DESCRIPTION

Method for calculating compensation coefficients of an analog reception chain, device and program therefor

1. Field of the invention

The invention relates to the reception of radio frequency signals. The invention relates more particularly to the reception of radio frequency signals within a reception device comprising an analog chain and a digital chain. A subject matter of the invention is to compensate for distortions of an analog reception chain of a digital receiver comprising an analog reception chain and a digital reception chain.

2. Prior art

In many areas of communication, it is necessary to deal with a requirement to increase communication rates. Such is the case e.g. in the civilian field but also in the military field. Such increase in bit rates is made necessary by the ever-increasing use of communication devices to receive ever greater amounts of data.

The increase in throughput should be achieved by ensuring constant reception qualities. Thereby, the continuous increase in the rates of military communications requires the use of waveforms that are increasingly demanding in terms of managing radio distortions, more particularly in the analog chain of the digital receiver.

It is recalled that in a digital receiver, the analog chain refers to the series of analog components and processes that manage the reception and the conversion of the analog radio frequency (RF) signal into a digital format. Thereof is the initial step during which the incoming RF signal is conditioned and prepared for a subsequent digital processing.

An analog chain often consists of the following components:

- an antenna responsible for capturing the RF signal from the waves and converting same into an electrical signal
- a low noise amplifier (LNA) that amplifies the weak electrical signal received from the antenna while adding in a minimum of noise.
- a bandpass filter used to eliminate unwanted signals and noise outside the desired frequency range;

- a mixer that combines the filtered RF signal with a local oscillator signal to perform frequency down-conversion: same produces an intermediate frequency (IF) signal that is easier to process and digitize.
- an intermediate frequency (IF) filter that removes unwanted signals and noise outside the desired frequency range, improving signal quality and isolating the specific frequency band of interest;
- an automatic gain control (AGC) circuit that adjusts the gain of the received signal to maintain a constant signal strength;
- an analog-to-digital converter (ADC) that converts the analog IF signal into a digital representation that can be processed by digital signal processing (DSP) components of the digital chain.

Once the analog chain has processed the signal and converted same to digital format, the digital chain performs the necessary signal processing (DSP). The analog chain thereby plays an essential role in reception by guaranteeing the fidelity and the quality of the received signal. Same minimizes noise, filters out unwanted frequencies and optimizes signal strength before digital processing takes place, ultimately contributing to the overall performance of the digital receiver. The digital chain processes the signal e.g. by performing a PSK phase demodulation or a quadrature amplitude demodulation such as 64QAM or higher, or any other type of phase and amplitude demodulation of a signal.

To meet such requirements at the analog chain, complex and expensive implementations are required. Such implementations can be costly in terms of the implementation thereof: e.g., to guarantee a minimum of distortions at the radio reception chain, a known method consists in performing sorting and selection of electronic components. The sorting is often done manually, testing one by one the electronic components to be used to manufacture the radio receiver. The additional cost generated is hence significant.

The implementations can be technologically complex, e.g. in terms of gain configuration in the analog chain; in terms of filtering and equalization (use of static equalizers, which are mainly analog all-pass filters to compensate for phase but not for amplitude and only in baseband); or in terms of the physical implementation and design of the analog chain, which requires minimizing the length of the signal path and limiting the sources of interference.

The document US10998926 proposes a solution for compensating the flattening of the gain of a reception chain by using a filter, the compensation being limited to the amplitude. The solution, however, requires the use of a plurality of signals and is complex to implement, since same requires working on a large number of points, which increases both the computational cost and the processing latency time under operational conditions, which is prohibitive for processing that needs to take place in real time.

It is thus necessary to propose a solution for compensating distortions of an analog reception chain that is simple and economical in terms of implementation, while responding to the problem of increasing the throughput to be processed, and that is suitable for real-time constraints.

3. Summary of the invention

To this end, the subject matter of the invention is a method for calculating static compensation coefficients of an analog reception chain, which method being implemented by an electronic device (2), during a calibration phase of a reception device (3) comprising the analog reception chain and a digital compensation module implementing an FIR filter the coefficients of which are the calculated static compensation coefficients, the method comprising:

- an estimation step (A01), using a predetermined calibration signal, of a frequency response H of the analog reception chain;
- a selection step (A02), using the frequency response, of a passband BP over which the static compensation coefficients are calculated;
- a calculation step (A03) for the static compensation coefficients of the analog receiver chain, in the time domain, as a function of the passband BP and of the inverse of the frequency response HBP over the passband.

According to particular embodiments used, the method includes one or a plurality of the following features:

- the estimation step (A01) for the frequency response H of the analog reception chain comprises:
    - a step of determining a reference wobbled signal, in a frequency band of a predetermined width;

- a step of obtaining a signal spectrum at the output of the analog reception chain, based on the reference wobbled signal;
- a calculation step for the frequency response H of the analog reception chain as a function of the signal spectrum at the output of the analog reception chain and the reference wobbled signal spectrum;

- the selection step (A02) for the passband BP on which the static compensation coefficients are calculated comprises a selection step for selecting, within the frequency response H, a portion $\mathrm{H_{BP}}$ corresponding to the portion of the frequency response H the amplitude of which exceeds a predetermined amplitude value;
- said amplitude value is predetermined by:
  - calculating the mean value of the amplitude response of the frequency response H of the analog reception chain;
  - subtracting a predefined tolerance value from the average value of the amplitude response of the frequency response H of the analog reception chain;
- the calculation step (A03) for the static compensation coefficients of the analog reception chain as a function of the passband BP and of the inverse of the frequency response HBP over the passband comprises at least one iteration of a calculation of coefficients of a finite impulse response (FIR) filter currently used in the time domain as a function of the inverse Hi of the frequency response HBP on the passband BP, said static compensation coefficients consisting of the coefficients of the FIR filter;
- the calculation of the coefficients of a Finite Impulse Response (FIR) filter comprises:
  - a step of obtaining a current number N_{FFTk} of points of an inverse Fourier transform, N_{FFTk} being a power of 2 comprised between a minimum value and a maximum value;
  - with the current number of points N_{FFTk} of the inverse Fourier transform *ifft*, a step of solving the equation:

$$h_{ik}(t) = ifft(H_i(f), N_{FFTk})$$

with:

$h_{ik}(t)$ being the impulse response;

$H_i(f)$ being the inverse response of the transfer function in the passband BP;

the calculation giving N_{FFTk} values of filter coefficients FIR;

- a step of digital filtering of a predetermined signal using the filter FIR the $N_{FFTk,}$ coefficients of which were obtained in the preceding step;
- an evaluation step for the quality of the FIR filter according to a performance criterion EVM;

- the number of points N_{FFTk} is comprised between 16 and 512.

The invention also relates to an electronic device (2) for calculating static compensation coefficients of an analog reception chain, device intended to be used during a calibration phase of a reception device (3) comprising the analog reception chain and a digital compensation module implementing an FIR filter the coefficients of which are the calculated static compensation coefficients, the electronic device comprising:

- means of estimating, using a predetermined calibration signal, a frequency response H of the analog reception chain;
- means of selecting, using the frequency response H, a passband BP over which the static compensation coefficients are calculated;
- means of calculating the static compensation coefficients of the analog reception chain, in the time domain, as a function of the passband BP and of the inverse of the frequency response H_{BP} over the passband.

A further subject matter of the invention is a computer program including software instructions which, when executed by a programmable electronic device, implement a method for calculating the static compensation coefficients of an analog reception chain, as defined hereinabove.

Finally, the invention relates to a reception device (3) comprising an analog reception chain and a module for digital compensation of distortion of the analog chain implementing an FIR filter, the digital compensation module being configured from coefficients calculated according to the method as defined hereinabove.

4. Brief description of figures

The invention will be clearer upon reading the following description, given only as an example, but not limited to, and making reference to the drawings wherein:

- [Figure 1] illustrates a system implemented to calibrate a receiver using a device for calculating the coefficients of a compensation filter;

- [Figure 2] illustrates a device for calculating coefficients of a compensation filter;
- [Figure 3] explains the different steps used to calculate the coefficients of the correction filter;
- [Figure 4] illustrates a wobbled signal of constant amplitude (between -1 in 1) and variable frequency in time;
- [Figure 5] illustrates a comparison of the spectra of the input wobbled signal with the output signal;
- [Figure 6] illustrates an amplitude response of a wobbled output signal of the analog chain, response obtained from a frequency response *H* of the analog chain;
- [Figure 7] illustrates an in-phase response of a wobbled output signal of the analog chain, response obtained from a frequency response *H* of the analog chain;
- [Figure 8] illustrates the number of points needed (16) to correct a FIRst analog chain in a first example of situation;
- [Figure 9] illustrates the number of points needed (256) to correct a second analog chain in a second example of situation;
- [Figure 10] illustrates the distribution of symbols for an output signal of an analog chain before correction by the method of the invention; and
- [Figure 11] illustrates the distribution of symbols for an output signal of an analog chain after correction by the method of the invention.

5. Description of an embodiment

5.1. Recalling the principles

As indicated hereinabove, a subject matter of the invention is to achieve digital compensation of distortions of an analog reception chain. Such compensation, according to the invention, is simple to implement insofar as same does not entail any significant modification of the digital chain and is carried out over a small passband. The compensation thus requires little calculation and is compatible with real-time processing requirements.

The compensation of such distortions reduces the cost of implementation by relaxing the linearity requirements of the filters of the analog chain. The solution implemented consists overall in inverting the amplitude response and/or the phase response (the responses being estimated during calibration) in the digital chain in order to correct the distortion of the overall analog chain and thereby comply with an imposed template (e.g., in

terms of EVM – *error vector magnitude*) and avoids the need for more expensive methods such as component sorting.

According to the invention, the digital compensation is carried out using a compensation filter which is defined from static compensation coefficients. More particularly, the compensation filter is a finite impulse response (FIR) filter. The static compensation coefficients of the FIR filter are calculated only once for a current radio chain. The calculation of the coefficients is carried out in an initial calibration phase of the analog chain. Such calculation can be performed again during a subsequent recalibration of the chain (e.g. during maintenance). Thereby, the static compensation coefficients are fixed once and for all at the end of the calibration. The advantage is that it is not necessary to modify the digital chain significantly, which is the case when a dynamic compensation is desired, wherein the coefficients are automatically recalculated according to the received signal. The other advantage, as explained hereinabove, is to limit not only the implementation cost but also the processing latency caused by the compensation in the receiver that implements the analog chain and the FIR filter.

According to the invention, the static compensation coefficients are calculated on the basis of a frequency response of the entire reception chain. The aim is to compensate for the overall distortions of the analog reception chain, not only the distortions added in by analog filters, although the latter are mainly responsible for all distortions.

More precisely, according to the invention, distortions are compensated only in the passband defined at −2 dB of the central frequency. Then, the inverse response to be applied in the time domain is calculated. Indeed, a correction directly in the frequency domain would be simpler but more expensive to implement and hence prohibitive. Instead, the inventors estimated an equivalent time filter, which is more easily implemented and thus more economical for the digital chain.

Figure 1 illustrates a system that can be implemented to calibrate a receiver 3 using the coefficient calculation device 2 of a compensation filter. For example, the calculation device 2 is implemented in a calibration phase and receives the analog chain RX of the receiver 3. Signals are transmitted to the analog chain RX and transformed signals are analyzed by a network analyzer AR, in particular the signals at the output of the analog chain RX of the receiver 3 being calibrated. The signals entering and leaving the analog chain RX are supplied to a component CC for calculating the static coefficients of the

compensation filter, which calculates these coefficients according to the method of the invention. When the coefficients of the filter are obtained, same are introduced into the filter of the receiver 3.

The receiver 3 comprises the analog chain RX, the digital filter that performs the compensation, using the components that are installed at the end of the calibration, so that the corrected signal can be processed by the rest of the digital baseband chain.

Figure 2 describes more precisely an electronic device for calculating static compensation coefficients of an analog reception chain. Such device can be implemented in a calibration phase of a reception device comprising the analog reception chain and a digital compensation module.

In the example shown in Figure 2, the on-board electronic device 2 comprises an electronic memory unit 10, at least one calculation processor 70 and an interface 80 for communication with remote devices, by a chosen communication protocol, e.g. a wired protocol and/or a radio communication protocol. The elements of the device 2 are suitable for communicating via a communication bus 90.

In the example shown in Figure 2, the estimation module for frequency response 20, the determination module for passband 40, and a calculation module for the static compensation coefficients 60 are each produced in the form of a software program, or a software brick, which can be run by the processor 70. The memory 10 of the electronic device for calculating static compensation coefficients 2 is then able to store an estimation software for the frequency response 20, determination software for the passband 40 and calculation software for static compensation coefficient 60. The processor 70 is then apt to execute each of said software.

In a variant (not shown), the estimation module for frequency response 20, the determination module for the passband 40, the calculation module for the static compensation coefficients 60 are each produced in the form a programmable logic component, such as an FPGA (*“Field Programmable Gate Array*), or else of an integrated circuit, such as an ASIC “(*Application Specific Integrated Circuit*).

When the electronic device for calculating static compensation coefficients 2 is produced in the form of one or a plurality of software programs, i.e. in the form of a computer program, same is further apt for being recorded on a computer-readable medium (not shown). The computer-readable medium is e.g. a medium apt to store the electronic

instructions and to be coupled to a bus of a computer system. As an example, the readable medium is an optical disk, a magneto-optical disk, a ROM, a RAM, any type of non-volatile memory (e.g. FLASH or NVRAM) or a magnetic card. A computer program containing software instructions is then stored on the readable medium.

The method used to calculate the static compensation coefficients is described in relation to Figure 3 and comprises the following steps:

- estimation A01, using a predetermined calibration signal SCP, of a frequency response H of the analog reception chain, delivering an amplitude response and/or a phase response;
- selection A02, using the frequency response H, of a passband BP over which the static compensation coefficients CSC will be calculated;
- (iterative) calculation A03 of the static compensation coefficients of the analog reception chain, in the time domain, as a function of the passband BP and of the inverse of the frequency response H_{BP} in the passband: thereof is an inverse filter calculation in the passband, time-dependent, allowing e.g. a minimization of the EVMs, in an example of embodiment.

Once the static compensation coefficients [are calculated], same are stored in a memory and made accessible to the FIR filter implemented in the digital chain of the receiver the calibration of which is being performed. The FIR filter will then, under operational conditions, filter the signal same receives by means of the calculated coefficients.

The FIR filter, calculated iteratively in the step of calculating the static compensation coefficients, represents a non-symmetrical i.e. non-real response. Thereby, it is not possible to apply the filter directly in baseband. The inventors have thus found it judicious to apply the filter *either* at intermediate frequency before returning to baseband *or* artificially at a low frequency sufficient to cover the band of the filter. The artificial intermediate frequency is denoted by f_i such that $f_i \geq Bw/2$ and *Bw* represents the passband of the filter.

In some implementation cases, however, it is not possible to operate on the artificial intermediate frequency f_i. In such case, an alternative is to use a real (i.e. symmetrical) filter that is less suitable and less precise but nevertheless allows to gain in terms of performance.

To obtain the time response, the assumed known frequency response is used. The global frequency response is reduced to the passband thereof and is then inverted. The impulse response thereof is then deduced therefrom via inverse Fourier transform processing. Such processing requires a certain number of points, ranging e.g. from 16 to 512, in order to be sufficiently accurate.

It should be noted that one of the particularities of the method described hereinabove lies in the radio application thereof which aims to optimize the particular performance criterion that is the EVM. Other equalization techniques are more studied but very oriented toward waveforms and propagation chain, and hence are not suitable for the problem solved by the invention.

5.2. Example of implementation

An example of implementation of the method described hereinabove is presented thereafter.

5.2.1. Estimation of the frequency response of the chain

Firstly, as indicated above, the device 2 estimates the frequency response of the analog chain. In order to obtain the global frequency response, it is necessary to calibrate the radio equipment 3 for which the coefficients of the FIR filter are calculated. Such calibration is implemented by the electronic device 2 by injecting a so-called wobbled or "chirp" waveform, i.e. a signal the frequency of which varies over time. In doing so, the different frequencies of interest in the band used by the analog chain of the receiver 3 are excited. The electronic device 2 then recovers the signal at the output of the analog chain of the receiver 3 tainted by the effect of the (only) distortions.

According to the invention, the wobbled signal c(t) that is implemented is defined as follows:

$$\mathrm{c(t)} = \mathrm{Aexp}\left(\mathrm{j2pi}\left(\mathrm{f_1} + \frac{\Delta_f}{2} \times t - \frac{\Delta_f}{2}\right) \times t\right)$$

With f_1 the starting frequency of the wobbled signal and Δf the frequency band covered. An example of such a signal is shown in Figure 4.

In the present case, it is interesting to know the passband to be estimated in order to determine the correct width of the wobbled signal. For a chain with a global filtering of band B, $\Delta f = B$ and $f_1 = fc_{theo} - \frac{B}{2}$ are thereby defined, where fc_{theo} is the theoretical central frequency of the chain to be characterized. For example, the passband

to be estimated may be comprised between 1.2 and 2 MHz, in a real case described with reference to Figures 4 and 5.

A comparison of the spectra of the input wobbled signal, denoted by wobbled In, with that of the output, denoted by wobbled Out, is exemplified in Figure 5. In the example shown in Figure 5, the attenuations at the band edge are noted which represent the filtering effects of the analog reception chain.

The following operation consists in performing a division of the two spectra to obtain the frequency response *H* of the chain. The frequency response includes both the phase and the amplitude:

$$H(f) = \frac{fft(S_{out}[n])}{fft(S_{in}[n])}$$

Wherein:

- fft is the fast Fourier transform function defined so that: $fft(k) = \sum_{n=0}^{N_{FFT}-1} x(n)e^{-j2\pi k\ /N_{FFT}}$ with k, n = 0… N_{FFT}-1 and N_{FFT} the number of FFT points used is a power of 2. In the calibration phase, there is no limit value on the number of points which should be chosen to obtain the most faithful possible representation of the frequency response;
- $S_{in}[n] = S_{in}(nT_e)$ represents the discrete signal obtained by sampling at the rate of $f_e = 1/T_e$ the time signal transmitted at the input of the analog reception chain;
- $S_{out}[n] = S_{out}(nT_e)$ represents the discrete signal obtained by sampling at the rate of $f_e = 1/T_e$ the time signal received at the output of the analog reception chain.

Equipped with the frequency response *H*, it is possible to obtain the amplitude response and the phase response. Figures 6 and 7 illustrate the two types of response (amplitude and phase), as an example.

5.2.2. Calculation of filter coefficients

Secondly, the electronic device 2 calculates the static compensation coefficients used in the digital domain in order to compensate for the distortions of the analog radio chain. The static compensation coefficients are determined from the frequency response *H.*

To this end, let us take the transfer function of the following Finite Impulse Response (FIR) filter with a frequency response H:

$$H(z) = b_0 + \sum_{k=1}^{M} b_k z^{-k}$$

Wherein:

- z^{-k} represents the period of k samples
- $B_0, b_1, b_2, \ldots, b_M$ are the coefficients of the FIR filter;
- M is the order of the FIR filter.

The order of the FIR filter informs on the number of coefficients N needed for the implementation thereof with $N=M+1$.

The static compensation coefficients are, according to the invention, the coefficients of an FIR filter, and one seeks to determine the FIR filter coefficients, within the framework of the present implementation. It is thus necessary to calculate the different coefficients $b_0, b_1, b_2, \ldots, b_M$ of such a filter.

Such calculation, however, is not carried out immediately from the frequency response H previously obtained, but from a portion H_{BP} of the latter which has to be determined during the selection step A02, the portion being limited to a so-called correction passband, within which it is sought to correct the distortions caused by the analog chain.

5.2.2.1. Determining the correction passband

The passband B_w of the filter defined at −2 dB of the amplitude value corresponding to a real central frequency denoted by *fc*, is selected.

Indeed, the real central frequency fc can be (and is often) different from the theoretical central frequency of the filter, denoted by fc_{theo} precisely because of the dispersions of the analog filters that it is desired to correct.

To this end, let us first consider the mean value of the amplitude response in dB:

$$\overline{H} = 20\log_{10}\frac{1}{L}\sum_{i=0}^{L}|H(i)|$$

As well as the overall amplitude response in dB:

$$G_{dB}(f) = 20\log_{10}|H(f)|$$

With |.| the absolute value operator. The response and associated frequencies are then selected that satisfy a certain tolerance with respect to the mean value:

$$\mathrm{H_{BP}(f)} = \mathrm{H(f_{\mathit{BP}})}$$

with

$$\mathrm{f_{BP}} = \mathrm{f\left(G_{\mathit{dB}}(f) > \overline{H} - tolerence\right)}$$

The value of the tolerance is e.g. set at 2 dB. The central frequency is then calculated as the average frequency of this frequency range set at 2 dB of the amplitude average value:

$$\mathrm{fc} = \frac{(\mathrm{f_{BP}[K]} - \mathrm{f_{BP}[0]})}{2} + \mathrm{f_{BP}[0]}$$

Wherein:

- $f_{BP}[K]$ represents the last frequency in the frequency range $\mathrm{f_{BP}}$
- $\mathrm{f}_{BP}[0]$ represents the first frequency in the frequency range $\mathrm{f_{BP}}$

Thereby, at the end of the calculation step, the correction passband and the central frequency of the correction passband are available.

5.2.2.2. Calculation of coefficients minimizing the EVM

The transfer function in the correction passband is denoted by H_{BP}. The inverse response of the transfer function in the passband is then calculated.

$$H_i(f) = \frac{1}{H_{BP}(f)}$$

The impulse response that can be applied in the time domain is deduced therefrom.

As explained hereinabove, the goal is to calculate the coefficients of an FIR filter (and not an IIR filter) for simplicity and ease, in particular in terms of implementation. The calculation of the coefficients of the FIR filter is implemented as follows:

$$h_i[n] = ifft(H_i(f), N_{FFT})$$

With:

- $h_i(t)$ being the discrete impulse response in the time domain;
- $H_i(f)$ being the inverse response of the transfer function in the passband BP;
- $ifft$ the inverse Fourier transform function defined so that:

- $ifft(n, N_{fft}) = \sum_{k=0}^{N_{FFT}-1} X[k]e^{j2\pi kn/N_{FFT}}$ with k, n = 0… N_{FFT}-1 and N_{FFT} the number of FFT points used is a power of 2. N_{FFT} the number of points to use to calculate the impulse response.

The number of N_{FFT} points of the inverse Fourier transform directly influences the resolution and the response estimate, as does the value thereof. The number of coefficients of the FIR filter used is equal to the number of points N_{FFT} of the inverse Fourier transform. The values of the coefficients of the FIR filter used are equal to the resolution values of the inverse Fourier transform.

Since the compensation is static, it is possible to establish the number of points needed in the calibration phase in order to meet a certain EVM criterion. In the case of saw type analog filters, this number of points can vary, e.g., from 16 to 512 depending on the performance EVM criterion.

In other words, depending on the performance criterion, it can be determined that it will be necessary to calculate e.g. between 16 and 512 points of the inverse Fourier transform i.e. e.g. between 16 and 512 static compensation coefficients. According to the invention, the number of points of the inverse Fourier transform (i.e. the number of static compensation coefficients) is a power of 2. Moreover, in order not to add in a time delay which would be due to the use of a filter comprising too many coefficients (i.e. e.g. 2048 and higher), the number of points of the inverse Fourier transform is necessarily limited. Thereby, the determination of the static compensation coefficients to be introduced into the filter for a current analog radio reception chain comprises at least one iteration of the following coefficient calculation steps, consisting in:

- Obtaining a number of points N_{FFTk} of the current inverse Fourier transform (i.e. 16, 32, etc.), where N_{FFTk} is a power of 2, between a minimum value and a maximum value (e.g. between 16 and 512), such that $N_{FFTk} = 2^k$ (k being then comprised between 5 and 9);
- With the number of points N_{FFTk} of the current inverse Fourier transform, solving:

$$h_{ik}[n] = ifft(H_i(f), N_{FFTk})$$

the calculation gives N_{FFTk} values of FIR filter coefficients;

- Digitally filtering a predetermined signal, such as a modulated signal representative of the final application such as a PSK or a QAM modulation e.g. using the FIR filter the N_{FFTk} coefficients of which were obtained in the preceding step;

- Evaluating the quality of the filter according to an EVM performance criterion.

The iterations of the coefficient calculation steps are repeated until the EVM performance criterion is met (e.g. to achieve an EVM of less than 5 %). When the performance criterion is reached, the calculation stops: one then has the number of coefficients N_{FFTk} and the value of the coefficients due to the resolution of the inverse Fourier transform.

In the present example of embodiment, the performance criterion is defined as being an EVM criterion. Other performance criteria could of course be used as needed. The EVM performance criterion is defined, for a given constellation, with a given frame:

$$e_k = \left(I_k - \widetilde{I_k}\right)^2 + \left(Q_k - \widetilde{Q_k}\right)^2$$

$$EVM_k = 100\sqrt{\frac{e_k}{P_{avg}}}$$

- With P_{avg} the mean power of the constellation in question;
- I_k the in-phase measurement of the *kth* symbol of the frame;
- $\widetilde{I_k}$ the expected in-phase value of the *kth* symbol of the frame;
- Q_k the quadrature measurement of the kth symbol of the frame;
- $\widetilde{Q_k}$ the expected quadrature value of the *kth* symbol of the frame.

Thereby, the number of points needed (i.e. the number and value of the coefficients) is determined during calibration as a function of the analog filter(s) necessary to correct (hence as a function of the radio chain to which the filter(s) relate). For example, with reference to Figures 8 and 9, for a FIRst radio chain, in order to correct the distortions, the number of points may be equal to 16 whereas for a second radio chain, the number of points may be equal to 256. More particularly, Figure 8 illustrates that for a FIRst radio chain, only 16 coefficients are needed to fall below a threshold of 5% of EVM. Figure 9 illustrates that for a second radio chain, distinct from the first, no less than 256 coefficients are needed to approach the 5% EVM threshold. The number of coefficients calculated is thus variable and is a function of the radio chain and of the uniqueness thereof, in particular as a function of the electronic components of which same is composed. Figures 10 and 11 illustrate the result obtained, in terms of EVM, using an FIR filter comprising coefficients calculated using

the calculation method explained hereinabove. The advantage of such implementation is that one is able to accept a certain variability of the radio chain, variability which is compensated by the static filter, of finite length.

5.2.3. Application of calculated filter coefficients

Provided with the static filter coefficients introduced into the FIR of the digital compensation module of the receiver, a correction is applied to the received signal. The correction is carried out, in operational conditions, i.e. when using the radio receiver, more particularly when using the receiving device comprising the analog reception chain and the digital distortion compensation module of the analog chain, the digital module being configured from coefficients calculated during the calibration.

However, in some cases, to be able to apply the baseband correction, it is necessary to transpose the impulse response of an intermediate frequency *fi* in order to be able to filter in the real domain:

$$\mathrm{h_{nco}[n]} = h_i[\mathrm{n}] \times \exp(\mathrm{j2pi} \times \mathrm{f}_i \times \mathrm{n})$$

- $h_i[\mathrm{n}]$ is the discrete impulse response used as baseband correction;
- $\mathrm{h_{nco}[n]}$ is the impulse response $h_i[\mathrm{n}]$ transposed to an intermediate frequency;
- N represents discrete samples in the time domain

The intermediate frequency *fi* is selected at the minimum necessary, i.e. at $\frac{Bw}{2}$ in order not to resort to excessive oversampling. Indeed, to apply the correction, the sampling frequency (intermediate frequency *fi*) of the signal to which the correction is applied has to be greater than or equal to the passband of the filter:

$$\mathrm{f}_i \geq \frac{Bw}{2}$$

The impulse response h_{nco} is then applied by convolution to the signal to be corrected, the signal as such being shifted by the frequency *fi*:

$$\mathrm{S_{nco}}[n] = \mathrm{S_{out}}[n] \times \exp(\mathrm{j2pi} \times \mathrm{f}_i \times \mathrm{n})$$

$$\mathrm{S_{comp|nco}[n]} = \mathrm{S_{nco}[n]} * \mathrm{h_{nco}[n]}$$

with: * the convolution operator.

One then goes back down to baseband:

$$S_{comp}[n] = S_{comp|nco}[n] \times \exp(-j2pi \times f_i \times n)$$

The filtered signal is then processed by the next module of the digital chain.

CLAIMS

1. A method for calculating static compensation coefficients of an analog reception chain, method implemented by an electronic device (2), during a calibration phase of a reception device (3) comprising the analog reception chain and a digital compensation module implementing an FIR filter the coefficients of which are the calculated static compensation coefficients, the method comprising:

- an estimation step (A01), using a predetermined calibration signal, of a frequency response H of the analog reception chain;
- a selection step (A02), using the frequency response, of a passband BP over which the static compensation coefficients are calculated;
- a calculation step (A03) for the static compensation coefficients of the analog receiver chain, in the time domain, as a function of the passband BP and of the inverse of the frequency response H_{BP} over the passband.

2. The method for calculating static compensation coefficients of an analog reception chain according to claim 1, characterized in that the estimation step (A01) for the frequency response H of the analog reception chain comprises:

- a step of determining a reference wobbled signal, in a frequency band of a predetermined width;
- a step of obtaining a signal spectrum at the output of the analog reception chain, based on the reference wobbled signal;
- a calculation step for the frequency response H of the analog reception chain as a function of the signal spectrum at the output of the analog reception chain and the reference wobbled signal.

3. The method for calculating static compensation coefficients of an analog reception chain according to claim 1 or 2, characterized in that the selection step (A02) for the passband BP on which the static compensation coefficients are calculated comprises a selection step for selecting, within the frequency response H, a portion $\mathrm{H_{BP}}$ corresponding to the portion of the frequency response H the amplitude of which exceeds a predetermined amplitude value.

4. The method for calculating static compensation coefficients of an analog reception chain according to claim 3, characterized in that said amplitude value is predetermined by:

- calculating the mean value of the amplitude response of the frequency response H of the analog reception chain;
- subtracting a predefined tolerance value from the average value of the amplitude response of the frequency response H of the analog reception.

5. The method for calculating static compensation coefficients of an analog reception chain according to any of the preceding claims, characterized in that the calculation step (A03) for the static compensation coefficients of the analog reception chain as a function of the passband BP and of the inverse of the frequency response H_{BP} over the passband comprises at least one iteration of a calculation of coefficients of a filter of finite impulse response (FIR) current in the time domain as a function of the inverse H_i of the frequency response H_{BP} over the passband BP, said static compensation coefficients consisting of the coefficients of the FIR filter.

6. The method for calculating static compensation coefficients of an analog reception chain according to claim 5, characterized in that the calculation of the coefficients of a finite impulse response (FIR) filter comprises:

- A step of obtaining a current number N_{FFTk} of points of an inverse Fourier transform, N_{FFTk} being a power of 2 comprised between a minimum value and a maximum value;
- with the current number of points N_{FFTk} of the inverse Fourier transform *ifft*, a step of solving the equation:

$$h_{ik}(t) = ifft(H_i(f), N_{FFTk})$$

with:

$h_{ik}(t)$ being the impulse response;

$H_i(f)$ being the inverse response of the transfer function in the passband BP;

the calculation giving N_{FFTk} values of FIR filter coefficients;

- a step of digital filtering of a predetermined signal using the filter FIR the N_{FFTk} coefficients of which were obtained in the preceding step;
- an evaluation step for the quality of the FIR filter according to a performance criterion EVM.

7. The method for calculating static compensation coefficients of an analog reception chain according to claim 6, characterized in that the number of points N_{FFTk} is comprised between 16 and 512.

8. An electronic device (2) for calculating static compensation coefficients of an analog reception chain, device intended to be used during a calibration phase of a reception device (3) comprising the analog reception chain and a digital compensation module implementing an FIR filter the coefficients of which are the calculated static compensation coefficients, the electronic device comprising:

- means of estimating, using a predetermined calibration signal, a frequency response H of the analog reception chain;
- means of selecting, using the frequency response H, a passband BP over which the static compensation coefficients are calculated;
- means of calculating the static compensation coefficients of the analog reception chain, in the time domain, as a function of the passband BP and of the inverse of the frequency response H_{BP} over the passband.

9. A computer program including software instructions which, when executed by a programmable electronic system, implement a method for calculating static compensation coefficients of an analog reception chain according to any of claims 1 to 7.

10. A reception device (3) comprising an analog reception chain and a module for digital compensation of distortion of the analog chain implementing an FIR filter, the digital compensation module being configured from coefficients calculated according to the method of any of claims 1 to 7.