

METHOD AND ARRANGEMENT FOR RECONSTRUCTION OF A RECEIVED
SPEECH SIGNAL

FIELD OF INVENTION

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The present invention relates to a method of reconstructing a speech signal that had been transmitted over a radio channel. The radio channel transmits either fully analogous speech information or digitally encoded speech information. In this latter case, however, the speech information is not speech encoded with linear predictive coding; in other words, it is not assumed that the speech information has been processed in a linear predictive speech encoder on the transmitter side. More specifically, the invention relates to a method for recreating from a received speech signal that may possibly have been subjected to disturbances, such as noise, interference or fading, a speech signal with which the effects of these disturbances have been minimized.

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The invention also relates to an arrangement for carrying out the method.

DESCRIPTION OF THE BACKGROUND ART

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It is known in the transmission of digitalized speech information from a transmitter to a receiver to encode and decode on the transmitter side and to decode the speech information on the receiver side in accordance with a linear predictive method. LPC (LPC = Linear Predictive Coding) is an energetic method of analyzing speech information, since it enables good speech quality to be achieved already at low bit rates. Linear predictive coding, LPC, generates reliable

estimates of speech parameters while being relatively effective calculatively at the same time. GSM EFR (GSM = Global System for Mobile communication; EFR = Enhanced Full Rate), the GSM standards improved speech encoder for full rate, constitute an example of linear predictive coding, LPC. This coding enables the receiver of a speech signal, which may have been transmitted by radio for instance, to correct certain types of errors that have occurred in the transmission and to conceal other types of error. The methods of frame substitution and error muting or suppression described in Draft GSM EFR 06.61, "Substitution and muting of lost frames for enhanced full rate speech traffic channels", ETSI, 1966, and ITU Study Group 15 contribution to question 5/15, "G.728 Decoder Modifications for Frame Erasure Concealment", AT&T, February 1995, based on the standard G.728, "Coding of speech at 16 kbps using Low Delay - Code Excited Linear Prediction (LD-CELP)", ITU, Geneva, 1992 can be mentioned as examples of procedures of this kind. For instance, U.S. Patent Specification 5,233,660 teaches a digital speech encoder and speech decoder that operate in accordance with the LD-CELP principle.

Because speech information is encoded in accordance with alternative coding algorithms, such as pulse code modulation, PCM, for instance, it is known to repeat the preceding data word when an error occurs in a given data word. The article "Waveform Substitution Techniques for Recovering Missing Speech Segments in Packet Voice Communications", IEEE Transactions on Acoustics, Speech and Signal Processing, Vol. ASSP-34, No. 6, Dec. 1986, pp. 1440-1447 by David J. Goodman et al, describes how speech information that has been lost in a PCM transmission between a transmitter and a receiver is

replaced on the receiver side with information that has been extracted from earlier received information.

In the case of systems in which speech information is modulated in accordance with adaptive differential pulse code modulation, ADPCM, several methods are known for suppressing errors and restricting high signal amplitudes, wherein the state in decoding filters is modified. M. Suzuki and S. Kubota describe in the article, "A Voice Transmission Quality Improvement Scheme for Personal Communication Systems - Super Mute Scheme", NTT Wireless Systems Laboratories, Vol. 4, 1995, pp. 713-717, a method of damping the received signal in the ADPCM transmission of speech information when data has been transmitted erroneously.

SUMMARY OF THE INVENTION

The present invention provides a solution to those problems that are caused in analog radio communications systems and in certain digital cordless telecommunications systems, such as DECT (DECT = Digital European Cordless Telecommunications), in which the radio signal is subjected to disturbances. The clicking sound that occurs when a received analog radio signal becomes too weak and is deluged in noise, for instance due to fading, is an example of one such problem.

The clicking and "bangs" that are generated when repeating a preceding data word in a digitalized speech signal due to registration of an error in the last received data word is an example of another problem.

A further problem concerns the interruption that occurs when a received digitalized speech signal is muted or suppressed because the error rate in the received data words is much too high.

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Accordingly, an object of the present invention is to create from a received speech signal that may have been subjected to disturbances during its transmission from a transmitter to a receiver a speech signal with which the effects of these disturbances is minimized. These disturbances may have been caused by noise, interference or fading, for instance.

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This object is achieved in accordance with the proposed invention by generating from the received speech signal with the aid of signal modelling an estimated signal which is dependent on a quality parameter that denotes the quality of the received speech signal. The received speech signal and the estimated speech signal are then combined in accordance with a variable relationship which is also determined by said quality parameter and forms a reconstructed speech signal. When reception conditions cause a change in the speech quality of the received speech signal, the aforesaid relationship is changed and the quality of the reconstructed speech signal restored, thereby obtaining an essentially uniform or constant quality. The inventive method is characterized by the features set forth in the following Claim 1.

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A proposed arrangement functions to reconstruct a speech signal from a received speech signal. The arrangement includes a signal modelling unit in which an estimated speech signal corresponding to anticipated future values of the

received speech signal are created, and a signal combining unit in which the received signal and the estimated speech signal are combined in accordance with a variable relationship which is determined by a quality parameter. The proposed apparatus is characterized by the features set forth in Claim 20.

By reconstructing a received analog or digitalized speech signal, utilizing statistical properties of the speech signal, the speech quality experienced by the receiver can be improved considerably in comparison with the speech quality that it has hitherto been possible to achieve with the aid of the earlier known solutions in analog systems and digital systems respectively that utilize PCM transmission or ADPCM transmission.

Because reconstruction of the received speech signal takes into account the statistical properties of the speech signal, it is also possible to avoid the clicking and banging sound generated in PCM transmissions and ADPCM transmissions for instance, when a preceding data word in the speech signal is repeated due to registration of an error in the data word that was last received.

The interruptions that occur when a received digitalized speech signal is muted because the error rate in the received data word is excessively high can also be avoided by using instead on such occasions solely the estimated speech signal obtained with the proposed method.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 illustrates coding and decoding of speech information with the aid of linear predictive coding (LPC) in a known manner;

Figure 2 illustrates in principle how speech information is transmitted, received and reconstructed in accordance with the proposed method;

Figure 3 illustrates an example of a channel model that can be used with the inventive method;

Figure 4 is a block schematic illustrating the signal reconstruction unit in Figure 2;

Figure 5 is a block schematic illustrating the proposed signal modelling unit in Figure 4;

Figure 6 is a block schematic illustrating the excitation generating unit in Figure 5;

Figure 7 is a block schematic illustrating the proposed signal combining unit in Figure 4;

Figure 8 is a flowchart illustrating a first embodiment of the inventive signal combining method applied in the signal combining unit in Figure 7;

Figure 9 illustrates an example of a result that can be obtained when following the flowchart in Figure 8;

Figure 10 is a flowchart illustrating a second embodiment of the inventive signal combining method applied in the signal combining unit in Figure 7;

5 Figure 11 illustrates an example of a result that can be obtained when following the flowchart in Figure 10;

10 Figure 12 illustrates an example of how a quality parameter for a received speech signal varies over a sequence of received speech samples;

Figure 13 is a diagram illustrating the signal amplitude of the received speech signal referred to in Figure 12;

15 Figure 14 is a diagram illustrating the signal amplitude of the speech signal shown in Figure 13, said speech signal having been reconstructed in accordance with the proposed method;

20 Figure 15 is a block schematic illustrating application of the inventive signal reconstruction unit in an analog transmitter/receiver unit; and

25 Figure 16 is a block schematic illustrating the application of the inventive signal reconstruction unit in a transmitter/receiver unit which is intended for transmitting and receiving digitalized speech information.

30 The invention will now be described in more detail with reference to proposed embodiments thereof and also with reference to the accompanying drawings.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

Figure 1 illustrates coding of human speech in the form of speech information S with the aid of linear predictive coding, LPC, in a known manner. The linear predictive coding, LPC, assumes that the speech signal S can conceivably be generated by a tone generator 100 located in a resonance tube 110. The tone generator 100 finds correspondence in the human vocal cords and trachea which together with the oral cavity constitute the resonance tube 110. The tone generator 100 is characterized by the parameters intensity and frequency and is designated in this speech model excitation e and is represented by a source signal K . The resonance tube 110 is characterized by its resonance frequencies, the so-called formants, which are described by short-term spectrum $1/A$.

In the linear predictive coding process, LPC, the speech signal S is analyzed in an analyzing unit 120 by estimating and eliminating the underlying short-term spectrum $1/A$ and by calculating the excitation e of the remaining part of the signal, i.e. the intensity and frequency. Elimination of the short-term spectrum $1/A$ is effected in a so-called inverse filter 140 having transfer function $A(z)$, which is implemented with the aid of coefficients in a vector a that has been created in an LPC analyzing unit 180 on the basis of the speech signal S . The residual signal, i.e. the inverse filter output signal, is designated residual R . Coefficients $e(n)$ and a side signal c that describes the residual R and short-term spectrum $1/A$ respectively are transferred to a synthesizer 130. The speech signal $\hat{S}$ is reconstructed in the synthesizer 130 by a process which is the reverse of the process that was used when coding in the analyzing unit 120.

The excitation $e(n)$, obtained by analysis in an excitation analyzing unit 150 is used to generate an estimated source signal $\hat{K}$ in an excitation unit 160, $\hat{e}$. The short-term spectrum $1/A$, described by the coefficients in the vector A , is created in an LPC-synthesizer 190 with the aid of information from the side signal c . The vector A is then used to create a synthesis filter 170, with transfer function $1/A(z)$, representing the resonance tube 110 through which the estimated source signal $\hat{K}$ is sent and wherewith the reconstructed speech signal $\hat{S}$ is generated. Because the characteristic of the speech signal S varies with time, it is necessary to repeat the aforescribed process from 30 to 50 times per second in order to achieve acceptable speech quality and good compression.

The basic problem with linear predictive coding, LPC, resides in determining a short-term spectrum $1/A$ from the speech signal S . The problem is solved with the aid of a differential equation that expresses the sample concerned as a linear combination of preceding samples for each sample of the speech signal S . This is why the method is called linear predictive coding, LPC. The coefficients a in differential equations which describe a short-term spectrum $1/A$ must be estimated in the linear predictive analysis carried out in the LPC analyzing unit 180. This estimation is made by minimizing the square mean value of the difference δS between the actual speech signal S and the predicted speech signal $\hat{S}$. The minimizing problem is solved by the following two steps. There is first calculated a matrix of the coefficient values. An array of linear equations, so-called predictor

equations, are then solved in accordance with a method that guarantees convergence and a unique solution.

When generating voiced sounds, a resonance tube 110 is well able to represent the trachea and oral cavity, although in the case of nasal sounds the nose forms a lateral cavity which cannot be modelled into the resonance tube 110. However, some parts of these sounds can be captured by the residual R, while remaining parts cannot be transmitted correctly with the aid of simple linear predictive coding, LPC.

Certain consonant sounds are produced by a turbulent air flow which results in a whistling noise. This sound can also be represented in the predictor equations, although the representation will be slightly different because, as distinct from voiced sounds, the sound is not periodic. Consequently, the algorithm LPC must decide with each speech frame whether or not the sound is voiced, which it most often is in the case of vocal sounds, or unvoiced, as in the case of some consonants. If a given sound is judged to be a voiced sound, its frequency and intensity are estimated, whereas if the sound is judged to be unvoiced, only the intensity is estimated. Normally, the frequency is denoted by one digit value and the intensity by another digit value, and information concerning the type of sound concerned is given with the aid of an information bit which, for instance, is set to a logic one when the sound is voiced and to a logic zero when the tone is unvoiced. These data are included in the side signal c generated by the LPC analyzing unit 180. Other information that can be created in the LPC analyzing unit 180 and included in the side signal c are coefficients

which denote the short-term prediction, STP, and the long term prediction, LTP, respectively of the speech signal S , the amplification values that relate to earlier transmitted information, information relating to speech sound and non-speech sound respectively, and information as to whether the speech signal is locally stationary or locally transient.

Speech sounds that consist of a combination of voiced and unvoiced sounds cannot be represented adequately by simple linear predictive coding, LPC. Consequently, these sounds will be somewhat erroneously reproduced when reconstructing the speech signal $\hat{S}$

Those errors that always unavoidably occur when the short-term spectrum $1/A$ is determined from the speech signal S result in more information being encoded into the residual R than is necessary theoretically. For instance, the earlier mentioned nasal sounds will be represented by the residual R . In turn, this results in the residual R containing essential information as to how the speech sound shall sound. Linear predictive speech synthesis would give an unsatisfactory result in the absence of this information. Thus, it is necessary to transmit the residual R in order to achieve high speech quality. This is normally effected with the aid of a so-called code book which includes a table covering the most typical residual signals R . When coding, each obtained residual R is compared with all the values present in the code book and the value that lies closest to the calculated value is selected. The receiver has a code book which is identical to the code book used by the transmitter, and consequently only the code VQ that denotes the relevant residual R need be transmitted. Upon receipt of the signal,

the residual value R corresponding to the code VQ is taken from the receiver code book and a corresponding synthesis filter $1/A(z)$ is created. This type of speech transmission is designated code excited linear prediction, CELP. The code book must be large enough to include all essential variants of residuals R while, at the same time, being as small as possible, since this will minimize code book search time and make the actual codes short. By using two small code books of which one is permanent and the other is adaptive enables many codes to be obtained and also enables searches to be carried out quickly. The permanent code book contains a plurality of typical residual values R and can therewith be made relatively small. The adaptive code book is originally empty and is filled progressively with copies of earlier residuals R , which have different delay periods. The adaptive code book will thus function as a shift register and the value of the delay will determine the pitch of the sound generated.

Figure 2 shows how speech information S is transmitted, received and reconstructed r_{rec} in accordance with the proposed method. An incoming speech signal S is modulated in a modulating unit 210 in a transmitter 200. A modulated signal S_{mod} is then sent to a receiver 220, over a radio interface, for instance. However, during its transmission the modulated signal S_{mod} will very likely be subjected to different types of disturbances D , such as noise, interference and fading, among other things. The signal S'_{mod} that is received in the receiver 220 will therefore differ from the signal S_{mod} that was transmitted from the transmitter 200. The received signal S'_{mod} is demodulated in a demodulating unit 230, therewith generating a received speech signal r . The demodulating unit 230 also generates a quality

parameter q which denotes the quality of the received signal S'_{mod} and therewith indirectly the anticipated speech quality of the received speech signal r . A signal reconstruction unit 240 generates a reconstructed speech signal r_{rec} of essentially uniform or constant quality, on the basis of the received speech signal r and the quality parameter q .

The modulated signal S_{mod} may be a radio frequency modulated signal, which is either completely analog modulated with frequency modulation, FM, for instance, or is digitally modulated in accordance with one of the principles FSK (FSK = Frequency Shift Keying), PSK (PSK = Phase Shift Keying), MSK (MSK = Minimum Shift Keying) or the like. The transmitter and the receiver may be included in both a mobile station and a base station.

The disturbances D to which a radio channel is subjected often derive from multi-path propagation of the radio signal. As a result of multi-path propagation, the signal strength will, at a given point, be comprised of the sum of two or more radio beams that have travelled different distances from the transmitter and are therefore time-shifted in relation to one another. The radio beams may be added constructively or destructively, depending on the time shift. The radio signal is amplified in the case of constructive addition and weakened in the case of destructive addition, said signal being totally extinguished in the worst case. The channel model that describes this type of radio environment is called the Rayleigh model and is illustrated in Figure 3. Signal strength γ is given in a logarithmic scale along the vertical axis of the diagram, while time t is given in a linear scale along the horizontal axis. The value γ_0 denotes the long-term

mean value of the signal strength γ , and γ_t denotes the signal level at which the signal strength γ is so low as to result in disturbance of the transferred speech signal. During respective time intervals t_A and t_B , the receiver is located in a point where two or more radio beams are added destructively and the radio signal is subjected to a so-called fading dip. It is, *inter alia*, during these time intervals that the use of an estimated version of the received speech signal is applicable in the reconstruction of said signal in accordance with the inventive method. If the receiver moves at a constant speed through a static radio environment, the distance Δt between two immediately adjacent fading dips t_A and t_B will be generally constant and t_A will be of the same order of magnitude as t_B . Both Δt and t_A and t_B are dependent on the speed of the receiver and the wavelength of the radio signal. The distance between two fading dips is normally one-half wavelength, i.e. about 17 centimetres at a carrier frequency of 900 Mhz. When the receiver moves at a speed of 1 m/s, Δt will be roughly equal to 0.17 seconds and a fading dip will seldomly have a duration of more than 20 milliseconds.

Figure 4 illustrates generally how the signal reconstruction unit 240 in Figure 2 generates a reconstructed speech signal r_{rec} in accordance with the proposed method. A received speech signal r is taken into a signal modelling unit 500, in which an estimated speech signal $\hat{f}$ is generated. The received speech signal r and the estimated speech signal $\hat{f}$ are received by a single signal combining unit 700 in which the signals r and $\hat{f}$ are combined in accordance with a variable ratio. The ratio according to which the combination is

5 effected is decided by a quality parameter q , which is also taken into the signal combining unit 700. The quality parameter q is also used by the signal modelling unit 500, where it controls the method in which the estimated speech
10 signal $\hat{r}$ is generated. The quality parameter q may be based on the measured received signal strength, RSS, an estimate of the signal level of the desired radio signal C (C = Carrier) at the ratio C/I to the signal level of a disturbance signal I (I = Interferer) or a bit error rate signal or bad frame
15 signal created from the received radio signal. The reconstructed speech signal r_{rec} is delivered from the signal combining unit 700 as the sum of a weighted value of the received speech signal r and a weighted value of the estimated speech signal $\hat{r}$ where the respective weights for r and $\hat{r}$ can be varied so as to enable the reconstructed speech
20 signal r_{rec} to be comprised totally of either one of the signals r or $\hat{r}$.

Figure 5 is a block schematic illustrating the signal modelling unit 500 in Figure 4. The received speech signal r
25 is taken into an inverse filter 510, in which the signal r is inversely filtered in accordance with a transfer function $A(z)$, wherein the short-term spectrum $1/A$ is eliminated and the residual R is generated. Inverse filter coefficients a are generated in an LPC/LTP analyzing unit 520 on the basis
30 of the received speech signal r . The filter coefficients a are also delivered to a synthesis filter 580 with transfer function $1/A(z)$. The LPC/LTP analyzing unit 520 analyses the received speech signal r and generates a side signal c and the values b and L which denote characteristics of the signal r and constitute control parameters of an excitation
generating unit 530 respectively. The side signal c includes information relating to short-term prediction, STP, and long

term prediction, LTP, respectively of the signal r , appropriate amplification values for the control parameter B , information relating to speech sound and non-speech sound respectively, and information relating to whether the signal r is locally stationary or transient, and is delivered to a state machine 540 and the values b and L are sent to the excitation generating unit 530, in which an estimated source signal $\hat{K}$ is generated.

The LPC/LTP analyzing unit 520 and the excitation generating unit 530 are controlled respectively by the state machine 540 through the medium of control signals s_1 and s_2 , s_3 and s_4 , the output signals s_1 - s_6 of the state machine 540 being dependent on the quality parameter q and the side signal c . The quality parameter q generally controls the LPC/LTP analyzing unit 520 and the excitation generating unit 530 through the medium of the control signals s_1 - s_4 in a manner such that the long term prediction, LTP, of the signal r will not be updated if the quality of the received signal r is below a specific value, and such that the amplitude of the estimated source signal $\hat{K}$ is proportional to the quality of the signal r . The state machine 540 also delivers weighting factors s_5 and s_6 to respective multipliers 550 and 560, in which the residual R and the estimated source signal $\hat{K}$ are weighted before being summated in a summing unit 570.

The quality parameter q controls, through the medium of the state machine 540 and the weighting factors s_5 and s_6 , the ratio according to which the residual R and the estimated source signal $\hat{K}$ shall be combined in the summing unit 570 and form a summation signal C , such that the higher the

quality of the received speech signal r , the greater the weighting factor s_5 for the residual R and the smaller the weighting factor s_6 for the estimated source signal $\hat{K}$. The weighting factor s_5 is reduced with decreasing quality of the received speech signal r and the weighting factor s_6 increased to a corresponding degree, so that the sum of s_5 and s_6 will always be constant. The summation signal C , where $C = s_5 R + s_6 \hat{K}$ is filtered in the synthesis filter 580, therewith forming the estimated speech signal $\hat{r}$. The signal C is also returned to the excitation generating unit 530, in which it is stored to represent historic excitation values.

Since the inverse filter 510 and the synthesis filter 580 have intrinsic memory properties, it is beneficial not to update the coefficients of these filters in accordance with properties of the received speech signal r during those periods when the quality of this signal is excessively low. Such updating would probably result in non-optimal setting of the filter parameters a , which in turn would result in an estimated signal R of low quality, even some time after the quality of the received speech signal r has assumed a higher level. Consequently, in accordance with a refined variant of the invention, the state machine 540 creates the weighted values of the received speech signal r and the estimated speech signal $\hat{r}$ respectively through the medium of a seventh and an eighth control signal, these values being summated and utilized in allowing the LPC/LPT analysis to be based on the estimated speech signal $\hat{r}$ instead of on the received speech signal r when the quality parameter q is below a predetermined value q_c , and to allow the LPC/LPT analysis to be based on the received speech signal r when the quality

parameter q exceeds the value q_c . When q is stable above q_c , the seventh control signal is always set to logic one and the eighth signal to logic zero, whereas when q is stable beneath q_c , the seventh control signal is set to logic zero and the eighth signal is set to logic one. During intermediate transmission periods, the state machine 540 allocates values between zero and one to the control signals in relation to the current value of the quality parameter q . The sum of said control signals, however, is always equal to one.

The transfer functions of the inverse filter 510 and the synthesis filter 580 are always an inversion of one another, i.e. $A(z)$ and $1/A(z)$. According to a simplified embodiment of the invention, the inverse filter 510 is a high-pass filter having fixed filter coefficients a , and the synthesis filter 580 is a low-pass filter based on the same fixed filter coefficients a . In this simplified variant of the invention, the LPC/LTP analyzing unit 520 thus always delivers the same filter coefficients a , irrespective of the appearance of the received speech signal r .

Figure 6 is a block schematic illustrating the excitation generating unit in Figure 5. The values b and L are taken into a control unit 610, which is controlled by the signal s_2 from the state machine 540. The value b denotes a factor by which a given sample $\hat{e}(n+1)$ from a memory buffer 620 shall be multiplied, and the value L denotes a shift corresponding to L sample steps backwards in the excitation history, from which a given excitation $\hat{e}(n)$ shall be taken. Excitation history $\hat{e}(n+1)$, $\hat{e}(n+2)$, ..., $\hat{e}(n+N)$ from the signal C is stored in the memory buffer 620. The storage capacity of the memory buffer 620 will correspond to at least 150 samples,

i.e. $N = 150$, and information from the signal C is stored in accordance with the shift register principle, wherein the oldest information is shifted out, i.e. in this case erased, when new information is shifted in.

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When the LPC/LTP analysis judges the sound concerned to be a voiced sound, the control signal s_2 gives the control unit 610 the consent to deliver the values b and L to the memory buffer 620. The value L , which is created from the long term prediction, LTP, of the speech signal r , denotes the periodicity of the speech signal r , and the value b constitutes a weighting factor by which a given sample $\hat{e}(n+i)$ from the excitation history shall be multiplied in order to provide an estimated source signal $\hat{K}$ which generates an optimal estimated speech signal $\hat{f}$, through the medium of the summation signal C . The values b and L thus control the manner in which information is read from the memory buffer 620 and thereby form a signal H_v .

If in the LPC/LTP analysis a current sound is judged to be unvoiced, the control signal s_2 delivers to the control unit 610 instead an impulse to send a signal n to a random generator 630, wherewith the generator generates a random sequence H_u .

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The signal H_v and the random signal H_u are weighted in multiplication units 640 and 650 with respective factors s_3 and s_4 and are summated in a summing unit 660, wherein the estimated source signal $\hat{K}$ is generated in accordance with the expression $\hat{K} = s_3 H_v + s_4 H_u$. If the current speech sound is voiced, the factor s_3 is set to a logic one and the factor s_4 ,

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is set to a logic zero, whereas if the current speech sound is unvoiced, the factor s_3 is set to a logic zero and the factor s_4 to a logic one. At a transition from a voiced to an unvoiced sound, s_3 is reduced during a number of mutually sequential samples and s_4 is increased to a corresponding degree, whereas in the transition from an unvoiced to a voiced sound, s_4 and s_3 are respectively reduced and increased in a corresponding manner.

10 The summation signal C is delivered to the memory buffer 620 and therewith updates the excitation history $\hat{e}(n)$ sample by sample.

Figure 7 illustrates the signal combining unit 700 in Figure 4, in which the received speech signal r and the estimated speech signal $\hat{r}$ are combined. In addition to these signals, the signal combining unit 700 also receives the quality parameter q . On the basis of the quality parameter q , a processor 710 generates weighting factors α and β by which the respective received speech signal r and estimated speech signal $\hat{r}$ are multiplied in multiplying units 720 and 730 prior to being added in the summation unit 740, and form the reconstructed speech signal r_{rec} . The respective weighting factors α and β are varied from sample to sample, depending on the value of the quality parameter q . When the quality of the received speech signal r increases, the weight factor α is increased and the weighting factor β decrease to a corresponding extent. The reversed applies when the quality of the received speech signal r falls. However, the sum of α and β is always one.

The flowchart in Figure 8 illustrates how the received speech signal r and the estimated speech signal $\hat{r}$ are combined in the signal combining unit 700 in Figure 7 in accordance with a first embodiment of the inventive method. The processor 710 of the signal combining unit 700 includes a counter variable n which can be stepped between the values -1 and n_t+1 . The value n_t gives the number of consecutive speech samples during which the quality parameter q of the received radio signal can fall beneath or exceed a predetermined quality level γ_m before the reconstructed signal r_{rec} will be identical with the estimated speech signal $\hat{r}$ for the received speech signal r respectively, and during which speech samples the reconstructed speech signal r_{rec} will be comprised of a combination of the received speech signal r and the estimated speech signal $\hat{r}$. Thus, the larger the value of n_t , the longer the transition period t_t between the two signals r and $\hat{r}$.

In step 800, the counter variable n is given the value $n_t/2$ in order to ensure that the counter variable n will have a reasonable value should the flowchart land in step 840 in the reconstruction of the first speech sample. In step 805, the signal combining unit 700 receives a first speech sample of the received speech signal r . In step 810, it is ascertained whether or not a given quality parameter q exceeds a predetermined value. In this example, the received signal quality is allowed to represent the power level γ of the received radio signal. The power level γ is therefore compared in step 810 with a power level γ_0 that comprises the long term mean value of the power level γ of the received radio signal. If γ is higher than γ_0 , the reconstructed speech

signal r_{rec} is made equal to the received speech signal r in step 815, the counter variable n is set to logic one in step 820, and a return is made to step 805 in the flowchart. Otherwise, it is ascertained in step 825 whether or not the power level γ is higher than a predetermined level γ_t , which corresponds to the lower limit of an acceptable speech quality. If γ is not higher than γ_t , the reconstructed speech signal r_{rec} is made equal to the estimated speech signal $\hat{r}$ in step 830, the counter variable n is set to n_t in step 835, and a return is made to step 805 in the flowchart. If it should be found in step 825 that γ is higher than γ_t , the reconstructed speech signal r_{rec} is calculated in step 840 as the sum of a first factor α multiplied by the received speech signal r and a second factor β multiplied by the estimated speech signal $\hat{r}$. In this example, $\alpha = (n_t - n)/n_t$ and $\beta = n/n_t$, and hence r_{rec} is given by the expression

$$r_{rec} = (n_t - n) \times r / n_t + n \times \hat{r} / n_t.$$

The next speech sample of the received speech signal is taken in step 845, and it is ascertained in step 850 whether or not the corresponding power level γ of the received radio signal is higher than the level γ_m , which denotes the arithmetical mean value of γ_0 and γ_t , i.e. $\gamma_m = (\gamma_0 + \gamma_t)/2$, and if such is the case the counter variable n is counted down one increment in step 855 and it is ascertained in step 860 whether or not the counter variable n is smaller than zero. If it is found in step 860 that the counter variable n is smaller than zero, this indicates that the power level γ has exceeded the value γ_m during n_t consecutive samples and that the reconstructive speech signal r_{rec} can therefore be made equal to the received speech signal r . The flowchart is thus followed to step 815. If, in step 860, the counter variable n is found to be

greater than or equal to zero, the flowchart is executed to step 840 and a new reconstructed speech signal r_{rec} is calculated. If in step 850 the power level γ is lower than or equal to γ_m , the counter variable n is increased by one in
5 step 865. It is then ascertained in step 870 whether or not the counter variable n is greater than the value n_t and if such is the case this indicates that the signal level γ has fallen beneath the value γ_m during n_t consecutive samples and that the reconstructed speech signal r_{rec} should therefore be
10 made equal to the estimated speech signal $\hat{r}$. A return is therefore made to step 830 in the flowchart. Otherwise, the flowchart is executed to step 840 and a new reconstructed speech signal r_{rec} is calculated.

15 Figure 9 illustrates an example of a result that can be obtained when executing the flowchart in Figure 8. n_t has been set to 10 in the example. The power level γ of the received radio signal exceeds the long-term mean value γ_0 during the first four received speech samples 1-4.
20 Consequently, because the flowchart in Figure 8 only runs through steps 800-820, the counter variable n will therefore be equal to one during samples 2-5. Thus, the reconstructed speech signal r_{rec} will be identical with the received speech signal r during samples 1-4. The reconstructed speech signal
25 r_{rec} will be comprised of a combination of the received speech signal r and the estimated speech signal $\hat{r}$ during the following twelve speech samples 5-16, because the power level γ of the received radio signal with respect to these speech samples will lie beneath the long-term mean value γ_0 of the
30 power level of the received radio signal. For instance, the reconstructed speech signal r_{rec} or speech sample 5 will be

given by the expression $r_{\text{rec}} = 0.9r + 0.1\hat{r}$, because $n=1$, and for speech sample 14 will be given by the expression $r_{\text{rec}} = 0.2r + 0.8\hat{r}$, because $n=8$. The reconstructed speech signal r_{rec} will be identical with the estimated speech signal $\hat{r}$ in the case of speech sample 17-23, since the power level γ of the received radio signal in respect of the ten ($n_t=10$) nearest preceding sample 7-16 has fallen beneath the value γ_m and the power level γ of the radio signal in respect of sample 17-22 is lower than the value γ_m . The reconstructed speech signal r_{rec} will again be comprised of a combination of the received speech signal r and the estimated speech signal $\hat{r}$ during the terminating two samples 24 and 25, because the power level γ of the received radio signal in respect of speech samples 23 and 24 exceeds the power level γ_m but falls beneath the long-term mean value γ_0 . It can be mentioned by way of example that the reconstructed speech signal r_{rec} for speech sample 25 is given by the expression $r_{\text{rec}} = 0.1r + 0.9\hat{r}$, because $n=9$.

The flowchart in Figure 10 shows how the received speech signal r and the estimated speech signal $\hat{r}$ are combined in the signal combining unit 700 in Figure 7 in accordance with a second embodiment of the inventive method. A variable n in the processor 710 can also be stepped between the values -1 and n_t+1 in this embodiment. The value n_t also in this case denotes the number of consecutive speech samples during which the quality parameter q of the received radio signal may lie beneath or exceed respectively a predetermined quality level B_m before the reconstructed signal r_{rec} is identical with the estimated speech signal $\hat{r}$ and the received speech signal r respectively, and during which speech samples the

reconstructed speech signal r_{rec} is comprised of a combination of the received speech signal r and the estimated speech signal $\hat{f}$.

5 The counter variable n is allocated the value $n_t/2$ in step 1000, so as to ensure that the counter variable n will have a reasonable value if step 1040 in the flowchart should be reached when reconstructing the first speech sample. In step 1005, the signal combining unit 700 takes a first speech
10 sample of the received speech signal r . In step 1010, it is ascertained whether or not the quality parameter q , in this example represented by the bit error rate, BER, in respect of a data word corresponding to a given speech sample, exceeds a given value, i.e. whether or not the bit error rate, BER,
15 lies beneath a predetermined value B_0 . The bit error rate, BER, can be calculated, for instance, by carrying out a parity check on the received data word that represents said speech sample. The value B_0 corresponds to a bit error rate, BER, up to which all errors can either be corrected or
20 concealed completely. Thus, B_0 will equal 1 in a system in which errors are not corrected and cannot be concealed. The bit error rate, BER, is compared with the level B_0 in step 1010. If the bit error rate, BER, is lower than B_0 , the reconstructed speech signal r_{rec} is made equal to the received
25 speech signal r in step 1015, the counter variable n is set to one in step 1020, and a return is made to step 1005 in the flowchart. Otherwise, it is ascertained in step 1025 whether or not the bit error rate, BER, is higher than a predetermined level B_t that corresponds to the upper limit of
30 an acceptable speech quality. If the bit error rate, BER, is found to be higher than B_t , the reconstructed speech signal r_{rec} is made equal to the estimated speech signal $\hat{f}$ in step

1030, the counter variable n is set to n_t in step 1035, and a return is made to step 1005 in the flowchart. If the bit error rate, BER, is found to be lower than or equal to B_t in step 1025, the reconstructed speech signal r_{rec} is calculated

5 in step 1040 as the sum of a first factor α multiplied by the received speech signal r and a second factor β multiplied by the estimated speech signal $\hat{r}$. In this example, $\alpha = (n_t - n)/n_t$ and $\beta = n/n_t$, and hence r_{rec} is given by the expression $r_{rec} = (n_t - n) \times r / n_t + n \times \hat{r} / n_t$. The next speech sample of the

10 received speech signal is taken in in step 1045 and it is ascertained in step 1050 whether or not a corresponding bit error rate, BER, of the received data signal is lower than a level B_m which, for example, denotes the arithmetical mean value of B_0 and B_t , i.e. $B_m = (B_0 + B_t)/2$, and if such is the

15 case the counter variable n is counted down one increment in step 1055 and it is ascertained in step 1060 whether or not the counter variable n is less than zero. If the counter variable n in step 960 is less than zero, this indicates that the bit error rate, BER, has fallen beneath the value B_m

20 during n_t consecutive speech samples and that the reconstructed speech signal r_{rec} can therefore be made equal to the received speech signal r . The flowchart is thus executed to step 1015. If the counter variable n in step 1060 is greater than or equal to zero, the flowchart is executed

25 to step 1040 and a new reconstructed speech signal r_{rec} is calculated. If the bit error rate, BER, in step 1050 is higher than or equal to B_m , the counter variable n is increased by one in step 1065. It is then ascertained in step 1070 whether or not the counter variable n is greater than

30 the value n_t . If such is the case, this indicates that the bit error rate, BER, has exceeded the value B_m during n_t consecutive samples and that the reconstructed speech signal

r_{rec} should therefore be placed equal with the estimated speech signal $\hat{r}$. A return is therefore made to step 1030 in the flowchart. Otherwise, the flowchart is executed to step 1040 and a new reconstructed speech signal r_{rec} is calculated.

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A special case of the aforescribed example is obtained when q is allowed to constitute a bad frame indicator, BFI, wherein q can assume two different values, instead of allowing the quality parameter q to denote the bit error rate, BER, for each data word. If the number of errors in a given data word exceeds a predetermined value B_t , this is indicated by setting q to a first value, for instance a logic one, and by setting q to a second value, for instance a logic zero, when the number of errors is lower than or equal to B_t .

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a soft transition between the received speech signal r and the estimated speech signal $\hat{r}$ is obtained in this case, by weighting the signals r and $\hat{r}$ together with respective predetermined weighting factors α and β during a predetermined number of samples n_t . For instance, n_t may be four samples during which α and β are stepped through the values 0.75, 0.50, 0.25 and 0.00, and 0.25, 0.50, 0.75 and 1.00 respectively, or vice versa.

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Figure 11 shows an example of a result that can be obtained when running through the flowchart in Figure 10. n_t has been set to 10 in the example. The bit error rate, BER, of a received data signal is shown along the vertical axis of the diagram in Figure 11, and samples 1-25 of the received data signal are shown along the horizontal axis of said diagram, said data signal having been transmitted via a radio channel and represents speech information. The bit error rate, BER,

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is divided into three levels B_0 , B_m and B_t . A first level, B_0 , corresponds to a bit error rate, BER, which results in a perceptually error-free speech signal. In other words, the system is able to correct and/or conceal up to B_0-1 bit errors in each received data word. A second level, B_t , denotes a bit error rate, BER, of such high magnitude that corresponding speech signals will have an unacceptably low quality. A third level B_m constitutes the arithmetical mean value $B_m = (B_t+B_0)/2$ of B_t and B_0 .

The bit error rate, BER, of the received data signal is below the level B_0 during the first four speech samples 1-4 received. Consequently, the counter variable n is equal to one during samples 2-5 and the reconstructed speech signal r_{rec} is identical to the received speech signal r . During the following twelve speech samples 5-16, the reconstructed speech signal r_{rec} will be comprised of a combination of the received speech signal r and the estimated speech signal $\hat{r}$, since the bit error rate, BER, of the received data signal with respect to these speech samples will lie above B_0 . The reconstructed speech signal r_{rec} will be identical to the estimated speech signal $\hat{r}$ in the case of speech samples 17-23, since the bit error rate, BER, of the received data signal with respect to the ten ($n_t=10$) nearest preceding samples 7-16 has exceeded the value B_m and the bit error rate in respect of samples 17-22 is higher than the value B_m . The reconstructed speech signal r_{rec} will again be comprised of a combination of the received speech signal r and the estimated speech signal $\hat{r}$ during the two terminating samples 24 and 25, since the bit error rate, BER, of the received data

signal with respect to speech samples 23 and 24 is below the level B_m , but exceeds the level B_0 .

In a first and a second embodiment of the invention, the quality parameter q has been based on a measured power level γ of the received radio signal and a calculated bit error rate, BER, of a data signal that has been transmitted via a given radio channel and which represents the received speech signal r . Naturally, in a third embodiment of the invention, the quality parameter q can be based on an estimate of the signal level of the desired radio signal C in a ratio C/I to the signal level of a interference signal I . The relationship between the ratio C/I and the reconstructed speech signal r_{rec} will then be essentially similar to the relationship illustrated in Figure 8, i.e. the factor β is increased and the factor α decreased to a corresponding extent in the case of decreasing C/I , and the factor α is increased at the cost of factor β in the case of increasing C/I . Corresponding flowcharts will, in principle, correspond to Figure 8. Step 810 would differ insomuch that instead $C/I > C_0$, step 825 would differ insomuch that $C/I > C_t$ and step 850 would differ insomuch that $C/I > C_m$, but the same conditions will apply in all other respects.

Figure 12 illustrates diagrammatically how a quality parameter q for a received speech signal r can vary over a sequence of received speech samples r_n . The value of the quality parameter q is shown along the vertical axis of the diagram, and the speech samples r_n are presented along the

horizontal axis of the diagram. The quality parameter q for speech sample r_n received during a time interval t_A lies beneath a predetermined level q_t that corresponds to the lower limit for acceptable speech quality. The received speech signal r will therefore be subjected to disturbance during this time interval t_A .

Figure 13 illustrates diagrammatically how the signal amplitude A of the received speech signal r , referred to in Figure 12, varies over a time t corresponding to speech samples r_n . The signal amplitude A is shown along the vertical axis of the diagram and the time t is presented along the horizontal axis of said diagram. the speech signal r is subjected to disturbance in the form of short discordant noises or crackling/clicking sound, this being represented in the diagram by an elevated signal amplitude A of a non-periodic character.

Figure 14 illustrates diagrammatically how the signal amplitude A varies over a time t corresponding to speech samples r_n of a version r_{rec} of the speech signal r illustrated in Figure 13 that has been reconstructed in accordance with the inventive method. The signal amplitude A is shown along the vertical axis of the diagram and the time t is presented along the horizontal axis thereof. During that time interval t_A , in which the quality parameter q lies beneath the level q_t , the reconstructed speech signal will be comprised, either totally or partially, of an estimated speech signal $\hat{r}$ that has been obtained by linear prediction of an earlier received speech signal r whose quality

parameter q has exceeded q_t . The estimated speech signal $\hat{f}$ is therefore probably of better quality than the received speech signal r concerned. Thus, the reconstructed speech signal r_{rec} , which is comprised of a variable combination of the received speech signal r and an estimated version $\hat{f}$ of said speech signal, will have a generally uniform or constant quality irrespective of the quality of the received speech signal r .

Figure 15 illustrates the use of the proposed signal reconstruction unit 240 in an analog transmitter/receiver unit 1500, designated TRX, in a base station or in a mobile station. A radio signal RF_R from an antenna unit is received in a radio receiver 1510 which delivers a received intermediate frequency signal IF_R . The intermediate frequency signal IF_R is demodulated in a demodulator 1520 and an analog received speech signal r_A and an analog quality parameter q_A are generated. These signals r_A and q_A are sampled and quantized in a sampling and quantizing unit 1530, which delivers corresponding digital signals r and q respectively that are used by the signal reconstruction unit 240 to generate a reconstructed speech signal r_{rec} in accordance with the proposed method.

A transmitted speech signal S is modulated in a modulator 1540 in which an intermediate frequency signal IF_T is generated. The signal IF_T is radio frequency modulated and amplified in a radio transmitter 1550, and a radio signal RF_T is delivered for transmission to an antenna unit.

Figure 16 illustrates the use of the proposed signal reconstruction unit 240 in a transmitter/receiver unit 1600, designated TRX, in a base station or a mobile station that communicate ADPCM encoded speech information. A radio signal RF_R from an antenna unit is received in a radio receiver 1610 which delivers a received intermediate frequency signal IF_R . The intermediate frequency signal IF_R is demodulated in a demodulator 1620 which delivers an ADPCM encoded baseband signal B_R and a quality parameter q . The signal B_R is decoded in an ADPCM decoder 1630, wherein a received speech signal r is generated. The quality parameter q is taken in to the ADPCM decoder 1630 so as to enable resetting of the state of the decoder when the quality of the received radio signal RF_R is excessively low. The signals r and q are finally used by the signal reconstruction unit 240 to generate a reconstructed speech signal r_{rec} in accordance with the proposed method.

A transmitted speech signal S is encoded in an ADPCM encoder 1640, the output signal of which is an ADPCM encoded baseband signal B_T . The signal B_T is then modulated in a modulator 1650, wherein an intermediate frequency signal IF_T is generated. The signal IF_T is radio frequency modulated and amplified in a radio transmitter 1660, from which a radio signal RF_T is delivered for transmission to an antenna unit.

Naturally, the ADPCM decoder 1630 and the ADPCM encoder 1640 may equally as well be comprised of a logarithmic PCM decoder and logarithmic PCM encoder respectively when this form of

speech coding is applied in the system in which the transmitter/receiver unit 1600 operate.

CLAIMS

1. A method of reconstructing a speech signal from a received signal (r), using a signal model (500) and a quality parameter (q), characterized by creating through the medium of said signal model (500) an estimated signal ($\hat{r}$) that corresponds to anticipated future values of the received signal (r); combining said received signal (r) and said estimated signal ($\hat{r}$) and forming a reconstructed speech signal (r_{rec}), wherein said quality parameter (q) determines the ratio (α , β) in pursuance of which the combination takes place.
2. A method according to Claim 1, characterized by basing the quality parameter (q) on a measured power level (RSS, γ) of the received signal (r).
3. A method according to Claim 1, characterized by basing the quality parameter (q) on an estimated received signal level (C) of said received signal (r) in proportion (C/I) to the signal level of a disturbance signal (I).
4. A method according to Claim 1, characterized by basing said quality parameter (q) on a bit error rate (BER) that has been calculated from a digital representation of said signal (r).



5. A method according to Claim 1, characterized by basing said quality parameter (q) on a bad frame indicator (BFI) that has been calculated from a digital representation of said signal (r).

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6. A method according to any one of Claims 1-5, characterized by basing said signal model (500) on a linear prediction (LPC/LTP) of said received signal (r).

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7. A method according to Claim 6, characterized in that said linear prediction (LPC/LTP) generates coefficients that denote a short-term prediction (STP) of said received signal (r).

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8. A method according to Claims 6 or 7, characterized in that said linear prediction (LPC/LTP) generates coefficients that denote a long-term prediction (LTP) of said received signal (r).

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9. A method according to any one of Claims 6-8, characterized in that said linear prediction (LPC/LTP) generates amplification values (b) that relate to a history ($\hat{e}(n+1)$, $\hat{e}(n+2)$, ..., $\hat{e}(n+N)$) of said estimated signal ($\hat{r}$).

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10. A method according to any one of Claims 6-9, characterized in that said linear prediction (LPC/LTP)



* C L M S *

includes information (c) as to whether the received signal (r) shall be assumed to represent speech information or to represent information of a non-speech kind.

5 11. A method according to any one of Claims 6-10, characterized in that said linear prediction (LPC/LTP) includes information (c) as to whether said received signal (r) shall be assumed to represent a voiced sound or to represent an unvoiced sound.

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12. A method according to any one of Claims 6-11, characterized in that said linear prediction (LPC/LTP) contains information (c) as to whether said received signal (r) shall be assumed to be locally stationary or locally
15 transient.

13. A method according to any one of Claims 1-12, characterized in that said received signal (r) is a sampled and quantized analog modulated and transmitted speech signal.

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14. A method according to any one of Claims 1-12, characterized in that said received signal (r) is a digitally modulated and transmitted encoded signal.

25 15. A method according to any one of Claims 1-12, characterized in that said received signal (r) is generated



by decoding an adaptive differential pulse code modulated (ADPCM) signal.

16. A method according to any one of Claims 1-12, characterized in that said received signal (r) is generated by decoding a logarithmically pulse code modulated (PCM) signal.

17. A method according to Claim 1, characterized in that said ratio (α , β) is variable from solely denoting said received signal (r) to solely denoting said estimated signal ($\hat{r}$).

18. A method according to Claim 17, characterized in that transition from solely said received signal (r) to solely said estimated signal ($\hat{r}$) takes place during a transition period (t_t) of at least a number (n_t) of consecutive samples of said received signal (r) during which the quality parameter (q) for said received signal (r) is below a predetermined quality value (γ_t).

19. A method according to Claim 17, characterized in that transition from solely said estimated signal ($\hat{r}$) to solely said received signal (r) takes place during a transition period (t_t) of at least a number (n_t) of consecutive samples of said received signal (r) during which the quality parameter (q) for said received signal (r) exceeds a predetermined quality value (γ_t).



20. A method according to Claim 17, characterized in that the duration of said transition period (t_t) is decided by a predetermined but variable transition value (n_t).

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21. An arrangement for reconstructing a speech signal from a received signal (r) and including a signal modelling unit (500), characterized in that the signal modelling unit (500) functions to create an estimated signal ($\hat{r}$) corresponding to anticipated future values of said received signal (r); in that the arrangement includes a signal combining unit (700) which functions to combine said received signal (r) and said estimated signal ($\hat{r}$), therewith to form a reconstructed speech signal (r_{rec}), wherein the ratio (α , β) according to which the combination is effected is determined by a quality parameter (q).

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22. An arrangement according to Claim 21, characterized in that a processor (710) in said signal combining unit (700) delivers a first weighting factor (α) and a second weighting factor (β) on the basis of the value of said quality parameter (q) for each sample of said received signal (r).

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23. An arrangement according to Claim 22, characterized in that the signal combining unit (700) functions to form a first weighted value (αr) of said received signal (r) by multiplying said received signal (r) with said first weighting factor (α) in a first multiplier unit (720), and

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to form a second weighted value ($\beta \hat{r}$) of said estimated signal ($\hat{r}$) by multiplying said estimated signal ($\hat{r}$) with said second weighting factor (β) in a second multiplier unit (730), wherein the first (αr) and the second ($\beta \hat{r}$) weighted values according to said ratio (α, β) are combined in a first summing unit (740), and wherein said reconstructed signal (r_{rec}) is formed as a first summation signal.

24. An arrangement according to Claim 23, characterized in that a transition value (n_t) stored in said processor (710) denotes a smallest number of consecutive samples of said received signal (r) during which said first weighting factor (α) can be decreased incrementally from a highest value to a lowest value and said second weighting factor (β) can be increased incrementally from a lowest value to a highest value.

25. An arrangement according to Claim 23, characterized in that a transition value (n_t) stored in said processor (710) denotes a smallest number of consecutive samples of said received signal (r) during which said first weighting factor (α) can be increased incrementally from a lowest value to a highest value and said second weighting factor (β) can be decreased incrementally from a highest value to a lowest value.

26. An arrangement according to Claim 24 or 25, characterized in that said highest value is equal to one; and



in that said lowest value is equal to zero; and in that the sum $(\alpha+\beta)$ of said first weighting factor (α) and said weighting factor (β) is equal to one.

5 27. An arrangement according to any one of Claims 21-26, characterized in that said signal modelling unit (500) includes an analyzing unit (520) which creates in accordance with a linear predictive signal model (LPC/LTP) parameters (a, b, c, L) that depend on certain properties of said
10 received signal (r).

28. An arrangement according to Claim 27, characterized in that said parameters (a, b, c, L) include filter coefficients (a) of a first digital filter (510) and a second digital
15 filter (580) whose respective transfer functions $(A(z), 1/A(z))$ are the inverse of each other.

29. An arrangement according to Claim 28, characterized in that the first digital filter (510) is an inverse filter
20 $(A(z))$; and in that the second digital filter (580) is a synthesis filter $(1/A(z))$.

30. An arrangement according to any one of Claims 21-26, characterized in that the signal modelling unit (500)
25 includes a first digital filter (510) and a second digital filter (580) whose respective transfer functions $((A(z), 1/A(z))$ are the inverse of each other.



31. An arrangement according to Claim 30, characterized in that the first digital filter (510) has the character of a high-pass filter; and in that the second digital filter (580) has the character of a low-pass filter.

32. An arrangement according to any one of claims 28-31, characterized in that said first digital filter (510) functions to filter said received signal (r), therewith generating a residual signal (R).

33. An arrangement according to Claim 32, characterized in that said signal modelling unit (500) includes an excitation generating unit (530) that functions to generate an estimated signal ($\hat{K}$) that is based on three of said parameters (b, c, L) and a second summation signal (C), and a state machine (540) that functions to generate control signals (s_1 - s_6) that are based on said quality parameter (q) and one of said parameters (c).

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34. An arrangement according to Claim 33, characterized in that said signal modelling unit (500) includes a second summation unit (570) that functions to combine a third weighted value ($s_5 R$) of said residual signal (R) with a fourth weighted value ($s_6 \hat{K}$), therewith to generate said second summation signal (C).

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35. An arrangement according to Claim 34, characterized in that said second digital filter (580) functions to filter said second summation signal (C), therewith to generate said estimated signal ($\hat{r}$).

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36. An arrangement according to any one of Claims 34-35, characterized in that said excitation generating unit (530) includes a memory buffer (620) and a random generator (630).

10 37. An arrangement according to Claim 36, characterized in that said memory buffer (620) functions to store the historic values ($\hat{e}(n+1)$, $\hat{e}(n+2)$, ..., $\hat{e}(n+N)$) of said second summation signal (C).

15 38. An arrangement according to Claim 37, characterized in that said memory buffer (620) functions to generate on the basis of two of said parameters (b, L) a first signal (H_v) that represents a voice speech sound.

20 39. An arrangement according to Claim 38, characterized in that said random generator (630) functions to generate on the basis of said control signals (s_2) a second signal (H_u) that represents an unvoiced speech sound.

25 40. An arrangement according to Claim 39, characterized by a third summation unit (660) which functions to combine a third



weighted value $(s_3 H_v)$ of said first signal (H_v) with a fourth weighted value $(s_4 H_u)$ of said second signal (H_u) , therewith forming said estimated signal $(\hat{K})$.

5 41. An arrangement according to any one of Claims 21-40, characterized in that said received signal (r) is a sampled and quantized analog transmitted speech signal.

10 42. An arrangement according to any one of Claims 21-40, characterized in that said received signal (r) is a digitally modulated and transmitted encoded signal.

15 43. An arrangement according to Claim 42, characterized in that said received signal (r) is generated by decoding an adaptive differential pulse code modulated (ADPCM) signal.

44. An arrangement according to Claim 42, characterized in that said received signal (r) is generated by decoding a logarithmic pulse code modulated (PCM) signal.

