(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
22 March 2001 (22.03.2001)

PCT

(10) International Publication Number
WO 01/20807 A1

(51) International Patent Classification[7]: H04B 7/005

(21) International Application Number: PCT/SE00/01703

(22) International Filing Date: 5 September 2000 (05.09.2000)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
9903235-1 10 September 1999 (10.09.1999) SE

(71) Applicant: **TELEFONAKTIEBOLAGET LM ERICSSON (publ)** [SE/SE]; S-126 25 Stockholm (SE).

(72) Inventors: **HE, Ning**; Polhemsvägen 1A, S-191 34 Sollentuna (SE). **KLANG, Göran**; Råsundavägen 121A, S-169 50 Solna (SE). **RIPSTRAND, Torbjörn**; Näckrosvägen 43, S-169 37 Solna (SE). **ÖSTMAN, Thomas**; Öregrundsgatan 6, S-115 59 Stockholm (SE).

(74) Agent: **NORIN, Klas**; Ericsson Radio Systems AB, Common Patent Department, S-164 80 Stockholm (SE).

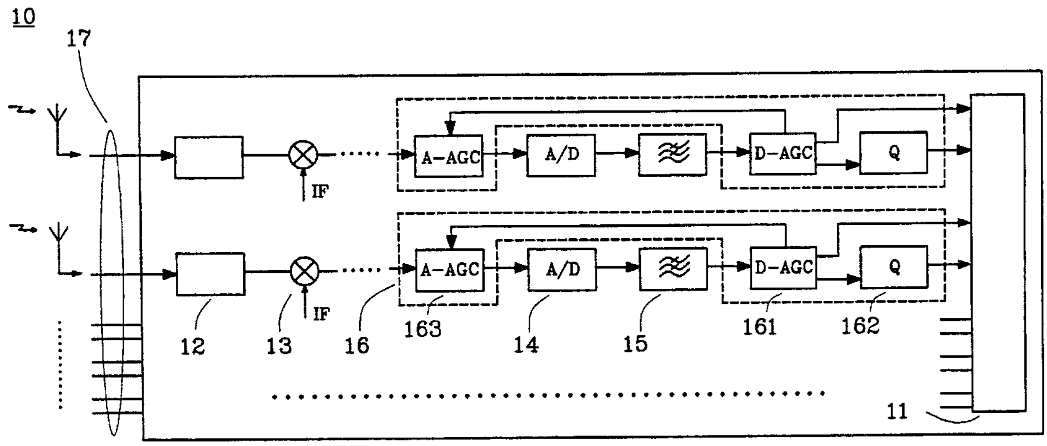
(81) Designated States (*national*): AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, BZ, CA, CH, CN, CR, CU, CZ, DE, DK, DM, DZ, EE, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, MZ, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, TZ, UA, UG, UZ, VN, YU, ZA, ZW.

(84) Designated States (*regional*): ARIPO patent (GH, GM, KE, LS, MW, MZ, SD, SL, SZ, TZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG).

Published:
— *With international search report.*
— *Before the expiration of the time limit for amending the claims and to be republished in the event of receipt of amendments.*

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

(54) Title: METHOD AND ARRANGEMENT FOR AUTOMATIC GAIN CONTROL OF RECEIVED SIGNALS



(57) Abstract: The present invention relates to a code division multiple access (CDMA) communication system, more particularly to an arrangement and a method in a receiver (10) to simplify and improve signal processing in the demodulator (11) of the receiver (10) by means of reducing the long term dynamic range of received signals and, thus, reducing the number of bits needed for coding of said signals. This is accomplished by a signal processing means (16) in the signal input paths in front of the demodulator (11), said means (16) including at least an improved digital AGC-unit (161), which scales incoming signal samples to a common and constant power level, and including a subsequent quantizer unit (162), which transforms the scaled signal samples to samples using a reduced number of bits. Due to this, the transmission bandwidth in the receiver unit is reduced and the signal processing in the demodulator is less complex.

WO 01/20807 A1

METHOD AND ARRANGEMENT FOR AUTOMATIC GAIN CONTROL OF RECEIVED SIGNALS

FIELD OF THE INVENTION

The present invention relates generally to telecommunication systems, more particularly to receiver units, preferably in code division multiple access (CDMA) communication systems, comprising means for automatic gain control of received signals.

BACKGROUND OF THE INVENTION

In code division multiple access (CDMA) based communication systems, a plurality of base stations cover each a certain geographic area and provide communication services to and from users within such areas. Characteristic for CDMA-systems is that the common transmission medium is shared between different users by assigning specific and unique code sequences, e.g. a pseudo-random code, to the uplink and downlink channels between base station and user equipment. These code sequences are used by the transmitters to transform signals into wideband spread spectrum signals. In the receiver units, e.g. base station or user equipment, a demodulator re-transforms said wideband signals from a specific transmitter into the original bandwidth by using the same code sequence as that transmitter while signals marked with different codes remain wideband signals and, thus, are interpreted by the receiver as part of the external background noise.

However, with regard to the physical radio communication channel, signals are subjected to, i.a., distance attenuation and propagation effects, e.g. multipath fading and shadowing due to obstacles in the radio path. Thus, the communication medium is to be modelled as a multipath channel with different time delays and attenuation factors in the different propagation paths.

Yet another aspect, specific to the multiple access technique used in CDMA, relates to the fact that all users transmit wideband signals potentially at the same time and using the same bandwidth. Due to this fact and the propagation effects as described above, a base station will receive wideband signals from different users within a large dynamic range, e.g. up to 80dB, depending on the received signal power. Additionally, as the received signal power from a single user may change during transmission and varies depending on the receiving antenna, each user transmission is subjected to an enlarged dynamic range. Therefore, in order to be able to receive signals from all users within the coverage area of, e.g., a base station, it is essential for said base station to provide means for supervision and control of the total received uplink transmission power per user as well as the total received uplink transmission power from all users in order to assure both that signals from all transmitting users, irrespective of signal propagation effects, arrive at the base station with virtually the same mean power and that the total interference level does not become too high with the result of a reduced system capacity. This is achieved by estimating the total received power at each base station antenna and using said estimate to regulate the output power of the transmitting user equipment for the respective base station.

Another measure to be able to handle signals with a large dynamic range is to apply an Automatic Gain Control (AGC-) unit in the analogue part of the receiver. JP 8-335 928 A shows, e.g., a receiver of a CDMA type mobile communication system including, i.a., an AGC-unit (30) controlling the gain according to the maximum amplitude of the analogue baseband signals and outputs a constant output signal (S30).

SUMMARY OF THE INVENTION

It is a first object of the present invention to achieve an arrangement and a method to simplify and improve processing of signals in the demodulator part of a receiver unit, preferably in a CDMA-based communication system.

It is another object of the present invention to simplify and improve said processing for sequences of digital signal samples by means of reducing the number of bits used for coding of said samples without loosing any information carried by said signals.

It is yet another object of the present invention to enable said demodulator to use basically identical signals that have been received from several signal inputs, e.g., base station antennas, in order to achieve a reliable signal to be used for demodulation.

It is still another object of the present invention to achieve an improved digital Automatic Gain Control (AGC-) unit which is less sensitive to external noise interference and which requires a minimum of hardware.

Briefly, these and other objects of the present invention are accomplished by signal processing means located in front of the demodulator in at least one of the signal input paths, said means including at least a digital AGC-unit and a subsequent quantizer unit. The digital AGC-unit scales said sequences of signal samples to a common and constant power level. Then, the quantizer unit can transform the scaled signal samples to samples using a reduced number of bits. In order to use the AGC efficiently, a control error is determined and used to maintain the signal quality at the demodulator input.

The invention is founded on the insight that input signals from the various antennas of a receiver have a large long term dynamic range, i.e. large differences in signal power due to varying reception properties, but normally a much smaller short term dynamic range, especially in CDMA-based communication systems. This implies that the corresponding digitised input signals must be coded as samples with a large number of redundant bits to be able to represent said large range. This number of coding bits can be reduced by transforming the input signals to a common power level that only represent the short term dynamic range.

As a first advantage, the present invention comprises an arrangement and a method that reduces the number of bits that is necessary to represent signal samples that are forwarded to a demodulator.

Thus, it is another advantage of the present invention that the required transmission bandwidth, i.e. the number of transmitted bits per second, in the receiver unit is reduced.

Thus, it is yet another advantage of the present invention that processing of said signal samples in the subsequent demodulator is significantly less complex, which implies decreasing power consumption and saving chip area.

Still another advantage of the present invention is an improved digital AGC-unit that is less sensitive to external noise interference. In a CDMA-based communication system, said AGC-unit can be implemented by reusing already existing power estimation means for received signals.

Yet another advantage of the present invention is that the demodulator can switch in a fast way to antennas that momentarily receive a signal best, e.g., in case of a softer handover.

Other objects, advantages and novel features of the invention will become apparent from the following detailed description of the invention when considered in conjunction with the accompanying drawings and claims.

BRIEF DESCRIPTION OF THE DRAWINGS

For a better understanding, reference is made to the following drawings and preferred embodiments of the invention.

Figure 1 shows an overview of a base station receiver and in particular the input uplink signal paths to the base station demodulator.

Figure 2 shows in greater detail the signal processing means for one of said signal paths according to the preferred embodiment of the present invention.

Figure 3a and 3b show two examples of assignment functions that elucidate the relationship between the power estimate for a signal sample and the scaling factor that is necessary to transform said signal sample to a certain target power level.

Figure 4 shows a flowchart that illustrates the main steps of the method according to the preferred embodiment of the present invention.

DETAILED DESCRIPTION

Figure 1 refers in general to a base station, e.g., in a CDMA-based communication system, and especially to that part of a base station receiver 10 that is responsible for forwarding of received signals to its demodulator 11. The base station provides communication services to user equipments within its cell, which is usually divided into a number of sectors, and is equipped with at least one antenna per sector, or preferably two antennas in order to improve signal reception by using diversity effects.

Transmissions from a user equipment within one of said sectors will be received mainly by the antennas that are directed to that specific sector but also from antennas of the neighbouring sectors. Thus, the base station can use various received copies of a transmitted signal from a user equipment. This is an important precondition, e.g., with respect to a softer handover, i.e. when a base station detects a better quality of received signals from another sector antenna and switches reception to that antenna. The received signals differ considerably in signal quality and are received at different power levels that cover a large dynamic range depending on, e.g., the different propagation and attenuation characteristics of the actual radio channel. In addition, said differences may also depend on the fact that the signal is received from an area outside the proper coverage area of an antenna. Therefore, the present invention intends to enable the base station receiver 10, and especially the demodulator part 11 of said receiver 10, to use the various received signal copies and retrieve an optimized signal that can be used for demodulation of the original user information.

Accordingly, the base station receiver part 10 comprises for each of its antennas a separate input path 17 for the uplink signals that are to be forwarded to the demodulator 11. Initially, the received high-frequency signals are pre-processed by amplifying and filtering means 12, down-converted to an intermediate frequency (IF) 13, and represented as sequences of 16-bit signal samples by an analog-to-digital converter 14 with a subsequent low-pass filter 15. Then, using the signal processing means 16 according to the present invention, the digitized input signals are transformed to a sufficiently high target power level that is common for each signal input paths 17 and, thus, making said signals applicable for the demodulator 11 as described above. For each of the signal input paths, said signal processing means 16 consists of at least a digital AGC-unit 161 for scaling the sequences of input signal samples and a quantizer unit 162 for reducing the number of bits that is necessary for coding of said signal samples. Optionally, the adaptation unit can also include a conventional analogue AGC-unit 163, which is applied, e.g., as a preceding signal attenuation unit for the analogue signal.

Figure 2 describes in more detail the different parts of the signal processing means 16 according to the present invention and how said parts co-operate according to the preferred embodiment of the invention. Said signal processing means can be divided into a number of blocks: The main blocks are a scaling block 21, which scales input signals in dependence of signal power estimates from a power estimating unit 22, and a quantiser block 23 that delivers a scaled output signal. Further, the signal adaptation unit contains offset compensation means 24 and, optionally, a conventional analogue AGC-unit 25.

The central part of said signal processing means consists of the scaling block 21 including means 211 for determining a scaling factor from estimated power values of a received input signal X[n] and means 212 for scaling said input signal X[n] to an output signal Y[n] that still carries the same information. When regarding said input signals as a sequence of digital signal samples $X_k[n]$, where k denotes the time interval index, scaling implies that each of these signal samples is adjusted to a common power target level σ_{ref}^2 by multiplication with an appropriately determined scaling factor α_k, i.e. $Y_k[n] = \alpha_k \cdot X_k[n]$. Thus, the scaling block 21 compensates the varying mean power levels of the input signal samples, i.e. reduces the long term dynamic range of a signal, which does not contain any user information but mainly represent the impact of the radio channel. This results in a scaled output signal Y[n] having a total dynamic range corresponding to the short term dynamic range of the input signal X[n], which, in general, is much smaller than said long term dynamic range.

Accordingly, coding of said signal samples does not longer need to represent the redundancy caused by the considerably different signal power levels. Instead, the quantiser unit 23 can reduce the number of coding bits per sample to a number that is sufficient for representing signal variations within the short term dynamic range.

When assuming a sequence of input signal samples $X_k[n]$ having an approximately constant variance σ_k^2 per sample, scaling can be performed by a multiplication with a scaling factor that is constant for the time interval of the k-th sample. Then, said scaling factor is defined as

$$\alpha_k = \sqrt{\frac{\sigma_{ref}^2}{\sigma_k^2}} .$$

Calculation of the scaling factor α_k for an input signal $X_k[n]$ requires buffering of the input data stream during the whole time interval while the variance σ_k^2 is calculated. However, if it can be assumed that the probability of significant changes in the average received signal power level, i.e. the variance, between two consecutive frames is low, said buffering can be avoided by using the variance estimate from the preceding frame to calculate $Y_k[n]$. Hence, in the method according to the preferred embodiment of the present invention, the following equation is used to derive the scaled output signal $Y_k[n]$:

$$Y_k[n] = \alpha_{k-1} \cdot X_k[n]$$

In the equation above, the scaling function has been described as a multiplication. Since the scaling in the AGC is performed on sample basis, hard demands will be raised on the multiplicators performing the actual scaling. However, these demands can be substantially relaxed if the scaling factor is quantized to a value that can be represented as a power of two. In this case, the scaling factor α_k can be interpreted as a shift factor β_k, i.e.

$$\alpha_k = 2^{\beta_k} \Leftrightarrow \beta_k = \log_2 \alpha_k \ .$$

Scaling is done by means of left-shifting a signal sample according to the shift factor β_k. In order to avoid too large steps of the scaling factor α_k due to that quantization, it is possible to quantize said factor to a value that can be represented as sum of powers of two. In this case, a scaling factor α_k is represented as a set of shift values $\{\beta_{k,i}\}$ consisting of the exponents of the sum terms. Scaling is done by means of left-shifting a signal sample for the number of positions that is indicated by each of the shift factors in said set and adding these shifted parts.

The scaling block as described above assumes input signals X[n] having zero mean value. This is necessary because the scaling operation is done by means of a multiplication. However, in practice, the hardware of the radio receiver might introduce a bias to the input signal, which can be considered as a time variant process with a very large time constant. This deviation is removed by introducing an offset compensation block 24 comprising means 241 for estimating the signal mean value by using a one tap low-pass filter with large time constant. A possible signal offset is removed by subtracting the estimated mean value from the input signal.

Optionally, an analogue AGC-unit 25 can be introduced to decrease the total dynamic range of an incoming signal. Said unit is designed as an attenuator that is controlled by means of power estimates from the power estimation unit 22. A control unit 251 activates a certain level of attenuation in response to the estimated signal power level. Said analogue AGC-unit should be designed at least as a step attenuator comprising active states with a certain attenuation and a non-active state with low attenuation. The control unit 251 activates an attenuation level if the signal power level exceeds a certain threshold level and deactivates when said signal power level decreases below said threshold. By means of said attenuator, incoming signals comprising a very large dynamic range can be attenuated, e.g., in order to avoid an undesired clipping in the subsequent analogue-to-digital converter.

As described above, the scaling factor for each signal sample is defined by using estimates of the variance σ_k^2 of the input signal sample $X_k[n]$. The scaling factor is calculated by using signal power estimates that are derived from a power estimation block 22. However, in order to reduce the hardware complexity needed for this operation, the non-normalized variance P_k is calculated instead of said variance estimates $\hat{\sigma}_k^2$, i.e. $P_k = 2N\hat{\sigma}_k^2$. Thus, the power estimation block 22 consists of estimation means 221 to calculate a power estimate and filtering means 222, viz. a low-pass one tap filter, in order to avoid sharp changes. Hardware complexity for calculating power estimates could be reduced furthermore by reusing an already implemented function block for power estimation: As it is crucial to minimize the total interference between sectors and between different base stations in a CDMA-based communication system, means are provided for estimating the total received power at the antenna in each sector and by that controlling and minimizing the output power of the mobiles within the sector. Said means needed for good system performance could also be combined with the arrangement and method according to the present invention and, thus, replace said estimation means 221.

Figures 3a and 3b are logarithmic presentations of examples for assignment functions of an appropriate scaling factor α_k with respect to a calculated variance level P_k for a certain signal sample $X_k[n]$. Said functions are applied in said means 211 to determine and assign discrete scaling factors in accordance with certain intervals of the estimated variance levels P_k. The diagram in figure 3a represents a first approach of a scaling factor assignment based on the calculated variance value within certain intervals. When assuming that the common signal power target level is chosen higher than the levels of the received signal power, it becomes apparent that the scaling factor must be chosen the higher the lower said received power level is.

As the power estimation values are sensitive to noise, an assigned scaling factor may frequently change between two values at the borders of two adjacent power intervals and, thus, causing an undesired oscillation in the scaled signal. Therefore, according to another preferred solution, as shown in figure 3b, the assignment of scaling factors also includes a relay control at the border between power intervals that prevents the assignment of oscillating scaling factors. The jump value of the power for changing the scaling factor varies depending from which side the power value approaches the ordinary jump value, i.e. the border line of the two adjacent power intervals. For decreasing power values the jump value is slightly minor than the ordinary jump value and, correspondingly, for increasing power values the jump value is slightly higher than the ordinary jump value. Thus, within a narrow power interval around the border the change of the scaling factor is delayed in order to prevent an oscillation between two scaling factors. The insensitivity against power fluctuations depends on the width of said narrow power interval.

Figure 4 shows a flowchart illustrating the main steps of the method according to the invention for transforming an input signal sequence $X_k[n]$, block 41, to a common and constant target variance level σ_{ref}^2. In a first step, block 42, the received signals may be adjusted, if necessary, to a common zero mean value in order to be able to perform a scaling operation by means of a multiplication. Next, for each signal sample $X_k[n]$ a power estimate P_k is calculated, block 43, and a scaling factor α_k is determined depending on said calculated power estimate P_k and the target power level σ_{ref}^2, block 44. Then, each of the received signal samples $X_k[n]$ is scaled to a signal $Y_k[n] = \alpha_{k-1} \cdot X_k[n]$ at said target power level, block 45, and forwarded to the demodulator, block 46. In an optional step, block 47, the estimated power level P_k of the AGC-unit can be applied to initiate an attenuation of the analogue received signal if said power estimates exceed threshold values.

From what has been described above, it becomes apparent that the scaled signal is marred by a signal error. This error results in the first place from the fact that the scaling factor is updated only at common time intervals and, secondly, from the fact that the scaling factor is quantized with respect to distinct intervals of signal power estimates and, possibly, quantized to a value that can be represented as a sum of powers of two. This implies that a subsequent unit, e.g. the demodulator, which assumes signal samples that are scaled to a common variance level, receives and processes a signal that might have a considerably large deviation from said common variance level. As a first solution, the scaling factor can be updated faster. However, as this increases the complexity in the subsequent demodulator, in the preferred alternative embodiment of the present invention, the scaling block 21 of the digital AGC-unit according to figure 2 generates side information 26 of the quality of the scaling and quantization that is updated faster than the scaling factor and forwarded to the demodulator 11 together with the scaled signal samples. This side information 26 is typically the control error of the AGC-algorithm due to the quantization of the scaling factor. When using said side information, the demodulator can easily and in a fast way switch to the antenna that momentarily receive the input signals best, e.g., when performing a softer handover.

CLAIMS

1. Arrangement in a receiver unit (10) of a communication system, said receiver unit (10) being equipped with at least one signal input (17) for reception of signals from a transmitter in said communication system, said signals having an essentially smaller short term dynamic range compared with the long term dynamic range, the receiver unit (10) including

means (11) for demodulating received signals,

a separate input path from each signal input (17) to said demodulator (11),

first processing means (12,13) in each of said input paths including at least means (13) for converting said received signals to baseband signals and means (14) for converting said baseband signals to sequences of digital signal samples,

c h a r a c t e r i z e d i n

second processing means (16,20) in at least one of said input paths for adapting said signal samples corresponding to a signal power adaptation to a common power level that only represents the short term dynamic range and forwarding said signal samples to the demodulator (11).

2. Arrangement according to claim 1,

c h a r a c t e r i z e d i n

said second processing means (16,20) including at least a digital Automatic Gain Control (AGC-) unit (161) and a subsequent quantizer unit (162).

3. Arrangement according to claim 2,

c h a r a c t e r i z e d i n

said digital AGC-unit (161) including

first means (211) for determining and assigning scaling factors from power estimates for each of said digital signal samples,

second means (212) for scaling a digital signal sample to a common power level by help of said assigned scaling factor.

4. Arrangement according to claim 2,

c h a r a c t e r i z e d i n

said digital AGC-unit (161) including

means (26) for providing side information about the quality of the scaling directly to the demodulator (11).

5. Arrangement according to claim 2,

c h a r a c t e r i z e d i n

said second processing means (16,20) including means (221) for determining power estimates of said digital signal samples and means (222) for low-pass filtering said power estimates.

6. Arrangement according to claim 2,

c h a r a c t e r i z e d i n

said second processing means (16,20) including means (24) for adjusting received sequences of digital signal samples to a mean value zero.

7. Arrangement according to claim 2,

c h a r a c t e r i z e d i n

said second processing means (16) including an attenuator in form of an analogue AGC-unit (163), said attenuator being controlled in response to power estimates of the digital signal samples.

8. Communication system consisting of a plurality of transmitter and receiver units,

c h a r a c t e r i z e d i n

at least one receiver unit comprising an arrangement according to one of claims 1-5.

9. Communication system according to claim 8,

c h a r a c t e r i z e d i n

said at least one receiver unit being part of a radio base station.

10. Communication system according to claim 8 or 9,

c h a r a c t e r i z e d i n

said system being a CDMA-based communication system.

11. Communication system according to claim 10, wherein each base station is equipped with a mandatory function block for power estimation of received uplink signals,

c h a r a c t e r i z e d i n

said mandatory function block for power estimation being re-used for retrieval of signal power estimates for the digital AGC-unit.

12. Method in a receiver unit of a communication system receiving signals from a transmitter in said communication system by at least one signal input, comprising for each of said signal inputs the steps of

processing received signals at least by means of converting said signals to baseband signals and converting said signals to a sequence of digital signal samples,

c h a r a c t e r i z e d i n

transforming for at least one of said signal inputs said digital signal samples to signals at a common and constant power level resulting in a reduced dynamic range,

quantizing the transformed signal samples to signals with a reduced number of bits,

forwarding the transformed and quantized signal samples to a demodulator.

13. Method according to claim 12,

c h a r a c t e r i z e d i n

transforming a digital signal sample by means of

determining a scaling factor from power estimates of said signal samples,

quantizing said scaling factor with respect to distinct intervals of power estimates,

scaling by means of multiplying said signal sample with said quantized scaling factor.

14. Method according to claim 12,

c h a r a c t e r i z e d i n

transforming a digital signal sample by means of

determining a scaling factor from power estimates of said signal sample, quantizing said scaling factor with respect to distinct intervals of power estimates to a value that can be represented as sum of powers of two,

determining shift values representing the exponents of the sum terms,

scaling by means of left-shifting said signal sample according to said shift values and adding the shifted parts.

15. Method according to claim 13 or 14,

c h a r a c t e r i z e d i n

determining one scaling factor per signal sample.

16. Method according to one of the claims 12-15,

c h a r a c t e r i z e d i n

providing side information about the quality of the scaling directly to the demodulator (11).

17. Method according to one of the claims 12-16,

c h a r a c t e r i z e d i n

assigning said quantized scaling factors with an overlapping for increasing and decreasing values of the power estimates at the borders of said distinct power intervals.

18. Method according to one of the claims 12-17,

c h a r a c t e r i s e d i n

adjusting received sequences of digital signal samples to a mean value zero by subtracting a calculated mean estimation value from said signal samples.

19. Method according to one of the claims 12-18,

c h a r a c t e r i z e d i n

attenuating the received signals in the analogue part of the receiver in response to power estimates of signal samples of said received signals in the digital part.

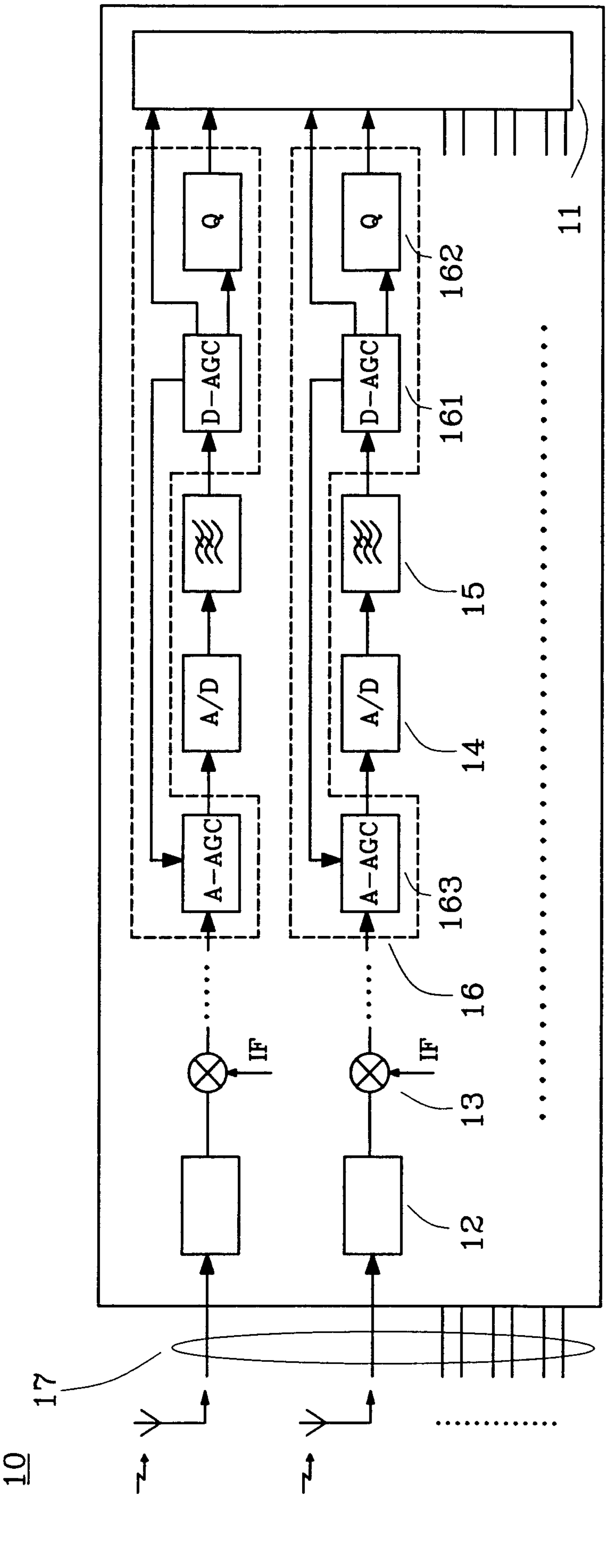

10
17
12
13
IF
IF
16
163
A-AGC
A-AGC
A/D
A/D
14
15
D-AGC
D-AGC
161
Q
Q
162
11

Fig. 1

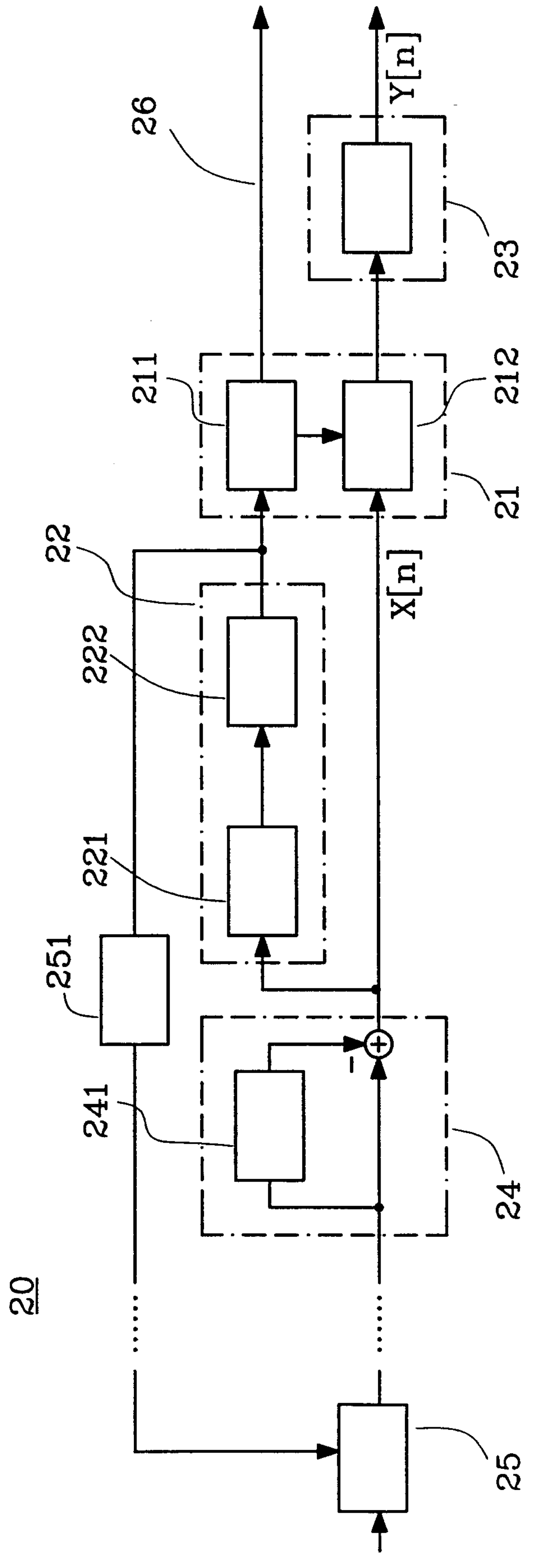


Fig. 2

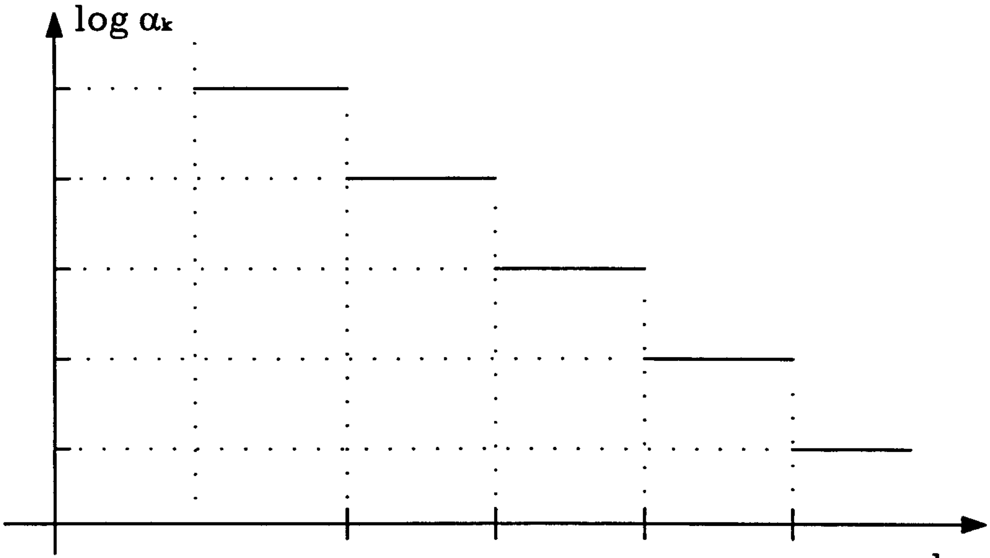


Fig. 3a

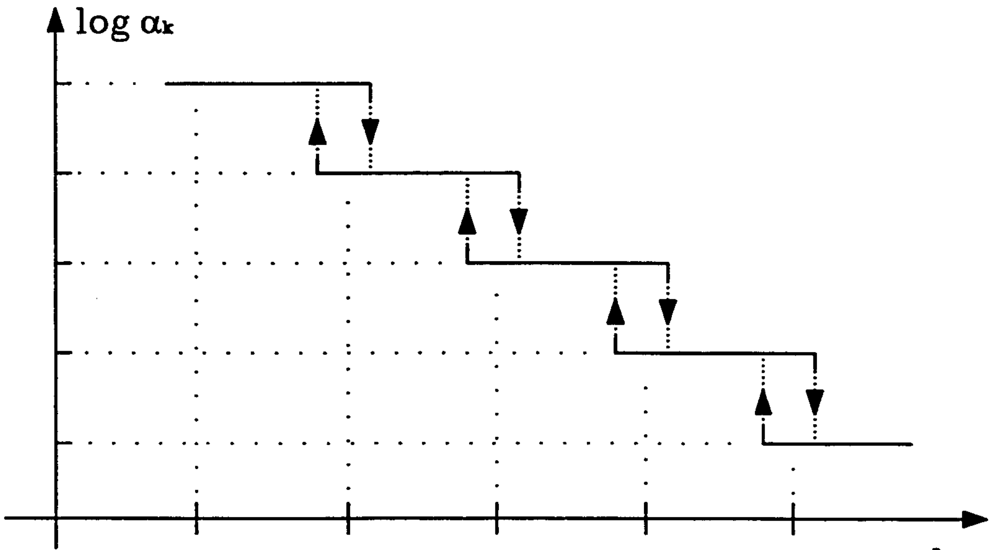
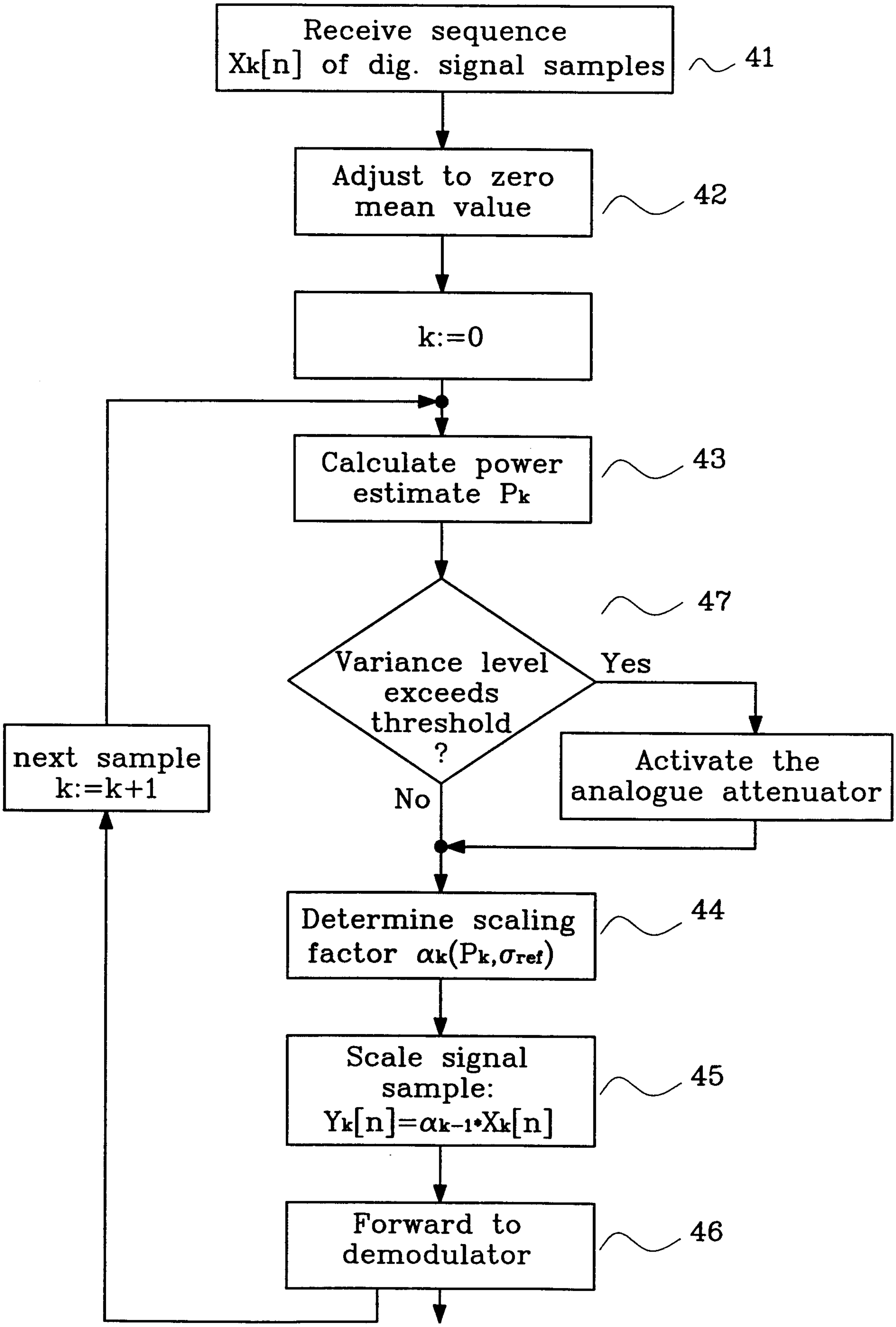


Fig. 3b



Receive sequence
$X_k[n]$ of dig. signal samples
41
Adjust to zero
mean value
42
k:=0
Calculate power
estimate P_k
43
47
Variance level
exceeds
threshold
?
Yes
No
Activate the
analogue attenuator
next sample
k:=k+1
Determine scaling
factor $\alpha_k(P_k, \sigma_{ref})$
44
Scale signal
sample:
$Y_k[n]=\alpha_{k-1} \cdot X_k[n]$
45
Forward to
demodulator
46

Fig. 4

INTERNATIONAL SEARCH REPORT

International application No.
PCT/SE 00/01703

A. CLASSIFICATION OF SUBJECT MATTER

IPC7: H04B 7/005
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC7: H04B

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

SE,DK,FI,NO classes as above

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 5566201 A (KJELL ÖSTMAN), 15 October 1996 (15.10.96), abstract --	1,12
A	US 5835527 A (GARY R. LOMP), 10 November 1998 (10.11.98), abstract --	1,12
A	WO 9310609 A1 (INTERDIGITAL COMMUNICATIONS CORPORATION), 27 May 1993 (27.05.93), abstract -- --------	12

☐ Further documents are listed in the continuation of Box C. ☒ See patent family annex.

* Special categories of cited documents:
"A" document defining the general state of the art which is not considered to be of particular relevance
"E" earlier application or patent but published on or after the international filing date
"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed
"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance: the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance: the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
12 December 2000	08 -01- 2001
Name and mailing address of the ISA/ Swedish Patent Office Box 5055, S-102 42 STOCKHOLM Facsimile No. +46 8 666 02 86	Authorized officer Fredrik Blomqvist/AE Telephone No. +46 8 782 25 00

INTERNATIONAL SEARCH REPORT
Information on patent family members

04/12/00

International application No.
PCT/SE 00/01703

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
US 5566201 A	15/10/96	AU 3682895 A	19/04/96
		BR 9509158 A	11/11/97
		CA 2198595 A	04/04/96
		CN 1158677 A	03/09/97
		EP 0783801 A	16/07/97
		JP 10506764 T	30/06/98
		WO 9610298 A	04/04/96
US 5835527 A	10/11/98	US 5920590 A	06/07/99
		US 5995538 A	30/11/99
		AU 4529796 A	24/07/96
		EP 0801856 A	22/10/97
		EP 1041727 A	04/10/00
		EP 1041728 A	04/10/00
		EP 1043844 A	11/10/00
		FI 972241 A	04/09/97
		IL 116444 A	17/08/99
		JP 10512113 T	17/11/98
		US 5563907 A	08/10/96
		US 5574747 A	12/11/96
		US 5673286 A	30/09/97
		WO 9621295 A	11/07/96
		ZA 9510969 A	08/07/96
WO 9310609 A1	27/05/93	CA 2100793 A	20/05/93
		EP 0567638 A	03/11/93
		SG 43891 A	14/11/97
		US 5299226 A	29/03/94
		US 5535238 A	09/07/96
		US 5631921 A	20/05/97