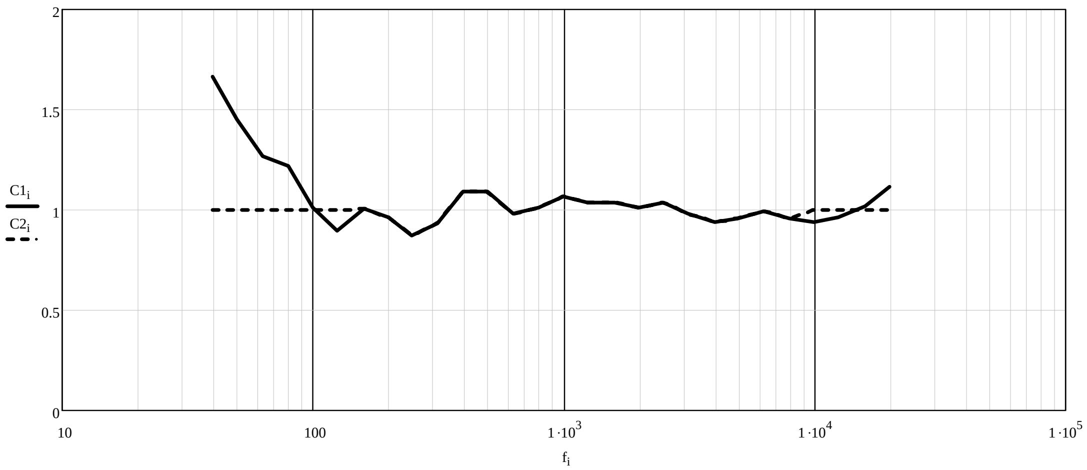

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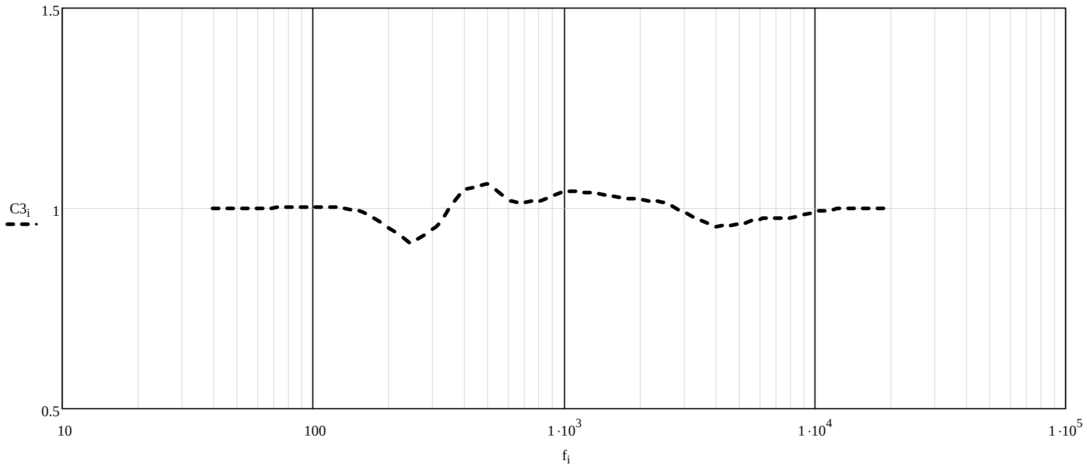

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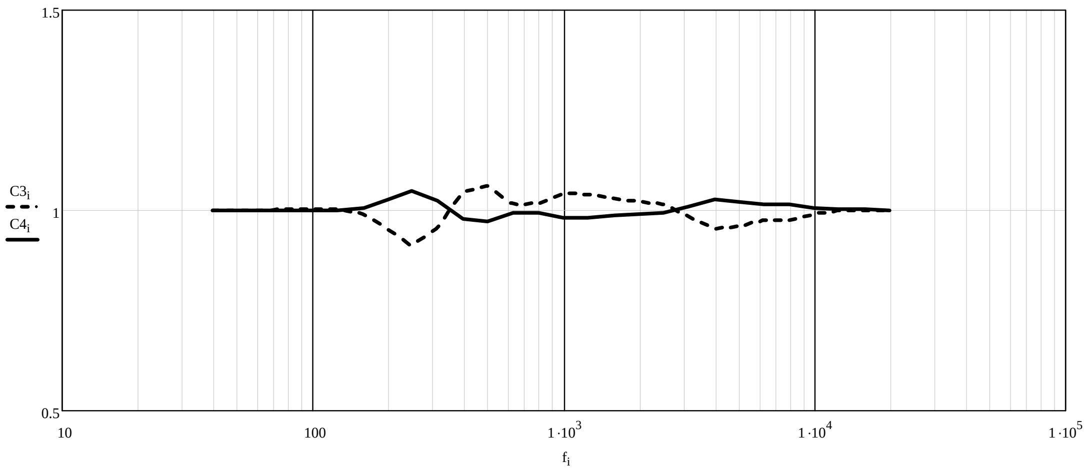

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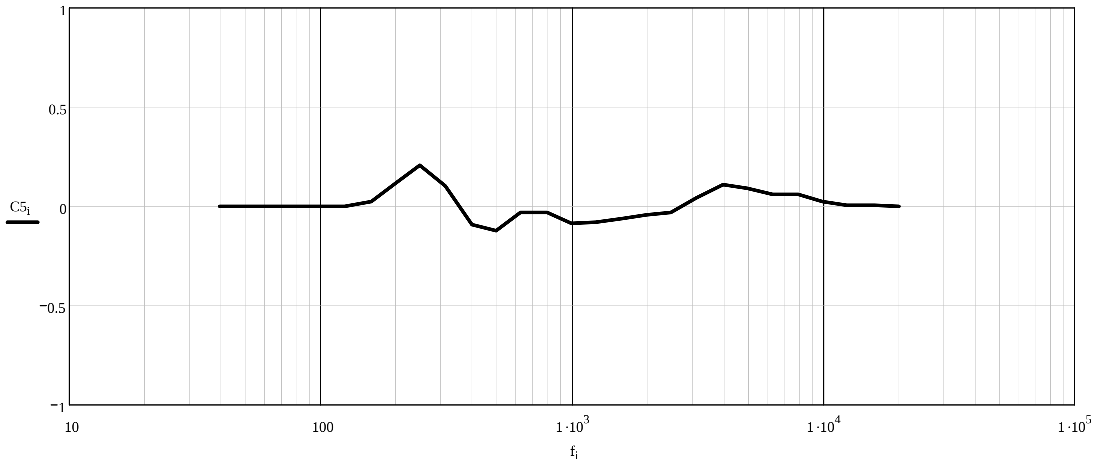

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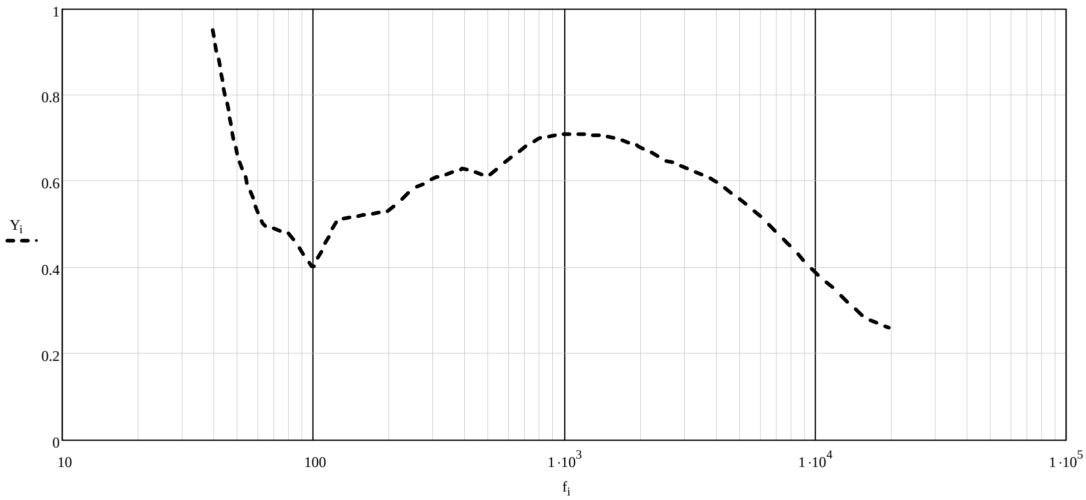

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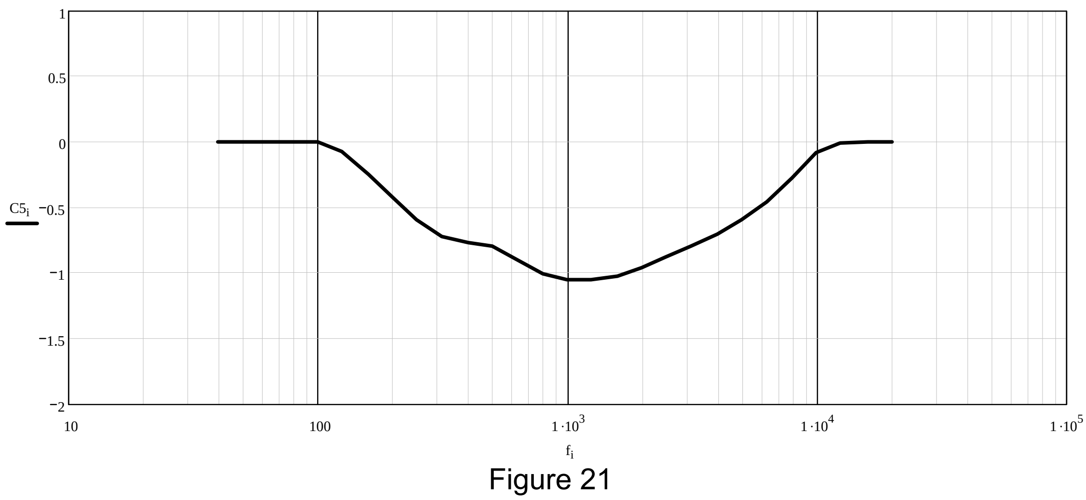

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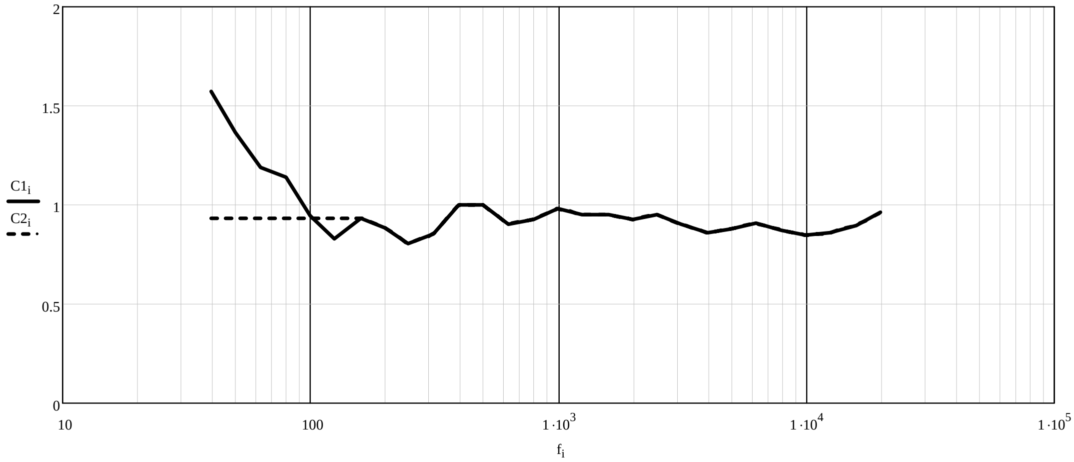

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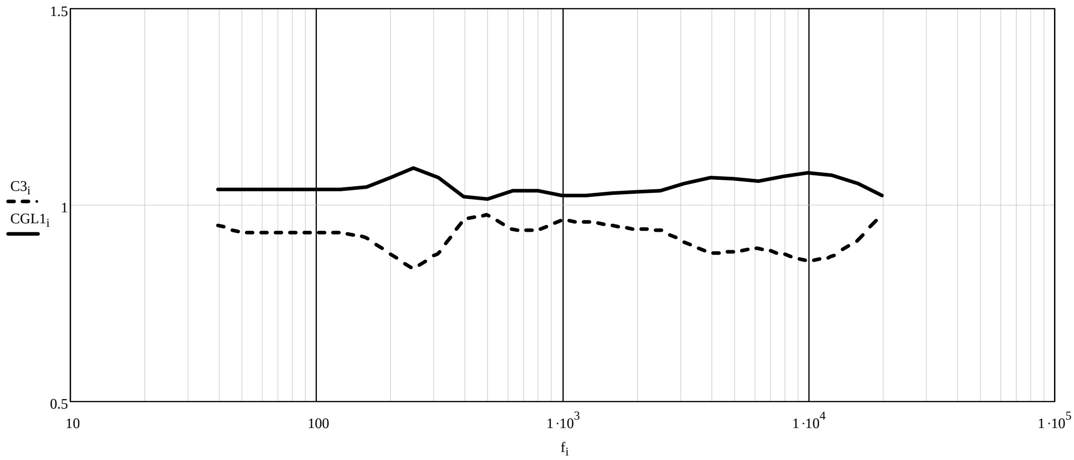

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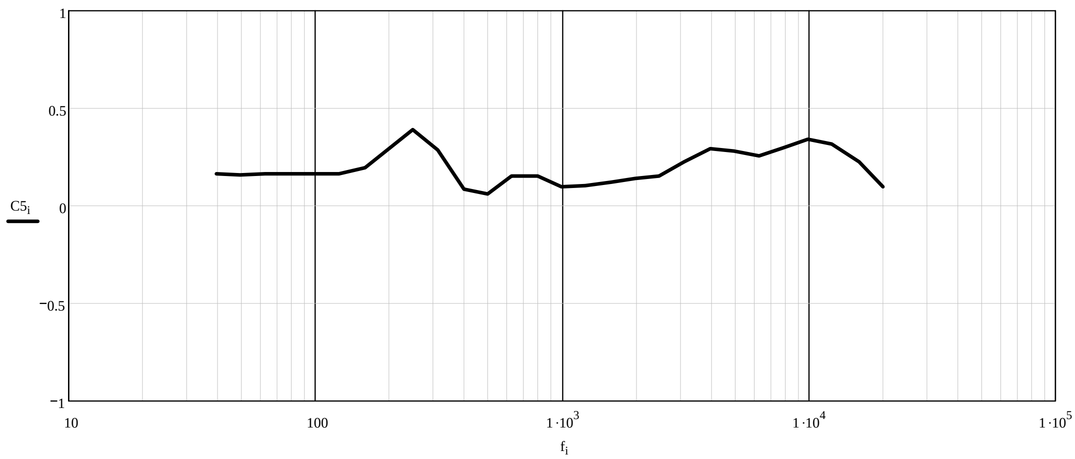

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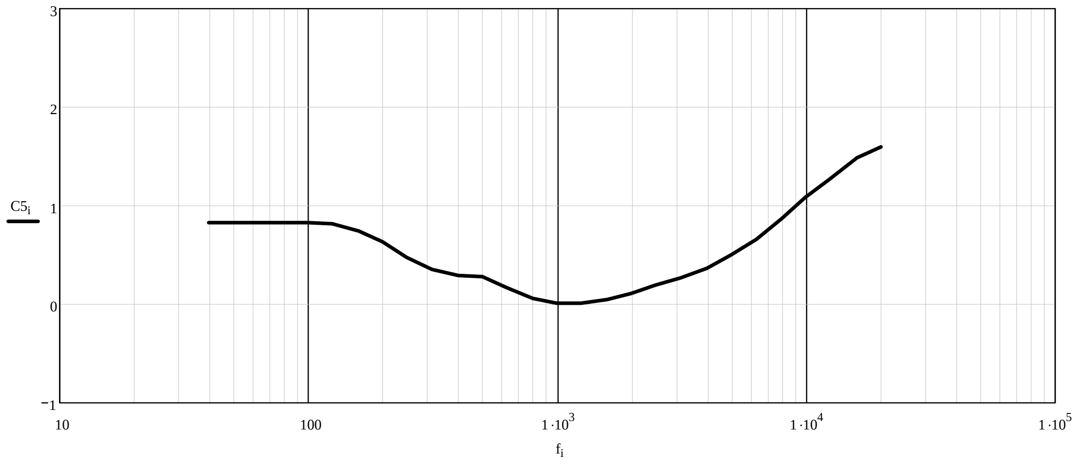

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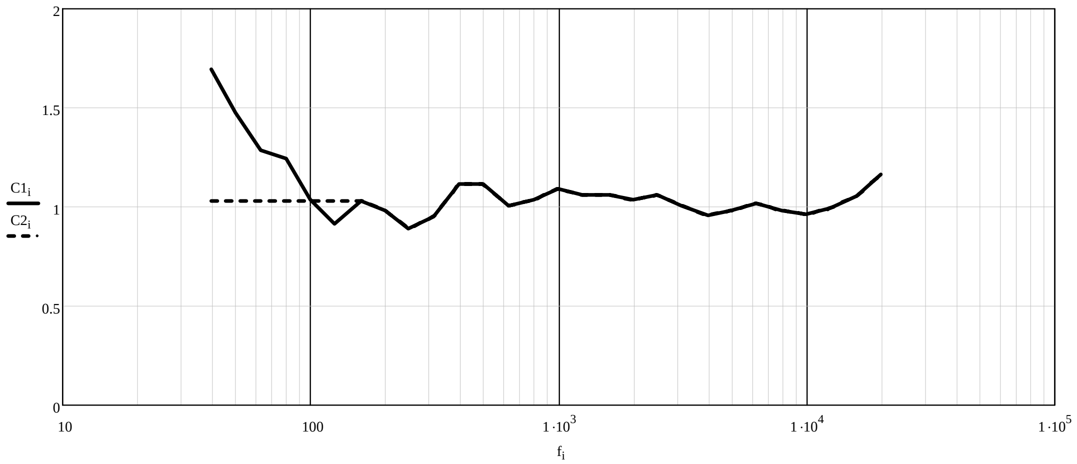

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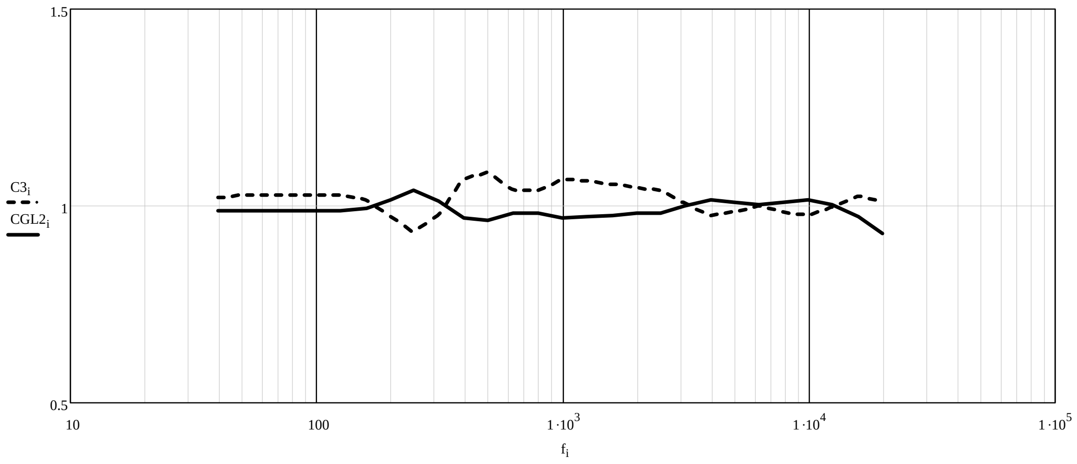

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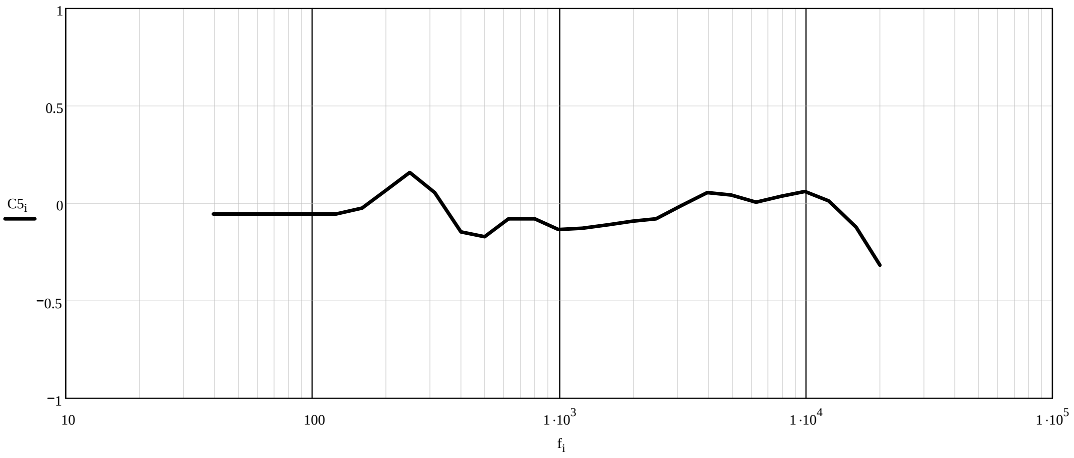

Figure 28

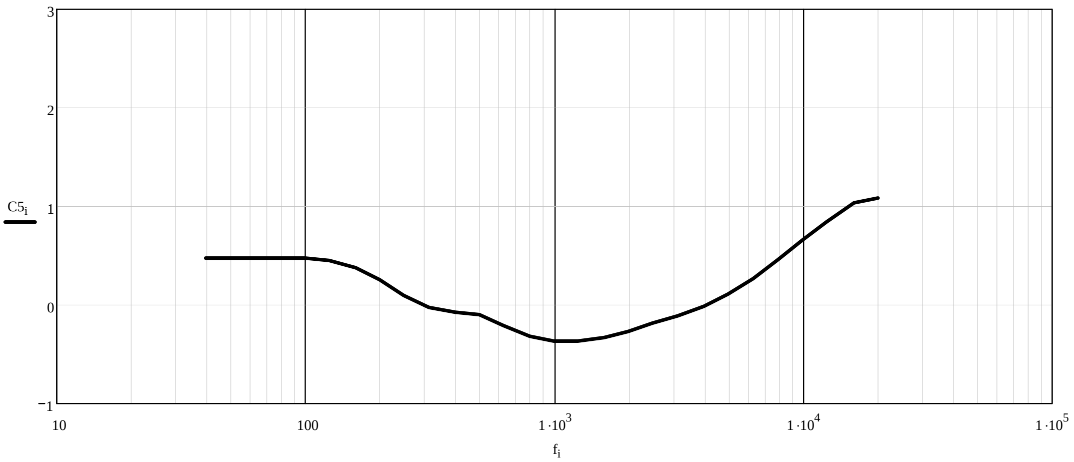

Figure 29