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Technical Notes

Method for signal conditioning and data acquisition system, based on variable amplification and feedback technique

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ABSTRACT

An original method of signal conditioning and adaptive amplification is proposed for data acquisition systems of analog signals, conceived to obtain a high resolution spectrum of any input signal. The procedure is based on a feedback scheme of the signal amplification with aim at maximizing the dynamic range and resolution of the data acquisition system. The paper describes the signal conditioning, digitization, and data processing procedures applied to an *a priori* unknown signal in order to enucleate its amplitude and frequency content for applications in different environments: on the ground, in space, or in the laboratory. An electronic board of the conditioning module has also been constructed and described. In the paper are also discussed the main fields of application and advantages of the method with respect to those known today.

1. Introduction

In many applications the amplitude and frequency content of analog signals to be detected is *a priori* unknown and can lie in a very large range of values. This is a particularly challenging problem to be managed, especially when the data acquisition and signal analysis must be carried out in extreme environments (such as in space exploration) where large data processing resources are requested together with a low power consumption, radiation-tolerant electronics and high efficiency of the device. The method proposed in the present paper is specifically useful when the power spectrum of measured signals exhibits a very large variability in different frequency bands (typically associated with several coexisting different parameters or fields), thus making particularly difficult the runtime multichannel signal analysis of the phenomenon under study (see for instance [1]). Examples can be found in laboratory, medical applications (Neurology, Cardiology, etc.), space science, aviation, telecommunications, etc. (see for instance [2–8]). Nowadays, there are many methods and devices available for spectral analysis such as *general purpose* laboratory instruments (conceived to be used in the widest range of signal analysis) [9,10] or *processing devices* made for specific needs ([11,12] and references therein). These two classes of instruments have obvious drawbacks with respect to a device based on our method. Thus, for example, laboratory spectrum analyzers, while allowing very sophisticated frequency analyses (but redundant with respect to specific uses for which our method is proposed), are oversized and generally not suitable for many in situ measurements and application areas, being bulky complex instruments of considerable overall dimensions and power consumption. The paper is organized as follows. Section 2 describes the proposed method to sample an analog signal and to reconstruct its spectrogram by using an original three-stage scheme: conditioning module (Section 2.1), acquisition module (Section 2.2) and processing module (Section 2.3). Section 3 introduces the electronic board of the signal conditioning module that has been constructed and tested for the case of 8 frequency bands. Finally, the discussion and conclusions are reported in Section 4.

2. Description of the method

The method is based on three (conditioning, acquisition and processing) interconnected modules, performing the following three main functions of data analysis procedures as illustrated in Fig. 1:

I. To perform a signal conditioning. The analog input signal is pre-amplified and sub-divided into a fixed number of frequency

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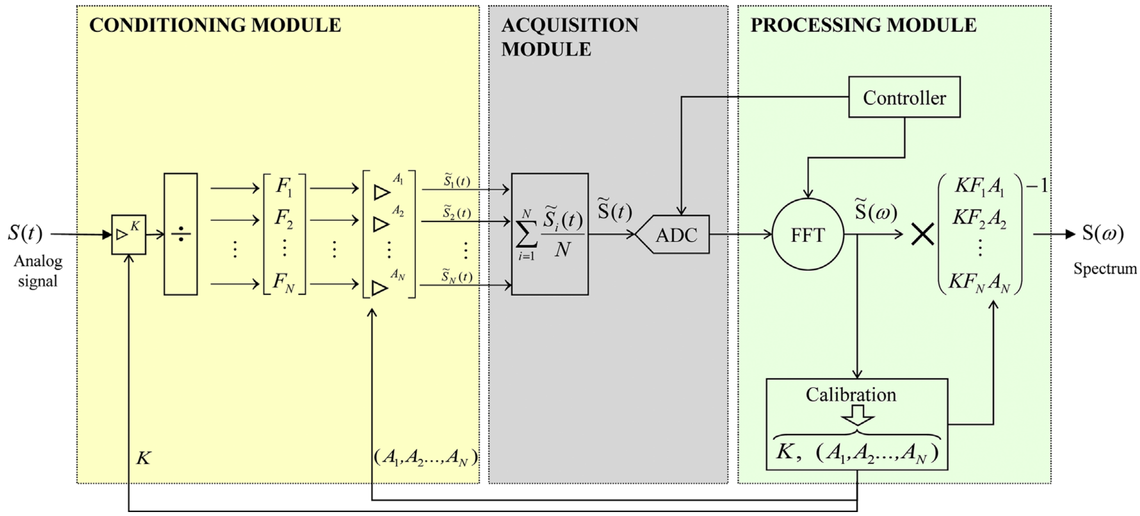

Fig. 1. Scheme of the signal conditioning, data acquisition and data processing modules.

channels, at each one of which a different tunable amplification is applied, defined as a function of weights estimated by a calibration procedure.

II. To sum up the conditioned channel signals. The resulting signal is digitized by a single analog-to-digital converter (ADC) to perform an appropriate spectrum analysis.

III. To execute the calibration procedure mentioned in item I. The calibration is accomplished on the basis of the spectrum analysis of the ADC output signal (with the aim at separately optimizing the amplification of each frequency channel) and then the spectrum of the initial input signal is reconstructed.

The method may be applied to any analog signal $S(t)$ to reconstruct its spectrogram (*i.e.* the spectrum $S(\omega)$ as a function of time) through the continuous optimization of the differential amplification of each individual signal frequency band, so that all the spectral amplitudes of the various frequency bands are made comparable between them. Once known the amplification gains (that in general vary over time), the true $S(\omega)$ spectrum can be reconstructed. The possibility to separately vary the gain of each frequency channel allows exploiting the resolution of a single ADC. The change in amplification, through a calibration procedure, optimizes the dynamic resolution of the ADC sampling in all frequency bands without "wasting bits" [13–15]. This flexibility of the method is particularly useful for signals in which the spectral content varies considerably from one frequency band to another. It follows that the actual dynamic resolution of digitization procedure, achieved by the proposed method, is higher than that of the ADC static one.

Concerning the configuration, it must be underlined that the subdivision into modules, shown schematically in Fig. 1, is to be considered only as an example, since the functional units can be placed and grouped in different configurations depending on the application or technological needs. The functional scheme described for each module may be constructed through both physical devices (hardware) and processing and control algorithms (software). All modules can be installed on a single electronic board or on separated interconnected units. In the basic scheme (described below) the method is applied to an analog input signal to produce its spectrogram. However, it is possible to exploit the modularity of the method to achieve different architectures. In particular, in case one wants to simultaneously analyze more analog input signals, it is possible to build an architecture consisting of a single processing module that simultaneously handles many subsystems. Each subsystem will consist of a conditioning module and an acquisition module and will be able to separately acquire several incoming analog signals. The single processing module will analyze all the various signals simultaneously and return all the spectra of the input signals in parallel outgoing.

2.1. The conditioning module

The conditioning module (Fig. 1) consists of an amplifier K (including one or more stages with programmable gain), a divider (with N outputs), N filters with transfer functions (F_1, …, F_N), N blocks of amplification (A_1, …, A_N). The conditioning module is connected to the acquisition module and the processing module. The output signal of K is divided by the divider block into N separate signals $S_i(t)$ ($i=1$, …, N) of equal amplitude and identical frequency content to the input one: $S_1(t)=S_2(t)=\ldots=S_N(t)=S(t)$. The N signals are sent in parallel and separately to the N filters, each one of which is chosen so as to select a particular frequency range (for its input signal), which is relevant for the purpose of the signal analysis and the specific application of the device. Also the number N of filters depends on the number of frequency bands needed to analyze the $S(t)$ signal. The limit of $N=1$ is possible and corresponds to an implementation of the method with only one frequency range analyzed. The N filters may be passive or active elements (with different stages depending on the requested filtering quality factor) and with characteristic parameters (cut off frequencies, attenuation factor, bandwidth, etc.) fixed or variable depending on the desired configuration. A possible choice for the filters is constituted by N band-pass filters centered on adjacent and consecutive bands so as to cover the whole frequency range of the input signal $S(t)$. Another possibility is to choose filters that suppress specific intervals of unwanted frequencies in the input signal. Each of the N channels is sent separately to one of the N blocks of amplification (with variable gains A_1, …, A_N) that can be constituted by one or more stages.

The output signal $\tilde{S}_i(t)$ from the ith block of amplification is given by $\tilde{S}_i(t)=S(t)\times K\times F_i\times A_i$. The gains ($A_1$, …, A_N) are set and varied (as described below) during the calibration phase by the processing module, that is activated at the start of the acquisition and that can be triggered cyclically during the acquisition, depending on the temporal evolution of the $S(t)$ input signal or on a defined schedule. Once the calibration of the amplification is performed, the blocks maintain their settings until the next calibration cycle and the N conditioned signals $\tilde{S}_i(t)$ ($i=1$, …, N) are sent from the conditioning module to the acquisition module.