

HEARING AID WITH AUDIO CODEC AND METHOD

The present invention relates to hearing aids. More specifically, it relates to a hearing aid having a time-domain audio codec for decoding and encoding digital audio signals. The invention further relates to a method of decoding and encoding audio signals.

A hearing aid is embodied as a small, wearable unit comprising one or more microphones, a signal processor, and means for acoustically reproducing sound signals. A hearing aid may additionally comprise means for receiving, processing and reproducing sound signals from other sources, such as a telecoil or an FM receiver. In order to alleviate a hearing loss of a user, the signal processor of the hearing aid is configured to amplify selected frequency bands based on a prerecorded audiogram of the user's hearing loss. For flexibility reasons, the signal processor is preferably a digital signal processor.

Modern day hearing aids are typically equipped with means for one- or two-way wireless communication, i.e. radio communication. Such wireless communication may carry sound signals, such as speech, suitable for being transmitted to and from the hearing aid in a digital form, e.g. between two hearing aids or between a hearing aid and another device. In such radio communication, there is a desire for keeping the transmission bit rate as low as possible, one of the reasons for this being that an increase in bandwidth of a radio communication leads to an increased power consumption, which, in turn, is undesired in a hearing aid.

One way to reduce the bit rate in a digital audio signal is to encode and decode the signals using an encoder/decoder unit or processor, commonly referred to as a codec, implemented as a combination of software and more or less dedicated hardware. However, such reduction of the bit rate comes at a cost, in terms of audio bandwidth, reproduction quality, computational complexity and delay.

One attempt to reduce the bandwidth and the delay time is described in the article: 'A Low-Delay CELP Coder for the CCITT 16 kb/s

Speech Coding Standard', Juin-Hwey Chen et al, IEEE JOURNAL ON SELECTED AREAS IN COMMUNICATIONS, Vol. 10, No. 5, June 1992. The audio bandwidth, reproduction quality and computational complexity described in that article, however, do not meet the needs in a hearing aid.

According to one aspect, there is provided a hearing aid comprising an audio codec, said codec being a time domain codec comprising a decoder adapted to generate a decoded output signal based on an input quantization index and comprising a predictor and a predictor adaptation, an encoder for generating an output quantization index based on a digital audio input signal, said encoder comprising said decoder and a predictor receiving an excitation signal and outputting a prediction signal, wherein the output quantization index is determined by repeated decoding of the quantization indices in a trial-and-error process yielding a number of different quantization indices in order to minimize the error between the digital audio input signal and the prediction signal in a first difference node, and selecting from said number of different quantization indices the quantization index yielding the least error in the difference node as the output quantization index, wherein said predictor adaptation uses a recursive autocorrelation estimate, basing its prediction on previous excitation signals corresponding to the previous output quantization indices, for updating the predictor, characterized in that the encoder incorporates the decoder.

According to another aspect, there is provided a method of encoding an audio signal in a hearing aid according to the above aspect, said method incorporating the steps of providing a digital audio input signal, providing said decoder adapted to generate a decoded output signal based on an input quantization index and comprising a predictor and a predictor adaptation, providing said encoder incorporating the decoder and said predictor adapted for receiving an excitation signal and outputting a prediction signal, generating an output quantization index based on the digital audio input signal, using said encoder, determining the output quantization index by repeated decoding of quantization indices in a trial-and-error process yielding a number of different quantization indices in order to minimize the error between the digital audio input signal and the prediction signal in a first difference node, and selecting from said number of different quantization indices the quantization index yielding the least error in the

difference node as the output quantization index, and updating said predictor using a recursive autocorrelation estimate basing its prediction on previous excitation signals corresponding to the previous output quantization indices.

It is the object of the present invention to provide a hearing aid having a codec which overcomes the bandwidth problems mentioned above while keeping the computational complexity low and still achieving an acceptable reproduction quality.

According to the invention this object is achieved by a hearing aid, in one embodiment, comprising an audio codec, said codec being a time domain codec comprising a decoder adapted to generate a decoded output signal based on an input quantization index and comprising a predictor and a predictor adaptation, an encoder for generating an output quantization index based on an input signal, said encoder comprising said decoder and said predictor receiving an excitation signal and outputting a prediction signal, wherein the output quantization index is determined by repeated decoding of the quantization indices in order to minimize the error between the input signal and the prediction signal, and wherein said predictor uses a recursive autocorrelation estimate for the error minimization.

By implementing such a codec in a hearing aid, the above criteria as to bandwidth and signal quality may be fulfilled while keeping complexity relatively low due to the fact that the operations necessary for the decoding are similar to those necessary for the encoding. Thus, large parts of the hardware, as implemented on a processing circuit chip, i.e. the dedicated processing parts of the chip used for either encoding or decoding, as the case may be, may be reused. This in turn saves physical space on the chip, as compared to designs having dedicated encoding units and decoding units, thus leading to an overall saving of space in the hearing aid.

According to a preferred embodiment of the invention, the codec comprises means for selectively switching between a scalar quantization mode, and a vector quantization mode.

In the scalar quantization mode, the signal is synthesized from a scalar in a codebook representing the signal shape. In the vector quantization mode, the signal is synthesized from a vector in a codebook representing e.g. a signal shape, a signal gain, and a signal sign.

5 Having means for operating in one of two different quantization modes, including means for selectively switching between these modes, allows for flexible utilization of the bandwidth during use, e.g. the use of the available bandwidth for the transmission of a mono signal in the scalar quantization mode, or the use of the available bandwidth for the
10 transmission of e.g. a stereo-encoded signal in the vector quantization mode.

 According to a further preferred embodiment of the invention, the hearing aid comprises a memory adapted for storing at least one predetermined sequence of quantization indices, and means for feeding
15 at least one such sequence to the codec.

 This feature allows the codec to be used not only for reproducing audio signals from a data stream received from an external device, e.g. a corresponding hearing aid, or a dedicated streaming device, but also for selectively switching the codec between a streaming mode and a
20 playback mode in order to play back sounds such as predetermined messages based on a sequence of quantization indices stored in a memory in the hearing aid. Storing a sequence of quantization indices rather than a sampled signal enables the signal to be reconstructed from the sequence of quantization indices when read out to the codec, thus saving
25 valuable space in the hearing aid memory.

 In a further preferred embodiment of the invention, the encoder comprises a codebook comprising a plurality of quantization indices, where said quantization indices are arranged in a searchable manner in a tree structure with a number of quantization indices arranged in different
30 branches, and where each individual quantization index is unique to a specific branch. This allows the codebook to be searched in a fast and efficient manner based on classified quantization indices, when repeatedly searching through the codebook in search of the optimum quantiza-

tion index.

In another preferred embodiment of the invention, the encoder comprises a codebook comprising a plurality of quantization indices, where said quantization indices are arranged in a searchable manner in
5 a tree structure with a number of quantization indices arranged in different branches, and where at least one individual quantization index is found in more than one branch. By overpopulating the search branches with quantization indices from other branches, i.e. other classes, the precision in finding the optimum quantization index, may be greatly improved at very little extra computational complexity.
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In an alternative embodiment, the encoder comprises a computing device adapted to calculate quantization indices directly from the input signal and the prediction signal. Calculating the indices in the codebook rather than simply having them tabulated and looking them up,
15 eliminates the need for memory capacity for a codebook in the hearing aid.

According to yet to another preferred embodiment of the invention, the said decoder comprises a shape codebook and a gain codebook, respectively, for providing a quantization vector representing a shape
20 value and a gain value, respectively. This embodiment enables the shape values in the codebook to be normalized, and utilizes gain values from the gain codebook to scale the normalized, synthesized output signal properly.

In a particularly preferred embodiment of the invention, said
25 gain adaptor is a backward-adaptive gain adaptor. This allows the gain adaptor to adapt to the overall dynamics of the sound signal.

In another preferred embodiment of the invention, the predictor is adapted using a recursive autocorrelation estimate based on a second- or higher-order autocorrelation model. This has the advantage that little
30 memory capacity is needed to store historical values as compared to correlation models involving a non-recursive part. Moreover, the computational complexity is significantly reduced.

In a particular embodiment of the invention, the hearing aid

comprises a sample rate converter for altering the sample rate of an audio signal prior to being encoded by the codec. This enables the encoder of the codec to operate on a signal with a sample rate different from the sample rate employed in the hearing aid signal processor. Thus a further
5 reduction in bandwidth requirement for the wireless link may be obtained if the sample rate of the coded signal is less than the sample rate of the hearing aid processor. The conversion from the sample rate of the hearing aid to the sample rate of the codec is performed prior to encoding the signal as a part of the pre-processing, and the conversion from
10 the sample rate of the codec to the sample rate of the hearing aid is performed after decoding as a part of the post-processing before the decoded signals are converted back into audio signals.

In a preferred embodiment of the invention, the hearing aid comprises means for detecting differences in clock frequencies between
15 the transmitter and receiver in the transmitted signal and means for modifying the decoded audio signal in order to compensate for the detected differences. This feature enables the receiving hearing aid to accommodate and compensate for the differences in clock frequencies between the received signal and the hearing aid in a way which is inaudible
20 to the wearer of the hearing aid.

In a more preferred embodiment of the invention, the means for detecting said differences in clock signal frequencies is a digital phase-locked loop (PLL). This embodiment enables an asynchronous conversion of the sample rate where the sample rate conversion factor is controlled
25 by said digital PLL. This simplifies reception of the signal, as no synchronization signals needs to be transmitted in order to get a correctly compensated clock frequency for the sample rate conversion.

In a further preferred embodiment of the invention the hearing aid comprises means for detecting, in the streaming mode of said codec,
30 in which a data stream is received via a streaming channel, bit errors in said data stream, means for estimating a bit error rate in said data stream, and means for fading the audio signal output from the codec. This allows the output signal from the codec to be faded rather than be-

ing abruptly disrupted, which would otherwise be disturbing to the user of the hearing aid.

In an particularly preferred embodiment the codec decoder comprises means for receiving the number of detected errors from the channel decoder and means for setting the excitation signal to the predictor to zero or the null-vector when errors are detected, zero representing the specific case of a one dimensional null-vector. This minimizes the effect of the transmission error on the predictor.

The invention also provides a method for encoding and decoding audio signals, said method incorporating the steps of providing a digital audio signal, providing a decoder adapted to generate a decoded output signal based on an input quantization index and comprising a predictor and a predictor adaptation, providing an encoder comprising a decoder and a predictor adapted for receiving an excitation signal and outputting a prediction signal, generating an output quantization index based on an input signal, using said encoder, determining the output quantization index by repeated decoding of quantization indices in order to minimize the error between the input signal and the prediction signal, and updating said predictor using a recursive autocorrelation estimate.

Preferred embodiments of the method are found in the dependent claims and provide advantages corresponding to those described above.

The invention will now be described, based on non-limiting exemplary embodiments and with reference to the drawings where:

fig. 1 schematically illustrates two hearing aids according to the invention and an external device,

fig. 2a and 2b show block diagrams of the functionality of the codec in either one of the hearing aids in fig. 1,

fig. 3 is a schematic diagram of a memory holding prerecorded indices,

fig. 4a shows a first example of a tree search,

fig. 4b shows a second example of a tree search,

fig. 5 is shows the gain fading as a function of the bit error rate,

and

fig. 6 shows a second-order recursive window used in the auto-correlation estimation.

Fig. 1 shows a first hearing aid 1a, a second hearing aid 1b and
5 an external device 2. The first hearing aid 1a is shown in a schematic form and the hearing aid 1b is suggested by its physical outline. Both hearing aids 1a and 1b are adapted to communicate with each other via a short range wireless radio communications link 3. Likewise they are adapted to communicate with an external unit 2 via the short range
10 wireless radio communications link 3.

The hearing aid 1a hearing aid comprises an antenna 51, a wireless transceiver 52, a hearing aid processor 50, a microphone 54, and an acoustic output transducer 55. The wireless transceiver 52 is capable of receiving and transmitting a digitally encoded signal. The hearing aid processor 50 comprises an audio signal processor 53, an input
15 channel decoder 56, an audio decoder 57, a post-processing block 58, an audio preprocessing block 59, an audio encoder 60 and an output channel encoder 61.

In reception mode, the audio signal processor 53 receives an
20 input signal from the microphone 54 and conditions and amplifies it for reproduction by the acoustic output transducer 55 according to a hearing aid prescription. When the antenna 51 receives a wireless signal, the transceiver 52 demodulates the received signal into a channel stream for further processing by the hearing aid processor 50.

25 The demodulated channel stream is used as the input for the input channel decoder 56 of the hearing aid processor 50, where the channel stream is decoded. The decoded channel stream is used as the input bit stream for the audio decoder 57. The audio decoder 57 decodes the bit stream by synthesizing the corresponding audio signals using the
30 codebook indices obtained from the bit stream and outputting a digital audio signal with a relatively low sample rate. The digital audio signal is used as the input for the post-processing block 58, where post-processing is performed on the digital audio signal. The post-processing

involves filtering, conditioning and asynchronous sample rate conversion into a digital audio signal having a relatively higher sample rate in order for the received signal to be compatible with the audio signal processing in the audio signal processor 53. In this way, the sample rate of the received audio signal may be lower than the sample rate in the hearing aid 1a, allowing for a more efficient transmission because fewer bits have to be transmitted via the wireless transceiver 52.

In transmission mode, the audio processor 53 prepares a digital audio signal for transmission by the wireless transceiver 52 in the following way: A digital audio signal is fed to the audio preprocessing block 59 where the digital audio signal is resampled and converted into a digital audio stream with a lower sample rate. The digital audio stream is encoded into a bit stream in the encoder 60. This bit stream comprises a sequence of codebook quantization indices representing the digital audio signal. The bit stream is used as input for the output channel encoder 61, where a channel stream is generated. The channel stream from the output channel encoder 61 is fed to the input of the wireless transceiver 52 for modulation, and transmitted wirelessly via the antenna 51.

The bandwidth of the short range wireless radio communications link 3 is limited because the power consumption of the radio circuit in the hearing aid 1 has to be kept down due to the limited power resources in a hearing aid. A typical bandwidth for wireless signals would be from 100 kbit/s to 400 kbit/s.

One purpose for which the short range wireless radio communications link 3 is used is streaming of audio signals, e.g. audio signals may be streamed from one hearing aid to another, i.e. from one side of the head to another, in what is referred to as Contralateral Routing of Signals, or CROS. Signals may also be streamed to a hearing aid from an external device 2, e.g. in order to transmit, via the external device 2, the audio from other sources, such as TV-sets, radios or the like.

Because of the limited bandwidth of the short-range wireless radio communications link 3 it is, however, necessary to compress the audio signals to be transmitted. The hearing aid 1 therefore comprises a

codec according to the invention. The codec is illustrated in Fig. 2a and Fig. 2b as an encoder and a decoder, respectively. However, as will be readily appreciated by comparison of Figs. 2a and 2b, and as explained in further detail below, the encoder incorporates the decoder. Thus, the hardware of the codec, i.e. the parts of the circuit chip on which the functionality of the codec is executed, may serve two purposes. This, in turn, means that the very same parts of hardware may constitute the hardware used with the encoding and decoding functionality, and redundancy of these parts of the chip is avoided. Valuable circuit chip space is thus saved in the hearing aid.

Fig. 2a is a block schematic showing an encoder according to the invention. The encoder comprises a first difference node 5, a filter adaptation block 6, a perceptual weighing block 7, a vector quantization block 8a, a scalar quantization block 8b, a codebook block 9, and a decoding sub-block 20. The decoding sub-block 20 comprises a gain adaptation block 10, an amplifier 12, a predictor block 4, and a predictor adaptation block 11.

A digital audio input signal enters the filter adaptation block 6 and the first difference node 5, and the output from the difference block 5 is fed either to the scalar quantization block 8b or to the input of the perceptual weighing block 7 for conditioning according to a perceptual weighing function. The perceptually weighted signal is then quantized into vectors in the vector quantization block 8a.

Depending on whether a scalar quantization or a vector quantization is used, the quantized vector or scalar indices, respectively, are fed to the corresponding input of the codebook block 9. The codebook block 9 outputs a shape index approximation and a gain index approximation from the indices to the decoding sub-block 20. In the decoding sub-block 20, a synthetic approximation of the instantaneous input signal is generated by repeatedly adapting the gain and the shape of the synthetic signal to the actual input signal. This approximation is performed by minimizing the error signal from the first difference node 5. Once the error signal is minimized, a vector quantization index or a sca-

lar quantization index, as the case may be, is output from the encoder for transmission.

As mentioned, error minimization is done by repeatedly comparing the input signal to a synthesized signal in a trial-and-error process yielding a number of different quantization indices as an output. Each of these different quantization indices is fed to the codebook 9. The output signal from the decoder sub-block 20 serves as an excitation signal for the predictor 4. At the end of the trial-and-error process, the quantization index yielding the least error in the difference node 5 is then selected as the output quantization index. The process is then performed repeatedly to provide a resulting output data stream suitable for transmission over the short range wireless radio communication link. This data stream is compressed as compared to the original sampled input signal as it is only necessary to transmit the quantization indices for the codebook 9. The gain adaptor 10 scales the signal from the codebook 9 and controls the amplifier 12 in order to provide an amplified, decoded output signal for the predictor 4.

The predictor 4 is controlled by the predictor adaptation block 11. The predictor adaptation block 11 is autorecursive, i.e. bases its prediction on previous excitation signals corresponding to the previous output quantization indices. Fig. 6 illustrates the weight applied to signal samples versus time in a window function as used in accordance with the present invention. The window function $W_m(n)$ being defined as:

$$W_m(n) = \begin{cases} 0 & \text{for } n \geq m \\ b_0 & \text{for } n = m-1 \\ -\sum_{k=1}^K a_k \cdot W_m(n-k) & \text{for } n < m-1 \end{cases}$$

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Window-weighted signal s_m at time m thus being:

$$s_m(n) = s(n) \cdot W_m(n)$$

Autocorrelation at time m for lag τ is:

$$R_m(\tau) = \sum_{n=-\infty}^{\infty} s_m(n) \cdot s_m(n-\tau)$$

- 5 Where R_m is used as an input for a Levinson-Durbin algorithm yielding the predictor adaptation coefficients.

For values larger than m , $W_m(n) = 0$ and consequently $s_m(n) = 0$. Causal autocorrelation at time m for lag τ is thus given by the formula:

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$$R_m(\tau) = \sum_{n=-\infty}^m s_m(n) \cdot s_m(n-\tau)$$

For the specific case of a second-order recursive window, the above formula reduces to:

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$$\begin{aligned} R_m(\tau) &= r_m(\tau) - (a_1 \cdot r_{m-1}(\tau) + a_2 \cdot r_{m-2}(\tau)), \text{ where} \\ r_m(\tau) &= s_m(m-1) \cdot s_m(m-1-\tau) \\ &= s(m-1) \cdot W_m(m-1) \cdot s(m-1-\tau) \cdot W_m(m-1-\tau) \end{aligned}$$

- 20 If the auto recursive window is based on frames rather than single samples the second-order autocorrelation window is given by:

$$R_m(\tau) = r_m(\tau) - (a_1 \cdot r_{m-L}(\tau) + a_2 \cdot r_{m-2L}(\tau))$$

- 25 Where

$$\begin{aligned} r_m(\tau) &= \sum_{l=1}^L s_m(m-l) \cdot s_m(m-l-\tau) \\ &= \sum_{l=1}^L s(m-l) \cdot W_m(m-l) \cdot s(m-l-\tau) \cdot W_m(m-l-\tau) \end{aligned}$$

and where L is the frame-length. The predictor 4 is only updated once for every frame, thus saving time.

In order to limit the number of vectors that have to be kept in the codebook and searched through within the available timeframe, the vector quantization codebook preferably holds only normalized vectors, i.e. vectors of a unit length. The normalized vectors must subsequently be multiplied by a suitable gain factor in order to provide the correctly scaled vector. In the gain multiplication node 12, the normalized vector output from the encoding codebook 9 is multiplied by the gain factor from the gain adaptation block 10 in order to yield the excitation signal for the predictor 4.

The gain factor derivation is preferably based on a separate gain codebook, yielding a separate gain index to be included in the output quantization index.

The excitation signal $X(t)$, which is presented to the predictor 4 thus follows the formula:

$$X(t) = (s(scbi_t) \cdot g(gcbi_t)) \cdot G(t)$$

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Where s is the normalized shape vector from the shape code book, g is the instantaneous gain from the separate gain codebook and G is the global gain factor.

As can be seen from Figs. 2a and 2b, the gain factor is also controlled adaptively by the gain adaptation block 10. When normalized gain indices are used, the gain adaption could e.g. follow the recursive formula:

$$G(t) = \alpha \cdot T_g(gcbi_t) + (1 - \alpha) \cdot G(t-1)$$

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Where G is the gain value, t is the current sample, t-1 is the previous sample, α is a decay factor, and $T_g(gcbi_t)$ is a mapping and/or function of

the gain values, g_{cbi} , in the gain codebook. By appropriate choice of α , the historical emphasis of the gain adaptation can be adjusted. The function T_g is preferably a non-linear function, such as the power of 3. This allows the gain values of the gain codebook to cover a wide dynamic
5 range though stored in only a few bits, thus three bits cover the range from 0 to 343, or 72 dB, rather than just the range from 0 to 7, or 26 dB.

As mentioned above, the available time for searching the codebook and trying out the resulting excitation signals is limited. It may
10 therefore be difficult or even impossible to search through all quantization vectors in the encoder codebook within a given timeframe. It is therefore preferred to classify the vectors in a tree structure and perform a tree search of first an appropriate class, and then the best quantization vector in that class. As illustrated in Fig. 4a, the $M \cdot N$ quantiza-
15 tion vectors V_{11} to V_{MN} have been arranged in classes C_1 to C_M . The maximum number of searches to be performed is hereby reduced from $M \cdot N$ to $M + N$.

However, classifying the vectors in this manner potentially excludes the best vector because it may actually be in a different class. If
20 sufficient time is available, this drawback may be mitigated if some redundancy is introduced in the classes, that is, some classes contain copies of vectors from other classes. This is illustrated in Fig. 4b, where the class C_1 has a copy of the element V_{21} from the class C_2 . Thus, unlike the codebook illustrated in Fig. 4a, where each individual quantization
25 index is unique to a specific branch of the searching tree, at least one individual quantization index, such as V_{21} , is found in more than one branch of the searching tree.

If the hearing aid, or the chip on which the codec hardware is realized, has sufficient processing power, it is possible to calculate the
30 quantization vector analytically as an alternative to looking up the vector in a codebook. Thus, instead of containing the vectors in a tabulated form, the codebook 9 stores a function calculating the vectors based on the input quantization index. This reduces the memory capacity neces-

sary to store the codebook.

Evidently, the skilled person will understand that the embodiment having a structured search tree codebook structure, the embodiment having a redundancy search tree codebook structure, and the embodiment having means for calculating the quantization vector analytically are preferred embodiments, but that an embodiment incorporating a full search in the encoding codebook 9 is not excluded.

As can be seen from Fig. 1, the hearing aid 1b may comprise a post-processing stage 58. The same is the case for the hearing aid 1b, but not visible in the figure. This post-processing stage 58 may comprise various kinds of post-processing, such as sample rate conversion, output fading and other post-filtering operations.

When operating in the streaming mode, the quality of the output data stream of indices received depends on the objective signal quality of the short range wireless radio communications link. If the signal received becomes too weak, or becomes disturbed by interfering radio signals or the like, the data stream of indices will contain more and more errors as the signal deteriorates. In order to avoid having the reproduced output signal breaking down in a disturbing manner due to the presence of too many errors in the data from the output data stream, the hearing aid comprises means for detecting errors in the output data stream received over the short range wireless radio communications link 3. If the error rate becomes higher than a predetermined error rate, the post-processing block 58 fades out the signal in a graceful manner, i.e. it turns down the output signal level over a short period of time. Thus, the potentially rather disturbing noise produced by other digital streaming signal systems when the error rate becomes too high, is avoided. Preferably, as illustrated in Fig. 5 this fading is performed by constantly measuring the bit error rate (BER) in the data stream and using the BER to control a gain reduction based on a hysteresis. Whenever the BER is above, say, 0.01 errors per bit, i.e. the signal quality is poor, the output gain is reduced to the low value G_0 . If the BER falls below 0.001 errors per bit, i.e. the signal quality is good, then the output gain is increased

to the nominal value G_n .

The channel encoder 61 for the streaming is preferably a Forward Error Correction code (FEC code). The FEC code error correction (ec) and detection capability (dc) is determined by the Hamming distance d_{min} , where the relationship $2*ec + dc < d_{min}$. From this relationship it is seen that detection is a simpler scheme. In this invention we may set the excitation signal, i.e. the input to the predictor 4 to zero or the null-vector whenever errors are detected. This has the effect that the transmission error has minimal influence on the predictor 4, because the erroneous input is not introduced. Furthermore, the gain is updated with a zero in the gain adaptation block 10, which results in the fading of the gain in case of consecutive transmission errors.

To obtain very low computational complexity a Hamming code is applied in the preferred embodiment Using e.g. Ham(24,18) having a Hamming distance of 4 hence allows the detection of up to two errors and in case of only one error the correction thereof.

P A T E N T C L A I M S

1. A hearing aid comprising an audio codec, said codec being a time domain codec comprising

a decoder adapted to generate a decoded output signal based on an input quantization index and comprising a predictor and a predictor adaptation,

an encoder for generating an output quantization index based on a digital audio input signal, said encoder comprising said decoder and a predictor receiving an excitation signal and outputting a prediction signal,

wherein the output quantization index is determined by repeated decoding of the quantization indices in a trial-and-error process yielding a number of different quantization indices in order to minimize the error between the digital audio input signal and the prediction signal in a first difference node, and selecting from said number of different quantization indices the quantization index yielding the least error in the difference node as the output quantization index,

wherein said predictor adaptation uses a recursive autocorrelation estimate, basing its prediction on previous excitation signals corresponding to the previous output quantization indices, for updating the predictor,

characterized in that the encoder incorporates the decoder.

2. A hearing aid according to claim 1, wherein the codec comprises means for selectively switching between a scalar quantization mode and a vector quantization mode.

3. A hearing aid according to claim 1 or 2, comprising a memory adapted for storing at least one predetermined sequence of quantization indices, and means for feeding at least one such sequence to the codec.

4. A hearing aid according to any one of the preceding claims, wherein said encoder comprises a codebook comprising a plurality of quantization indices, where said quantization indices are arranged in a searchable manner in a tree structure with a number of quantization in-

dices arranged in different branches, and where each individual quantization index is unique to a specific branch.

5. A hearing aid according to any one of the claims 1-3, wherein said encoder comprises a codebook comprising a plurality of quantization indices, where said quantization indices are arranged in a searchable manner in a tree structure with a number of quantization indices arranged in different branches, and where at least one individual quantization index is found in more than one branch.

6. A hearing aid according to any one of the claims 1-3, wherein said encoder comprises a computing device adapted to calculate quantization indices directly from the digital audio input signal and the prediction signal.

7. A hearing aid according to any one of the preceding claims, wherein said decoder comprises a shape codebook and a gain codebook, respectively, for providing a quantization vector representing a shape value and a gain value, respectively.

8. A hearing aid according to claim 7, wherein said gain adaptor is a backward adaptive gain adaptor.

9. A hearing aid according to claim 1, wherein said predictor is adapted using a recursive autocorrelation estimate based on a second- or higher-order autocorrelation model.

10. A hearing aid according to any one of the preceding claims, wherein the hearing aid comprises a sample rate converter for altering the sample rate of an audio signal prior to being encoded by the codec.

11. A hearing aid according to any one of the preceding claims, wherein the hearing aid comprises means for detecting differences in digital clock frequencies between the transmitter and receiver in the transmitted signal and means for modifying the decoded audio signal in order to compensate for the detected differences.

12. A hearing aid according to claim 11, wherein the means for detecting said differences in clock signal frequencies is a digital phase-locked loop (PLL).

13. A hearing aid according to claim 1, comprising means for

detecting in the streaming mode of said codec, in which a data stream is received via a streaming channel, bit errors in said data stream,

means for estimating a bit error rate in said data stream, and

means for fading the audio output from the codec.

14. A hearing aid according to claim 13, wherein the codec decoder comprises means for receiving the number of detected errors from the channel decoder and means for setting the excitation signal to the predictor to zero or the null-vector when uncorrectable errors are detected.

15. Method of encoding an audio signal in a hearing aid according to claim 1, said method incorporating the steps of

providing a digital audio input signal,

providing said decoder adapted to generate a decoded output signal based on an input quantization index and comprising a predictor and a predictor adaptation,

providing said encoder incorporating the decoder and said predictor adapted for receiving an excitation signal and outputting a prediction signal,

generating an output quantization index based on the digital audio input signal, using said encoder,

determining the output quantization index by repeated decoding of quantization indices in a trial-and-error process yielding a number of different quantization indices in order to minimize the error between the digital audio input signal and the prediction signal in a first difference node, and selecting from said number of different quantization indices the quantization index yielding the least error in the difference node as the output quantization index, and

updating said predictor using a recursive autocorrelation estimate basing its prediction on previous excitation signals corresponding to the previous output quantization indices.

16. Method according to claim 15, wherein said repeated decoding comprises the searching of a codebook comprising a plurality of quantization indices, where said quantization indices are arranged in a

searchable manner in a tree structure with a number of quantization indices arranged in different branches, and where each individual quantization index is unique to a specific branch.

17. Method according to claim 15, wherein said repeated decoding comprises the searching of a codebook comprising a plurality of quantization indices, where said quantization indices are arranged in a searchable manner in a tree structure with a number of quantization indices arranged in different branches, and where at least one individual quantization index is found in more than one branch.

10 18. Method according to claim 15, wherein said repeated decoding comprises calculating quantization indices directly from the digital audio input signal and the prediction signal using a computing device.

15 19. Method according to any one of claims 15 to 18, wherein a quantization vector representing a shape value and a gain value is provided using a shape codebook and a gain codebook, respectively.

20 20. A method according to claim any one of claims 15 to 19, wherein said predictor is a backward adaptive gain adaptor, which is adapted using a recursive autocorrelation estimate based on a second- or higher-order autocorrelation model.

21. A method according to claim 15, in which a data stream is received via a streaming channel, bit errors in said data stream are detected, and the audio output from the codec is faded upon detection of bit errors.

22. A method according to claim 21, wherein the number of detected errors from the channel decoder is provided to the codec and the excitation signal to the predictor is set to zero or the null-vector when uncorrectable errors are detected.