

## U.S. Patent No. 7,260,521

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**HD** HD Voice Capable™ - Clear conversations and reduced background noise.

Claim 1

<p>A pitch analysis device for producing a set of pitch codebook parameters, comprising:</p>	<p>The accused products comprise a pitch analysis device for producing a set of pitch codebook parameters.</p> <p>For example, the accused products practice the Adaptive Multi-Rate – Wideband (AMR-WB) speech codec standard. See 3GPP TS26.190 v10.0.0 (2011-03). Under this standard, the accused products comprise a pitch analysis device for producing a set of pitch codebook parameters.</p> <h3>4.3 Principles of the adaptive multi-rate wideband speech encoder</h3> <p>The AMR-WB codec consists of nine source codecs with bit-rates of 23.85, 23.05, 19.85, 18.25, 15.85, 14.25, 12.65, 8.85 and 6.60 kbit/s.</p> <p>The codec is based on the code-excited linear predictive (CELP) coding model. The input signal is pre-emphasized</p> <p>The CELP speech synthesis model is shown in Figure 1. In this model, the excitation signal at the input of the short-term LP synthesis filter is constructed by adding two excitation vectors from adaptive and fixed (innovative) codebooks. The speech is synthesized by feeding the two properly chosen vectors from these codebooks through the short-term synthesis filter. The optimum excitation sequence in a codebook is chosen using an analysis-by-synthesis search procedure in which the error between the original and synthesized speech is minimized according to a perceptually weighted distortion measure.</p> <h3>5.7 Adaptive codebook</h3> <p>Adaptive codebook search is performed on a subframe basis. It consists of performing closed loop pitch search, and then computing the adaptive codevector by interpolating the past excitation at the selected fractional pitch lag.</p> <p>The adaptive codebook parameters (or pitch parameters) are the delay and gain of the pitch filter. In the search stage, the excitation is extended by the LP residual to simplify the closed-loop search.</p>
<p>a pitch codebook search device configured to generate a pitch code vector based on a digitized input audio data, wherein said</p>	<p>The accused products comprise a pitch codebook search device configured to generate a pitch code vector based on a digitized input audio data, wherein said digitized input audio data represents an input audio signal that has been sampled and digitized.</p> <p>For example, the accused devices comprise a pitch codebook search device that is configured to generate a pitch code vector based on a digitized input audio data.</p>

<p>digitized input audio data represents an input audio signal that has been sampled and digitized;</p>	<div data-bbox="436 191 953 240"> <h2>5.7 Adaptive codebook</h2> </div> <div data-bbox="436 264 1667 329"> <p><u>Adaptive codebook search</u> is performed on a subframe basis. It consists of performing closed loop pitch search, and then computing <u>the adaptive codevector</u> by interpolating the past excitation at the selected fractional pitch lag.</p> </div> <div data-bbox="436 354 1673 418"> <p><u>The adaptive codebook parameters (or pitch parameters) are the delay and gain of the pitch filter.</u> In the search stage, the excitation is extended by the LP residual to simplify the closed-loop search.</p> </div> <div data-bbox="436 459 1627 495"> <p>The <u>digitized input audio signal</u> represents an input audio signal that has been sampled and digitized.</p> </div> <div data-bbox="445 535 1337 587"> <h2>4.1 Functional description of audio parts</h2> </div> <div data-bbox="445 612 1686 647"> <p>The analogue-to-digital and digital-to-analogue conversion will in principle comprise the following elements:</p> </div> <div data-bbox="483 670 917 706"> <p>1) Analogue to uniform digital PCM</p> </div> <div data-bbox="520 729 1293 993"> <ul style="list-style-type: none"> <li>- microphone;</li> <li>- input level adjustment device;</li> <li>- input anti-aliasing filter;</li> <li>- <u>sample-hold device sampling at 16 kHz;</u></li> <li>- <u>analogue-to-uniform digital conversion to 14-bit representation.</u></li> </ul> </div> <div data-bbox="520 1018 1234 1052"> <p>The uniform format shall be represented in two's complement.</p> </div> <div data-bbox="445 1109 1199 1159"> <h2>4.2 Preparation of speech samples</h2> </div> <div data-bbox="445 1182 1728 1276"> <p><u>The encoder is fed with data comprising of samples with a resolution of 14 bits left justified in a 16-bit word. The decoder outputs data in the same format. Outside the speech codec further processing must be applied if the traffic data occurs in a different representation.</u></p> </div> <div data-bbox="172 1295 401 1421"> <p>a) at least two signal paths associated to respective sets of</p> </div> <div data-bbox="436 1295 1812 1357"> <p>The accused products comprise at least <u>two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data.</u></p> </div>
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<p>pitch codebook parameters representative of said digitized input audio data, wherein:</p>	<h2 data-bbox="470 253 978 298">5.7 Adaptive codebook</h2> <p data-bbox="470 321 1776 545">In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, <u>there are two signal paths associated to respective sets of pitch codebook parameters</u>, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</p> <p data-bbox="470 568 1747 633">The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>. Note that 1 bit is used to encode the chosen path.</p> <div data-bbox="483 659 1829 972" style="border: 1px solid blue; padding: 10px;"> <p data-bbox="512 682 1793 711">Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where <math>\mathbf{b}_{LP}=[0.18,0.64,0.18]</math>. The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> <math display="block">v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).</math> </div> <p data-bbox="441 1019 1873 1081">For example, the accused products must support at a minimum the AMR-WB codec set, including the 6.6, 8.85, and 12.65 Kbit/s codes under the HD Voice Logo License Agreement. See Annex C for minimum HD Voice requirements.</p>
<p>i) each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of said pitch codevector from said pitch codebook search device; and</p>	<p data-bbox="441 1084 1818 1143">For the accused products, each signal path comprises <b>a pitch prediction error calculating device for calculating a pitch prediction error of said pitch codevector from said pitch codebook search device.</b></p>

## 5.7 Adaptive codebook

In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.

The low pass filter used in the second path is in the form  $B_{LP}(z)=0.18z+0.64+0.18z^{-1}$ . Note that 1 bit is used to encode the chosen path.

Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook  $v(n)$ ,  $v(n) = v'(n)$  in the first path, or  $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)$  in the second path, where  $\mathbf{b}_{LP}=[0.18,0.64,0.18]$ . The path which results in minimum energy of the target signal  $x_2(n)$  defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes,  $v(n)$  is always

$$v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$$

ii) at least one of said at least two signal paths comprises a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of said at

For the accused products, at least one of said at least two signal paths comprises a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of said at least one signal path.

<p>least one signal path; and</p>	<h2 data-bbox="470 253 978 297">5.7 Adaptive codebook</h2> <p data-bbox="470 321 1776 545">In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. <u>One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</u></p> <div data-bbox="470 570 1755 634" style="border: 1px solid green; padding: 5px;"> <p>The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>. Note that 1 bit is used to encode the chosen path.</p> </div> <p data-bbox="506 683 1797 870">Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where <math>\mathbf{b}_{LP}=[0.18,0.64,0.18]</math>. The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$
<p>b) a selector for comparing the pitch prediction errors calculated in said at least two signal paths, for choosing the signal path having the lowest calculated pitch prediction error and for selecting the set of pitch codebook</p>	<p>The accused products include a selector for comparing the pitch prediction errors calculated in said at least two signal paths, for choosing the signal path having the lowest calculated pitch prediction error and for selecting the set of pitch codebook parameters associated to the chosen signal path.</p>



<p>parameters associated to the chosen signal path.</p>	<h2 data-bbox="470 253 978 297">5.7 Adaptive codebook</h2> <p data-bbox="470 321 1776 545">In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. <u>The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</u></p> <p data-bbox="470 570 1745 634">The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>. Note that 1 bit is used to encode the chosen path.</p> <p data-bbox="506 678 1787 870">Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where <math>b_{LP}=[0.18,0.64,0.18]</math>. <u>The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector.</u> For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$
<p style="text-align: center;"><b>Claim 2</b></p>	
<p>A pitch analysis device as defined in claim 1, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error</p>	<p>For the accused products, <u>one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.</u></p>

calculating device.	<h2 style="text-align: center;">5.7 Adaptive codebook</h2> <p>In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. <u>One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths.</u> The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</p> <p>The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>, Note that 1 bit is used to encode the chosen path.</p> <p>Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, <u>there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where</u>  <math>b_{LP}=[0.18,0.64,0.18]</math>. The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$
<b>Claim 5</b>	
A pitch analysis device as defined in claim 1, wherein each pitch prediction error calculating device comprises: a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter	<p>For the accused products, each pitch prediction error calculating device comprises a convolution unit for <b>convolving the pitch codevector</b> with <b>a weighted synthesis filter impulse response signal</b> and <b>therefore calculating a convolved pitch codevector</b>.</p> <p>For example, as defined in the technical specification for the AMR-WB speech codec, <math>v(n)</math> corresponds to the pitch codebook vector and <math>h(n)</math> corresponds to the impulse response of the weighted synthesis filter. The pitch codebook vector is convolved with the weighted synthesis impulse response to calculate <math>y(n)</math>, a convolved pitch code vector.</p>



<p>impulse response signal and therefore calculating a convolved pitch codevector;</p>	<div> <div>Release 10</div> <div>10</div> <div>3GPP TS 26.190 V10.0.0 (2011-03)</div> </div> <p> <math>res_{LP}(n)</math>      The LP residual signal  <math>c(n)</math>      The fixed codebook vector  <math>v(n)</math>      The adaptive codebook vector  <math>y(n) = v(n) * h(n)</math>      The filtered adaptive codebook vector </p> <p> <math>d_i</math>      The distance between the immittance spectral frequencies <math>f_{i+1}</math> and <math>f_{i-1}</math>  <math>h(n)</math>      The impulse response of the weighted synthesis filter  <math>H(z)W(z)</math>      The weighted synthesis filter </p>
<p>b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;</p>	<div> <div>Release 10</div> <div>26</div> <div>3GPP TS 26.190 V10.0.0 (2011-03)</div> </div> <p>The adaptive codebook gain is then found by</p> $g_p = \frac{\sum_{n=0}^{63} x(n) y(n)}{\sum_{n=0}^{63} y(n) y(n)}, \quad \text{bounded by } 0 \leq g_p \leq 1.2, \quad (39)$ <p>where <math>y(n) = v(n) * h(n)</math> is the filtered adaptive codebook vector (zero-state response of <math>H(z)W(z)</math> to <math>v_i(n)</math>). To insure stability, the adaptive codebook gain <math>g_p</math> is bounded by 0.95, if the adaptive codebook gains of the previous subframes have been small and the LP filters of the previous subframes have been close to being unstable.</p>

	<p>Then the following operations are repeated for each subframe:</p> <ul style="list-style-type: none"> <li>- The target signal <math>x(n)</math> is computed by filtering the LP residual through the weighted synthesis filter <math>W(z)H(z)</math> with the initial states of the filters having been updated by filtering the error between LP residual and excitation (this is equivalent to the common approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal).</li> </ul>
c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and	<p>For the accused products, each pitch prediction error calculating device comprises an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector.</p> <h3>5.8.3 Codebook search</h3> <p>The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is</p> $x_2(n) = x(n) - g_p y(n), \quad n = 0, \dots, 63, \quad (40)$
d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.	<p>For the accused products, each pitch prediction error calculating device comprises a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.</p> <h3>5.8.3 Codebook search</h3> <p>The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is</p> $x_2(n) = x(n) - g_p y(n), \quad n = 0, \dots, 63, \quad (40)$

	<p>Then the following operations are repeated for each subframe:</p> <ul style="list-style-type: none"> <li>- The target signal <math>x(n)</math> is computed by filtering the LP residual through the weighted synthesis filter <math>W(z)H(z)</math> with the initial states of the filters having been updated by filtering the error between LP residual and excitation (this is equivalent to the common approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal).</li> <li>- The impulse response, <math>h(n)</math> of the weighted synthesis filter is computed.</li> <li>- Closed-loop pitch analysis is then performed (to find the pitch lag and gain), using the target <math>x(n)</math> and impulse response <math>h(n)</math>, by searching around the open-loop pitch lag. Fractional pitch with 1/4th or 1/2nd of a sample resolution (depending on the mode and the pitch lag value) is used. The interpolating filter in fractional pitch search has low pass frequency response. Further, there are two potential low-pass characteristics in the the adaptive codebook and this information is encoded with 1 bit.</li> <li>- The target signal <math>x(n)</math> is updated by removing the adaptive codebook contribution (filtered adaptive codevector), and this new target, <math>x_2(n)</math>, is used in the fixed algebraic codebook search (to find the optimum innovation).</li> </ul>
<b>Claim 6</b>	
<p>A pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain <math>b^{(j)}</math> using the relation:</p> $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ <p>where <math>j=0, 1, 2, \dots, K</math>, and <math>K</math> corresponds to a number of signal paths, and where <math>x</math> is said pitch search target</p>	<p>The accused products comprise a pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain <math>b^{(j)}</math> using the relation:</p> $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ <p>Additionally, <math>j=0, 1, 2, \dots, K</math>, and <math>K</math> corresponds to a number of signal paths, and where <math>x</math> is said pitch search target vector and <math>y^{(j)}</math> is said convolved pitch codevector.</p>



<p>vector and <math>y^j</math> is said convolved pitch codevector.</p>	<div> <div>Release 10</div> <div>26</div> <div>3GPP TS 26.190 V10.0.0 (2011-03)</div> </div> <p>The adaptive codebook gain is then found by</p> $g_p = \frac{\sum_{n=0}^{63} x(n)y(n)}{\sum_{n=0}^{63} y(n)y(n)}, \quad \text{bounded by } 0 \leq g_p \leq 1.2, \quad (39)$ <p>where <math>y(n) = v(n) * h(n)</math> is the filtered adaptive codebook vector (zero-state response of <math>H(z)W(z)</math> to <math>v_i(n)</math>). To insure stability, the adaptive codebook gain <math>g_p</math> is bounded by 0.95, if the adaptive codebook gains of the previous subframes have been small and the LP filters of the previous subframes have been close to being unstable.</p>
<div>Claim 7</div>	
<p>A pitch analysis device as defined in claim 5, wherein:</p> <p>a) each of said filters of the plurality of signal paths is identified by a filter index;</p>	<p>For the accused products, each of the filters of the plurality of signal paths is identified by a filter index.</p> <h2>6.1 Decoding and speech synthesis</h2> <p>The decoding process is performed in the following order:</p> <p><b>Decoding of LP filter parameters:</b> The received indices of ISP quantization are used to reconstruct the quantized ISP vector. The interpolation described in Section 5.2.6 is performed to obtain 4 interpolated ISP vectors (corresponding to 4 subframes). For each subframe, the interpolated ISP vector is converted to LP filter coefficient domain <math>a_k</math>, which is used for synthesizing the reconstructed speech in the subframe.</p> <p>The following steps are repeated for each subframe:</p> <ol style="list-style-type: none"> <li><b>Decoding of the adaptive codebook vector:</b> <u>The received pitch index (adaptive codebook index) is used to find the integer and fractional parts of the pitch lag. The adaptive codebook vector <math>v(n)</math> is found by interpolating the past excitation <math>u(n)</math> (at the pitch delay) using the FIR filter described in Section 5.7. The received adaptive filter index is used to find out whether the filtered adaptive codebook is <math>v_1(n) = v(n)</math> or <math>v_2(n) = 0.18v(n) + 0.64v(n-1) + 0.18v(n-2)</math>.</u></li> </ol>

<p>b) said pitch codevector is identified by a pitch codebook index; and</p>	<p>For the accused products, the pitch codevector is identified by a pitch codebook index.</p> <h2 data-bbox="457 256 1243 305">6.1 Decoding and speech synthesis</h2> <p data-bbox="457 332 1104 362">The decoding process is performed in the following order:</p> <p data-bbox="457 389 1787 516"><b>Decoding of LP filter parameters:</b> The received indices of ISP quantization are used to reconstruct the quantized ISP vector. The interpolation described in Section 5.2.6 is performed to obtain 4 interpolated ISP vectors (corresponding to 4 subframes). For each subframe, the interpolated ISP vector is converted to LP filter coefficient domain <math>a_k</math>, which is used for synthesizing the reconstructed speech in the subframe.</p> <p data-bbox="457 544 1035 573">The following steps are repeated for each subframe:</p> <ol data-bbox="657 600 1787 764" style="list-style-type: none"> <li>1. <b>Decoding of the adaptive codebook vector:</b> <u>The received pitch index (adaptive codebook index) is used to find the integer and fractional parts of the pitch lag. The adaptive codebook vector <math>v(n)</math> is found by interpolating the past excitation <math>u(n)</math> (at the pitch delay) using the FIR filter described in Section 5.7. The received adaptive filter index is used to find out whether the filtered adaptive codebook is <math>v_1(n) = v(n)</math> or <math>v_2(n) = 0.18v(n) + 0.64v(n-1) + 0.18v(n-2)</math>.</u></li> </ol> <h2 data-bbox="470 829 1029 878">5.7 Adaptive codebook</h2> <p data-bbox="470 906 1812 976">Adaptive codebook search is performed on a subframe basis. It consists of performing closed loop pitch search, and then <u>computing the adaptive codevector by interpolating the past excitation at the selected fractional pitch lag.</u></p>
<p>c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.</p>	<p>For the accused products, the pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.</p>



In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.

The low pass filter used in the second path is in the form  $B_{LP}(z)=0.18z+0.64+0.18z^{-1}$ . Note that 1 bit is used to encode the chosen path.

Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook  $v(n)$ ,  $v(n) = v'(n)$  in the first path, or  $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)$  in the second path, where  $b_{LP}=[0.18,0.64,0.18]$ . The path which results in minimum energy of the target signal  $x_2(n)$  defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes,  $v(n)$  is always

$$v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$$

## 6.1 Decoding and speech synthesis

The decoding process is performed in the following order:

**Decoding of LP filter parameters:** The received indices of ISP quantization are used to reconstruct the quantized ISP vector. The interpolation described in Section 5.2.6 is performed to obtain 4 interpolated ISP vectors (corresponding to 4 subframes). For each subframe, the interpolated ISP vector is converted to LP filter coefficient domain  $a_k$ , which is used for synthesizing the reconstructed speech in the subframe.

The following steps are repeated for each subframe:

1. **Decoding of the adaptive codebook vector:** The received pitch index (adaptive codebook index) is used to find the integer and fractional parts of the pitch lag. The adaptive codebook vector  $v(n)$  is found by interpolating the past excitation  $u(n)$  (at the pitch delay) using the FIR filter described in Section 5.7. The received adaptive filter index is used to find out whether the filtered adaptive codebook is  $v_1(n)=v(n)$  or  $v_2(n)=0.18v(n)+0.64v(n-1)+0.18v(n-2)$ .



## 5.7 Adaptive codebook

Adaptive codebook search is performed on a subframe basis. It consists of performing closed loop pitch search, and then computing the adaptive codevector by interpolating the past excitation at the selected fractional pitch lag.

The adaptive codebook parameters (or pitch parameters) are the delay and gain of the pitch filter. In the search stage, the excitation is extended by the LP residual to simplify the closed-loop search.

The adaptive codebook gain is then found by

$$g_p = \frac{\sum_{n=0}^{63} x(n)y(n)}{\sum_{n=0}^{63} y(n)y(n)}, \quad \text{bounded by } 0 \leq g_p \leq 1.2,$$

### Claim 8

A pitch analysis device as defined in claim 1, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the

For the accused products, the pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

<p>signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.</p>	<h2 data-bbox="468 250 968 293">5.7 Adaptive codebook</h2> <p data-bbox="468 315 1751 537">In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, <u>wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device.</u> One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</p> <p data-bbox="468 558 1724 623">The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>. Note that 1 bit is used to encode the chosen path.</p> <p data-bbox="506 667 1766 699">Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where</p> <p data-bbox="506 786 1782 850"><math>b_{LP}=[0.18,0.64,0.18]</math>. <u>The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector.</u> For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$
<p align="center"><b>Claim 10</b></p>	
<p>An encoder having a pitch analysis device as in claim 1 for encoding a wideband input signal, said encoder comprising: a) a linear prediction synthesis filter calculator responsive to the wideband signal</p>	<p>The accused products comprise <b>a linear prediction synthesis filter calculator responsive to the wideband signal for producing linear prediction synthesis filter coefficients.</b></p>



<p>for producing linear prediction synthesis filter coefficients;</p>	<h3>4.3 Principles of the adaptive multi-rate wideband speech encoder</h3> <p>The AMR-WB codec consists of nine source codecs with bit-rates of 23.85, 23.05, 19.85, 18.25, 15.85, 14.25, 12.65, 8.85 and 6.60 kbit/s.</p> <p><u>The codec is based on the code-excited linear predictive (CELP) coding model. The input signal is pre-emphasized using the filter <math>H_{pre-emph}(z)=1-\mu z^{-1}</math>. The CELP model is then applied to the pre-emphasized signal. A 16th order linear prediction (LP), or short-term, synthesis filter is used which is given by:</u></p> $H(z) = \frac{1}{\hat{A}(z)} = \frac{1}{1 + \sum_{i=1}^m \hat{a}_i z^{-i}} \quad (1)$ <p><u>where <math>\hat{a}_i, i=1, \dots, m</math> are the (quantized) linear prediction (LP) parameters, and <math>m = 16</math> is the predictor order. The long-term, or pitch, synthesis filter is usually given by:</u></p> $\frac{1}{B(z)} = \frac{1}{1 - g_p z^{-T}} \quad (2)$ <p>where <math>T</math> is the pitch delay and <math>g_p</math> is the pitch gain. The pitch synthesis filter is implemented using the so-called adaptive codebook approach.</p> <p>The encoder performs the analysis of the LPC, LTP and fixed codebook parameters at 12.8 kHz sampling rate. The coder operates on speech frames of 20 ms. <u>At each frame, the speech signal is analysed to extract the parameters of the CELP model (LP filter coefficients, adaptive and fixed codebooks' indices and gains). In addition to these parameters, high-band gain indices are computed in 23.85 kbit/s mode. These parameters are encoded and transmitted. At the decoder, these parameters are decoded and speech is synthesized by filtering the reconstructed excitation signal through the LP synthesis filter.</u></p>
<p>b) a perceptual weighting filter, responsive to the wideband signal and the linear</p>	<p>The accused products comprise a perceptual weighting filter, responsive to the wideband signal and the linear prediction synthesis filter coefficients, for producing a perceptually weighted signal.</p>



<p>prediction synthesis filter coefficients, for producing a perceptually weighted signal;</p>	<h3>4.3 Principles of the adaptive multi-rate wideband speech encoder</h3> <p>The <u>perceptual weighting filter</u> used in the analysis-by-synthesis search technique is given by:</p> $W(z) = A(z/\gamma_1)H_{de-emph}(z), \quad (3)$ <p>where <math>A(z)</math> is the unquantized LP filter, <math>H_{de-emph} = \frac{1}{1 - 0.68z^{-1}}</math>, and <math>\gamma_1=0.92</math> is the perceptual weighting factor. <u>The weighting filter uses the unquantized LP parameters.</u></p> <h3>5.3 Perceptual weighting</h3> <p>The traditional perceptual weighting filter <math>W(z) = A(z/\gamma_1)/A(z/\gamma_2)</math> has inherent limitations in modelling the formant structure and the required spectral tilt concurrently. The spectral tilt is more pronounced in wideband signals due to the wide dynamic range between low and high frequencies. A solution to this problem is to introduce the preemphasis filter at the input, compute the LP filter <math>A(z)</math> based on the preemphasized speech <math>s(n)</math>, and <u>use a modified filter <math>W(z)</math> by fixing its denominator.</u> <u>This structure substantially decouples the formant weighting from the tilt.</u></p> <p><u>A weighting filter of the form <math>W(z) = A(z/\gamma_1)H_{de-emph}(z)</math> is used, where <math>H_{de-emph} = \frac{1}{1 - \beta_1 z^{-1}}</math> and <math>\beta_1=0.68</math>.</u></p> <p>Because <math>A(z)</math> is <u>computed based on the preemphasized speech signal <math>s(n)</math>,</u> the tilt of the filter <math>1/A(z/\gamma_1)</math> is less pronounced compared to the case when <math>A(z)</math> is computed based on the original speech. Since deemphasis is performed at the decoder end, it can be shown that the quantization error spectrum is shaped by a filter having a transfer function <math>W^{-1}(z)H_{de-emph}(z)=1/A(z/\gamma_1)</math>. Thus, the spectrum of the quantization error is shaped by a filter whose transfer function is <u><math>1/A(z/\gamma_1)</math>, with <math>A(z)</math> computed based on the preemphasized speech signal.</u></p>
<p>c) an impulse response generator responsive to said linear prediction synthesis filter</p>	<p>The accused products comprise <b>an impulse response generator responsive to said linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal.</b></p>

<p>coefficients for producing a weighted synthesis filter impulse response signal;</p>	<div style="margin-left: 40px;"> <math>d_i</math>                      The distance between the immittance spectral frequencies <math>f_{i+1}</math> and <math>f_{i-1}</math> </div> <div style="margin-left: 40px; border: 1px solid blue; padding: 5px; display: inline-block;"> <math>h(n)</math>                      The impulse response of the weighted synthesis filter </div> <div style="margin-left: 40px;"> <math>H(z)W(z)</math>              The weighted synthesis filter </div> <h3 style="text-align: center;">5.5              Impulse response computation</h3> <p><u>The impulse response, <math>h(n)</math>, of the weighted synthesis filter <math>H(z)W(z) = A(z/\gamma_1)H_{de-emph}(z)/\hat{A}(z)</math> is computed each subframe. This impulse response is needed for the search of adaptive and fixed codebooks. The impulse response <math>h(n)</math> is computed by filtering the vector of coefficients of the filter <math>A(z/\gamma_1)</math> extended by zeros through the two filters <math>1/\hat{A}(z)</math> and <math>H_{de-emph}(z)</math>.</u></p>
<p>d) a pitch search unit for producing pitch codebook parameters, said pitch search unit comprising:</p> <p>i) said pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector and an innovative search target vector; and</p>	<p>The accused products comprise a pitch search unit for producing pitch codebook parameters. The pitch search unit comprises the pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector and an innovative search target vector.</p> <h3 style="text-align: center;">5.7              Adaptive codebook</h3> <p><u>Adaptive codebook search is performed on a subframe basis. It consists of performing closed loop pitch search, and then computing the adaptive codevector by interpolating the past excitation at the selected fractional pitch lag.</u></p> <p><u>The adaptive codebook parameters (or pitch parameters) are the delay and gain of the pitch filter. In the search stage, the excitation is extended by the LP residual to simplify the closed-loop search.</u></p>

## 5.6 Target signal computation

The target signal for adaptive codebook search is usually computed by subtracting the zero-input response of the weighted synthesis filter  $H(z)W(z) = A(z/\gamma_1)H_{de-emph}(z)/\hat{A}(z)$  from the weighted speech signal  $s_w(n)$ . This is performed on a subframe basis.

An equivalent procedure for computing the target signal, which is used in this codec, is the filtering of the LP residual signal  $r(n)$  through the combination of synthesis filter  $1/\hat{A}(z)$  and the weighting filter  $A(z/\gamma_1)H_{de-emph}(z)$ . After determining the excitation for the subframe, the initial states of these filters are updated by filtering the difference between the LP residual and excitation. The memory update of these filters is explained in Section 5.10.

The residual signal  $r(n)$  which is needed for finding the target vector is also used in the adaptive codebook search to extend the past excitation buffer. This simplifies the adaptive codebook search procedure for delays less than the subframe size of 64 as will be explained in the next section. The LP residual is given by

$$r(n) = s(n) + \sum_{i=1}^{16} \hat{a}_i s(n-i), n = 0, \dots, 63. \quad (36)$$

- The target signal  $x(n)$  is updated by removing the adaptive codebook contribution (filtered adaptive codevector), and this new target,  $x_2(n)$ , is used in the fixed algebraic codebook search (to find the optimum innovation).

ii) said pitch analysis device responsive to the pitch codevector for selecting, from said sets of pitch codebook parameters, the set of pitch codebook parameters associated to the

The accused products comprise a pitch analysis device responsive to the pitch codevector for selecting, from said sets of pitch codebook parameters, the set of pitch codebook parameters associated to the signal path having the lowest calculated pitch prediction error.



<p>signal path having the lowest calculated pitch prediction error;</p>	<h2 data-bbox="468 250 972 297">5.7 Adaptive codebook</h2> <p data-bbox="468 318 1757 540">In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. <u>The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</u></p> <p data-bbox="468 565 1728 626">The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>. Note that 1 bit is used to encode the chosen path.</p> <p data-bbox="506 672 1766 703">Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the</p> <p data-bbox="506 711 1776 789">adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where</p> <p data-bbox="506 797 1787 860"><math>b_{LP}=[0.18,0.64,0.18]</math>. <u>The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector.</u> For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$
<p>e) an innovative codebook search device, responsive to a weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters; and</p>	<p data-bbox="436 984 1856 1044">The accused products comprise <b>an innovative codebook search device</b>, responsive to a <b>weighted synthesis filter impulse response signal</b>, and the <b>innovative search target vector</b>, for producing <b>innovative codebook parameters</b>.</p> <h2 data-bbox="457 1078 1215 1125">5.5 Impulse response computation</h2> <p data-bbox="457 1154 1776 1312">The impulse response, <math>h(n)</math>, of the weighted synthesis filter <math>H(z)W(z) = A(z/\gamma_1)H_{de-emph}(z)/\hat{A}(z)</math> is computed each subframe. <u>This impulse response is needed for the search of adaptive and fixed codebooks.</u> The impulse response <math>h(n)</math> is computed by filtering the vector of coefficients of the filter <math>A(z/\gamma_1)</math> extended by zeros through the two filters <math>1/\hat{A}(z)</math> and <math>H_{de-emph}(z)</math>.</p>

	<ul style="list-style-type: none"> <li>- The target signal <math>x(n)</math> is updated by removing the adaptive codebook contribution (filtered adaptive codevector), and this new target, <math>x_2(n)</math>, is used in the fixed algebraic codebook search (to find the optimum innovation).</li> </ul> <h2>5.8 Algebraic codebook</h2> <h3>5.8.3 Codebook search</h3> <p>The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is</p> $x_2(n) = x(n) - g_p y(n), \quad n = 0, \dots, 63, \quad (40)$ <p>where <math>y(n) = v(n) * h(n)</math> is the filtered adaptive codebook vector and <math>g_p</math> is the unquantized adaptive codebook gain.</p> <p>The matrix <math>\mathbf{H}</math> is defined as the lower triangular Toeplitz convolution matrix with diagonal <math>h(0)</math> and lower diagonals <math>h(1), \dots, h(63)</math>, and <math>\mathbf{d} = \mathbf{H}^T \mathbf{x}_2</math> is the correlation between the target signal <math>x_2(n)</math> and the impulse response <math>h(n)</math> (also known as the backward filtered target vector), and <math>\Phi = \mathbf{H}^T \mathbf{H}</math> is the matrix of correlations of <math>h(n)</math>.</p>
f) a signal forming device for producing an encoded wideband signal comprising the set of pitch codebook parameters associated to the signal path having the lowest pitch prediction error, said innovative codebook parameters, and said linear prediction	<p>The accused products comprise a signal forming device for producing an encoded wideband signal comprising the set of pitch codebook parameters associated to the signal path having the lowest pitch prediction error, said innovative codebook parameters, and said linear prediction synthesis filter coefficients.</p>

synthesis filter coefficients.

### 4.3 Principles of the adaptive multi-rate wideband speech encoder

The CELP speech synthesis model is shown in Figure 1. In this model, the excitation signal at the input of the short-term LP synthesis filter is constructed by adding two excitation vectors from adaptive and fixed (innovative) codebooks. The speech is synthesized by feeding the two properly chosen vectors from these codebooks through the short-term synthesis filter. The optimum excitation sequence in a codebook is chosen using an analysis-by-synthesis search procedure in which the error between the original and synthesized speech is minimized according to a perceptually weighted distortion measure.

The perceptual weighting filter used in the analysis-by-synthesis search technique is given by:

$$W(z) = A(z/\gamma_1)H_{de-emph}(z), \quad (3)$$

where  $A(z)$  is the unquantized LP filter,  $H_{de-emph} = \frac{1}{1 - 0.68z^{-1}}$ , and  $\gamma_1 = 0.92$  is the perceptual weighting factor. The weighting filter uses the unquantized LP parameters.

The encoder performs the analysis of the LPC, LTP and fixed codebook parameters at 12.8 kHz sampling rate. The coder operates on speech frames of 20 ms. At each frame, the speech signal is analysed to extract the parameters of the CELP model (LP filter coefficients, adaptive and fixed codebooks' indices and gains). In addition to these parameters, high-band gain indices are computed in 23.85 kbit/s mode. These parameters are encoded and transmitted. At the decoder, these parameters are decoded and speech is synthesized by filtering the reconstructed excitation signal through the LP synthesis filter.



## 5.7 Adaptive codebook

In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.

The low pass filter used in the second path is in the form  $B_{LP}(z)=0.18z+0.64+0.18z^{-1}$ . Note that 1 bit is used to encode the chosen path.

Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook  $v(n)$ ,  $v(n) = v'(n)$  in the first path, or  $v(n) = \sum_{i=1}^1 b_{LP}(i+1)v'(n+i)$  in the second path, where  $\mathbf{b}_{LP}=[0.18,0.64,0.18]$ . The path which results in minimum energy of the target signal  $x_2(n)$  defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes,  $v(n)$  is always

$$v(n) = \sum_{i=1}^1 b_{LP}(i+1)v'(n+i).$$

### Claim 11

An encoder as defined in claim 10, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating

For the accused products, one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

device.	<h2 style="text-align: center;">5.7 Adaptive codebook</h2> <p>In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. <u>One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</u></p> <p>The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>. Note that 1 bit is used to encode the chosen path.</p> <p>Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, <u>there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where <math>\mathbf{b}_{LP}=[0.18,0.64,0.18]</math>. The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</u></p> $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$
<b>Claim 14</b>	
An encoder as defined in claim 10, wherein each pitch prediction error calculating device comprises: a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and	<p>For the accused products, each pitch prediction error calculating device comprises a convolution unit for <b>convolving the pitch codevector</b> with <b>a weighted synthesis filter impulse response signal</b> and <b>therefore calculating a convolved pitch codevector</b>.</p> <p>For example, as defined in the technical specification for the AMR-WB speech codec, <math>v(n)</math> corresponds to the pitch codebook vector and <math>h(n)</math> corresponds to the impulse response of the weighted synthesis filter. The pitch codebook vector is convolved with the weighted synthesis impulse response to calculate <math>y(n)</math>, a convolved pitch code vector.</p>

<p>therefore calculating a convolved pitch codevector;</p>	<div> <div>Release 10103GPP TS 26.190 V10.0.0 (2011-03)</div> <div> <math>res_{LP}(n)</math> The LP residual signal         </div> <div> <math>c(n)</math> The fixed codebook vector         </div> <div> <math>v(n)</math> The adaptive codebook vector         </div> <div> <math>y(n) = v(n) * h(n)</math> The filtered adaptive codebook vector         </div> <div> <math>d_i</math> The distance between the immittance spectral frequencies <math>f_{i+1}</math> and <math>f_{i-1}</math> </div> <div> <math>h(n)</math> The impulse response of the weighted synthesis filter         </div> <div> <math>H(z)W(z)</math> The weighted synthesis filter         </div> </div>
<p>b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;</p>	<div> <div>Release 10263GPP TS 26.190 V10.0.0 (2011-03)</div> <div>           The adaptive codebook gain is then found by         </div> <div> <math display="block">g_p = \frac{\sum_{n=0}^{63} x(n)y(n)}{\sum_{n=0}^{63} y(n)y(n)}, \quad \text{bounded by } 0 \leq g_p \leq 1.2, \quad (39)</math> </div> <div>           where <math>y(n) = v(n) * h(n)</math> is the filtered adaptive codebook vector (zero-state response of <math>H(z)W(z)</math> to <math>v_i(n)</math>). To insure stability, the adaptive codebook gain <math>g_p</math> is bounded by 0.95, if the adaptive codebook gains of the previous subframes have been small and the LP filters of the previous subframes have been close to being unstable.         </div> </div>



	<p>Then the following operations are repeated for each subframe:</p> <ul style="list-style-type: none"> <li>- The target signal <math>x(n)</math> is computed by filtering the LP residual through the weighted synthesis filter <math>W(z)H(z)</math> with the initial states of the filters having been updated by filtering the error between LP residual and excitation (this is equivalent to the common approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal).</li> </ul>
c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and	<p>For the accused products, each pitch prediction error calculating device comprises an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector.</p> <h3>5.8.3 Codebook search</h3> <p>The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is</p> $x_2(n) = x(n) - g_p y(n), \quad n = 0, \dots, 63, \quad (40)$
d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.	<p>For the accused products, each pitch prediction error calculating device comprises a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.</p> <h3>5.8.3 Codebook search</h3> <p>The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is</p> $x_2(n) = x(n) - g_p y(n), \quad n = 0, \dots, 63, \quad (40)$

	<p>Then the following operations are repeated for each subframe:</p> <ul style="list-style-type: none"> <li>- The target signal <math>x(n)</math> is computed by filtering the LP residual through the weighted synthesis filter <math>W(z)H(z)</math> with the initial states of the filters having been updated by filtering the error between LP residual and excitation (this is equivalent to the common approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal).</li> <li>- The impulse response, <math>h(n)</math> of the weighted synthesis filter is computed.</li> <li>- Closed-loop pitch analysis is then performed (to find the pitch lag and gain), using the target <math>x(n)</math> and impulse response <math>h(n)</math>, by searching around the open-loop pitch lag. Fractional pitch with 1/4th or 1/2nd of a sample resolution (depending on the mode and the pitch lag value) is used. The interpolating filter in fractional pitch search has low pass frequency response. Further, there are two potential low-pass characteristics in the the adaptive codebook and this information is encoded with 1 bit.</li> <li>- The target signal <math>x(n)</math> is updated by removing the adaptive codebook contribution (filtered adaptive codevector), and this new target, <math>x_2(n)</math>, is used in the fixed algebraic codebook search (to find the optimum innovation).</li> </ul>
<b>Claim 15</b>	
<p>An encoder as defined in claim 10, wherein said pitch gain calculator comprises a means for calculating said pitch gain <math>b^{(j)}</math> using the relation:</p> $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ <p>where <math>j=0, 1, 2, \dots, K</math>, and <math>K</math> corresponds to a number of signal paths, and where <math>x</math> is said pitch search target vector and <math>y^{(j)}</math> is</p>	<p>The accused products comprise a pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain <math>b^{(j)}</math> using the relation:</p> $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ <p>Additionally, <math>j=0, 1, 2, \dots, K</math>, and <math>K</math> corresponds to a number of signal paths, and where <math>x</math> is said pitch search target vector and <math>y^{(j)}</math> is said convolved pitch codevector.</p>

said convolved pitch codevector.	<div data-bbox="470 217 1803 248" data-label="Page-Header"> <div>Release 10</div> <div>26</div> <div>3GPP TS 26.190 V10.0.0 (2011-03)</div> </div> <div data-bbox="470 297 978 332" data-label="Text"> <p>The adaptive codebook gain is then found by</p> </div> <div data-bbox="785 358 1724 470" data-label="Equation-Block"> <math display="block">g_p = \frac{\sum_{n=0}^{63} x(n)y(n)}{\sum_{n=0}^{63} y(n)y(n)}, \quad \text{bounded by } 0 \leq g_p \leq 1.2, \quad (39)</math> </div> <div data-bbox="470 505 1803 609" data-label="Text"> <p>where <math>y(n) = v(n) * h(n)</math> is the filtered adaptive codebook vector (zero-state response of <math>H(z)W(z)</math> to <math>v_i(n)</math>). To insure stability, the adaptive codebook gain <math>g_p</math> is bounded by 0.95, if the adaptive codebook gains of the previous subframes have been small and the LP filters of the previous subframes have been close to being unstable.</p> </div>
Claim 17	
An encoder as defined in claim 10, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest	<p>For the accused products, the pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.</p>



<p>calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.</p>	<h2 style="text-align: center;">5.7 Adaptive codebook</h2> <p>In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, <u>wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device.</u> One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</p> <p>The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>. Note that 1 bit is used to encode the chosen path.</p> <p>Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where <math>b_{LP}=[0.18,0.64,0.18]</math>. <u>The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector.</u> For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$
<b>Claim 28</b>	
<p>A cellular mobile transmitter/receiver unit, comprising:</p> <p>a) a transmitter including an encoder for encoding a wideband signal as recited in claim 10 and a transmission circuit for</p>	<p>The accused products comprise a cellular mobile transmitter/receiver unit, comprising:</p> <p>a) <b>a transmitter including an encoder for encoding a wideband signal as recited in claim 10 and a transmission circuit for transmitting the encoded wideband signal; and</b></p> <p>b) <b>a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.</b></p>

<p>transmitting the encoded wideband signal; and</p> <p>b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.</p>	<h3 data-bbox="464 204 1646 293">4.3 Principles of the adaptive multi-rate wideband speech encoder</h3> <p data-bbox="464 321 1766 505">The encoder performs the analysis of the LPC, LTP and fixed codebook parameters at 12.8 kHz sampling rate. The coder operates on speech frames of 20 ms. At each frame, the speech signal is analysed to extract the parameters of the CELP model (LP filter coefficients, adaptive and fixed codebooks' indices and gains). In addition to these parameters, high-band gain indices are computed in 23.85 kbit/s mode. <u>These parameters are encoded and transmitted. At the decoder, these parameters are decoded and speech is synthesized by filtering the reconstructed excitation signal through the LP synthesis filter.</u></p>
<p align="center"><b>Claim 29</b></p>	
<p>29. A cellular mobile transmitter/receiver unit as defined in claim 28, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.</p>	<p>For the accused products, <u>one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.</u></p>



## 5.7 Adaptive codebook

In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.

The low pass filter used in the second path is in the form  $B_{LP}(z)=0.18z+0.64+0.18z^{-1}$ . Note that 1 bit is used to encode the chosen path.

Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook  $v(n)$ ,  $v(n) = v'(n)$  in the first path, or  $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)$  in the second path, where

$\mathbf{b}_{LP}=[0.18,0.64,0.18]$ . The path which results in minimum energy of the target signal  $x_2(n)$  defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes,  $v(n)$  is always

$$v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$$

### Claim 32

A cellular mobile transmitter/receiver unit as defined in claim 28, wherein each pitch prediction error calculating device comprises:  
a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter

For the accused products, each pitch prediction error calculating device comprises a convolution unit for **convolving the pitch codevector** with **a weighted synthesis filter impulse response signal** and **therefore calculating a convolved pitch codevector**.

For example, as defined in the technical specification for the AMR-WB speech codec,  $v(n)$  corresponds to the pitch codebook vector and  $h(n)$  corresponds to the impulse response of the weighted synthesis filter. The pitch codebook vector is convolved with the weighted synthesis impulse response to calculate  $y(n)$ , a convolved pitch code vector.



<p>impulse response signal and therefore calculating a convolved pitch codevector;</p>	<div> <div>Release 10103GPP TS 26.190 V10.0.0 (2011-03)</div> <div> <div> <math>res_{LP}(n)</math> The LP residual signal </div> <div> <math>c(n)</math> The fixed codebook vector </div> <div> <math>v(n)</math> The adaptive codebook vector </div> <div> <math>y(n) = v(n) * h(n)</math> The filtered adaptive codebook vector </div> <div> <math>d_i</math> The distance between the immittance spectral frequencies <math>f_{i+1}</math> and <math>f_{i-1}</math> </div> <div> <math>h(n)</math> The impulse response of the weighted synthesis filter </div> <div> <math>H(z)W(z)</math> The weighted synthesis filter </div> </div> </div>
<p>b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;</p>	<div> <div>Release 10263GPP TS 26.190 V10.0.0 (2011-03)</div> <div> <div>The adaptive codebook gain is then found by</div> <div> <math display="block">g_p = \frac{\sum_{n=0}^{63} x(n)y(n)}{\sum_{n=0}^{63} y(n)y(n)}, \quad \text{bounded by } 0 \leq g_p \leq 1.2, \quad (39)</math> </div> <div> <div>where <math>y(n) = v(n) * h(n)</math> is the filtered adaptive codebook vector</div> <div>(zero-state response of <math>H(z)W(z)</math> to <math>v_i(n)</math>). To insure stability, the adaptive codebook gain <math>g_p</math> is bounded by 0.95, if the adaptive codebook gains of the previous subframes have been small and the LP filters of the previous subframes have been close to being unstable.</div> </div> </div> </div>

	<p>Then the following operations are repeated for each subframe:</p> <ul style="list-style-type: none"> <li>- The target signal <math>x(n)</math> is computed by filtering the LP residual through the weighted synthesis filter <math>W(z)H(z)</math> with the initial states of the filters having been updated by filtering the error between LP residual and excitation (this is equivalent to the common approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal).</li> </ul>
c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and	<p>For the accused products, each pitch prediction error calculating device comprises an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector.</p> <h3>5.8.3 Codebook search</h3> <p>The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is</p> $x_2(n) = x(n) - g_p y(n), \quad n = 0, \dots, 63, \quad (40)$
d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.	<p>For the accused products, each pitch prediction error calculating device comprises a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.</p> <h3>5.8.3 Codebook search</h3> <p>The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is</p> $x_2(n) = x(n) - g_p y(n), \quad n = 0, \dots, 63, \quad (40)$

	<p>Then the following operations are repeated for each subframe:</p> <ul style="list-style-type: none"> <li>- The target signal <math>x(n)</math> is computed by filtering the LP residual through the weighted synthesis filter <math>W(z)H(z)</math> with the initial states of the filters having been updated by filtering the error between LP residual and excitation (this is equivalent to the common approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal).</li> <li>- The impulse response, <math>h(n)</math> of the weighted synthesis filter is computed.</li> <li>- Closed-loop pitch analysis is then performed (to find the pitch lag and gain), using the target <math>x(n)</math> and impulse response <math>h(n)</math>, by searching around the open-loop pitch lag. Fractional pitch with 1/4th or 1/2nd of a sample resolution (depending on the mode and the pitch lag value) is used. The interpolating filter in fractional pitch search has low pass frequency response. Further, there are two potential low-pass characteristics in the the adaptive codebook and this information is encoded with 1 bit.</li> <li>- The target signal <math>x(n)</math> is updated by removing the adaptive codebook contribution (filtered adaptive codevector), and this new target, <math>x_2(n)</math>, is used in the fixed algebraic codebook search (to find the optimum innovation).</li> </ul>
<b>Claim 33</b>	
<p>A cellular mobile transmitter/receiver unit as defined in claim 32, wherein said pitch gain calculator comprises a means for calculating said pitch gain <math>b^{(j)}</math> using the relation:</p> $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ <p>where <math>j=0, 1, 2, \dots, K</math>, and <math>K</math> corresponds to a number of signal paths, and where <math>x</math> is said pitch search target</p>	<p>The accused products comprise a pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain <math>b^{(j)}</math> using the relation:</p> $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ <p>Additionally, <math>j=0, 1, 2, \dots, K</math>, and <math>K</math> corresponds to a number of signal paths, and where <math>x</math> is said pitch search target vector and <math>y^{(j)}</math> is said convolved pitch codevector.</p>



<p>vector and <math>y^j</math> is said convolved pitch codevector.</p>	<div data-bbox="470 215 1803 248"> <div>Release 10</div> <div>26</div> <div>3GPP TS 26.190 V10.0.0 (2011-03)</div> </div> <div data-bbox="470 297 978 329"> <div>The adaptive codebook gain is then found by</div> </div> <div data-bbox="785 362 1724 472"> <div> <math display="block">g_p = \frac{\sum_{n=0}^{63} x(n)y(n)}{\sum_{n=0}^{63} y(n)y(n)}</math> </div> <div> <div>bounded by <math>0 \leq g_p \leq 1.2</math>,</div> <div>( 39 )</div> </div> </div> <div data-bbox="470 505 1803 610"> <p>where <math>y(n) = v(n) * h(n)</math> is the filtered adaptive codebook vector (zero-state response of <math>H(z)W(z)</math> to <math>v_i(n)</math>). To insure stability, the adaptive codebook gain <math>g_p</math> is bounded by 0.95, if the adaptive codebook gains of the previous subframes have been small and the LP filters of the previous subframes have been close to being unstable.</p> </div>
<div>Claim 35</div>	
<p>A cellular mobile transmitter/receiver unit as defined in claim 28, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the</p>	<p>For the accused products, the pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.</p>

<p>signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.</p>	<h2 style="text-align: center;">5.7 Adaptive codebook</h2> <p>In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, <u>wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device.</u> One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</p> <p>The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>. Note that 1 bit is used to encode the chosen path.</p> <p>Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where <math>b_{LP}=[0.18,0.64,0.18]</math>. <u>The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector.</u> For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$
<b>Claim 37</b>	
<p>A network element, comprising: a transmitter including an encoder for encoding a wideband signal as recited in claim 10 and a transmission circuit for transmitting the</p>	<p>The accused products comprise <b>a transmitter including an encoder for encoding a wideband signal as recited in claim 10 and a transmission circuit for transmitting the encoded wideband signal.</b></p>



<p>encoded wideband signal.</p>	<h3>4.3 Principles of the adaptive multi-rate wideband speech encoder</h3> <p>The encoder performs the analysis of the LPC, LTP and fixed codebook parameters at 12.8 kHz sampling rate. The coder operates on speech frames of 20 ms. At each frame, the speech signal is analysed to extract the parameters of the CELP model (LP filter coefficients, adaptive and fixed codebooks' indices and gains). In addition to these parameters, high-band gain indices are computed in 23.85 kbit/s mode. <u>These parameters are encoded and transmitted. At the decoder, these parameters are decoded and speech is synthesized by filtering the reconstructed excitation signal through the LP synthesis filter.</u></p>
<p align="center"><b>Claim 38</b></p>	
<p>38. A network element as defined in claim 37, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.</p>	<p>For the accused products, <u>one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.</u></p> <h3>5.7 Adaptive codebook</h3> <p>In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. <u>One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths.</u> The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</p> <p>The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>. Note that 1 bit is used to encode the chosen path.</p> <p>Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, <u>there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where</u></p> <p><math>\mathbf{b}_{LP}=[0.18,0.64,0.18]</math>. The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$



# Claim 41

<p>A network element as defined in claim 37, wherein each pitch prediction error calculating device comprises:</p> <p>a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;</p>	<p>For the accused products, each pitch prediction error calculating device comprises a convolution unit for <b>convolving the pitch codevector</b> with <b>a weighted synthesis filter impulse response signal</b> and <b>therefore calculating a convolved pitch codevector</b>.</p> <p>For example, as defined in the technical specification for the AMR-WB speech codec, <math>v(n)</math> corresponds to the pitch codebook vector and <math>h(n)</math> corresponds to the impulse response of the weighted synthesis filter. The pitch codebook vector is convolved with the weighted synthesis impulse response to calculate <math>y(n)</math>, a convolved pitch code vector.</p> <p><b>Release 10</b> <span style="float: right;"><b>10</b> <b>3GPP TS 26.190 V10.0.0 (2011-03)</b></span></p> <p><math>res_{LP}(n)</math>      The LP residual signal</p> <p><math>c(n)</math>      The fixed codebook vector</p> <p><span style="border: 2px solid red; padding: 2px;"><math>v(n)</math></span>      The adaptive codebook vector</p> <p><span style="border: 2px solid green; padding: 2px;"><math>y(n) = v(n) * h(n)</math></span>      The filtered adaptive codebook vector</p> <p><math>d_i</math>      The distance between the immittance spectral frequencies <math>f_{i+1}</math> and <math>f_{i-1}</math></p> <p><span style="border: 2px solid blue; padding: 2px;"><math>h(n)</math></span>      The impulse response of the weighted synthesis filter</p> <p><math>H(z)W(z)</math>      The weighted synthesis filter</p>
<p>b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;</p>	<p>For the accused products, each pitch prediction error calculating device comprises <b>a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector</b>.</p>

The adaptive codebook gain is then found by

$$g_p = \frac{\sum_{n=0}^{63} x(n)y(n)}{\sum_{n=0}^{63} y(n)y(n)}, \quad \text{bounded by } 0 \leq g_p \leq 1.2, \quad (39)$$

where  $y(n) = v(n) * h(n)$  is the filtered adaptive codebook vector (zero-state response of  $H(z)W(z)$  to  $v_i(n)$ ). To insure stability, the adaptive codebook gain  $g_p$  is bounded by 0.95, if the adaptive codebook gains of the previous subframes have been small and the LP filters of the previous subframes have been close to being unstable.

Then the following operations are repeated for each subframe:

- The target signal  $x(n)$  is computed by filtering the LP residual through the weighted synthesis filter  $W(z)H(z)$  with the initial states of the filters having been updated by filtering the error between LP residual and excitation (this is equivalent to the common approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal).

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

For the accused products, each pitch prediction error calculating device comprises an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector.

### 5.8.3 Codebook search

The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is

$$x_2(n) = x(n) - g_p y(n), \quad n = 0, \dots, 63, \quad (40)$$

d) a combiner circuit for combining the amplified convolved pitch codevector with

For the accused products, each pitch prediction error calculating device comprises a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.



<p>the pitch search target vector to thereby produce the pitch prediction error.</p>	<h3>5.8.3 Codebook search</h3> <p>The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is</p> $\underline{x_2(n)} = \underline{x(n)} - \underline{g_p y(n)}, \quad n = 0, \dots, 63, \quad (40)$ <p>Then the following operations are repeated for each subframe:</p> <ul style="list-style-type: none"> <li>- The target signal <math>x(n)</math> is computed by filtering the LP residual through the weighted synthesis filter <math>W(z)H(z)</math> with the initial states of the filters having been updated by filtering the error between LP residual and excitation (this is equivalent to the common approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal).</li> <li>- The impulse response, <math>h(n)</math> of the weighted synthesis filter is computed.</li> <li>- Closed-loop pitch analysis is then performed (to find the pitch lag and gain), using the target <math>x(n)</math> and impulse response <math>h(n)</math>, by searching around the open-loop pitch lag. Fractional pitch with 1/4th or 1/2nd of a sample resolution (depending on the mode and the pitch lag value) is used. The interpolating filter in fractional pitch search has low pass frequency response. Further, there are two potential low-pass characteristics in the the adaptive codebook and this information is encoded with 1 bit.</li> <li>- The target signal <math>x(n)</math> is updated by removing the adaptive codebook contribution (filtered adaptive codevector), and this new target, <math>x_2(n)</math>, is used in the fixed algebraic codebook search (to find the optimum innovation).</li> </ul>
<p align="center"><b>Claim 42</b></p>	
<p>A network element as defined in claim 41, wherein said pitch gain calculator comprises a means for calculating said pitch gain <math>b^{(j)}</math> using the relation:</p>	<p>The accused products comprise a pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain <math>b^{(j)}</math> using the relation:</p> $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ <p>Additionally, <math>j=0, 1, 2, \dots, K</math>, and <math>K</math> corresponds to a number of signal paths, and where <math>x</math> is said pitch search target vector and <math>y^{(j)}</math> is said convolved pitch codevector.</p>



$b^{(j)} = x^T y^{(j)} / \ y^{(j)}\ ^2$ <p>where <math>j=0, 1, 2, \dots, K</math>, and <math>K</math> corresponds to a number of signal paths, and where <math>x</math> is said pitch search target vector and <math>y^{(j)}</math> is said convolved pitch codevector.</p>	<div> <div>Release 10</div> <div>26</div> <div>3GPP TS 26.190 V10.0.0 (2011-03)</div> </div> <div> <div>The adaptive codebook gain is then found by</div> <div> <math display="block">g_p = \frac{\sum_{n=0}^{63} x(n)y(n)}{\sum_{n=0}^{63} y(n)y(n)}</math> <div> <div>bounded by <math>0 \leq g_p \leq 1.2</math>,</div> <div>( 39 )</div> </div> </div> <div> <p>where <math>y(n) = v(n) * h(n)</math> is the filtered adaptive codebook vector (zero-state response of <math>H(z)W(z)</math> to <math>v_i(n)</math>). To insure stability, the adaptive codebook gain <math>g_p</math> is bounded by 0.95, if the adaptive codebook gains of the previous subframes have been small and the LP filters of the previous subframes have been close to being unstable.</p> </div> </div>
<div>Claim 44</div> <div> <div> <p>A network element as defined in claim 37, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.</p> </div> </div>	

<p>calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.</p>	<h2 style="text-align: center;">5.7 Adaptive codebook</h2> <p>In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, <u>wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device.</u> One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</p> <p>The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>. Note that 1 bit is used to encode the chosen path.</p> <p>Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where <math>\mathbf{b}_{LP}=[0.18,0.64,0.18]</math>. <u>The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector.</u> For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> $v(n) = \sum_{i=1}^1 b_{LP}(i+1)v'(n+i).$
<b>Claim 55</b>	
<p>A pitch analysis method for producing a set of pitch codebook parameters:</p>	<p>The accused products practice a pitch analysis method for producing a set of pitch codebook parameters.</p> <p>For example, the accused products practice the Adaptive Multi-Rate – Wideband (AMR-WB) speech codec standard. See 3GPP TS26.190 v10.0.0 (2011-03). Under this standard, the accused products comprise <b>a pitch analysis device</b> for producing <b>a set of pitch codebook parameters</b>.</p>



	<p><b>4.3 Principles of the <u>adaptive multi-rate wideband speech encoder</u></b></p> <p>The AMR-WB codec consists of nine source codecs with bit-rates of 23.85, 23.05, 19.85, 18.25, 15.85, 14.25, 12.65, 8.85 and 6.60 kbit/s.</p> <p><u>The codec is based on the code-excited linear predictive (CELP) coding model.</u> The input signal is pre-emphasized</p> <p><u>The CELP speech synthesis model</u> is shown in Figure 1. In this model, the excitation signal at the input of the short-term LP synthesis filter is constructed by adding two excitation vectors from adaptive and fixed (innovative) codebooks. The speech is synthesized by feeding the two properly chosen vectors from these codebooks through the short-term synthesis filter. The optimum excitation sequence in a codebook is chosen <u>using an analysis-by-synthesis search procedure</u> in which the error between the original and synthesized speech is minimized according to a perceptually weighted distortion measure.</p> <p><b>5.7 Adaptive codebook</b></p> <p><u>Adaptive codebook search</u> is performed on a subframe basis. It consists of performing closed loop pitch search, and then computing <u>the adaptive codevector</u> by interpolating the past excitation at the selected fractional pitch lag.</p> <p><u>The adaptive codebook parameters (or pitch parameters) are the delay and gain of the pitch filter.</u> In the search stage, the excitation is extended by the LP residual to simplify the closed-loop search.</p>
<p>generating a pitch code vector from a pitch codebook search device based on a digitized input audio data, wherein said digitized input audio data represents an input audio signal that has been sampled and</p>	<p>The accused products generate a pitch code vector from a pitch codebook search device based on a digitized input audio data, wherein said digitized input audio data represents an input audio signal that has been sampled and digitized.</p> <p>For example, the accused devices comprise <b>a pitch codebook search device</b> that is configured to generate <b>a pitch code vector</b> based on a digitized input audio data.</p> <p><b>5.7 Adaptive codebook</b></p> <p><u>Adaptive codebook search</u> is performed on a subframe basis. It consists of performing closed loop pitch search, and then computing <u>the adaptive codevector</u> by interpolating the past excitation at the selected fractional pitch lag.</p> <p><u>The adaptive codebook parameters (or pitch parameters) are the delay and gain of the pitch filter.</u> In the search stage, the excitation is extended by the LP residual to simplify the closed-loop search.</p>



digitized;	<p>The digitized input audio signal represents an input audio signal that has been sampled and digitized.</p> <h2>4.1 Functional description of audio parts</h2> <p>The analogue-to-digital and digital-to-analogue conversion will in principle comprise the following elements:</p> <ol style="list-style-type: none"> <li>1) Analogue to uniform digital PCM <ul style="list-style-type: none"> <li>- microphone;</li> <li>- input level adjustment device;</li> <li>- input anti-aliasing filter;</li> <li>- <u>sample-hold device sampling at 16 kHz;</u></li> <li>- <u>analogue-to-uniform digital conversion to 14-bit representation.</u></li> </ul> </li> </ol> <p>The uniform format shall be represented in two's complement.</p> <h2>4.2 Preparation of speech samples</h2> <p><u>The encoder is fed with data comprising of samples with a resolution of 14 bits left justified in a 16-bit word. The decoder outputs data in the same format. Outside the speech codec further processing must be applied if the traffic data occurs in a different representation.</u></p>
a) in at least two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data, calculating, for each signal path, a pitch prediction	<p>In at least two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data, the accused products calculate, for each signal path, a pitch prediction error of said pitch codevector from said pitch codebook search device.</p>

error of said pitch codevector from said pitch codebook search device:

## 5.7 Adaptive codebook

In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.

The low pass filter used in the second path is in the form  $B_{LP}(z)=0.18z+0.64+0.18z^{-1}$ . Note that 1 bit is used to encode the chosen path.

Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook  $v(n)$ ,  $v(n) = v'(n)$  in the first path, or  $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)$  in the second path, where  $\mathbf{b}_{LP}=[0.18,0.64,0.18]$ . The path which results in minimum energy of the target signal  $x_2(n)$  defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes,  $v(n)$  is always

$$v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$$

For example, the accused products must support at a minimum the AMR-WB codec set, including the 6.6, 8.85, and 12.65 Kbit/s codes under the HD Voice Logo License Agreement. See Annex C for minimum HD Voice requirements.

## 5.7 Adaptive codebook

In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.

The low pass filter used in the second path is in the form  $B_{LP}(z)=0.18z+0.64+0.18z^{-1}$ . Note that 1 bit is used to encode the chosen path.

Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook  $v(n)$ ,  $v(n) = v'(n)$  in the first path, or  $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)$  in the second path, where  $\mathbf{b}_{LP}=[0.18,0.64,0.18]$ . The path which results in minimum energy of the target signal  $x_2(n)$  defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes,  $v(n)$  is always

$$v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$$

b) in at least one of said at least two signal paths, filtering the pitch codevector before supplying said pitch codevector for calculation of said pitch prediction error of said at least one signal path; and

For the accused products, in at least one of said at least two signal paths, the accused products filter the pitch codevector before supplying said pitch codevector for calculation of said pitch prediction error of said at least one signal path.



## 5.7 Adaptive codebook

In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.

The low pass filter used in the second path is in the form  $B_{LP}(z)=0.18z+0.64+0.18z^{-1}$ . Note that 1 bit is used to encode the chosen path.

Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook  $v(n)$ ,  $v(n) = v'(n)$  in the first path, or  $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)$  in the second path, where  $\mathbf{b}_{LP}=[0.18,0.64,0.18]$ . The path which results in minimum energy of the target signal  $x_2(n)$  defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes,  $v(n)$  is always

$$v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$$

c) comparing the pitch prediction errors calculated in said at least two signal paths, choosing the signal path having the lowest calculated pitch prediction error, and selecting the set of pitch codebook parameters

The accused products compare the pitch prediction errors calculated in said at least two signal paths, choosing the signal path having the lowest calculated pitch prediction error, and selecting the set of pitch codebook parameters associated to the chosen signal path.

<p>associated to the chosen signal path.</p>	<h2 data-bbox="470 253 978 297">5.7 Adaptive codebook</h2> <p data-bbox="470 321 1776 545">In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. <u>The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</u></p> <p data-bbox="470 570 1745 634">The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>. Note that 1 bit is used to encode the chosen path.</p> <p data-bbox="506 678 1793 870">Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where <math>b_{LP}=[0.18,0.64,0.18]</math>. <u>The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector.</u> For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$
<p align="center"><b>Claim 56</b></p>	
<p>38. A pitch analysis method as defined in claim 55, wherein, in one of said at least two signal paths, no filtering of the pitch codevector is performed before supplying said pitch codevector to a pitch</p>	<p>For the accused products, <u>one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.</u></p>



prediction error calculating device.	<h2 style="text-align: center;">5.7 Adaptive codebook</h2> <p>In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. <u>One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths.</u> The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.</p> <p>The low pass filter used in the second path is in the form <math>B_{LP}(z)=0.18z+0.64+0.18z^{-1}</math>, Note that 1 bit is used to encode the chosen path.</p> <p>Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, <u>there are two possibilities to generate the adaptive codebook <math>v(n)</math>, <math>v(n) = v'(n)</math> in the first path, or <math>v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)</math> in the second path, where</u></p> <p><math>\mathbf{b}_{LP}=[0.18,0.64,0.18]</math>. The path which results in minimum energy of the target signal <math>x_2(n)</math> defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes, <math>v(n)</math> is always</p> $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i).$
<b>Claim 59</b>	
A pitch analysis method as defined in claim 55, wherein calculating a pitch prediction error in each signal path comprises: a) convolving the pitch codevector with a weighted synthesis filter impulse response	<p>The accused products <b>convolve the pitch codevector</b> with <b>a weighted synthesis filter impulse response signal</b> and <b>therefore calculate a convolved pitch codevector</b>.</p> <p>For example, as defined in the technical specification for the AMR-WB speech codec, <math>v(n)</math> corresponds to the pitch codebook vector and <math>h(n)</math> corresponds to the impulse response of the weighted synthesis filter. The pitch codebook vector is convolved with the weighted synthesis impulse response to calculate <math>y(n)</math>, a convolved pitch code vector.</p>

<p>signal and therefore calculating a convolved pitch codevector;</p>	<div> <div>Release 10</div> <div>10</div> <div>3GPP TS 26.190 V10.0.0 (2011-03)</div> </div> <p> <math>res_{LP}(n)</math>      The LP residual signal  <math>c(n)</math>      The fixed codebook vector  <math>v(n)</math>      The adaptive codebook vector  <math>y(n) = v(n) * h(n)</math>      The filtered adaptive codebook vector </p> <p> <math>d_i</math>      The distance between the immittance spectral frequencies <math>f_{i+1}</math> and <math>f_{i-1}</math>  <math>h(n)</math>      The impulse response of the weighted synthesis filter  <math>H(z)W(z)</math>      The weighted synthesis filter </p>
<p>b) calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;</p>	<div> <div>Release 10</div> <div>26</div> <div>3GPP TS 26.190 V10.0.0 (2011-03)</div> </div> <p>The accused products calculate a pitch gain in response to the convolved pitch codevector and a pitch search target vector.</p> <p>The adaptive codebook gain is then found by</p> $g_p = \frac{\sum_{n=0}^{63} x(n) y(n)}{\sum_{n=0}^{63} y(n) y(n)}, \quad \text{bounded by } 0 \leq g_p \leq 1.2, \quad (39)$ <p>where <math>y(n) = v(n) * h(n)</math> is the filtered adaptive codebook vector (zero-state response of <math>H(z)W(z)</math> to <math>v_i(n)</math>). To insure stability, the adaptive codebook gain <math>g_p</math> is bounded by 0.95, if the adaptive codebook gains of the previous subframes have been small and the LP filters of the previous subframes have been close to being unstable.</p>



	<p>Then the following operations are repeated for each subframe:</p> <ul style="list-style-type: none"> <li>- The target signal <math>x(n)</math> is computed by filtering the LP residual through the weighted synthesis filter <math>W(z)H(z)</math> with the initial states of the filters having been updated by filtering the error between LP residual and excitation (this is equivalent to the common approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal).</li> </ul>
c) multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and	<p>The accused products multiply the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector.</p> <h3>5.8.3 Codebook search</h3> <p>The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is</p> $x_2(n) = x(n) - g_p y(n), \quad n = 0, \dots, 63, \quad (40)$
d) combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.	<p>The accused products combine the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.</p> <h3>5.8.3 Codebook search</h3> <p>The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is</p> $x_2(n) = x(n) - g_p y(n), \quad n = 0, \dots, 63, \quad (40)$

	<p>Then the following operations are repeated for each subframe:</p> <ul style="list-style-type: none"> <li>- The target signal <math>x(n)</math> is computed by filtering the LP residual through the weighted synthesis filter <math>W(z)H(z)</math> with the initial states of the filters having been updated by filtering the error between LP residual and excitation (this is equivalent to the common approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal).</li> <li>- The impulse response, <math>h(n)</math> of the weighted synthesis filter is computed.</li> <li>- Closed-loop pitch analysis is then performed (to find the pitch lag and gain), using the target <math>x(n)</math> and impulse response <math>h(n)</math>, by searching around the open-loop pitch lag. Fractional pitch with 1/4th or 1/2nd of a sample resolution (depending on the mode and the pitch lag value) is used. The interpolating filter in fractional pitch search has low pass frequency response. Further, there are two potential low-pass characteristics in the the adaptive codebook and this information is encoded with 1 bit.</li> <li>- The target signal <math>x(n)</math> is updated by removing the adaptive codebook contribution (filtered adaptive codevector), and this new target, <math>x_2(n)</math>, is used in the fixed algebraic codebook search (to find the optimum innovation).</li> </ul>
<b>Claim 60</b>	
<p>A pitch analysis method as defined in claim 59, wherein said pitch gain calculator comprises a means for calculating said pitch gain <math>b^{(j)}</math> using the relation:</p> $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ <p>where <math>j=0, 1, 2, \dots, K</math>, and <math>K</math> corresponds to a number of signal paths, and where <math>x</math> is said pitch search target</p>	<p>The accused products comprise a pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain <math>b^{(j)}</math> using the relation:</p> $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ <p>Additionally, <math>j=0, 1, 2, \dots, K</math>, and <math>K</math> corresponds to a number of signal paths, and where <math>x</math> is said pitch search target vector and <math>y^{(j)}</math> is said convolved pitch codevector.</p>

<p>vector and <math>y^j</math> is said convolved pitch codevector.</p>	<div data-bbox="470 215 1803 248"> <div>Release 10</div> <div>26</div> <div>3GPP TS 26.190 V10.0.0 (2011-03)</div> </div> <div data-bbox="470 297 978 329"> <div>The adaptive codebook gain is then found by</div> </div> <div data-bbox="785 362 1724 472"> <div> <math display="block">g_p = \frac{\sum_{n=0}^{63} x(n)y(n)}{\sum_{n=0}^{63} y(n)y(n)}</math> </div> <div> <div>bounded by <math>0 \leq g_p \leq 1.2</math>,</div> <div>( 39 )</div> </div> </div> <div data-bbox="470 505 1803 610"> <p>where <math>y(n) = v(n) * h(n)</math> is the filtered adaptive codebook vector (zero-state response of <math>H(z)W(z)</math> to <math>v_i(n)</math>). To insure stability, the adaptive codebook gain <math>g_p</math> is bounded by 0.95, if the adaptive codebook gains of the previous subframes have been small and the LP filters of the previous subframes have been close to being unstable.</p> </div>
<div>Claim 62</div>	
<p>A pitch analysis method as defined in claim 55, wherein calculating said pitch prediction error, in each signal path, comprises calculating an energy of the corresponding pitch prediction error, and wherein comparing the pitch prediction errors comprises comparing the energies of said pitch prediction errors of the different signal paths and choosing as the</p>	<p>For the accused products, wherein calculating said pitch prediction error, in each signal path, comprises <b>calculating an energy of the corresponding pitch prediction error</b>, and wherein comparing the pitch prediction errors comprises <b>comparing the energies of said pitch prediction errors of the different signal paths and choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error</b>.</p>



signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

## 5.7 Adaptive codebook

In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity doesn't necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated to respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device. One of these two paths comprises a low-pass filter for filtering the pitch codevector and the pitch prediction error is calculated for these two signal paths. The signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain.

The low pass filter used in the second path is in the form  $B_{LP}(z)=0.18z+0.64+0.18z^{-1}$ . Note that 1 bit is used to encode the chosen path.

Thus, for 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s modes, there are two possibilities to generate the adaptive codebook  $v(n)$ ,  $v(n) = v'(n)$  in the first path, or  $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)$  in the second path, where  $b_{LP}=[0.18,0.64,0.18]$ . The path which results in minimum energy of the target signal  $x_2(n)$  defined in Equation (40) is selected for the filtered adaptive codebook vector. For 6.60 and 8.85 kbit/s modes,  $v(n)$  is always

$$v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)$$