HOW DOES AN SDR TRANSCEIVER WORK?

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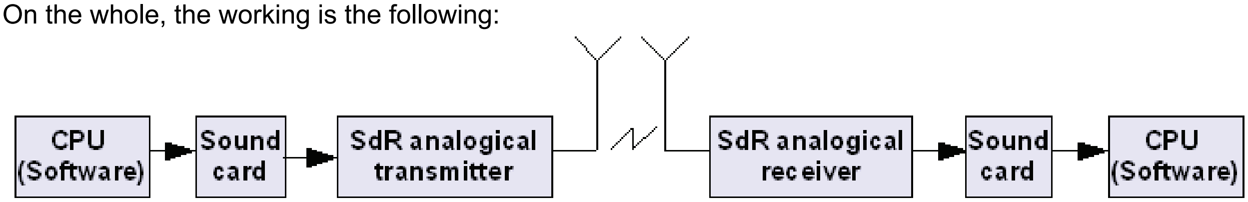
This paper will describe an SdR transceiver, from an algebraic point of view (simple mathematics). Following are several principle electronic diagrams and two snapshots of the Multipsk SdR demodulator/modulator.

Introduction

Following are two simplified diagrams related, in one hand, to an SdR transmitter and, in the other hand, to an SdR receiver using the PC for the AF part. Each diagram is accompanied with an explanation, expressed algebraically. The diagrams presented are as straightforward as possible. The Multidem and Multipsk software's from the author (http://f6cte.free.fr) which allow the management of the AF part of the SdR transceivers, either on transmission or in reception, are much more complex but fundamentally, it is not more that an extension of the principles exposed below, through different digital tools (Fourier transform, decimation, interpolation...).

On the whole, the working is the following:



The sound card is the digital → analogical interface (in reception) or the analogical → digital one (in transmission).

The scenario followed by the description is as follows: the software transmits an AF signal, which after analogical conversion, is transformed in HF USB signal in the transmitter before being driven to the antenna. The SdR receiver of the other Ham receives the HF signal on a frequencial band located on both sides (LSB and USB) of the HF central carrier. This HF band is transformed in an AF band, by a heterodyne process. After conversion in digital, the initial AF signal is extracted from the received BF band.

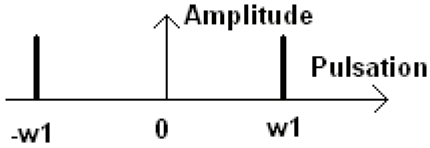
Transmission

Agreement: « x » can be written « . » to simplify the writing.

Some useful formulas:

- $\mathrm{Cos}(\theta-\pi/2)=\mathrm{Sin}(\theta)$
- $\mathrm{Cos}(\theta)=\mathrm{Cos}(-\theta)$
- $\mathrm{Sin}(\theta-\pi/2)=-\mathrm{Cos}(\theta)$
- $\mathrm{Sin}(\theta)=-\mathrm{Sin}(-\theta)$
- $\mathrm{Cos}(a).\mathrm{Cos}(b)=(\mathrm{Cos}(a+b)+\mathrm{Cos}(a-b))\ /\ 2$
- $\mathrm{Sin}(a).\mathrm{Sin}(b)=(\mathrm{Cos}(a-b)-\mathrm{Cos}(a+b))\ /\ 2$
- $\mathrm{Cos}(a).\mathrm{Sin}(b)=(\mathrm{Sin}(a+b)-\mathrm{Sin}(a-b))\ /\ 2$

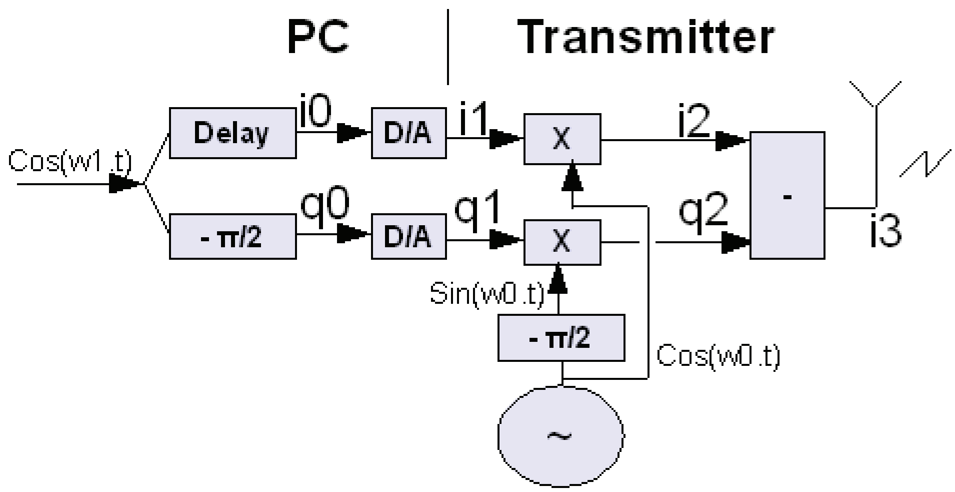
Representation of a real signal in terms of carriers defined by their frequencies :
A real signal under the $\mathrm{Cos}(w1.t)$ form can also be written :
$((\mathrm{Cos}(w1.t)+i.\mathrm{Sin}(w1.t)) + (\mathrm{Cos}(w1.t)-i.\mathrm{Sin}(w1.t)))\ /\ 2$ or
$(\exp(i.w1.t)+\exp(-i.w1.t))\ /\ 2$,
which corresponds to two rotating vectors, one at the pulsation w1 and the other at the pulsation –w1, the vectors rotating in the opposed direction. Their modules are represented on the spectrum :



The AF signal to be transmitted can be either a PSK31 signal, a pure carrier (« Tune »), a CW signal or a phone signal. It will be supposed, here, an AF carrier of f1 frequency and w1 pulsation (with w1 = 2 x π x f1) written y=Cos(w1.t).

It is desired to modulate, in USB, a HF carrier of f0 frequency and w0 pulsation.

The SdR transmitter diagram is the following :



- The « Delay » is done with an « all-pass filter ». It simply compensates the delay (φ) introduced by the « -π/2 » phase shifter. This one introduces a necessary delay, because the goal is not to shift a pure carrier but to shift an AF band. Consequently, it is necessary to use a phase shifter filter, introducing some inevitable delay.
  <u>Note :</u> the phase shift is equal to « -π/2 » for the positive frequencies (exp(i.w.t)) but is equal to « π/2 » for the negative frequencies (exp(-i.w.t)), because the q0 signal is real.
  <u>Proof :</u>
  Cos(w.t)=(exp(i.w.t)+exp(-i.w.t)) / 2
  Cos(w.t-π/2) =(exp(i.w.t-π/2)+exp(-i.w.t+π/2)) / 2=(exp(i.w.t) x exp(-π/2))+(exp(-i.w.t) x exp(π/2)) / 2
  exp(-π/2) is the phase difference of -π/2 applied to exp(i.w.t) and
  exp(π/2) is the phase difference of -π/2 applied to exp(-i.w.t)

- In i0, it is found Cos(w1.t- φ)
- In q0, it is found Sin(w1.t- φ) because Cos(θ-π/2)=Sin(θ)
  <u>Proof :</u>
  Cos(θ-π/2)= (exp(i.θ) x exp(-π/2))+(exp(-i. θ) x exp(π/2)) / 2
  Cos(θ-π/2)= (exp(i. θ) x -i)+(exp(-i. θ) x i) / 2 = (exp(i. θ)-exp(-i. θ)) / (2.i) = Sin(θ)

- « D/A » means « Digital/Analogical ». The conversion is done by the sound card.
- In i1 and q1, it is found the same as in i0 and q0 but in analogical.

- The HF carrier is introduced through a double signal in quadrature (Cos(w0.t) / Sin(wo.t))
- In i2, it is found Cos(w1.t- φ) x Cos(w0.t) = (Cos((w1+w0).t-φ) + Cos((w1-w0).t-φ)) / 2
- In q2, it is found Sin(w1.t- φ) x Sin(w0.t) = (Cos((w1-w0).t-φ) - Cos((w1+w0).t-φ)) / 2

- In i3, it is found i2-q2
  = (Cos((w1+w0).t-φ) + Cos((w1-w0).t-φ)) / 2 –((Cos((w1-w0).t-φ) - Cos((w1+w0).t-φ))) / 2
  = Cos((w0+w1).t-φ) <u>It is the wished USB signal (w0+w1).</u>

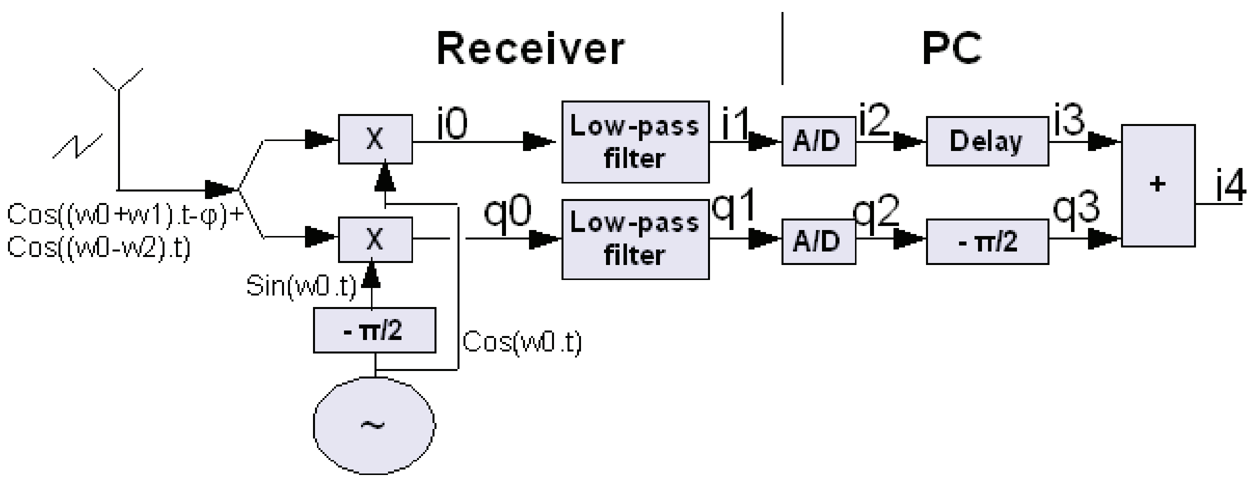
  It must be noted that:
  - if the operator « + » had been applied (instead of « - »), it would have been found Cos((w1-w0).t-φ)=Cos((w0-w1).t+φ), hence a LSB signal (w0-w1),

  - if the initial phase shift had been equal to π/2 instead of -π/2, due to the equality Sin(θ)=-Sin(-θ), the necessary operator to transmit in USB, would have been a « + », which is much easier to realize in analogical than a « - ».

Reception

The start signal is the USB one at the output of the transmitter : Cos((w0+w1).t-φ), φ being any phase and being supposed to integrate the transmission delay between the transmitter and the receiver. However, the receiver can also receive, in the inferior pass band (LSB), a parasitic signal which will be supposed equal to Cos((w0-w2).t). The goal will be to eliminate it.
Consequently, in the pass band of the receiver input, there is the composite signal:
Cos((w0+w1).t-φ) + Cos((w0-w2).t)

The SdR receiver diagram is the following :



<u>Note :</u> the system being supposed linear, the USB signal (Cos((w0+w1).t-φ) can be considered independently of the LSB signal (Cos((w0-w2).t).

- The HF transmission is heterodyned through a double signal in quadrature (Cos(w0.t) / Sin(wo.t))
- In i0, it is found for the signal Cos((w0+w1).t-φ) :
  Cos((w0+w1).t- φ) x Cos(w0.t) = (Cos((2xw0+w1).t-φ) + Cos(w1.t-φ)) / 2
  In i0, it is found for the signal Cos((w0-w2).t) :
  Cos((w0-w2).t) x Cos(w0.t) = (Cos((2xw0-w2).t) + Cos(-w2.t)) / 2
- In q0, it is found for the signal Cos((w0+w1).t-φ) :
  Cos((w0+w1).t- φ) x Sin(w0.t) = (Sin((2xw0+w1).t-φ) - Sin(w1.t-φ)) / 2
  In q0, it is found for the signal Cos((w0-w2).t) :
  Cos((w0-w2).t) x Sin(w0.t) = (Sin((2xw0-w2).t) - Sin(-w2.t)) / 2

- The low-pass filters delete the HF components (in 2xw0).

- In i1, it is found for the signal Cos((w0+w1).t-φ) : Cos(w1.t-φ) / 2
  In i1, it is found for the signal Cos((w0-w2).t) : Cos(-w2.t) / 2= Cos(w2.t) / 2
- In q1, it is found for the signal Cos((w0+w1).t-φ) : -Sin(w1.t-φ) / 2
  In q1, it is found for the signal Cos((w0-w2).t) : -Sin(-w2.t) / 2=Sin(w2.t) / 2

- « A/D » means « Analogical/ Digital ». The conversion is done by the sound card.
- In i2 and q2, the signals are the same than in i1 and q1 but in digital.

- The « Delay » is done with an « all-pass filter ». It simply compensates the delay (φ) introduced by the « -π/2 » phase shifter. This one introduces a necessary delay, because the goal is not to shift a pure carrier but to shift an AF band. Consequently, it is necessary to use a phase shifter filter, introducing some inevitable delay.
  <u>Note :</u> the phase shift is equal to « -π/2 » for the positive frequencies (exp(i.w.t)) but is equal to « π/2 » for the negative frequencies (exp(-i.w.t)), because the q2 signal is real (see the “Transmission” part).
- In i3, it is found for the signal Cos((w0+w1).t-φ) : Cos(w1.t-φ2) / 2
  In i3, it is found for the signal Cos((w0-w2).t) : Cos(w2.t-φ1) / 2
- In q3, it is found for the signal Cos((w0+w1).t-φ) : Cos(w1.t-φ2) / 2
  In q3, it is found for the signal Cos((w0-w2).t-φ) : -Cos(w2.t-φ1) / 2

- In i4, it is found i4 = i3+q3, i.e. :
  * for the signal Cos((w0+w1).t-φ) : Cos(w1.t-φ2) / 2 + (Cos(w1.t-φ2) / 2)= Cos(w1.t-φ2), <u>which is the initial signal</u> (apart to the φ2 phase, which role is without importance).
  * for the signal Cos((w0-w2).t) : Cos(w2.t-φ1) / 2 + (-Cos(w2.t-φ1) / 2)= 0. <u>The LSB signal has been removed, which was the wished goal.</u>

  It must be noted that:
  - if the operator « - » had been applied (instead of « + »), it would have been obtained the LSB signal and the USB signal would have been removed,
  - if the initial phase shift had been equal to π/2 instead of -π/2, due to the equality Sin(θ)=-Sin(-θ), the necessary operator to receive in USB, would have been a « - ».

Conclusion

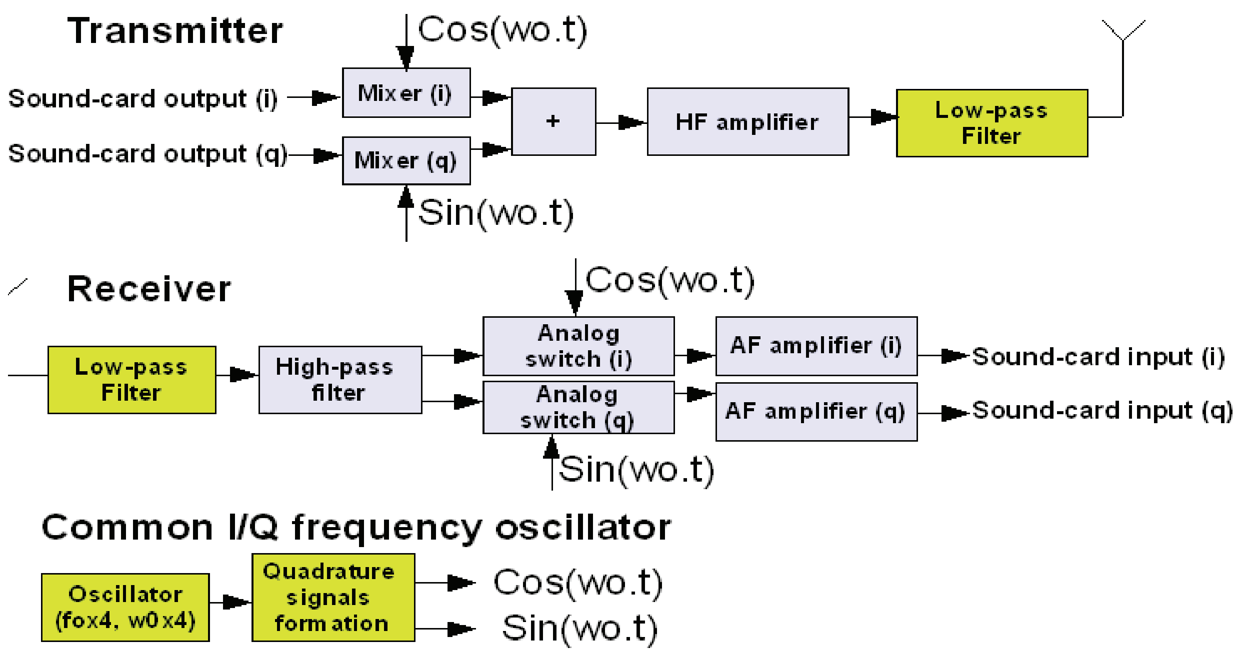
The initial signal has been recovered and the parasitic LSB signal has been removed. Thus, it will be possible to send the AF recovered signal to the speaker and/or to process it with a software (digital decoding, for example).
This system works well in permanent base band (+/-3 KHz) reception, i.e. if the SdR received is driven by a VFO (DDS or other system). So, the sampling frequency can be weak (8000 or 11025 samples/s).
But if the central frequency of reception of the SdR receiver is fixed (i.e. in a common way, driven by a crystal), the reception band will be wide (up to +/-96 KHz) and, consequently, the sampling frequency must be high (192000 samples/s for +/-96 KHz). In that case, the AF signal must be digitally heterodyned in base band by the software and, afterwards, decimated before being processed, but this is another story...

ELECTRONIC DIAGRAMS

It is proposed, below, several principle or simplified electronic diagrams. They can't be used as is. They are just intended to give ideas for a future project. They are widely inspired by the Tasa YU1LM ones (http://yu1lm.qrpradio.com/). It is supposed that the project is a 3.5 MHz SdR transceiver, having a central frequency of: 14.31818 / 4 = 3.579545 MHz. It would permit a reception/transmission from 3555 KHz to 3603 KHz , i.e. part of the CW band and digimode bands.

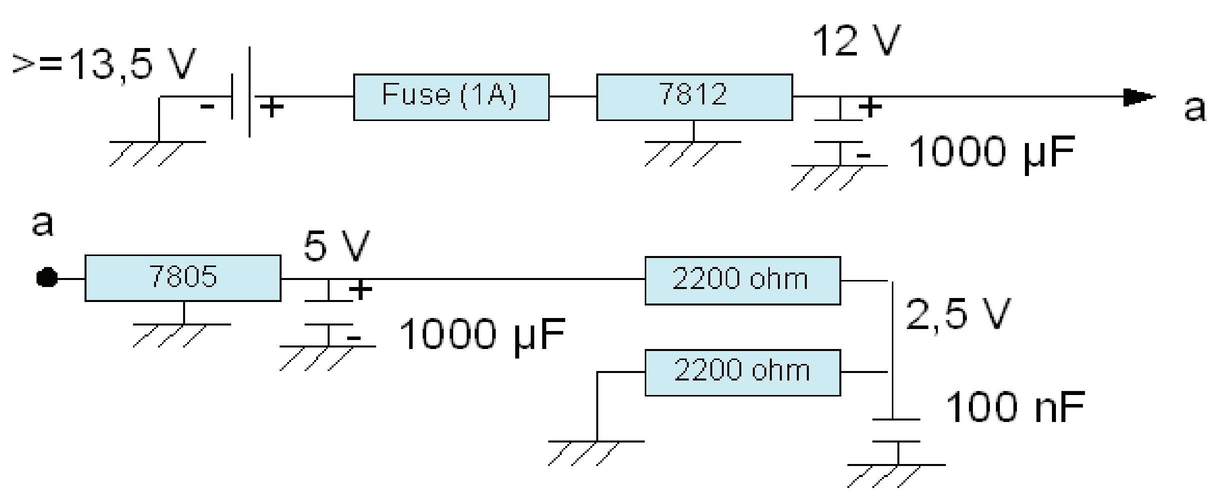
SYNOPTIC

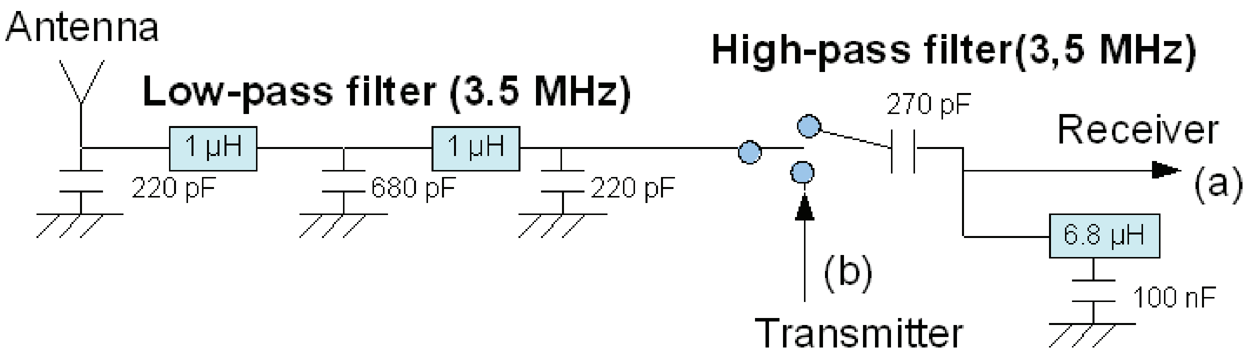
Notes: the yellow modules are common to the receiver and the transmitter. Analog switches are used for mixing.



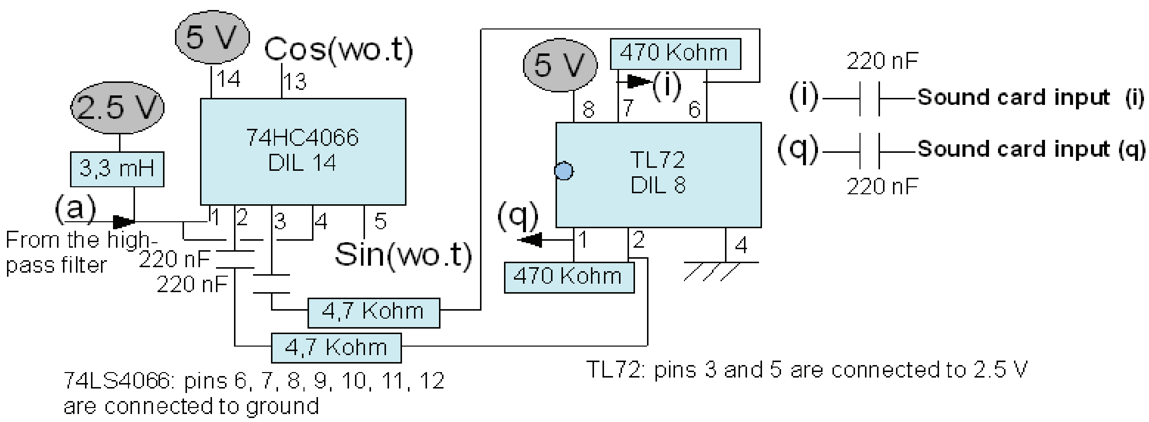
FILTERS

Supply Diagram



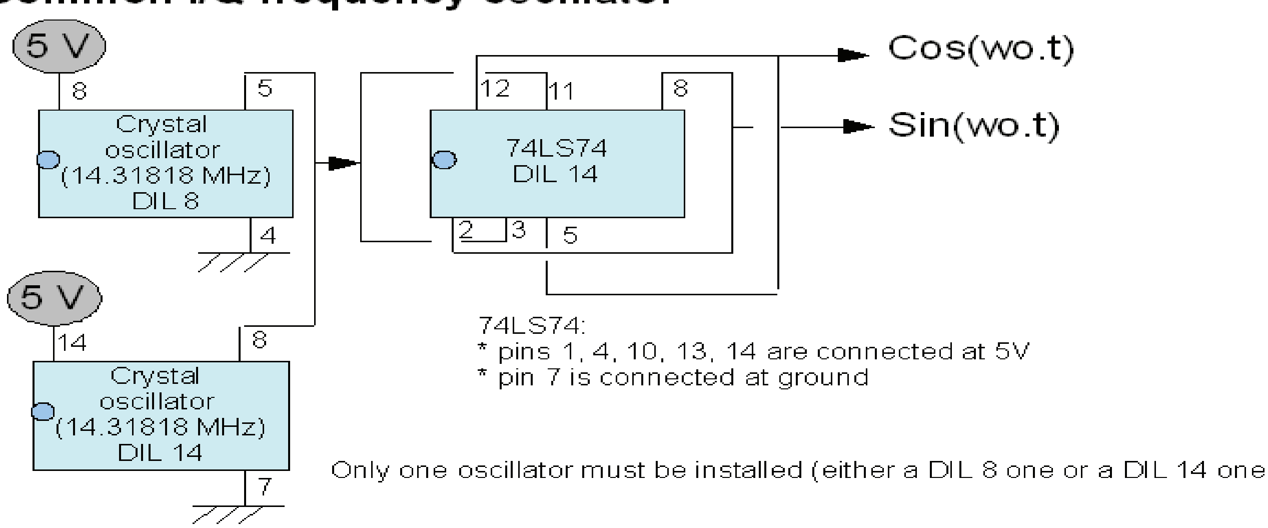


Analog switches and AF amplifiers

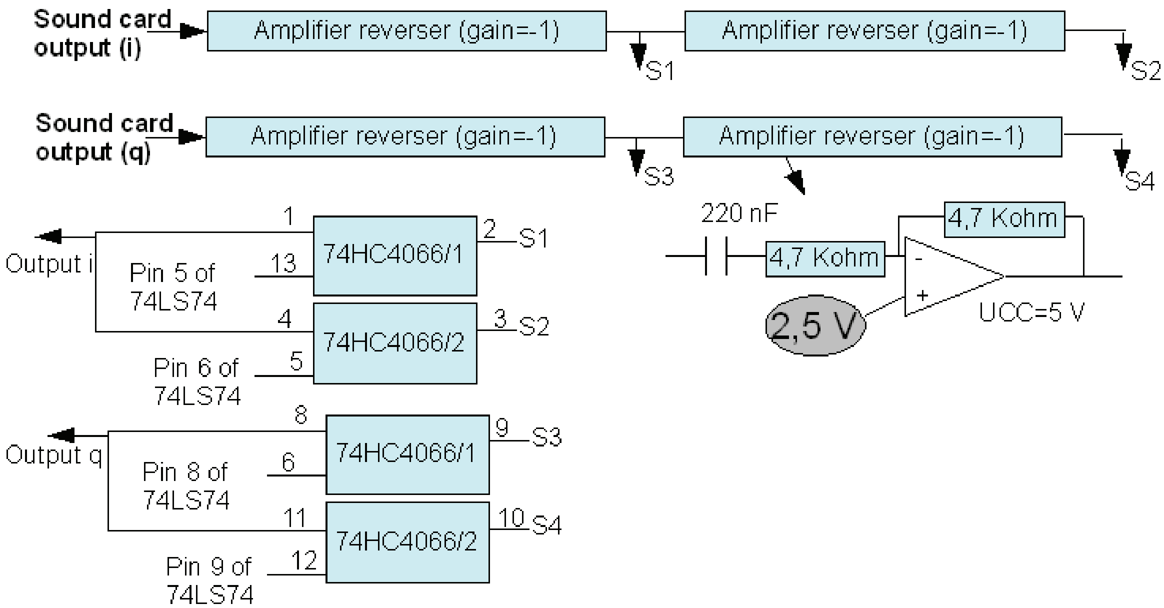


The system below supplies two signal in quadrature at a frequency f0 equal to the quarter of the frequency of the crystal oscillator (i.e. here : 14.31818/4=3.579545 MHz). These signal are not, strictly, sinusoidal (Cos(wot)/Sin(wot)) but rather TTL signals, so with harmonics which are not annoying because removed downstream (either by the TL72 amplifier and the anti-folding filter of the sound card in RX or by the low-pass filter in TX).

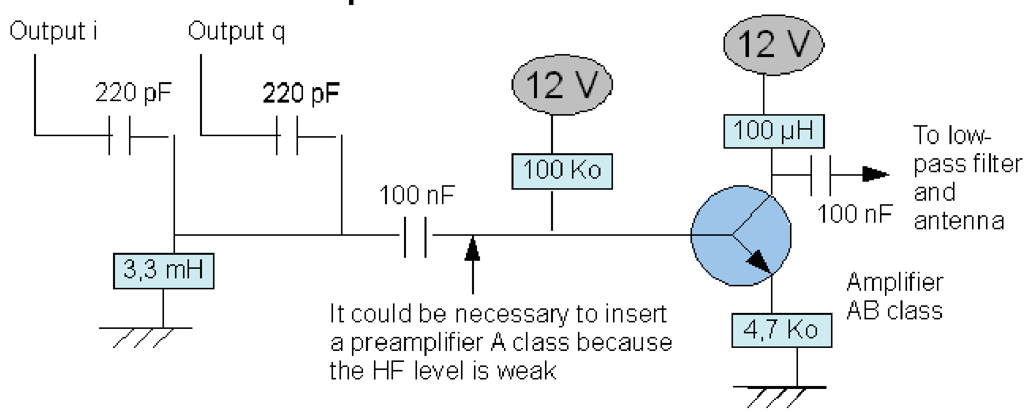
Common I/Q frequency oscillator



Mixers



Mixture and HF amplifier



The transistor used could be a 2N2219

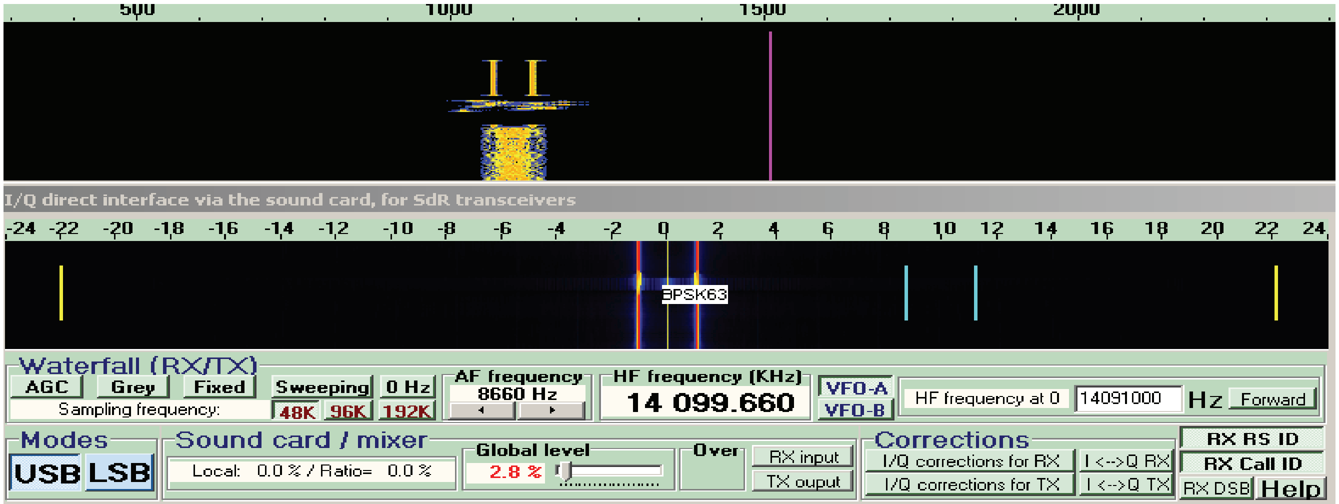
Below are given two snapshots (Multipsk 4.14) which show the automatic detection on all the SdR band:

- of a RS ID identifier, which permits to automatically determine the mode name and the frequency, of a call done in an given digital mode (exotic in general, i.e. neither PSK31 nor RTTY),
- of a Call ID (and precisely a Prop ID), to signal oneself on the band.

These 2 applications permit, thanks to a SdR receiver, to handle not the traditional 3 KHz bandwidth but 44 KHz.

For a quick guide to use RS ID, Call ID or Prop ID, download:
http://f6cte.free.fr/The_RS_ID_easy_with_Multipsk.doc
http://f6cte.free.fr/The_Call_ID_and_Prop_ID_easy_with_Multipsk.doc

RS ID



Call ID (Prop ID)