A Method of Simultaneous Signals Spectrum Analysis for Instantaneous Frequency Measurement Receiver

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Abstract. In this paper the simplified problem of frequency determination for multiple simultaneously present harmonic oscillations through subsampling is considered. The proposed (used) subsampling is realized by Dirac function comb with principal frequency much less than input signal frequencies. So all signal frequencies are transformed to first Nyquist zone. We consider subsampling is implemented in three parallel channels when comb principal frequencies are differing but close one to another. Each channel includes also ADC, FFT unit and digital processing unit. Output channel information is a set of possible input frequencies, and these sets intersection is searched for finding input frequencies. Proposed system math model was developed for estimation of ambiguity of recovering values for original input frequencies. The subsampler with three parallel channels was realized as a small unit. It works as mixer with comb type heterodyne in superheterodyne receiver. The subsampler has analog bandwidth up to 5 GHz with 100 MHz principal frequencies. Its experimental characteristics are presented.

Keywords: Subsampling · Multicomponent signal
Frequency recovering · Nyquist zone · Dirac function comb

1 Introduction

Input signal spectrum analysis is an important task in many processing systems (receivers). Nowadays many types of complicated signals are used in various radioelectronics systems, such as phase and frequency modulation signal, chirp and so on. They occupied wide frequency range from RF to microwave, approx. 0.1–20 GHz, and may be present at the same time at a receiver input. One approach for such input signal spectral analysis is traditional superheterodyne receiver with tunable local oscillator, but there is a drawback – missing of pulse signal when local oscillator scans throw total frequency band. Another approach for spectral analysis is division of total frequency band to many subbands with parallel function of many receivers, but there is a drawback – an equipment will have large size and cost. One can use an instantaneous frequency measurement

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(IFM) receiver. Conventional IFM receiver consists of a delay line, mixers and RF couplers. Built on this architecture receiver works in wide bandwidth and has high performance [1]. However, this approach has one serious disadvantage - well-known problem of frequency determination in case of multicomponent input signal. To overcome such problem Prony and Pisarenko method was applied in [2]. In addition, some other methods like MUSIC (Multiple Signal Classification) assist to distinguish the frequencies in multicomponent signal spectrum [3]. But these methods require digital processor with large processing power. Next approaches are based on input signal conversion in digital form and wide usage of digital techniques – one can say this is main way in modern processing systems realization. But it has become prohibitive to sample modern wideband signals because their Nyquist rates (for sampling) may exceed specifications of the best analog-to-digital converters (ADCs) by orders of magnitude. There are approaches with decreasing the sampling frequency – subsampling approaches with parallel work of some ADCs, but they to our opinion are not simple because of acquiring samples from a periodic but nonuniform grid or multi-coset sampling with specific strategy of this type [4–6].

The goal of this paper is to propose a more simple method of multicomponent input signal spectrum analysis which combines an approach of superheterodyne receiver and a concept of signal multi-rate subsampling in some parallel channels with constant but differing from channel to channel sampling rates. We consider channel synchronous subsampling as heterodyning with Dirac function comb.

2 Theory of Operation

For the beginning let's consider a frequency shift via heterodyning. Heterodyning is a radio signal processing technique that creates new two frequencies by combining two frequencies [7]. The down-conversion operation is illustrated in Fig. 1. The frequency of input radio signal $\mathbf{F_{RF}}$ and the image frequency $\mathbf{F_{RF}}$ also are transposed to the intermediate frequency $\mathbf{F_{IF}}$. The superheterodyne receiver operation over the wide frequency range is achieved by tunable local oscillator. Such tuning takes a certain time. In a conventional approach of heterodyning an image frequency signal must be filtered out before the frequency conversion.

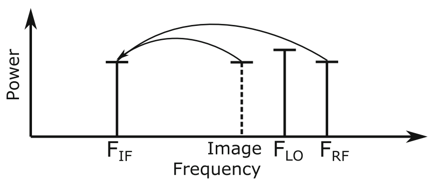

Fig. 1. Description of a conventional frequency down-conversion heterodyning.

The idea of parallel spectrum processing is based on using frequency Dirac comb generator instead of harmonic oscillator. Mathematically Dirac comb is a periodic tempered distribution constructed from Dirac delta functions [8]. The explanation of spectrum transform is depicted in Fig. 2.

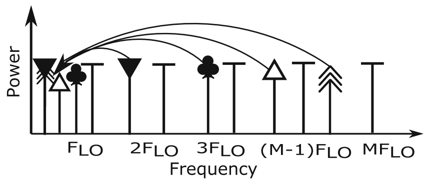

Fig. 2. Description of a parallel spectrum transform.

Here multiple harmonics from F_{LO} separate the spectrum into frequency zones. Every zone shifts to the beginning of spectrum. Thus with a knowledge of a law of spectrum transform it becomes possible to observe the wide frequency range in parallel mode. Performance of a receiver in this case depends only on Fourier transform speed and a time of frequency matrix calculation.

3 Basic Mathematical Relations

At first let's introduce key terms:
$s(t)$ is the incoming multicomponent RF signal; $\mathbf{s_{LO}(t)}$ is a signal frequency from comb generator; $\mathbf{F_{udf}}$ is a frequency range of unambiguous definition of frequency; **N** is a number of components of input signal; **M** is a number of comb generator's harmonics.

The expression for input signal is given by

$$\mathbf{s(t) = \sum_{i=1}^{N} A_i cos(2\pi f_i t)} \tag{1}$$

The signal from frequency comb generator can be written as

$$\mathbf{s_{LO}(t) = \sum_{k=1}^{M} B_k cos(2\pi k f_{LO} t)} \tag{2}$$

The signal on the mixer's output is equal to a multiplication of expressions for signal and local oscillator

$$\mathbf{s_{mixer}(t) = s(t) \cdot s_{LO}(t)} \tag{3}$$

A time domain representation of a signal on the mixer's output in general form is given by (5). Here for every input signal component like for conventional heterodyning we have two output components. The first term corresponds to intermediate frequency and the second is an image frequency.

$$\mathbf{s_{mixer}(t)} = \frac{\mathbf{1}}{\mathbf{2}}\Bigg[\underbrace{\sum_{i=1}^{M} A_i \sum_{k=1}^{N} B_k cos\{(2\pi(f_i + kf_{LO}))t\} +}_{\textbf{Intermediate}} \tag{4}$$

$$\underbrace{+\sum_{i=1}^{M} A_i \sum_{k=1}^{N} B_k cos\{2\pi(f_i - kf_{LO}))t\}}_{Image}\Bigg] \tag{5}$$

We continue the discussion under the assumption of zero image frequency components. We note that there are some technical approaches to realize filtration of image frequencies in conventional approach of heterodyning.

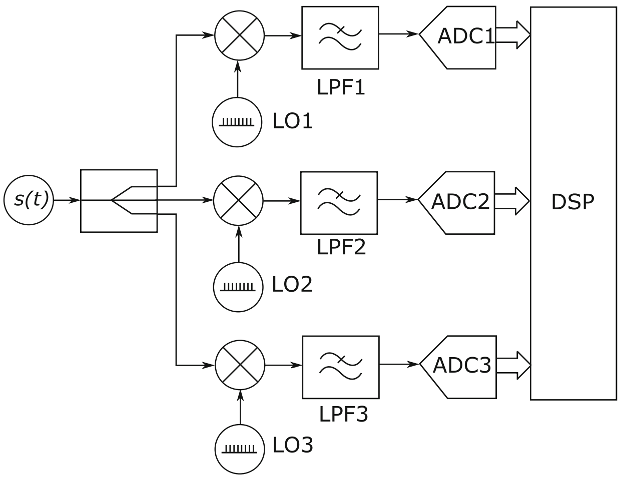

Fig. 3. Proposed IFM Receiver block diagram.

3.1 IFM Receiver operation algorithm

The block diagram of proposed instantaneous frequency measurement receiver is illustrated in Fig. 3.

The proposed IFM Receiver consists of three equivalent RF channels. The incoming signal is passed through in-phase power divider. On the next step the input signal is mixed with a signal from comb generator. Mixer output signals go to low-pass filter. The cutoff frequency is equal to $\mathbf{F_{LO}}$. Afterwards the response is sampled by analog-to-digital converter and is processed in digital form in DSP. First of all FFT is accomplished in DSP, so one can receive spectrum information: frequencies $\mathbf{F_{IF}}$ of all components and also its amplitude [9]. It should be marked that every channel gives the possibility of extracting additional information about incoming signal spectrum according to next assertion:

1. One channel provides frequency determination for one-tone signal for value not more than LO's principal frequency ($\mathbf{F_{LO1}}$).
2. Adding the second channel allows to uniquely identifying the frequency of one-tone signal in range up do $\mathbf{F_{udf}}$.
3. Three channels provide the frequency determination for N tones with values up to $\mathbf{F_{udf}}$.

where $\mathbf{M}$ is a maximum LO's harmonic number which is defined by (6) under condition (8).

$$\mathbf{M} = \frac{\mathbf{F_{LO1}}}{\mathbf{F_{LO2}} - \mathbf{F_{LO1}}} \tag{6}$$

where $\mathbf{F_{LO1}}$ and $F_{\mathbf{LO2}}$ are the principal frequencies of two comb generators

There is a requirement for values of LO's frequencies

$$\mathbf{F_{LO1}} < \mathbf{F_{LO2}} < \mathbf{F_{LO3}} \tag{7}$$

The frequency range of unambiguous frequency definition is given by

$$\mathbf{F_{udf}} = \mathbf{M} \cdot \mathbf{F_{LO1}} \tag{8}$$

With taking into account of described relations the values of possible frequency components from first channel can be calculated as

$$\begin{pmatrix} F_{LO1} - F_{IF_1} & 2F_{LO1} - F_{IF_1} & \dots & MF_{LO1} - F_{IF_1} \\ F_{LO1} - F_{IF_2} & 2F_{LO1} - F_{IF_2} & \dots & MF_{LO1} - F_{IF_2} \\ \dots & \dots & \dots & \dots \\ F_{LO1} - F_{IF_{N-1}} & 2F_{LO1} - F_{IF_{N-1}} & \dots & MF_{LO1} - F_{IF_{N-1}} \\ F_{LO1} - F_{IF_N} & 2F_{LO1} - F_{IF_N} & \dots & MF_{LO1} - F_{IF_N} \end{pmatrix} \tag{9}$$

where F_{LO1} is a first harmonic of comb generator and $F_{IF_1} \dots F_{IF_N}$ are the intermediate frequencies.The values from the second and third channel can be obtained by the same way.

In order to get the initial frequencies of input signal the intersection between elements of matrices should be found.

4 Simulation

The simulation was performed with MATLAB. The graphical user interface (GUI) with main IFM receiver functional blocks according to above described functional diagram (see Fig. 3) was developed and it is shown in Fig. 4.

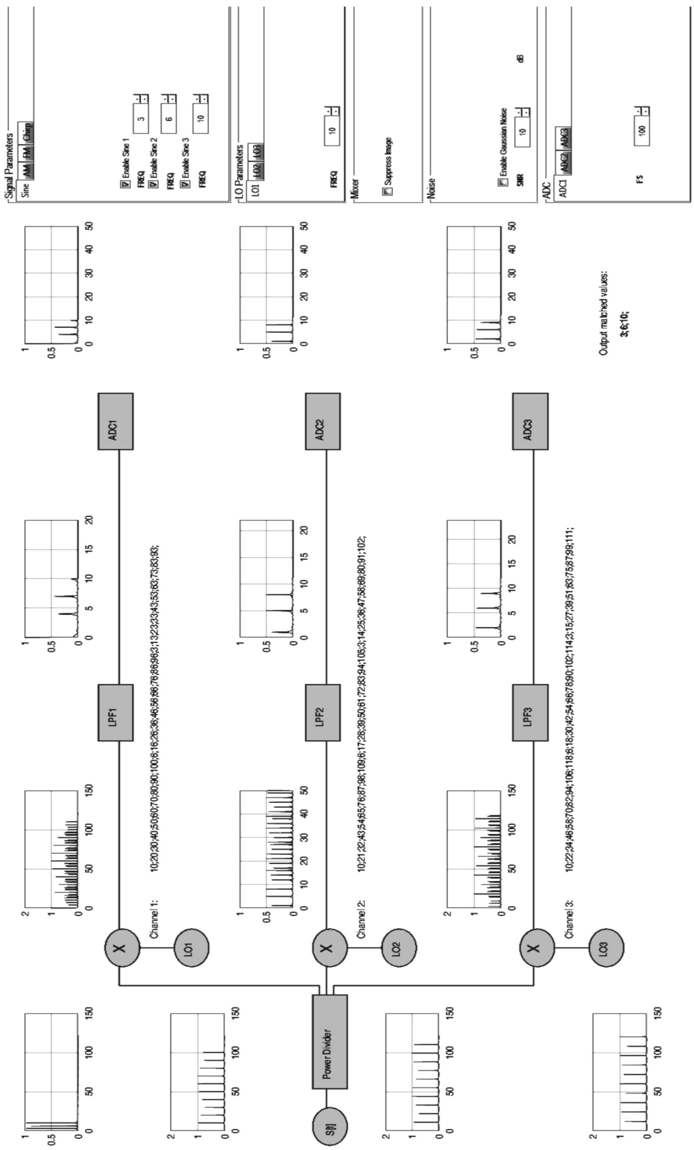

Fig. 4. Simulator GUI.

We choose 3 parallel working channels in IFM receiver model functioning. Moreover, the library of various simple signal types like sine, AM, chirp was also added. The spectrum after every block can be evaluated, see insertions - spectrum diagrams in Fig. 4 in different flow paths. The algorithm of peak search and frequency values intersection was encapsulated. In general the model of instantaneous frequency measurement has confirmed the assumption about the possibility of spectrum analysis in parallel mode. Some details about our approaches of IFM receiver model functioning simulation one can find in [9].

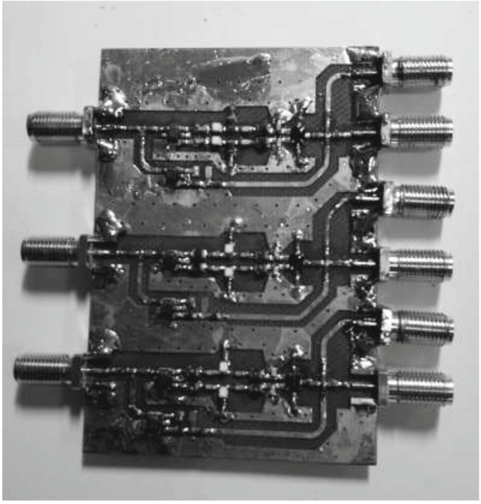

Fig. 5. Mixer with comb generator prototype.

5 Measurements

We developed some key units of IFM receiver. For frequency comb generation a circuit with step recovery diode (SRD) was chosen. The board of the instantaneous frequency receiver main part - mixer was manufactured on FR4 substrate. The photo of our mixer with comb generator prototype is shown in Fig. 5. One can see 3 inputs for connection to in-phase power divider (left side), 3 outputs for connection to 3 ADC and 3 inputs to feed LO principal frequencies (right side).

The spectrum of outputs of the comb generators were evaluated with Keysight N9010A EXA signal analyzer. The spectrogram of one channel comb is presented in Fig. 6, in this case frequency $\mathbf{F_{LO}}$ is 100 MHz. The unevenness of harmonics power was not more than 5 dB in frequency range up to 5 GHz.

On the input of the mixer prototype two-tone sine signal was applied with frequencies $\mathbf{f_1}$ = 2359 MHz and $\mathbf{f_2}$ = 1261 MHz. In the Fig. 7 the mixer one channel output spectrum is shown.

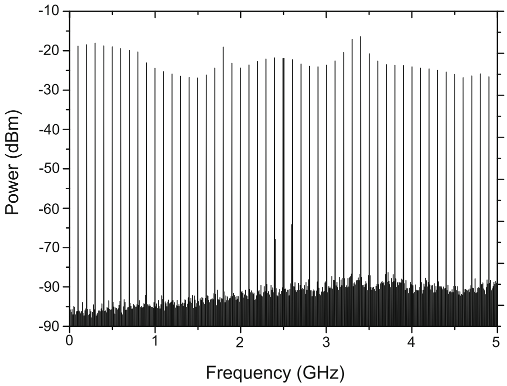

Fig. 6. The comb generator output spectrum.

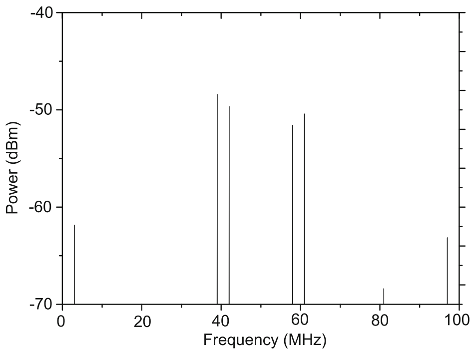

Fig. 7. The filter's output signal's spectrum for two-tone sine signal with frequencies 2359 and 1261 MHz.

It can be seen that there are four power peaks. Two of them are correspond to intermediate frequency and the two other are matched to image frequencies. We have to note that image signal frequency filtration not used here. From signal with frequency f_1 we received $f_{IF1} = f_1 - 230 \cdot F_{LO} = 59\,\text{MHz}$ and $f_{IM1} = f_1 - 231 \cdot F_{LO} = 41\,\text{MHz}$. From signal with frequency f_2 we received $f_{IF2} = f_2 - 120 \cdot F_{LO} = 61\,\text{MHz}$ and $f_{IM2} = f_2 - 121 \cdot F_{LO} = 39\,\text{MHz}$. This behavior is consistent with theoretical model. Now we are developing an algorithm of proper input signal frequencies identification (recovering) on processing data of 3 parallel receiver channels, see [9].

6 Conclusion

The idea of parallel spectrum analysis exploiting frequency comb generator as LO in receiver mixer was proposed. The algorithm for input signal frequencies determination was described with basic mathematical relations. The model of instantaneous frequency receiver with 3 parallel channels was developed. Simulation confirmed the assumption about the possibility of simultaneously presented signals (multicomponent signal) transformation into frequency baseband and its initial frequencies recovering. The receiver based on such approach has the following advantages. In comparison with the superheterodyne receiver it doesn't need any control of local oscillator. In addition, this approach eliminates the need of high-speed analog-to-digital converters. Compared to conventional approach of instantaneous frequency receiver the proposed receiver allows working with multicomponent signals. The developed mixer prototype has measured unevenness of spectral components power on the output of comb generator not more than 5 dB for frequency range up to 5 GHz. Nevertheless, the fundamental limitation of this parallel analysis approach is an existence of image channels. It leads to erroneous input frequencies recovering results. In the future work the image channel rejection will be made by using special algorithms.

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