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Description

The present invention relates to a method for compressing the frequency of an audio signal having a fundamental frequency and at least one harmonic by providing the audio signal in a plurality of frequency channels and shifting or mapping the harmonic of the audio signal from a first frequency channel of the plurality of frequency channels into a second frequency channel of the plurality of frequency channels. In addition, a corresponding device for frequency compression is described. A device of said kind can be used in particular in a hearing apparatus. In the present context a hearing apparatus is understood to mean any sound-emitting device that can be worn in or on the ear, in particular a hearing aid, a headset, headphones and the like.

Hearing aids are wearable hearing apparatuses which serve to provide hearing assistance to the hearing-impaired. In order to accommodate the multiplicity of individual requirements, hearing aids are provided in different designs, including behind-the-ear (BTE) hearing aids, hearing aids with external earpiece (RIC: Receiver In the Canal) and in-the-ear (ITE) hearing aids, e.g. including concha hearing aids or canal (ITE, CIC) hearing aids. The hearing aids cited by way of example are worn on the outer ear or in the auditory canal. In addition, however, bone conduction hearing aids and implantable or vibrotactile hearing aids are also commercially available. With these devices the damaged hearing is stimulated either mechanically or electrically.

Basically, hearing aids have as their main components an input transducer, an amplifier and an output transducer. The input transducer is generally a sound receiver, e.g. a microphone, and/or an electromagnetic receiver, e.g. an induction coil. The output transducer is mostly realised as an electroacoustic transducer, e.g. a miniature loudspeaker, or as an electromechanical transducer, e.g. a bone conduction earpiece. The amplifier is typically integrated into a signal processing unit. This basic layout is shown in FIG 1 taking a behind-the-ear hearing aid as an example. A hearing aid housing 1 that is designed to be worn behind the ear has incorporated into it one or more microphones 2 for recording ambient sound.

A signal processing unit 3 which is also integrated into the hearing aid housing 1 processes the microphone signals and amplifies them. The output signal from the signal processing unit 3 is transmitted to a loudspeaker or earpiece 4 which emits an acoustic signal. The sound is transmitted to the hearing aid wearer's eardrum, where appropriate by way of a sound tube that is fixed in the auditory canal by means of an earmould. The hearing aid and in particular the signal processing unit 3 are supplied with power by means of a battery 5 that is likewise integrated into the hearing aid housing 1.

Many forms of hearing loss can be compensated by means of frequency-dependent amplification in combination with dynamic compression. There are, however, forms of hearing loss in which amplification has no effect or is disadvantageous. An example of this are forms of hearing loss characterised by so-called "dead regions". Dead regions are frequency ranges in which it is no longer possible to make spectral components audible by means of amplification.

A possible technique for dealing with the above problem is frequency compression. With this approach spectral components from a source frequency range which typically lies at higher frequencies and in which no amplification is to be applied (e.g. dead region) are shifted into a lower-lying target frequency range. In said target frequency range audibility is usually guaranteed in principle, for which reason an amplification can be applied.

Hearing aids are known which support frequency compression of this kind. In the compression method the properties of a filter bank, for example, are used for a simple implementation. Individual channels are selectively copied, inter alia as a function of their instantaneous power, onto other channels so that the frequency components contained in these channels reappear, shifted at the output, in a different frequency range. An adjustable mapping rule determines where the channels are mapped to, with the result that different compression ratios can be realised.

FIG 2 shows the principle of frequency compression by simple copying of channels, a technique already used for hearing aids. For example, the channel 14' (characterised by its mid-band frequency 14) is copied or shifted onto the channel 11' (characterised by its mid-band frequency 11). Located in the channel 14' is a tone 14'' (e.g. a harmonic) which is shifted onto the tone 11'' in the target channel 11'. The distance of the tone 14'' from the mid-band frequency 14 is identical to the distance of the tone 11'' from the mid-band frequency 11.

This simple mapping rule is attended by problems in relation to harmonic signals. Harmonic signals occur e.g. in voiced sounds in speech, in vowels for example. In this case the uncompressed spectrum has a linear-like structure, with spectral lines occurring at the voice fundamental frequency and at its integral multiples. With the simple mapping rule according to the prior art, the pattern of the harmonic signals (line structure) is not taken into account and is therefore destroyed, i.e. the spectral lines are no longer guaranteed to occur on an integral multiple of the voice fundamental frequency. This expresses itself in clearly discernible artefacts (signal components which occur at integral multiples of the fundamental frequency are referred to in the present context as "harmonic" for short).

A device and a method for proportional audio compression are known from publication US 6 577 739 B1. For compression the input spectrum is multiplied by a factor of 0.5, 0.7 or 0.9 for example, so that a proportionality of the spectral peaks is retained.

In addition publication WO 2009/143898 A1 discloses a method for adapting a sound in a hearing device by frequency modification. Here too the modification is undertaken for example by multiplying the input spectrum by a compression factor. A shift might additionally be performed. The modification can in addition also be made logarithmically.

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The object of the present invention therefore consists in further reducing artefacts occurring during frequency compression.

This object is achieved according to the invention by means of a method for compressing the frequency of an audio signal having a fundamental frequency and at least one harmonic according to claim 1.

- 5 A harmonic correction is performed during or after the shifting or mapping of the harmonic into another frequency channel. This means that the harmonic is placed onto a frequency position which likewise represents an integral multiple of the fundamental frequency. Even after the shift the harmonic therefore still represents a harmonic. This reduces the artefacts significantly.

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- The first frequency channel is shifted completely into the second frequency channel. This enables for example a frequency channel from a dead region to be shifted into an audible range of a hearing aid wearer. If a harmonic is present in the first frequency channel, it will be shifted completely with the frequency channel. In the process its distance from the mid-band frequency of the channel remains initially unchanged.

- A second frequency assigned to the harmonic that is shifted with the frequency channel is estimated and the shifted harmonic is then shifted further onto the first frequency in the second frequency channel. This means that the shifting takes place in two steps. First the entire frequency channel is shifted and then the original harmonic is shifted again within the frequency channel onto a harmonic frequency position.

- 25 The further shifting onto the first frequency in the second shifting step is effected by means of amplitude modulation. This can be realised in the time domain by means of a simple multiplication by a factor $\exp(j \cdot \omega \cdot t)$.

- The harmonic in the first frequency channel preferably represents a dominant frequency. This allows its position before and after shifting to be estimated relatively accurately.

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The present invention is explained in more detail with reference to the attached drawings, in which:

FIG 1 shows the basic design of a hearing aid according to the prior art;

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FIG 2 shows the principle of frequency compression by simple copying of channels according to the prior art;

FIG 3 shows an example of compression according to the prior art;

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FIG 4 shows an example of compression according to the present invention; and

FIG 5 shows a section of an uncompressed spectrum and a section of a compressed spectrum.

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For a better understanding of the invention, however, frequency compression according to the prior art will first be explained in detail with reference to FIG 3. According thereto, frequencies conforming to a frequency mapping curve (e.g. SPINC, BARK, etc.) are compressed. The starting point is for example a line spectrum, as represented in the top part of FIG 3. The amplitude response α is plotted against the frequency f . The line spectrum has numerous harmonics 20 that form the spectral fine structure of the harmonic signal. The amplitudes of the harmonics 20 can be combined by means of a spectral envelope 21. The spacing f_0 between two harmonics 20 corresponds to the fundamental frequency in the entire spectral range. The aim is now to compress the spectrum above a frequency f_c . The compression is carried out channel by channel in that selected channels of the original spectrum are copied into lower-lying channels. However, the channels generally have a different bandwidth than the spacing f_0 between 25 the harmonics. As a result thereof, in the course of the shifting, the harmonics 20 land on frequency positions outside the line pattern shown in the top part of FIG 3. The bottom part of FIG 3 shows a compressed spectrum of said type. The 30

spacings f_1 , f_2 between the individual lines 22 which represent the shifted harmonics are no longer constant and in particular are not equal to f_0 . Although in the compressed range the envelope 23 of the compressed spectrum shows the shifted formants 24 and 25, as they appear from the original spectrum, the distance between the lines 22 is not uniform, so as a result thereof the spectral fine structure and hence the structure of the harmonic signal are destroyed. Corresponding artefacts are the consequence.

A significant improvement in particular for voice signals can be achieved if a harmonic correction is performed in addition to the simple mapping rule according to the prior art, as is explained in more detail with reference to FIG 4. In the top part of the figure the original spectrum with its harmonics 20 and the envelope 21 is shown once again as in the top part of FIG 3. Over the entire original spectrum the spacing of the individual harmonics 20 corresponds to the fundamental frequency f_0 .

The object sought to be achieved by means of the invention is shown in an exemplary manner in the bottom part of FIG 4. The spectrum is compressed above the cutoff frequency f_c . The envelope 23 of the compressed spectrum possesses the same shape as that shown in the bottom part of FIG 3. In other words the formants 24 and 25 can also be identified in the compressed range. The lines 26 of the spectrum in the compressed range above f_c have the same spacing f_0 relative to one another as the lines or harmonics 20 in the uncompressed range. This means that the fine structure of the spectrum of the harmonic signal is untouched by the compression. Accordingly fewer artefacts are generated.

For the purpose of frequency compression with harmonic correction the frequency structure of the harmonic pattern of the uncompressed signal is first estimated, i.e. the positions of the harmonics in the frequency range are determined. This shall be explained in more detail with reference to FIG 5, which again shows a section of an uncompressed spectrum above and a section of a compressed spectrum below. In this case the section of the spectrum shown has a line or harmonic 30. This lies in a frequency channel 31 which for its part has a mid-

band frequency f_{31} . Located below the first frequency channel 31 is a second frequency channel 32 which has the mid-band frequency f_{32} . For compression purposes the first frequency channel 31 is now shifted, copied or mapped onto the second frequency channel 32. This represents a first step 33 in the frequency compression. Said step 33 corresponds to the prior art compression as shown in FIG 3. According thereto the harmonic 30 of the first frequency channel 31 is shifted onto the line 34 to which a frequency f_{34} is assigned (henceforth also referred to as the second frequency). The distance Δf between the frequencies f_{31} and f_{30} is identical to the distance between the frequencies f_{32} and f_{34} . However, the frequency f_{34} does not correspond to a harmonic of the fundamental frequency. Rather, a harmonic would lie at the frequency position f_{35} in the second frequency channel 32. This can be determined for example by means of a first frequency estimation in the target frequency range, i.e. in the second frequency channel 32 onto which the first frequency channel 31 is mapped or shifted. The line 34 must therefore be shifted onto the frequency f_{35} in order to obtain the fine structure of the harmonic signal. To that end the frequency structure of the still uncorrected compressed spectral components is estimated in a second estimation. In the simplified example of FIG 5, in which only one channel is shifted, the frequency f_{34} of the line 34 is therefore estimated or determined after the shift in the first step 33. The frequency offset, i.e. the distance between the frequencies f_{34} and f_{35} , can be determined from the two frequency estimations. The offset is compensated for with the aid of a modulation in a second step 36, wherein the harmonic pattern is restored. In this case the line 34 is shifted onto the frequency f_{35} , producing the line 35 as a result.

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The modulation can be achieved for example on the basis of the analytical signal through multiplication by a suitable complex twiddle factor. Thus, the shift by an angular frequency ω_1 corresponds to a multiplication by the factor $\exp(j \cdot \omega_1 \cdot t)$. The resulting modulation corresponds to an amplitude modulation.

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This method can advantageously be used in the case of a polyphase filter bank which only generates the complex-valued analytical signal (only positive frequency component of a Fourier transform) in the channels. With this approach, by

means of modulation using the modulation term $\exp(j \cdot \omega_1 \cdot t)$, each channel can be modulated cyclically, with the result that the frequency components are shifted therein correspondingly cyclically by the angular frequency ω_1 .

5 Basically, two cases need to be distinguished in the estimation of the (dominant) frequency:

- 1) A dominant frequency exists which can be readily estimated, i.e. a strong tonal component exists in this channel. This enables a good correction of
10 the harmonic pattern to be achieved.
- 2) No dominant frequency exists, i.e. the signal in the channel is noise-like. The frequency estimation leads to a more or less random instantaneous frequency. During mapping onto a target frequency this leads in turn to a phase randomisation or random modulation in the channel, which in the
15 case of noise-like channels has scarcely any effect on the hearing impression.

The exemplary embodiment described above is based on the assumption that the harmonic 30 is actually shifted as a signal component of the audio signal. According to an example, which is not part of the invention, the compressed spectral components are generated half-synthetically. The information relating to the frequency position of the half-synthetically generated spectral components is acquired from the estimation of the uncompressed harmonic structure, i.e. the frequency 35 is determined as in the above example. However, a synthetic signal is
25 now generated at the frequency f_{35} . The amplitude of said synthetic signal is adjusted such that it corresponds to the amplitude of the original harmonic 30, i.e. the associated amplitude is obtained from the source spectrum. By this means, too, a frequency compression can be achieved in which the harmonic pattern is preserved.

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The source frequency to target frequency mapping rule for frequency compression is applied in the known manner in audiology. The harmonic correction or, as the case may be, the preservation of the harmonic structure of the compressed

spectral components is then achieved according to the invention. As a result the artefacts of the simple mapping rule according to the prior art are substantially reduced.

Patentkrav

1. Fremgangsmåde til frekvenskompression af et audiosignal, som har en grundfrekvens og mindst en harmonisk (20, 30), ved

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- til rådighedstillelse af audiosignalet i flere frekvenskanaler (31, 32) og
- forskydning eller afbildning af den harmoniske (20, 30) af audiosignalet fra en første frekvenskanal (31) ud af de flere frekvenskanaler i en anden frekvenskanal (32) ud af de flere frekvenskanaler,

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kendetegnet ved

- vurdering af en første frekvens (f_{35}), som ligeledes er harmonisk til grundfrekvensen, i den anden frekvenskanal (32), hvorved
- 15 - den harmoniske (20, 30) forskydes eller afbildes på den vurderede første frekvens (f_{35}),
- hvorved den første frekvenskanal (31) forskydes fuldstændigt ind i den anden frekvenskanal (32),
- hvorved en anden frekvens (f_{34}), som er tilknyttet den forskudte harmoniske,
- 20 vurderes, og den forskudte harmonisk (20, 30) forskydes videre til den første frekvens (f_{35}) i den anden frekvenskanal (32), og
- hvorved videreforskydningen til den første frekvens (f_{35}) sker ved amplitudemodulation.

FIG 1
Kendt Teknik

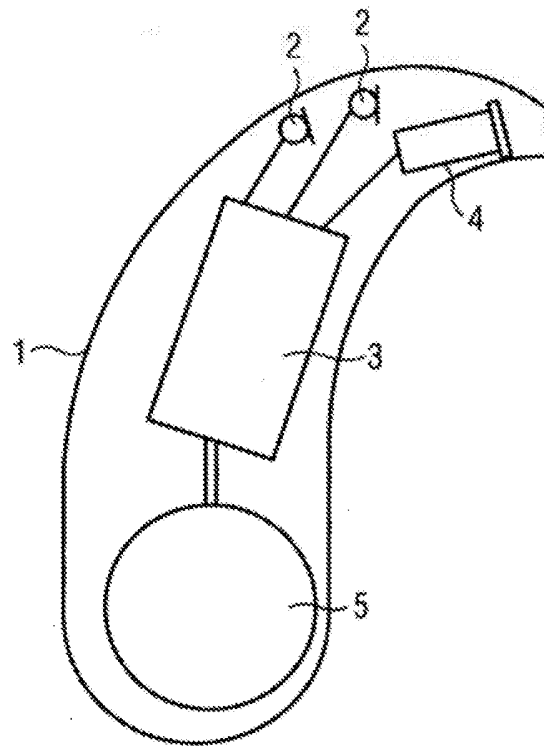
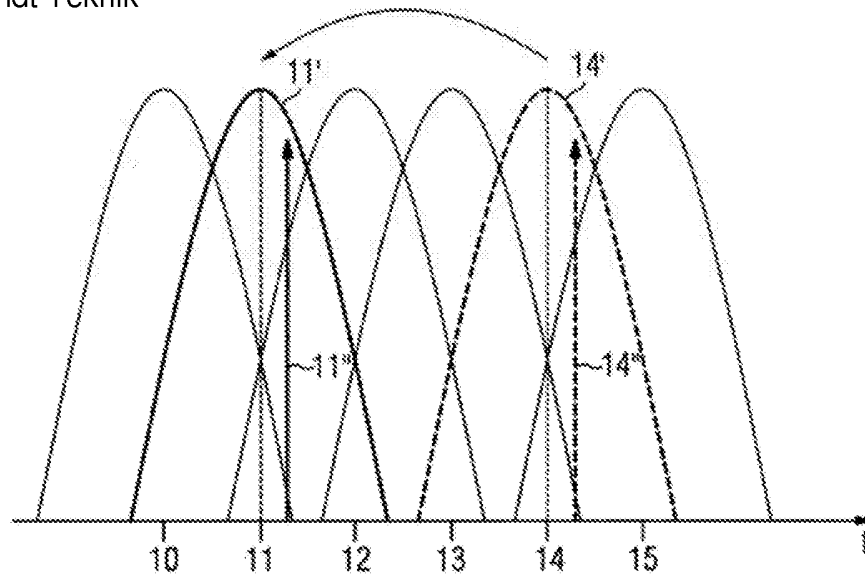


FIG 2

Kendt Teknik

**FIG 3**

Kendt Teknik

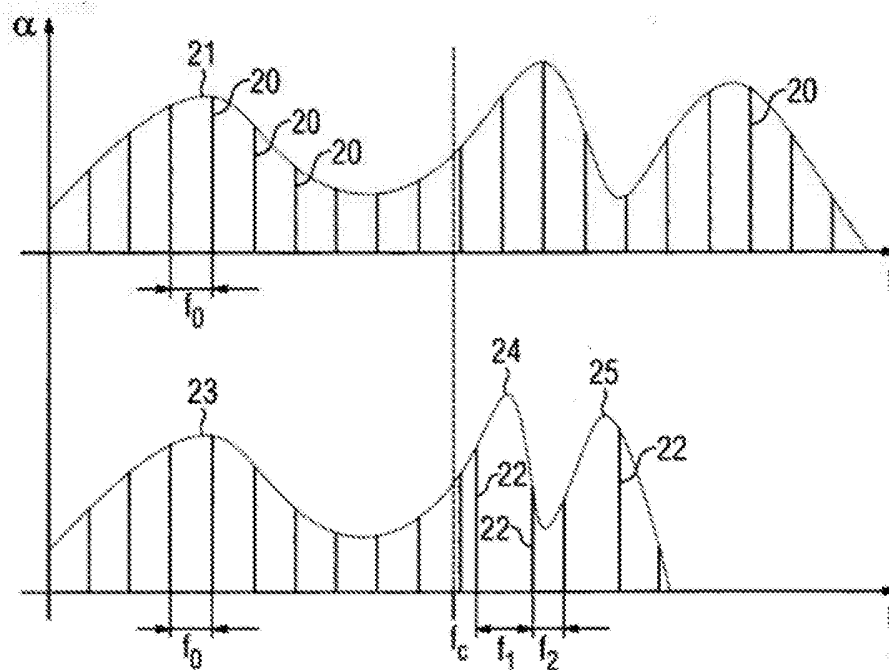


FIG 4

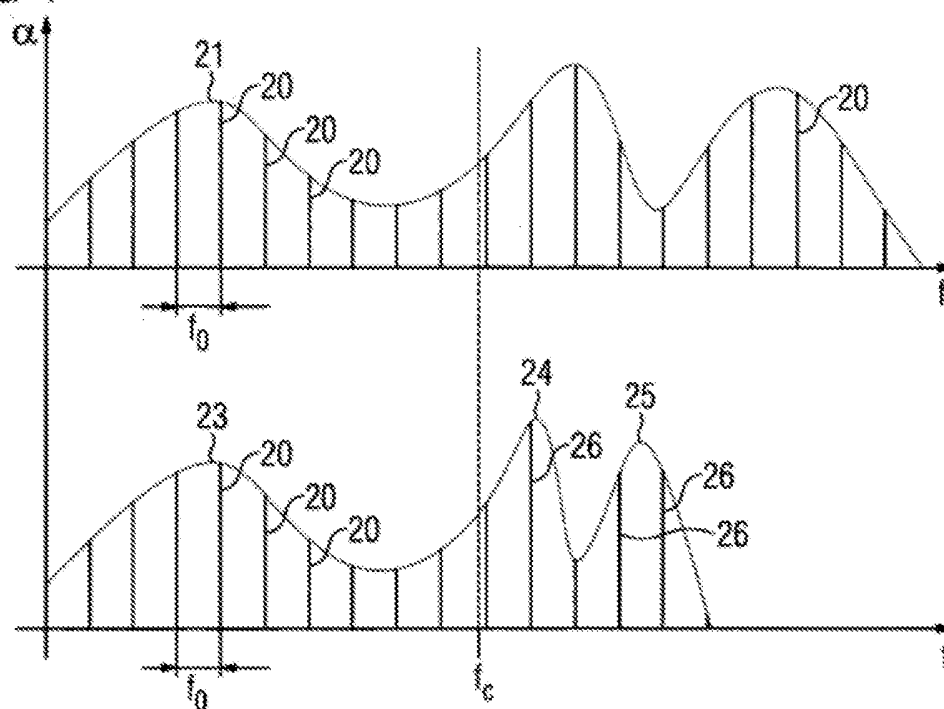


FIG 5

