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(54) Title: INTERPOLATIVE ANALOG/DIGITAL CONVERTER FOR BAND-PASS SIGNALS

(54) Bezeichnung: EIN INTERPOLATIVER A/D UMSETZER FÜR BANDPASSSIGNALE

(57) Abstract

The invention relates to an interpolative analog/digital converter for band-pass signals in which the amplitudes of the scanning values are quantized in 2 consecutive stages. The first stage contains the actual analog/digital interface and the second stage is purely digital. The amplitudes of the scanning values are not quantized, as in other processes, using finely quantizing elements at the analog/digital interface, but by overscanning, shaping of the quantization noise spectrum and digital interpolation. As a result, very coarsely resolving elements (with only 2 quantization stages in the extreme case) can be used at the actual analog/digital interface.

(57) Zusammenfassung

Bei dem erfindungsgemässen interpolativen Analog-Digital-Umsetzer für Bandpass-Signale wird die Amplitudenquantisierung der Abtastwerte in zwei hintereinandergeschalteten Stufen durchgeführt, wobei die 1. Stufe die eigentliche Analog-Digital-Schnittstelle enthält und die 2. Stufe rein digital ist. Die Amplitudenquantisierung der Abtastwerte erfolgt hier nicht wie bei anderen Verfahren durch den Einsatz von feinquantisierenden Elementen an der Analog-Digital-Schnittstelle, sondern durch Übertastung, spektrale Formung des Quantisierungsrauschens und digitale Interpolation. Dadurch wird erreicht, dass man an der eigentlichen Analog-Digital-Schnittstelle sehr grob auflösende Elemente (im Extremfall mit nur 2 Quantisierungsstufen) verwenden kann.

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Translation of New Specification Pages and Claims filed in European Patent Office on February 5, 1992:

INTERPOLATIVE ANALOG/DIGITAL CONVERTER

The invention relates to an interpolative analog/digital (A/D) converter as defined in the preamble of Claim 1, particularly for the conversion of analog bandpass filter signals in the carrier frequency or intermediate frequency domain in radio receivers.

It is known that the digitalization and digital processing of signals permits a reduction of circuitry expenditures and, in particular, the drawbacks of non-ideal component characteristics (noise, ageing, drift due to changes in temperature, DC-offset, etc.) can be avoided.

For example, the publication NTZ-Archiv, Volume 5, No. 12/1983, discloses at pages 353 et seq. a radio receiver in which the bandwidth limited input signal of a radio receiver is sampled and converted into a sequence of digital values. U.S. Patent No. 4,750,214 discloses a similar radio receiver in which the input signal is converted, before its digitalization, into an intermediate frequency position. In both cases, the digitized values are processed further in a digital quadrature conversion and digital demodulation unit. In a receiver disclosed in U.S. Patent No. 4,888,557, the digitalization and digital further processing of analog bandpass filter signals additionally includes advantageous dimensionsing for the sampling frequency of the A/D converter relative to the center frequency of the bandpass filter signals and, derived therefrom, a digital mixed frequency and a sorting signal for the in-phase and quadrature components.

The drawback of these prior art solutions is that the analog/digital converter converts the bandpass filter signal already at the A/D interface with full amplitude resolution and thus must be configured in a very expensive manner with many precision elements. Such a solution, on the one hand, involves high costs, on the other hand, it is necessary to observe the moment of sampling during the A/D conversion the more accurately the higher the signal components to be processed and the finer the resolution of the input signal is to be. Since certain fluctuations in the switching method of these components cannot be eliminated completely, the amplitude resolution attainable by conventional methods has upper limits.

Methods for the analog/digital conversion of signals employing only a few quantizing stages at the analog/digital interface with nevertheless high resolution are interpolative methods. The interpolative method for the analog/digital conversion performs the amplitude quantization of the sampled values in two series connected states. The first stage (stage

1) here includes the actual analog/digital interface, while the second stage (stage 2) performs purely digital functions. The amplitude quantization of the sampled values here takes place, in contrast to non-interpolative methods, not with finely quantizing elements at the analog/digital interface, but by oversampling, spectral shaping of the quantization noise in the first stage and digital interpolation in the second stage.

Interpolative analog/digital converters are known, for example, from IEEE Trans. Commun. Vol. Com-33, No. 3, pages 249 et seq. However, these converters are unsuitable for the A/D conversion of bandpass filter signals since only the signals whose spectral components lie far below the sampling frequency are converted with the desired resolution. For adaptation to the case of a bandpass filter, the publication Electronics Letters, Volume 25, No. 23, November 9, 1989, pages 1560-1561, proposes a lowpass/bandpass transformation in which a coarse quantization is performed in a first stage configured as a modulator loop including a time discreet loop filter and the fine quantization takes place in a subsequent second stage by means of a digital bandpass filter.

It is the object of the present invention to provide an advantageous A/D converter of the type defined in the preamble of Claim 1.

This is accomplished by the feature defined in the characterizing portion of Claim 1. Advantageous features and modifications of the invention are defined in the dependent claims.

The invention will now be described with reference to the drawing figures.

According to the sampling theorem for bandpass filter signals, an analog bandpass filter signal having a bandwidth B can be represented without information loss by sampled values $T = 1/f_A$ if the equation

$$f_A > 2B \qquad (1)$$

is observed between the sampling frequency f_A and the bandwidth B of the bandpass filter signal, and the equation

$$fm = (2m + 1) \cdot f_A/4 \tag{2}$$

where m = 0, 1, 2, ... is observed between the center frequency fm of the bandpass filter signal and the sampling frequency f_A. Hereinafter m represents the bandpass index and the sampling of bandpass filter signals is called bandpass sampling.

In real bandpass sampling by means of an analog/digital converter, an amplitude quantization is performed to correspond to the resolution of the analog/digital converter in addition to the sampling of the bandpass filter signal (time domain quantization). The interpolative analog/digital converter according to the invention converts the bandpass filter signals in two series connected stages, both operated at the clock frequency

$$f_A > 2NB \tag{3}$$

where N is assumed to be much greater than one (high oversampling); additionally, Equation (2) applies. Thus the following results for the sampling frequency:

$$f_A = \frac{4}{2m+1} fm > 2NB \tag{4}$$

Figure 1 shows the first stage of the A/D converter according to the invention. It is a control circuit composed of a coarsely quantizing A/D converter (in the extreme case with only one bit resolution), a D/A converter of corresponding resolution, reference junctions, linear networks H(p) and coefficient members a_i and b_i. The clock rate f_A for the A/D converter and for the D/A converter should be selected according to Equation (4).

The coefficients a_i and b_i should be selected so that the control circuit is stable and the quantizing noise in the useful frequency range is minimal. The stability criteria are solved by general control technology methods such as, for example, a Bode diagram. The noise is minimized in that the coefficients a_i and b_i are selected so that the circuit reacts approximately like a pure n^{th} order system, where n is the number of networks H(p) employed.

The linear networks H(p) are resonators whose resonant frequency is tuned to the center frequency or the close vicinity of the center frequency. Networks exhibiting the same or a similar behavior as networks that result from lowpass/bandpass filter transformation of integrating or approximately integrating first order networks are suitable as resonators. These are, for example, the following:

linear networks which, excited by a sinusoidal oscillation at frequency f_m switched in at time $t = 0$, react at the output with a sinusoidal oscillation whose amplitude rises in proportion with the time t;

or

linear networks which, excited by a sinusoidal oscillation at frequency f_m switched in at time $t = 0$, react at the output with a sinusoidal oscillation whose amplitude rises over a finite time interval $0 < t < t_0$ in proportion with the time t; t_0 should here be much less than the sampling period;

or

resonators whose resonant frequency exactly or approximately coincides with the center frequency of the input signal;

or

resonators whose resonant frequency exactly or approximately coincides with the center frequency of the input signal and which exhibit the same or similar or similar [sic] behavior as networks re-

resulting from lowpass/bandpass transformations from integrating or approximately integrating first order networks;

or

bandpass filters that are narrowbanded compared to the bandwidth of the input signal and whose center frequency exactly or approximately coincides [with the center frequency of*] the input signal;

or

bandpass filters that are narrowbanded compared to the bandwidth of the input signal and whose center frequency exactly or approximately coincides with the center frequency of the input signal and which exhibit the same or a similar behavior as networks resulting from a lowpass/bandpass transformation of approximately integrating first order networks.

Due to the feedback connection of the coarsely quantized sampled values and a comparison with the input signal and the signal curves at the output of the networks H(p), the signal

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*Translator's note: the bracketed portion of the text is not part of the German-language original but is believed to be intended and required for completion of the sentence (see immediately following paragraph).

components of the error signals are spectrally weighted or spectrally shaped, respectively, with respect to other signal components, in the vicinity of the resonant frequency. The output signals of the networks H(p) are added according to the weighting factors a_i and are fed to the A/D converter which performs a coarse amplitude quantization. The feedback connection produces a coarsely quantized sequence of sampled values at the output of the A/D converter so that it coincides in the useful frequency range with the ideally sampled input signal except for a residual noise component.

Figure 2 shows the curve of the quantization noise 1 and of the bandpass signal 2 which still includes a small amount of residual noise 3. It can be seen that the significant components of the quantization noise lie essentially outside of the useful frequency range.

In the second stage of the A/D converter for bandpass filter signals according to the invention the fine quantization of the previously coarsely quantized sampled values is effected by digital interpolation. This stage as well is operated at the same clock frequency f_A according to Equation (4) as the first stage. Two basic possibilities exist for performing this interpolation, namely either digital interpolation by means of a digital bandpass filter whose band center and bandwidth, for ideal bandpass sampling without amplitude quantization, exactly or approximately coincide with the useful frequency band of the bandpass filter signal, or a digital interpolation by means of a digital quadrature modulator.

In the above-described digital interpolation by means of a digital bandpass filter the significant components of the quantization noise lie outside of the useful frequency range. Thus finely quantized sampled values are available at the output of this bandpass filter.

Figure 3 depicts a digital quadrature modulator for the A/D converter according to the invention for bandpass filter signals. This modulator is composed of a quadrature mixer 4 including two digital multipliers 5 and 6 and two digital lowpass filters 7 and 8. The digital lowpass filters 7 and 8 are designed in such a way that their passband ranges exactly or approximately coincide with the useful frequency range of the equivalent lowpass filter signal (cutoff frequency f_g = B/2). Since the information content of a bandpass filter signal is not included in its carrier position (center frequency) but in its equivalent lowpass filter signal, it is possible to perform a fine quantization of the equivalent lowpass filter signal. The output signal of the first stage is multiplied with cos (π/2 x K) by digital multiplier 5 in digital quadrature mixer 4 and is then sent through digital lowpass filter 7, at whose output the finely quantized real component of the equivalent lowpass filter signal is available. Likewise the output signal of the first stage is multiplied with -sin (π/2 x K) by digital multiplier [6] in digital quadrature mixer 4 and is then sent through digital lowpass filter 8 at whose output the finely quantized imaginary component of the equivalent lowpass filter signal is available.

The sampling rate at the output of the two lowpass filters 5 and 6 can be reduced by a factor of 2N corresponding to the bandwidth of the digital interpolation filters.

Figure 4 depicts the digital quadrature mixer 4 and, for comparison, a simplified digital quadrature mixer 9. Since the two digital carrier sequences cos (k x π/2) and sin (k x π/2) take on only values from among (-1, 0, 1), digital multipliers 5 and 6 can be omitted in a realization of the digital quadrature mixer 4. Instead, switches 10 and 11 are employed which periodically pass through the switch positions

0, 1, 2, and 3 in synchronism with the sampling frequency f_A as well as digital elements which perform the trivial multiplications by the factors 0, 1 and -1.

Figure 5 depicts a circuit realization for a network H(p). It is composed of a parallel resonant circuit including a capacitor 12 and a coil 13, fed by a transistor 14 in common emitter connection as a voltage controlled current source. The ohmic resistances for setting the operating point are not shown.

Figure 6 depicts a circuit realization of the first stage including only one linear network H(p). It is composed of the parallel resonant circuit including capacitor 16 and coil 17, a comparator 18, a flip-flop 19, transistors 20 and 21, and a voltage controlled current source 22. Flip-flop 19 is operated at frequency f_A and thus takes over the switching states of comparator 18. The latter, in turn, is actuated by the parallel resonant circuit including capacitor 16 and coil 17. The collector currents of the two transistors are controlled by the difference between voltages U_x and U_y. Thus the differential amplifier composed of transistors 20 and 21 and a constant current source 22 simultaneously acts as a reference junction and as a voltage controlled current source for feeding the parallel resonant circuit composed of capacitor 16 and coil 17 as well as constant current source 22. Constant current source 22 may be realized, for example, by an ohmic resistance or a transistor circuit.

New Claims 1 to 6

1. An interpolative analog/digital converter including a first stage which coarsely quantizes the amplitude of an analog input signal by oversampling with a sampling frequency f_A and shapes the quantization noise produced by the coarse amplitude quantization in such a manner that the significant noise components lie outside of the useful frequency range, the converter further including a second stage which interpolates the signal that was coarsely quantized in the first stage, **characterized in that** the input signal is a bandpass filter signal of the bandwidth B with a center frequency fm, the sampling rate being

$$f_A = \frac{4}{2m + 1}$$

where m = 0, 1, 2, ..., and, according to $f_A > 2$ NB, where N >> 1, is significantly greater than the bandwidth of the input signal; and the coarsely quantized signal is initially digitally quadrature mixed in the second stage and digitally interpolated by means of two digital lowpass filters at whose output sampling values are available of the in-phase and quadrature components of the input signal.

2. A converter according to claim 1, characterized in that the passband range of the digital lowpass filters exactly or approximately coincides with the lowpass filter signal that is equivalent to the bandpass filter signal.

3. A converter according to claim 1 or 2, characterized in that the quadrature mixer is composed of two preferably electronic switches which periodically pass through the switch positions 0, 1, 2 and 3 and are associated with the respective switch positions of digital elements that perform a multiplication by 0, 1 and -1.

4. A converter according to claims 1 to 3, characterized in that the structure of the first stage is composed of an A/D converter of the corresponding resolution, at least one network H(p), coefficient members a_i and b_i, which may also take on the values 0 and 1, and of reference junctions.

5. A converter according to claim 4, characterized in that the coefficients a_i and b_i are selected in such a manner that the circuit of the first stage is stable and the quantization noise is shaped in an optimum manner.

6. A converter according to claim 4 or 5, characterized in that the network H(p) or the networks H(p) are composed of resonators whose resonant frequencies exactly or approximately coincide with the center frequency f_m of the input signal.