DESIGN OF A RADIOFREQUENCY RECEIVER FOR LOW FIELD MAGNETIC RESONANCE APPLICATIONS

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Abstract: Radiofrequency (RF) receivers are key components in magnetic resonance imaging (MRI) systems. Birdcage coils are widely used for their ability to achieve high signal-to-noise ratio (SNR), however, when they are used in low field scanners, both the detected signal SNR and the coil quality factor Q decrease with frequency, thus reducing the image quality. In this paper we initially describe the design of a high quality birdcage coil, demonstrating the possibility of obtaining high quality images even using low field scanners (0.18T). To convert the RF signal from the coil into a suitable form for an analog-to-digital (ADC) converter, a classic analog superheterodyne circuit is generally employed. This kind of receiver is simple but also presents many problems for filters and amplifiers design. In this study we implemented a phase-quadrature digital detector system using an alternative technique called undersampling. Our receiver is constituted by a passband filter, an ADC and a DDC (Digital Down Converter) for the frequency translation and filtering to process the signal using a PC. In this work we give all the detailed specifications and hardware design issues for the receiver that is designed to be used in a dedicated scanner guaranteeing top performances with low cost.

Introduction

In 1985 Hayes et al. [1] described the first application of the birdcage coil in MRI systems, explaining its ability to generate wide field of view (FOV) with high RF magnetic field homogeneity in transmission and to achieve high signal-to-noise ratio (SNR) in reception. However, when the birdcage coil is used in scanners at very low static magnetic field, both the SNR and the quality factor (Q, defined as the ratio of stored energy to energy lost per cycle) decrease with frequency, thus reducing the image quality. In this paper we initially show how to design a high quality birdcage coil making a correct choice of the electrical parameters (conductor geometry and capacitors quality factor) that affects the coil overall performance. Q factor and ratio between unloaded and loaded were evaluated to characterize the birdcage quality.

Simultaneously, we began the design of the RF receiver. A MRI receiver is employed to convert the received RF signal from the coil into a suitable form for an ADC converter. Generally, the receiver is a superheterodyne circuit which demodulates the RF signal into a low frequency band and this process is done with respect to a reference frequency equal to the emitted RF radiation [1]. The scheme of such classic analog MRI receiver is shown in Figure 1. The FID (Free Induction Decay) signal emitted by the sample is picking up by a RF coil and then it is processed by a circuit which maximizes the energy to transfer to the amplifier. The preamplifier is used to amplify the signal while minimizing the noise, successively the RF signal is taken to a phase-quadrature detector system [2].

This type of receiver is very simple but also presents many problems: in fact, filters and amplifiers must have the same phase and frequency characteristics and there must be an extreme accuracy on 90° phase difference between the two reference signals otherwise a first type distortion can occur (“ghost” artifacts on the image) [3].

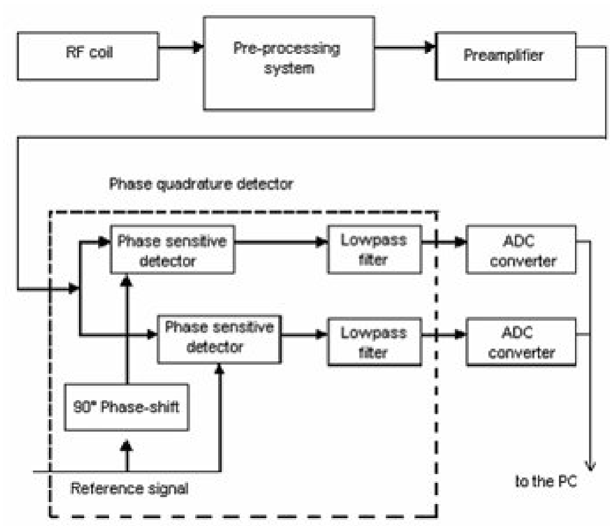

Figure 1: Block diagram of a MRI analog receiver

For these reasons we implemented a phase-quadrature digital detector system without any problem of phase errors, unbalance errors and ghost artifacts.

Our digital receiver is constituted by a passband filter (antialiasing filter), an ADC to digitalize the signal with the undersampling technique, and a DDC (Digital Down Converter) for the frequency translation and filtering to process the signal using a PC.

We initially explained the undersampling technique theory, then we reported the hardware specifics to realize the circuit and finally the test results of the device.

Materials and Methods

For the design of birdcage coils, we used a coils simulator previously developed by some of the authors [4], that allows to calculate the resonant frequency and the field pattern as function of conductor geometry and legs number.

Several birdcage coils, tuned at 7.66 MHz, were designed and built, using conductors with different shapes (strip or cylindrical rod) and capacitors with different quality factors. The testing of coils showed that high quality capacitors are absolutely needed to obtain high performance coils; moreover, the best performance was provided by birdcage coil made of cylindrical rod conductor. This fact is related to a better current distribution inside the cylindrical rod conductor with respect to the strip one [5].

The built birdcage coil (Fig. 2) is a low pass version with 11 cm height and 14 cm diameter; it was realised using ATC (American Technical Ceramics, USA) high quality capacitors and a cylindrical rod conductor of 4.5 mm diameter.

Figure 2: The built lowpass birdcage coil

Regarding the demodulator, the input of this detector is the MR radiofrequency signal represented by a frequency distribution around the transmission frequency f_0 (the Larmor's frequency); the detector translates it of a quantity equal to f_0 in order to centre it around the zero frequency. This is the classic based band detection and it is obtained by multiplying the components of the MR signal with a reference signal at f_0 frequency: the phase-sensitive detector output is the sum of a component centred around zero frequency with another component centred around $2f_0$.

Successively, a lowpass filter removes the components centred around $2f_0$. Finally the complex signal (two channels) is converted by two analog-to-digital converters for processing by PC.

Generally, the input signal of the MR receiver has the following form:

$$S(t) = f(t)\cos(2\pi f_0 t) + jg(t)\sin(2\pi f_0 t) \qquad (1)$$

where $f(t)$ and $g(t)$ are the phase and quadrature components of the RF signal.

The RF signal is emitted by the nuclei in the object to be imaged and it is picked up by a RF receive coil, in our case the built lowpass birdcage coil.

Generally, in a digital receiver the direct sampling (or pass band sampling) is used: the signal is sampled without reporting it around the zero frequency (base band). In this case the Signal-to-Noise Ratio (SNR) is calculated by the following equation [6]:

$$SNR = 6.02N + 1.76dB \qquad (2)$$

where N is the number of bit of the ADC converter; for a wanted SNR of 90-100 dB the ADC converter must have at least 14-16 bit.

To increase SNR an oversampling technique can be used. Oversampling methods for Analog-to-Digital Converters are based on sampling an analog signal at a much higher rate, filtering the samples with a digital lowpass filter and reducing the sample rate by decimation. In this case the equation for SNR is:

$$SNR = 6.02N + 1.76dB + 10\log\left(\frac{f_s}{2B}\right) \qquad (3)$$

where f_s is the sampling frequency, which must be greater than the double of the signal maximum frequency, according to Nyquist's theory, and B is the signal bandwidth.

When the sampling frequency becomes greater, the noise will tend to a lower value, because it is distributed on a greater range of frequency, so the SNR increases; the additional term in the previous expression is defined as *process gain* and it provides a 3 dB increment for each doubling of f_s frequency.

As described above, the output signal of the birdcage coil has a central frequency of 7.66 MHz and a bandwidth of about 55 KHz, therefore with oversampling the signal has to be sampled at a frequency greater than 2 * 7.66 MHz.

Otherwise an alternative technique [7] that is defined as *undersampling* or *Super-Nyquist* configuration can be used, where the sampling rate is determined only by the bandwidth of the signal.

The Shannon's theorem establishes that a signal with a maximum frequency of f_a must be sampled by a sampling frequency $f_s > 2f_a$ in order to avoid loss of information. A consequent of this theorem is the Nyquist's criteria: if $f_s < 2f_a$ you have *aliasing* that is the noise caused by the overlapped signal replicas.

This criteria is valid when the signal extends from zero frequency to f_a but if the signal does not extend from DC (Direct Current), the sampling rate depends on both its band and its position on the frequency spectrum.

Therefore, the Nyquist's criteria establishes that f_s must be greater than $2B$, where $B = f_2 - f_1$ is the signal bandwidth.

Since Nyquist's band is defined as the frequency range from zero to f_s/2, the frequency spectrum can be subdivided into many Nyquist's zones with amplitude of 0.5 f_s.

In general the sampling of a signal with B band at a rate of f_s causes two components of aliasing: one at f_s + B and another at f_s - B. By the effect of the sampling a signal replica or the signal itself falls into the first Nyquist's zone (0-f_s/2): for this reason every signal (noise or useful signal) that falls out of the interesting band must be filtered before the sampling.

Therefore, a very important part of the receiver is the bandpass filter.

In the undersampling technique the aliasing phenomenon is an advantage: the signal replica that falls in the first Nyquist's zone contents all original signal information. Obviously, the band of the original signal must be limited to a single Nyquist's zone by the bandpass filter (anti-aliasing filter).

The sampling frequency f_s must be chosen considering the following two expressions:

- $f_s > 2B$ that is the super Nyquist condition (B is the signal band);
- $f_s = \dfrac{4f_c}{2NZ-1}$ where f_c is the signal central frequency, NZ is the Nyquist's zone (NZ=1,2,3…).

The second equation assures that f_c is in the centre of a Nyquist's zone. For our signal, if we choose a sampling frequency of 2.7648 MSPS, the signal falls into the sixth Nyquist's zone.

In order to calculate where the aliased signal falls this equation can be used:

$$f_c \bmod f_s \tag{4}$$

and in our case

$$7.66\ \text{MHz} \bmod 2.7648\ \text{MHz} = 2.1304\ \text{MHz} \tag{5}$$

This frequency value lies in the second Nyquist's zone, therefore, the signal must be translated in the centre of the first Nyquist's zone, using a Digital Down Converter (DDC) which acts like a phase quadrature detector. The DDC control software calculates the translation frequency using the following expression:

$$3\,f_s - f_c = 634.4\ \text{KHz} \tag{6}$$

The chosen sampling frequency allows to space out the replicas in the frequency spectrum, in order to relax the anti-aliasing filter performance and to reach an oversampling factor equal to:

$$10\log\left(\frac{f_s}{2B}\right) = 14dB \tag{7}$$

This is the minimum value for the oversampling factor; in fact, the received signal bandwidth B is also related to the field of view (FOV) and the frequency encoding gradient, G_f, according to the equation [8]:

$$FOV = \frac{B}{\gamma\, G_f} \tag{8}$$

In our case, with a gradient magnetic field of 0.5 G/cm and γ =42.58 MHz/T for the Hydrogen atom, the field of view is equal to:

$$\frac{55KHz}{42.58MHz/T \times 0.5 \cdot 10^{-4}T/cm} = 25cm \tag{9}$$

with a process gain values of 14 dB (see Eq. 7).

If the FOV is smaller and the product γG_f remains constant, the signal bandwidth decreases while the process gain increases.

For our intended use of musculoskeletal limbs studies dedicated MRI, a standard FOV value is equal to 10 cm and the bandwidth calculation provides:

$$B = FOV \times \gamma G_f = 21.29KHz \tag{10}$$

and, according to Eq. 7, the process gain has a value of

$$10\log\frac{2.7648MHz}{2 \times 21.29KHz} \cong 18\ \text{dB} \tag{11}$$

Figure 3 shows the block diagram of the designed digital MR receiver. The FID signal has detected by the birdcage coil and, by a pre-processing system, it has taken to a preamplifier which maximises the SNR; successively, a variable gain amplifier adapts the signal to the input dynamic of the ADC converter

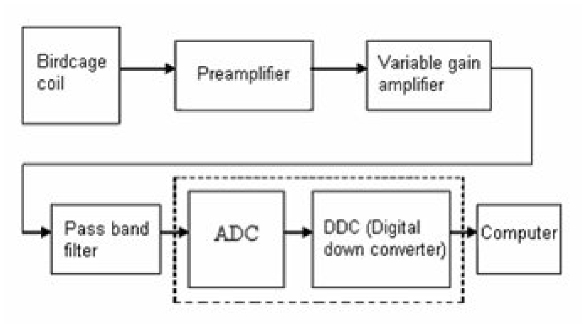

Figure 3: Block diagram of a MRI digital receiver

Next block is a passband filter which eliminates the undesired frequency outside the interesting band. The ADC converter digitalizes the signal for the last block, the digital demodulator DDC.

The sampling frequency we chose is f_s =2.7648 MHz, obtained by the division for 4 of a commercial oscillator frequency (11.0592 MHz). This f_s permits to centre the signal in the Nyquist's zone: after the sampling, the aliased signal falls in the second Nyquist's zone (using the (4)), therefore in order to obtain the exact position of the signal in the first Nyquist's zone, the DDC must perform a frequency translation of 634.4 KHz according to the Equation (6) (see Figure 4).

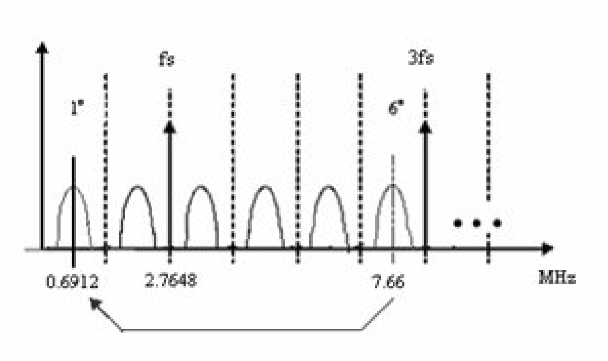

Figure 4: The aliased signals after undersampling application

Using a free downloaded software simulator (FilterMaster v. 1.0) [9], we designed a 6^{th} order elliptic passive passband filter in order to achieve a good attenuation in the Nyquist's zone corresponding to the interesting signal.
The simulated filter specifics are describe below:

- Lower pass band limit frequency: 7.5 MHz
- Upper pass band limit frequency: 7.8 MHz
- Lower stop band limit frequency: 6.573 MHz
- Upper stop band limit frequency: 8.9 MHz
- Pass band attenuation: 0.1 dB
- Return loss: 16.43 dB
- Stop band attenuation: 61.03 dB
- Degree: 6

To achieve good performance, it is necessary to use high quality factor capacities: we chose ATC (American Technical Ceramics, USA) capacitors, with high quality factor Q. In order to obtain the fine tuning of the frequency responsivity of the filter we used high quality capacitive trimmers. Regarding the inductors, we used windings of copper wires on air or on ferromagnetic support to achieve the desired value.

Figure 5 shows the passband filter scheme: the input/output resistors are characteristic of the coaxial cable used to transmit and capture the signal. The real specifics obtained by the built filter are:

- pass band = 300 kHz
- pass band attenuation = 2 dB
- stop band = 1.25 MHz
- stop band attenuation = 40 dB

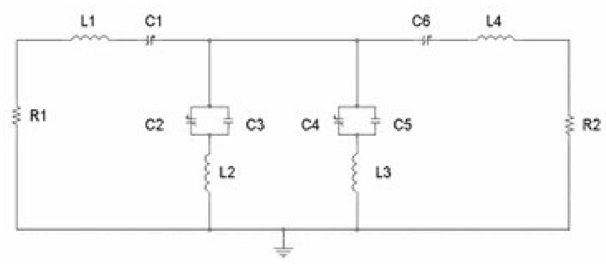

Figure 5: Electric scheme of the pass band filter

In this project we used Analog Device AD9243 analog-digital converter, characterized by a maximum sampling rate of 3 MSPS and a dynamic of 14 bit. This converter uses a four block pipeline architecture with an input sample-and-hold amplifier (SHA) with wide band.

It has a SINAD (Signal to Noise And Distortion) of 77 dB and a SNR of 80 dB at our frequency of interest (f_c = 7.66 MHz). These two parameters are defined in this way: the SINAD is the ratio between the rms value of the input signal and the rms value of the other spectral components under the Nyquist's frequency sum, including all harmonics without the DC components, while the SNR is defined as the ratio between the rms value of the input signal and the rms value of the first six harmonics sum included the DC component.

The DDC acts the following functions on the ADC output samples: frequency translation, frequency decimation and digital filter. We chose the Analog Device AD6620 that is characterized by four main signal processing stages:

- a Frequency Translator
- two Cascaded Integrator Combo FIR Filters (CIC2, CIC5), fixed coefficients decimator filters
- a RAM Coefficient FIR Filter (RCF), programmable coefficient decimator filter.

The first signal processing stage is a frequency translator consisting of two multipliers and a 32-bit complex Numerically Controlled Oscillator (NCO). The NCO serves as a quadrature local oscillator capable of producing any frequency between $-f_s/2$ and $+f_s/2$ with a resolution of $f_s/2^{32}$.

The control word, NCO_FREQ, is interpreted as a 32-bit unsigned integer. To translate a signal centred at f_c to zero frequency, NCO_FREQ can be calculated using the following equation:

$$NCO\_FREQ = 2^{32} \times \bmod \frac{f_c}{f_s} \qquad (12)$$

The second stage is constituted by a fixed coefficient decimator filter and the final signal processing stage is a sum-of-products decimating filter with programmable coefficients.

The final response of the AD6620 is the composition of three cascaded filters responses, where each stages allows to obtain more and more tight transition bands.

The clock signal is utilized to interrogate the device input port and to synchronize the successive processing stages. In our system we used a hybrid commercial oscillator Sunny Electronics SCO-020 with a frequency of 11.0592 MHz and a PLL (Phase Locked Loop) Motorola MC88915 with a frequency ratio of 1:2, 1:1, 2:1 respect to the input frequency.

The external loop filter is a home-made large band filter that decreases the device sensibility to power oscillation typical of a high frequency digital system.

Principal components of our digital receiver (AD6620 and AD9243) are available as evaluation boards. The input data of the AD6620 evaluation board are processed by two latch, successively they are available in real time to be elaborated by a microprocessor (serial or parallel output), or memorized in a FIFO (First Input First Output) memory and then downloaded on PC by the parallel port.

The Evaluation Board provides a Bypass modality: in this way the AD6620 is excluded and the data input flow through a second series of latch and are successively memorized by the FIFO memory to be processed later; this modality allows to verify the AD9243 function. Moreover, in the evaluation board there are some jumpers that allow to introduce some delay in order to obtain the right temporization of the device.

The AD6620 simulation software allows to set up the digital filters specifics (pass band, stop band and attenuations). The signal bandwidth is B=21.29 kHz, therefore the final decimation filter factor must be equal to

$$\frac{f_S}{2 \cdot B} = \frac{2.7648MHz}{2 \cdot 21.29KHz} = 64.9 \qquad (13)$$

Since the decimation factor must be an integer, we chose a bandwidth equal to 21.6 KHz. Moreover, we established that the final digital filter must have the following specifics:

- pass band = 21.6 kHz
- pass band attenuation = 0 dB

After the setting up of sampling parameters, the software calculates the decimation factors of the three filters CIC2, CIC5, RCF in order to obtain the requested specifics.

In order to control the AD6620 we used the evaluation board support software: the software graphic interface consists of two windows, the AD6620 Monitor and the AD6620 Controller which allow to set up the AD6620 parameters and download the data from the FIFO memory to the PC. The AD6620 Controller allows to set up all functional parameters as the data input modality (in our case Single Channel Real), the sampling rate, the shift frequency value of NCO (automatically set up) and the configuration of the digital filters (CIC2, CIC5, RCF). The AD6620 Monitor allows to control and process the FIFO memory data: in particular, it allows to analyze the data FFT (Fast Fourier Transform), their value in the time domain, the I vs Q graphic (phase and quadrature components of data), to evaluate the SINAD and SNR values, the second and third harmonic level of the wider signal in the spectrum (in dBc) and the level of the successive spurious wider signal measured respect to the full scale (in dBFS).

Results

The birdcage coil was tested using laboratory instrumentation consisting of an HP 3577A network analyzer and a dual-loop probe [10]. The first test consisted in the evaluation of unloaded quality factor, provided a good value of 477. Then, we measured the quality factor for the loaded coil, where the load consisted of a cylindrical homogeneous phantom of saline solution that simulates the knee conductivity (diameter 11cm, length 20cm, constituted of 55mM of NaCl and 5mM of $NiCl_2$). The ratio between the Q factor of unloaded and loaded coil was 2.93 and the corresponding RF signal bandwidth was about 55KHz.

For the receiver testing we used a function generator (HP3325A, Hewlett Packard) providing, as input signal, a sinusoidal wave at Larmor's frequency ($f_0 = 7.66$ MHz) with an amplitude near to the full scale value of the converter ($\cong$ - 0.5 dBFS on 2 Vpp) in order to maximize its performance.

In the first measurement, we bypassed the DDC for the evaluation of the ADC output signal FFT. The performance of the analog-digital converter are reported below:

- SINAD = 63.8 dB
- SNR = 65 dB
- second harmonic level respect to the principal component = -71 dBc
- third harmonic level respect to the principal component = -84 dBc

Successively, we evaluated the complete system composed by the ADC and the DDC, choosing a digital signal bandwidth (B) equal to 21.6 kHz.

The AD6620 centred the fundamental component of the signal around the zero frequency, guaranteeing a processing gain that provides a signal-to-noise ratio equal to

$$\text{SINAD}=\text{SNR}=83\ \text{dB} \qquad (14)$$

Another important result of using the digital demodulator is the fact that it permits to obtain a perfect quadrature of the two signals and to reduce the ghost artifacts.

Discussion

In the present paper we initially demonstrated that a high quality birdcage coil can be designed even for low field scanners. In particular, using high quality capacitors and a cylindrical rod conductor we realised an efficient birdcage coil, with an unloaded quality factor of 477 and a ratio between the quality factors (unloaded/loaded) of 2.93. It is a good result, considering that for a coil tuned at f_0, the SNR of the MR experiment is proportional to $\sqrt{Q}$ [11]. Successively, we used the undersampling technique for the design of a MR signals receiver system for the demodulation of the passband signal centred at the frequency of 7.66 MHz and with a band of 55 kHz.

The signal digitalize process acts as a method to elaborate the signal with excellent performance respect to an analog receiver system.

The ADC test shows dynamic performance characterized by a SINAD of 63.8 dB and a SNR of 65 dB. The use of the DDC as phase quadrature detector and digital filter provides a performance increasing, guaranteeing the perfect quadrature of the two signals and a high performance filtering; this permits to reduce the ghost artifacts that affect the MR images obtained with a classic receiver. Moreover, it is perfectly adaptable to each modification of the system thanks to its programmability.

With DDC, a value of SNR of about 85 dB was obtained: it is very good result considering the non ideal conditions of the tests.

The useful of undersampling technique in the acquisition of magnetic resonance signals has already been demonstrated in literature [7]: in our paper we reported the theory of this technique and described in detail all components of such a digital receiver, therefore the presented low cost architecture becomes completely home-made realizable.

Conclusions

A prototype of a RF receiver chain for low field MR applications was performed using an optimized high-quality birdcage coil. Undersampling technique acted as an excellent method for the demodulation of a passband RF signals at low frequency. The intended use of our sub-system should be as receiver chain in a low cost dedicated MRI scanner (e.g. in a machine for musculoskeletal limbs studies).

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