Master Thesis

Linear Zero-IF Direct Conversion Receiver

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December 2006

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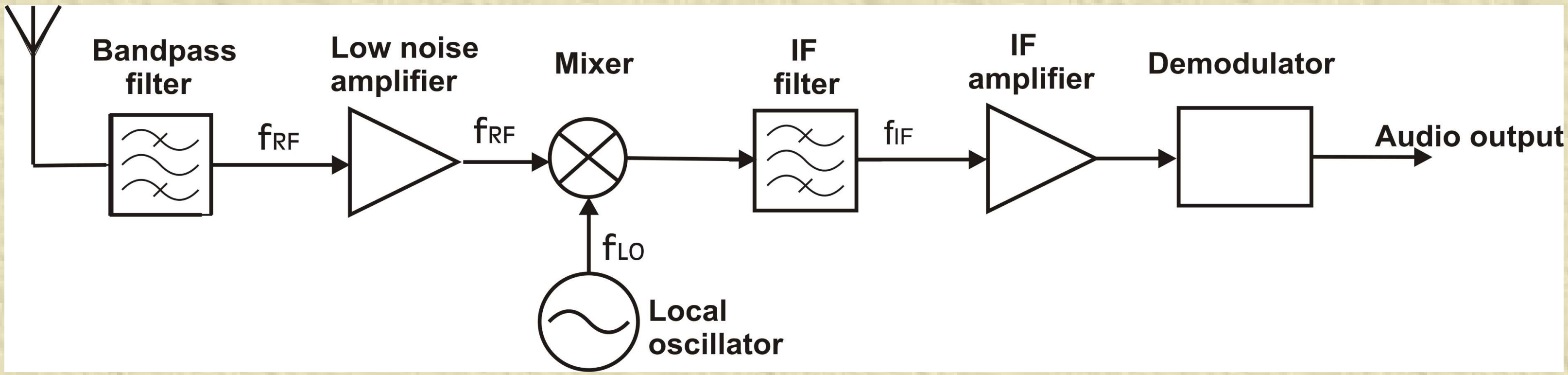
Contents

Introduction

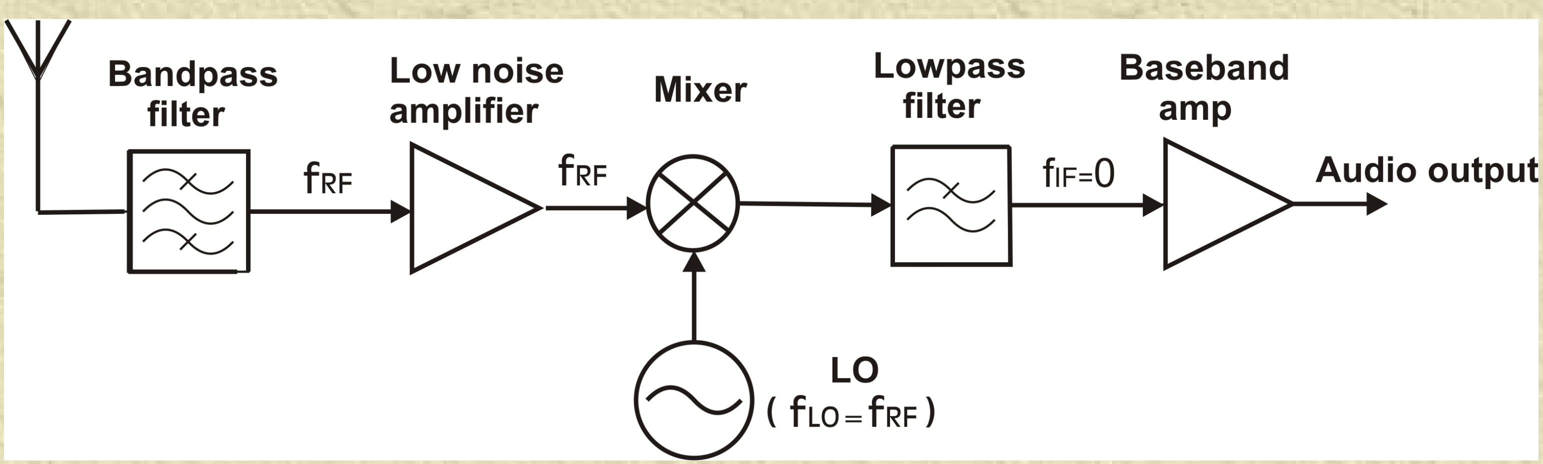
- Superheterodyne Receiver(Conventional Receiver):



- By far the most popular type of receiver used today, which was invented by E.H Armstrong 1918.
- Converting all signal frequencies to a constant lower frequency(IF) before detection.
- Tuning is conveniently accomplished by varying the frequency of the LO so that the IF remains constant.

Introduction

- Direct Conversion Receiver:

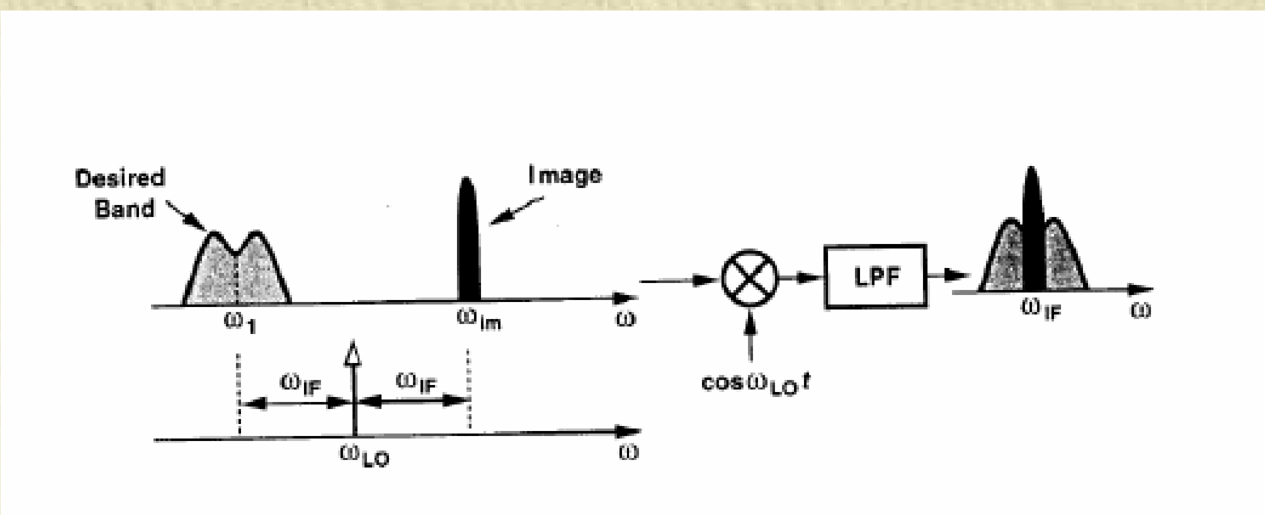


- Direct Conversion is not a new concept, which was first described by F.M colebrook in 1924.
- The intermediate frequency(IF)is eliminated by converting the input signal directly to the baseband(Zero IF receiver).
- The LO is set to the same frequency as the desired RF signal.

Introduction

- The mixer output contains the baseband signal and the signal at twice the carrier frequency which is then be removed with the low-pass filter.
- The advantages of direct conversion receiver :

1. Lower complexity and power consumption(no IF amplifier, no IF bandpass filter, or no IF local oscillator), which has the potential to reach the 'one chip goal'.



2. No image frequency

Introduction

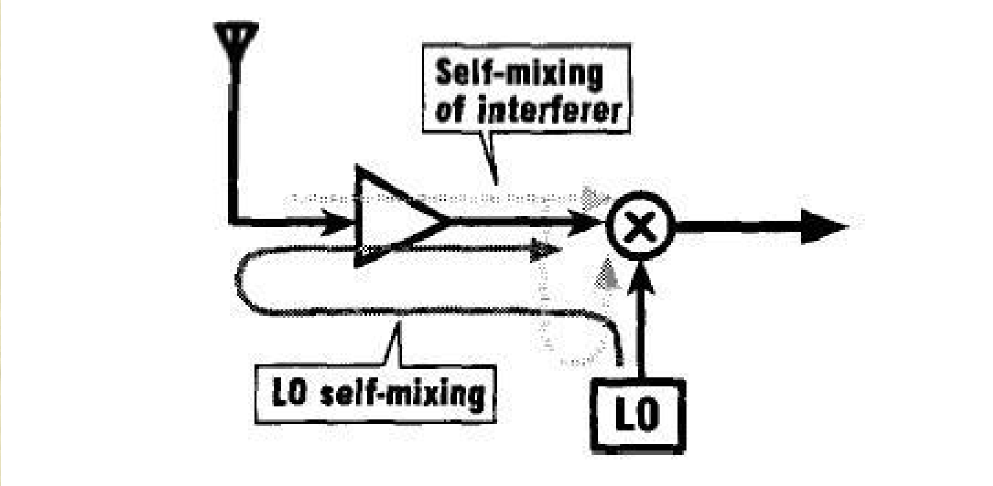
- The challenges of direct conversion receiver, which cause that the superheterodyne receivers are still the most widely used communication receiver in the second half of last century.

1. Frequency drift: theoretically the frequency of LO should be exactly the same as the input signal. Slight drift can cause the direct conversion receiver become unstable.

2. Flicker noise: occurs in almost all electronic devices at low frequencies with a 1/f character. More effecting on SNR than thermal noise (CMOS > bipolar).

Introduction

3.DC offset: the LO signal leaking to the antenna because of poor reverse isolation through the mixer and RF amplifier; a large near-channel interferer leaking into the LO port of the mixer.



4. Even order distortion:
Contrary to the superheterodyne receiver which nonlinearity is characterized by the third-order intermodulation distortion (IP3), the nonlinearity of DC receiver is characterized by the second-order intermodulation distortion(IP2):

Introduction

- For a two-tone input signals:

$$f_{IF1} = f_1 - f_{LO} \qquad f_{IF2} = f_2 - f_{LO}$$

The second-order product

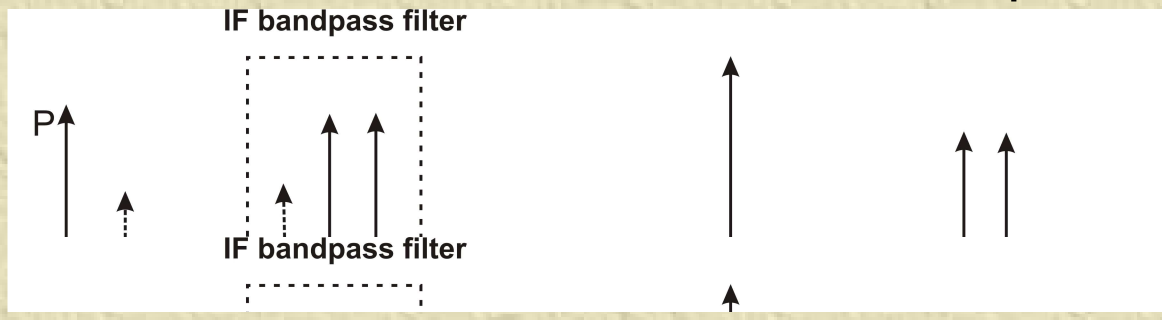
$$\begin{aligned} f_{IM2} &= f_{IF2} - f_{IF1} \\ &= (f_2 - f_{LO}) - (f_1 - f_{LO}) \\ &= f_2 - f_1 = f_d \end{aligned}$$

The third-order product

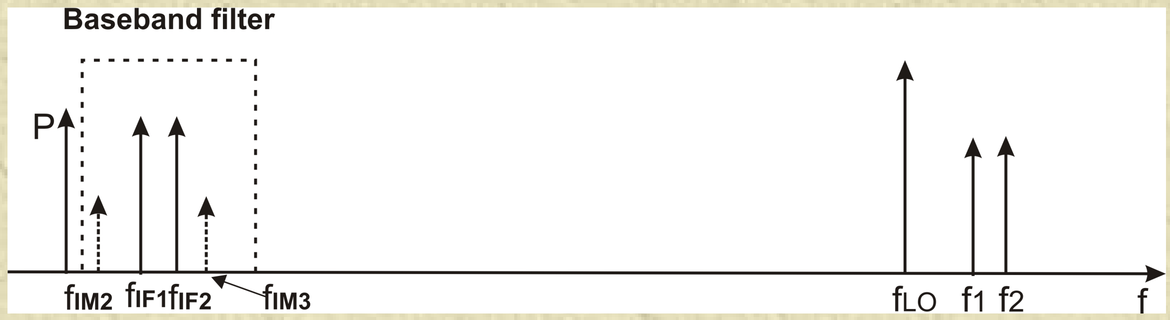
$$\begin{aligned} f_{IM3} &= 2 f_{IF1} - f_{IF2} \\ &= 2(f_1 - f_{LO}) - (f_2 - f_{LO}) \\ &= (f_1 - f_{LO}) + (f_1 - f_2) \\ &= f_{IF1} - f_d \end{aligned}$$

Introduction

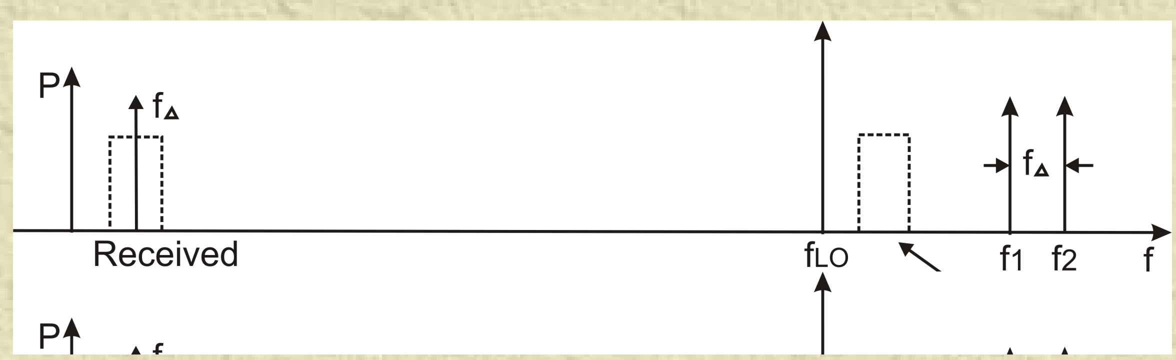
- The intermodulation distortion in the superheterodyne receiver



- The intermodulation distortion in the direct conversion receiver



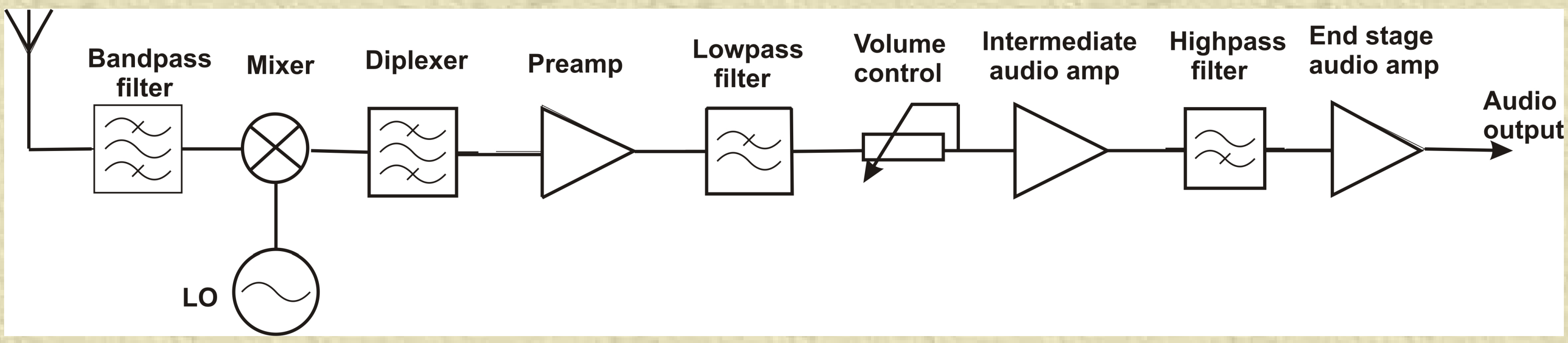
in-channel signals



neighbor channel signals

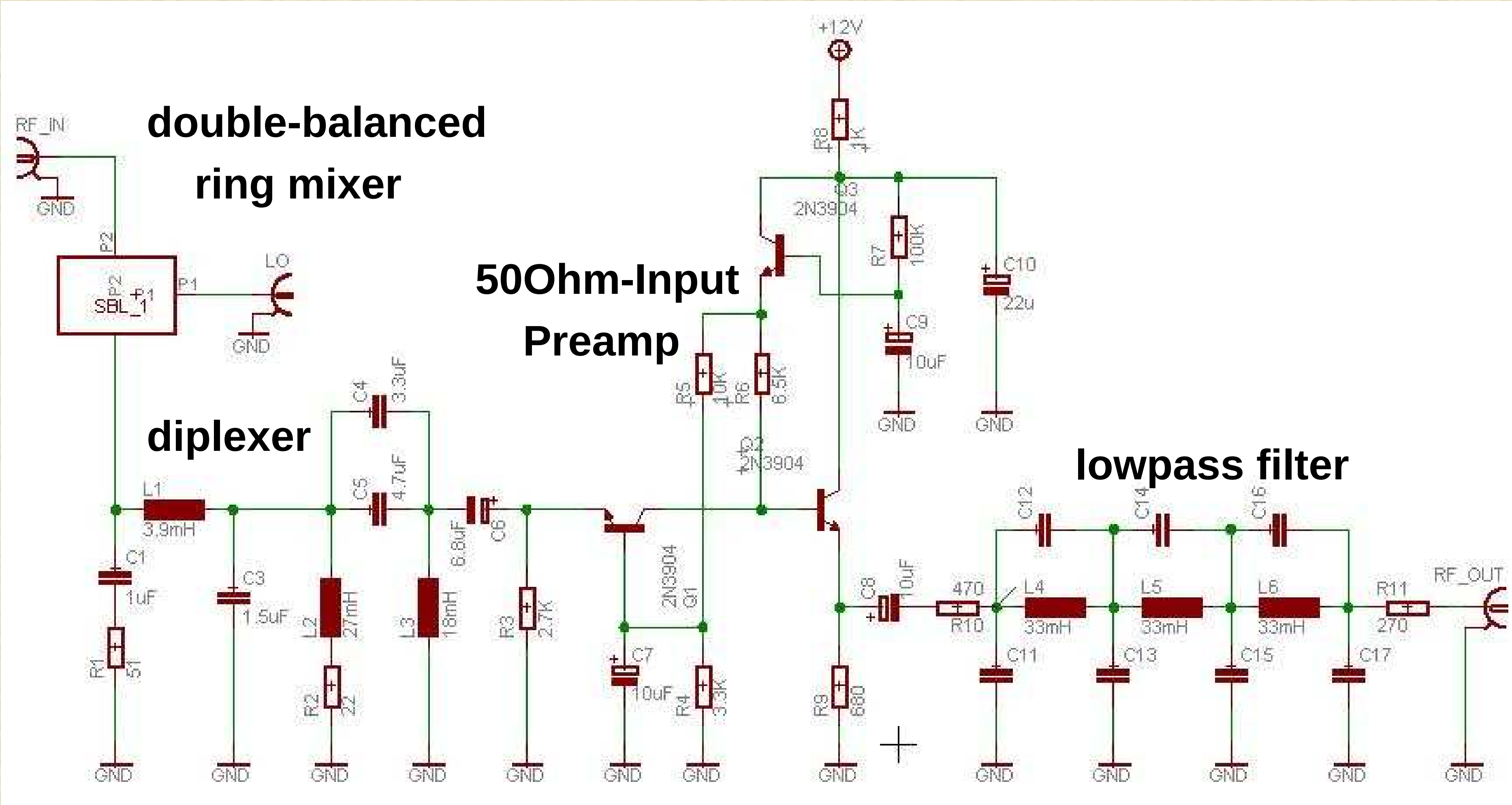
Circuit Design and Realization

- The circuit design in this thesis is based on a circuit description of a Zero-IF receiver for short wave application, designed by Rick Campbell, KK7B, which was further discussed and represented by Heinz Sarrasch, DJ7RC.



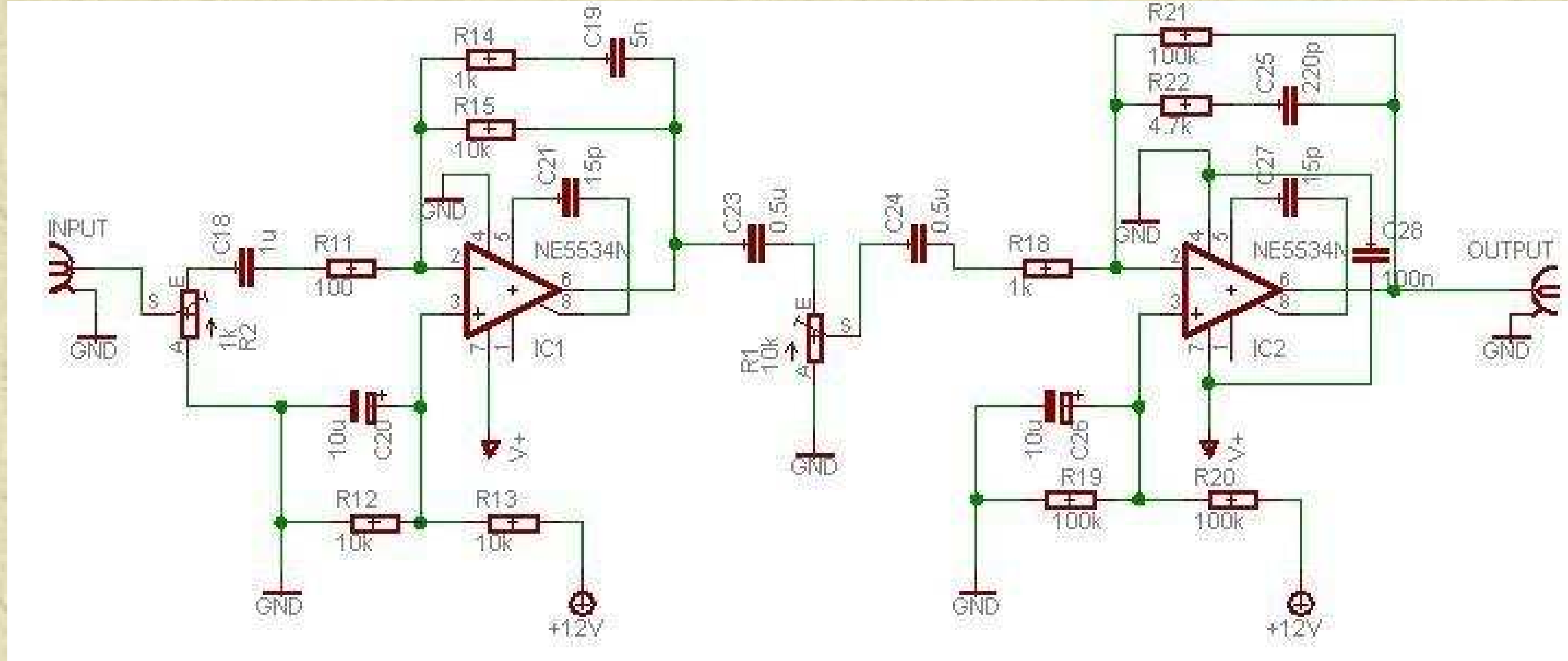
Block diagram of DC Receiver

Circuit Design and Realization



The schematic of the direct conversion receiver front-end

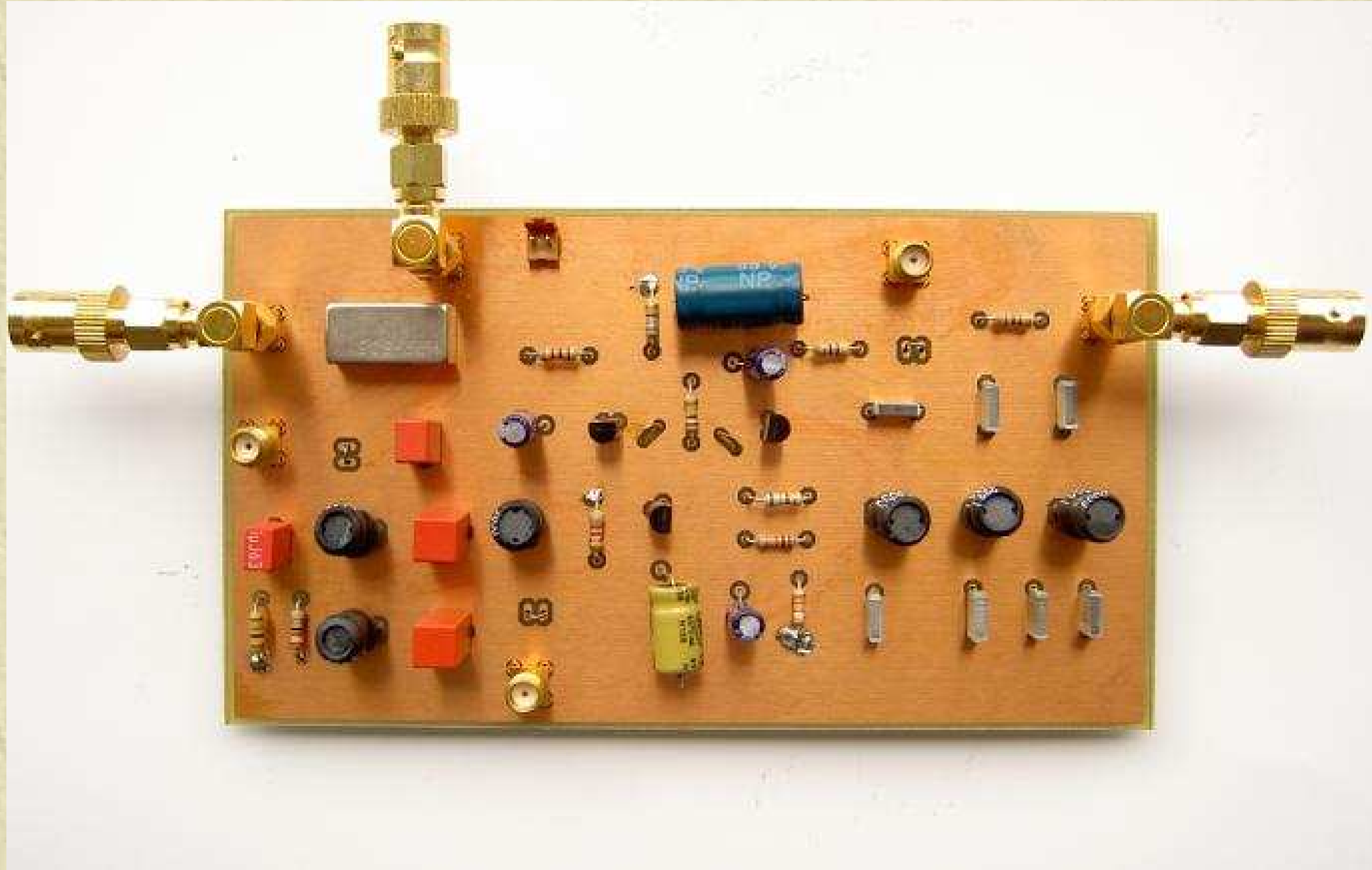
Circuit Design and Realization



The schematic of audio amplifier

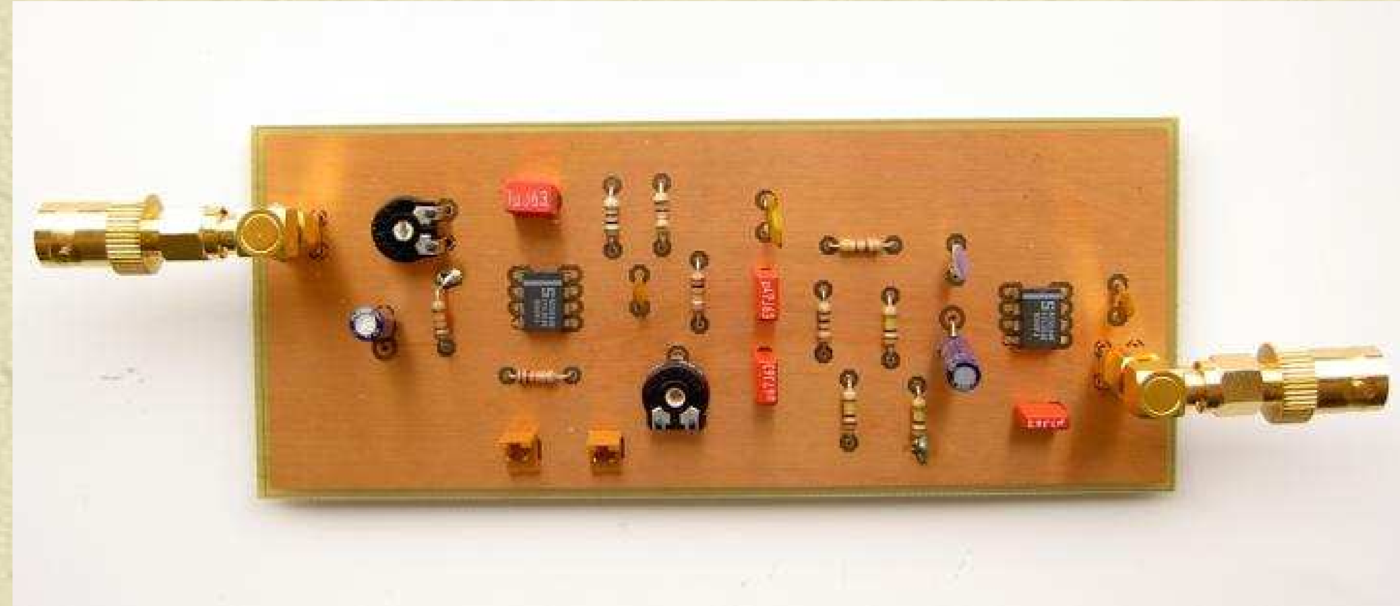
- NE5534: low-noise audio operational amplifer
- 300-Hz highpass filter: reducing the interference of flicker noise
- No Automatic Gain Control(AGC): to analyse the fundamental linear properties of receiver.

Circuit Design and Realization



The PCB for the direct conversion receiver front-end

Circuit Design and Realization

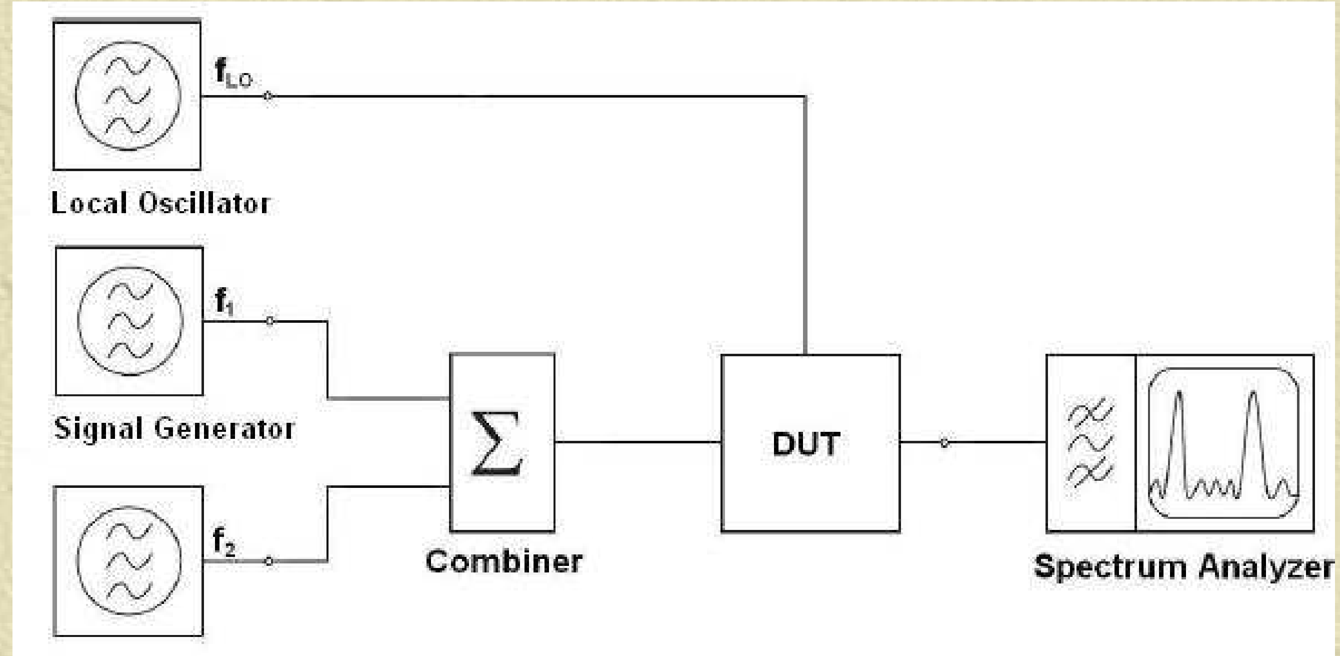


The PCB for the audio amplifier

- To design the PCB by using Easily Graphical Layout Editor(Eagle)
- Conventional Through the Hole Technology
- Only top and bottom layers
- all the areas except for signal tracks are connected to ground layer- **Metallization.**

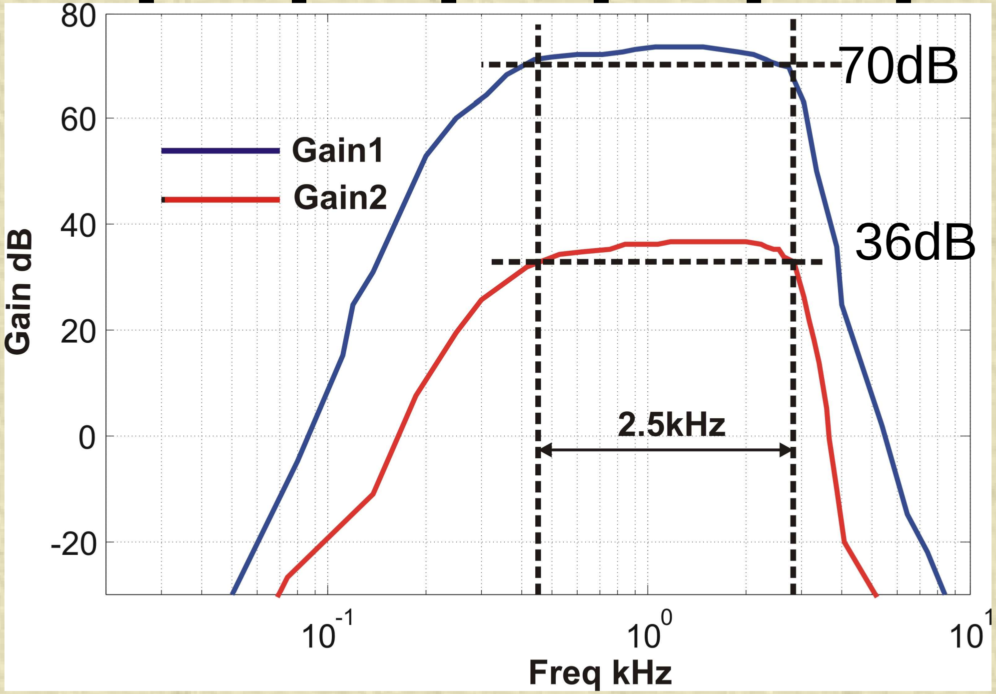
Measurement

1.The block diagram of measurement



- Two signal generators are used as two-tone signals(14MHz)
- Another signal generator as LO(difference frequency 0.3-3kHz)
- R9211B/C FFT SERVO Analyzer is used, which has the capability to dispaly the spectrum in the low frequency range clearly.

Measurement

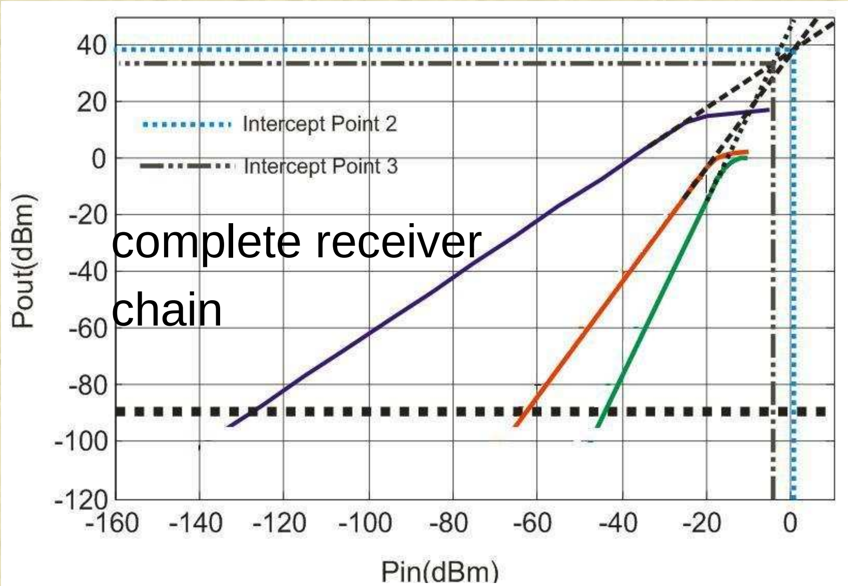


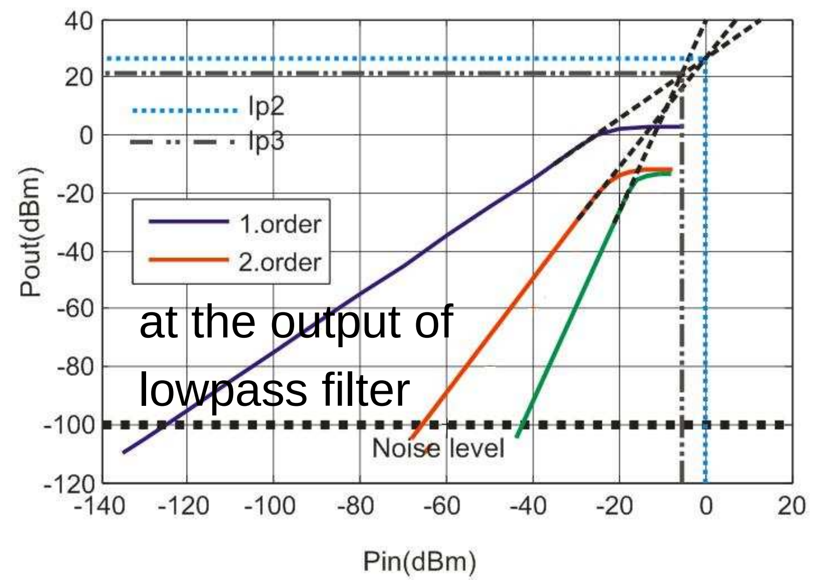
1. Gain

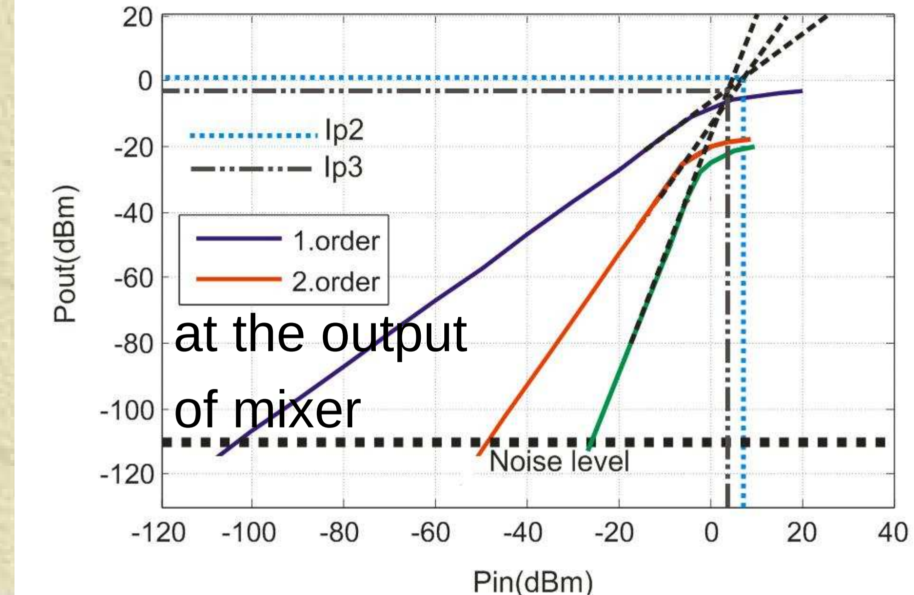
- The maximum available gain (Gain1)
- The maximum gain without clipping distortion at input level –10dBm(Gain2)

- A large Gain leads to clipping distortion
- Without AGC it is hard to obtain a good linearity
- Good Selectivity: A obvious bandpass effect corresponding to the bandwidth of diplexer from 300Hz to 3000Hz .

Measurement







2. IP2 and IP3

- second-and third-order product is created by the mixer, which are not changed except for the power level.
- second-order product is the most primary intermodulation product.

Taylor Series Modelling

- In order to model the nonlinearity of a complete receiving system, the coefficients of a Taylor Series can be derived from the measured intermodulation products.
- We assume that the diodes in the mixer are perfertly matched and transformers are ideal, the output of the double-ring mixer is:

$$V_o = \frac{R_L}{R_L + r_d} \frac{2A}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^n + 1}{n} \sin\frac{n\pi}{2} \{[\cos(n\omega_{LO} + \omega_{RF})t] + [\cos(n\omega_{LO} - \omega_{RF})t]\}$$

- If we let n=1 and assume R**L>>rd,** we get the ideal output signal:

$$V_o = \frac{2A}{\pi} \{[\cos(\omega_{LO} + \omega_{RF})t] + [\cos(\omega_{LO} - \omega_{RF})t]\}$$

Taylor Series Modelling

- Now we transfer the taylor series model to the baseband signals and consider the signals at the difference-frequencies of the two tones with the LO-signal as our ideal input signals.

$$
\begin{aligned}
y(t) &= y_0 + a_1 x(t) + a_2 x(t)^2 + a_3 x(t)^3 \\
&= y_0 + a_1(A_1\cos(\omega_1 t) + A_2\cos(\omega_2 t)) + a_2(A_1\cos(\omega_1 t) + A_2\cos(\omega_2 t))^2 \\
&\quad + a_3(A_1\cos(\omega_1 t) + A_2\cos(\omega_2 t))^3
\end{aligned}
$$

Where $A_1 = A_2 = \frac{2}{\pi}A$, and $\begin{aligned}\omega_1 &= \omega_{RF1} - \omega_{LO} \\ \omega_2 &= \omega_{RF2} - \omega_{LO}\end{aligned}$.

Taylor Series Modelling

- Taylor series 1: complete receiver chain

  we assume that our measurement is done at a standard impedence of 50 Ω

a) We know the ratio of output and input power as the linear gain

$$G[dB] = 20\log\frac{U_{out}}{U_{in}} \longrightarrow a_1 = \frac{U_{out}}{U_{in}} = 10^{G[dB]/20dB}$$

where the input power is the RF power and G is approximately 36dB at 1.6kHz, thus there is a ideal insertion loss:

$$G_{IL} = 20\log\left(\frac{A}{A1}\right) = 20\log\left(\frac{\pi}{2}\right) = 4dB$$

So the final $G' = G + 4dB = 40dB$, so we get

$$a_1 = 10^{40dB/20dB} = 100$$

Taylor Series Modelling

b) We can determine a_2 with $a_2 A_1 A_2 = A_1 a_1$ at intercept point(IP2).

The input IP2 can be read from the measurement, P**in** = 2 dBm.

For a standard system impedance of R= 50 Ω , we have

$$P[dBm] = 10\log\left(\frac{P}{1mW}\right) = 20\log\left(\frac{U}{0.316V}\right)$$

So $U = A \approx 0.4$, with $A_1 = A_2 = \frac{2}{\pi} A$ and $a_2 A_1 A_2 = A_1 a_1$

We can determine $a_2 = 400$.

c) We can solve a_3 at IP3 by .

$$a_1 A_1 = -\frac{3}{4} a_3 A_1^3$$

The input level with IP3, P**in** = -4 dBm, so we can solve $A \approx 0.2$

with $A_1 = A_2 = \frac{2}{\pi} A$ and $a_1 = 100$, we finally get $a_3 \approx -7.89 \times 10^3$

Taylor Series Modelling

The taylor series which can model the nonlinearity of our receiving system can be expressed by

$$y(t) = y_0 + a_1 x(t) + a_2 x(t)^2 + a_3 x(t)^3$$

$$= 100x(t) + 400x(t)^2 - 7.89\times 10^3 x(t)^3$$

where $y_0 = 0$ for the direct component has been eliminated by the DC blocking capacitor.

- The taylor series which can model the nonlinearity of the incomplete receiving system(referred to the lowpass filter output) can be derived as the same way and expressed by

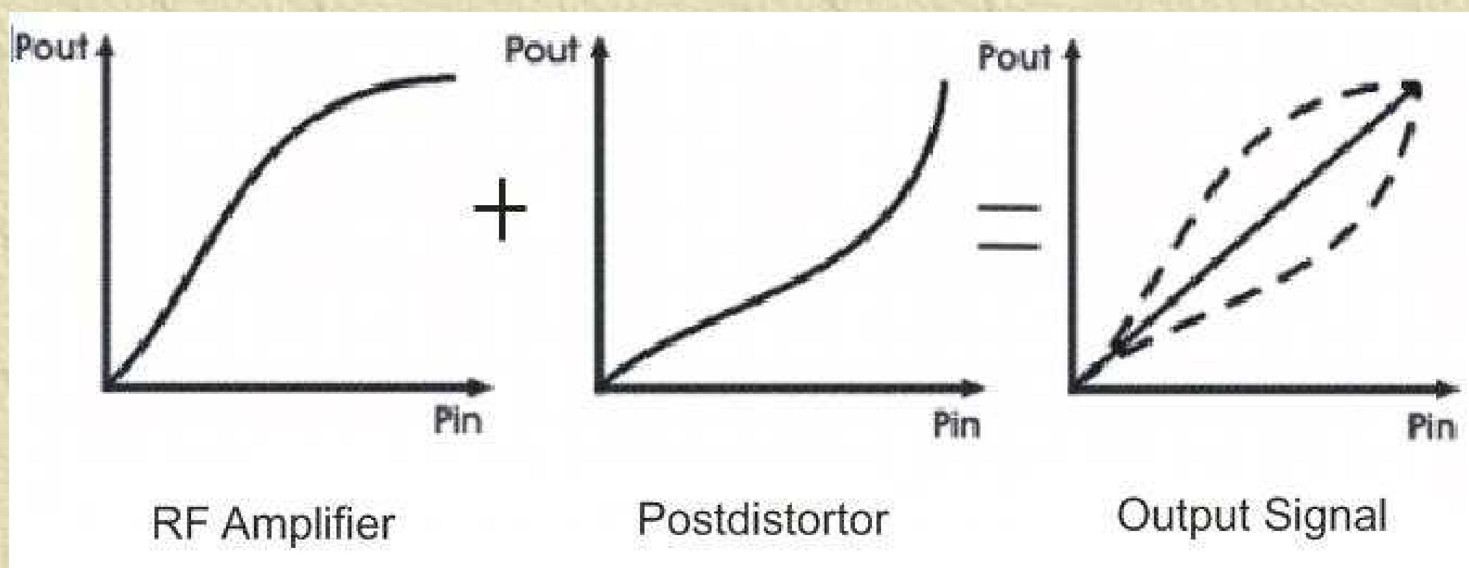
$$y(t) = y_0 + a_1 x(t) + a_2 x(t)^2 + a_3 x(t)^3$$

$$= 28.2x(t) + 1.38\times 10^2 x(t)^2 - 2.49\times 10^3 x(t)^3$$

which is to be applied to build the Postdistortion Circuit.

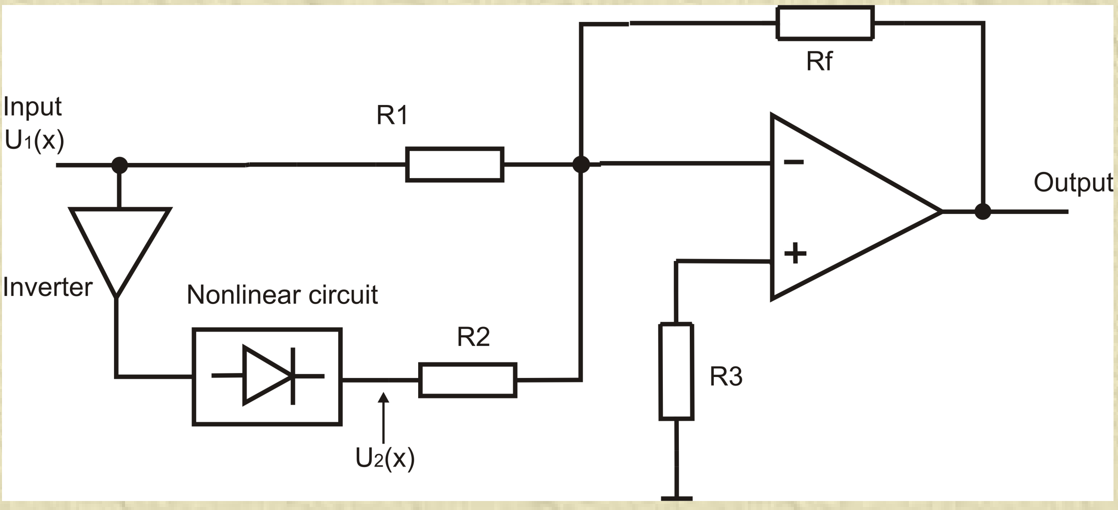
Postdistortion Technique

- Postdistortion circuit involves the creation of a distortion characteristic which is precisely complementary to the distortion characteristic of the RF component and cascading the two in order to ensure that the resulting system has little or no ínput-output distortion.
- If we cascade the complementary distortion element after RF component, it is referred to as postdistortion.



Postdistortion Technique

- Postdistortion Circuit Design:



- An operational amplifier is used as the linear component to provide us the linear subtraction of the two input signals(U**1**(x) and U**2**(x)).

- with $U_1(x) = c_1 x + c_2 x^2$ and $U_2(x) = a_1 U_1(x) + a_2 (U_2(x))^2$, we have

$$U_o(x) = b_1 U_1(x) - b_2 U_2(x)$$

$$= b_1 U_1(x) - b_2 [a_1 (c_1 x + c_2 x^2) + a_2 (c_1 x + c_2 x^2)^2]$$

$$= (b_1 a_1 - b_2 a_1 c_1) x + (b_1 a_2 - b_2 a_1 c_2 - b_2 a_2 c_1) x^2 + \ldots$$

Postdistortion Technique

- We let $b_1 a_1 - b_2 a_1 c_1 = c_1$ to keep the power of the

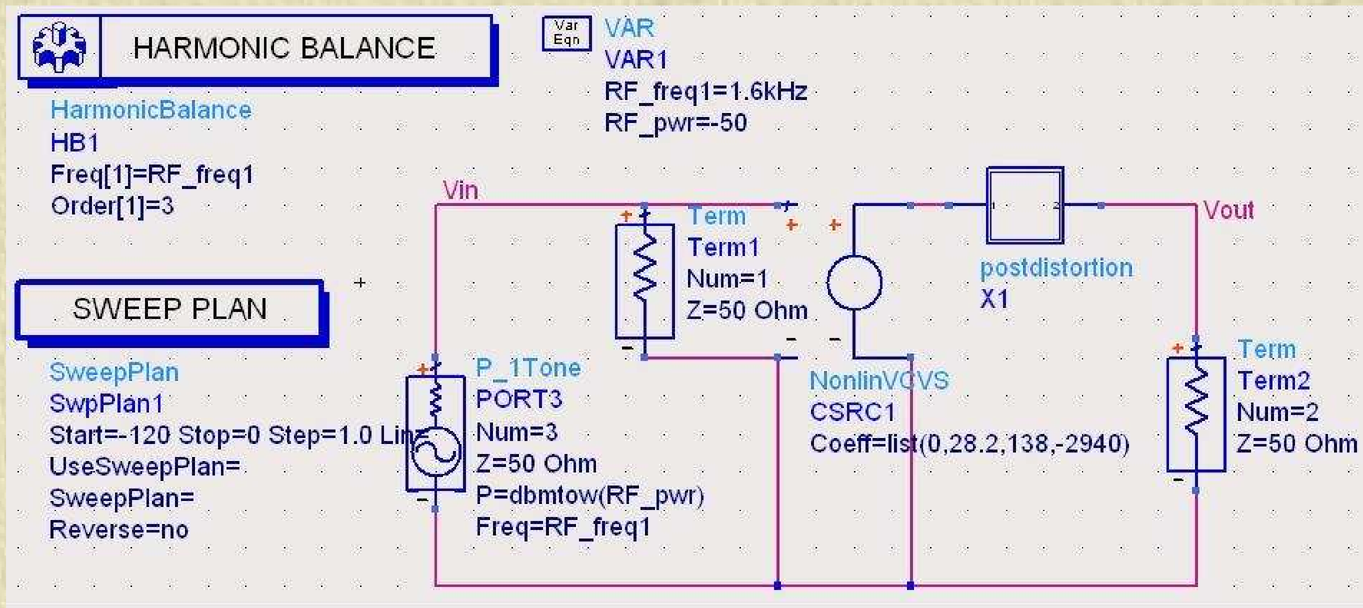
  $$b_1 a_2 - b_2 a_1 c_2 - b_2 a_2 c_1 = 0$$

  fundamental product of the output signal unchanged and cancel the second-order intermodulation product.
- This circuit provides us much flexibility since it releases the critical requirement on a_1 and a_2 through the linear control of b_1 and b_2.

- Simulation in ADS(Advanced Design System)
  - In ADS a model use to describe the nonlinearity of a system is the nonlinear Voltage-Control-Voltage-Source(VCVS) which nonlinearity is determined by the coefficients of taylor series.
  - The taylor series describing the receiving system before the audio amplifier is $y(t) = a_1 x(t) + a_2 x(t)^2 + a_3 x(t)^3$

    $$= 28.2x(t) + 1.38 \times 10^2 x(t)^2 - 2.49 \times 10^3 x(t)^3$$

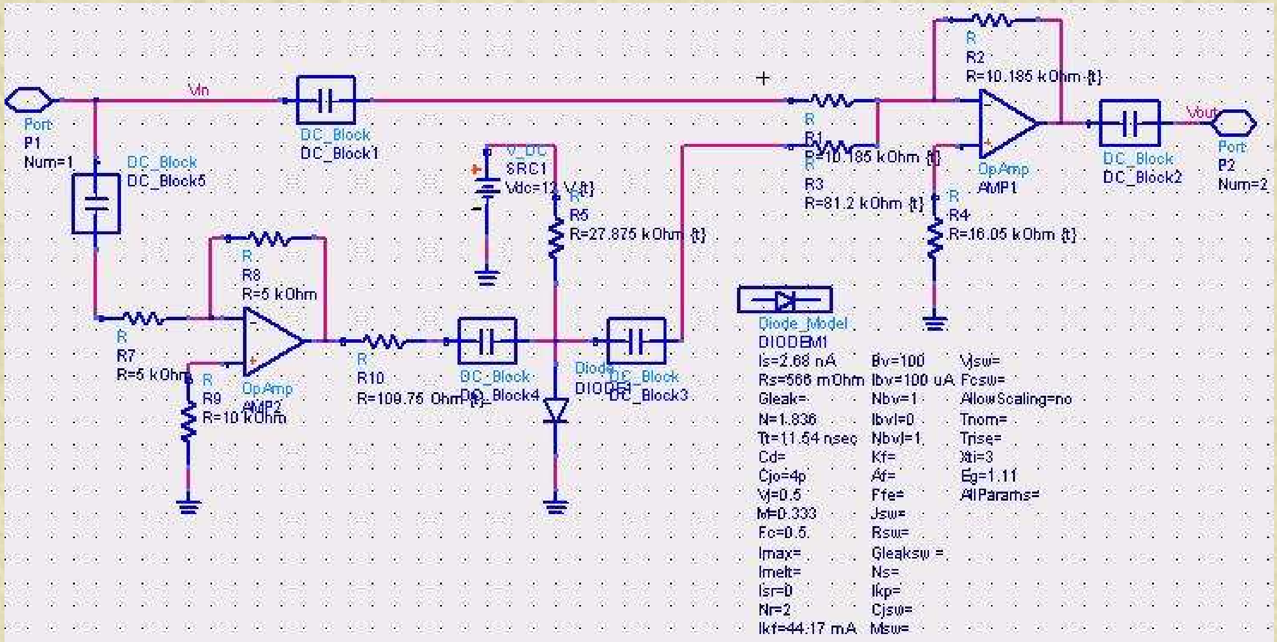
Postdistortion Technique



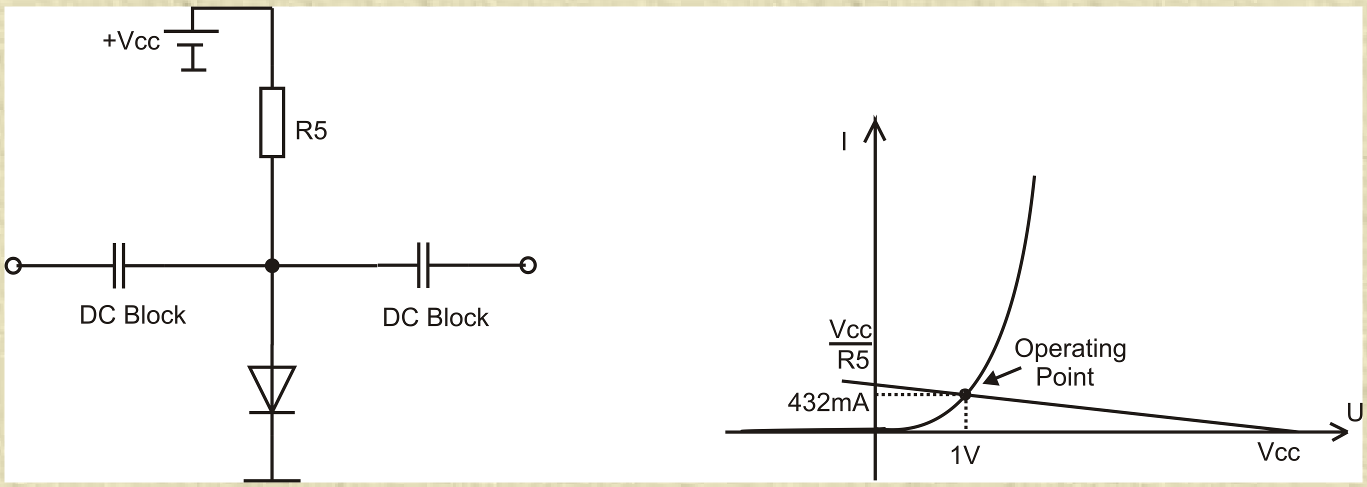
Simulation In ADS

- We put the postdistortion circuit behind the lowpass filter instead of directly after the mixer in order to obtain a better NF.

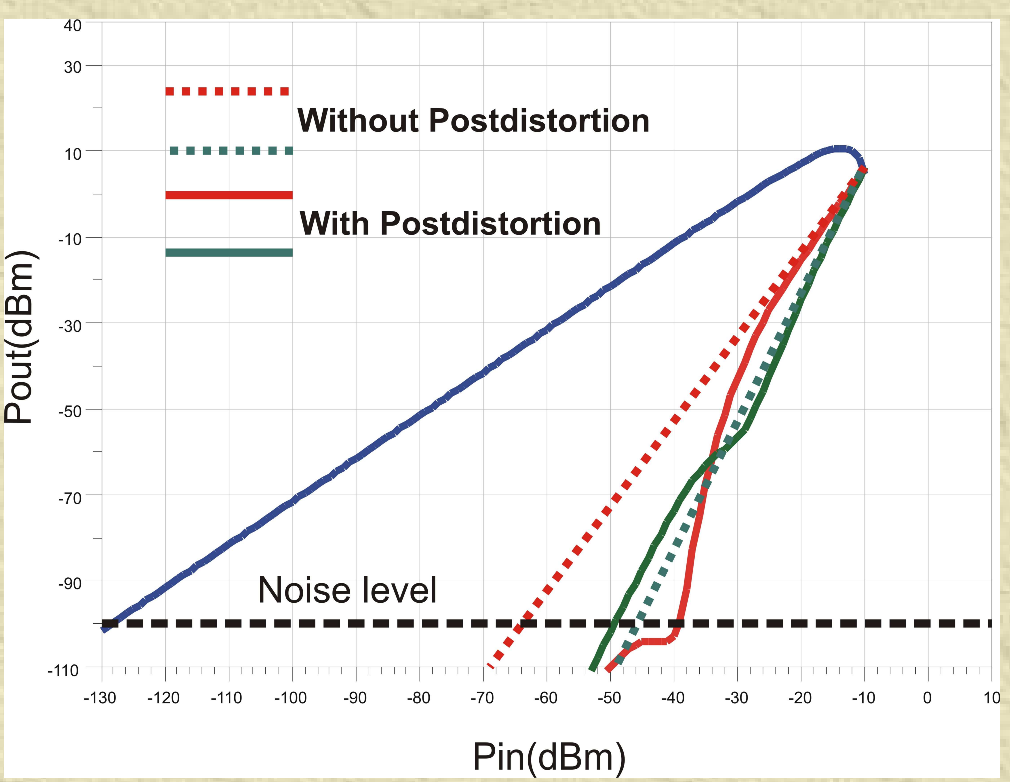
Postdistortion Technique



- A small signal diode is used, which is biased through R_5 and V_{cc}.
- let $R_1=R_2$, $R_3<<R_1$ to keep the fundamental product unchanged.
  - We can adjust R3, R5, and R10 to obtain the best linearity.



Postdistortion Technique

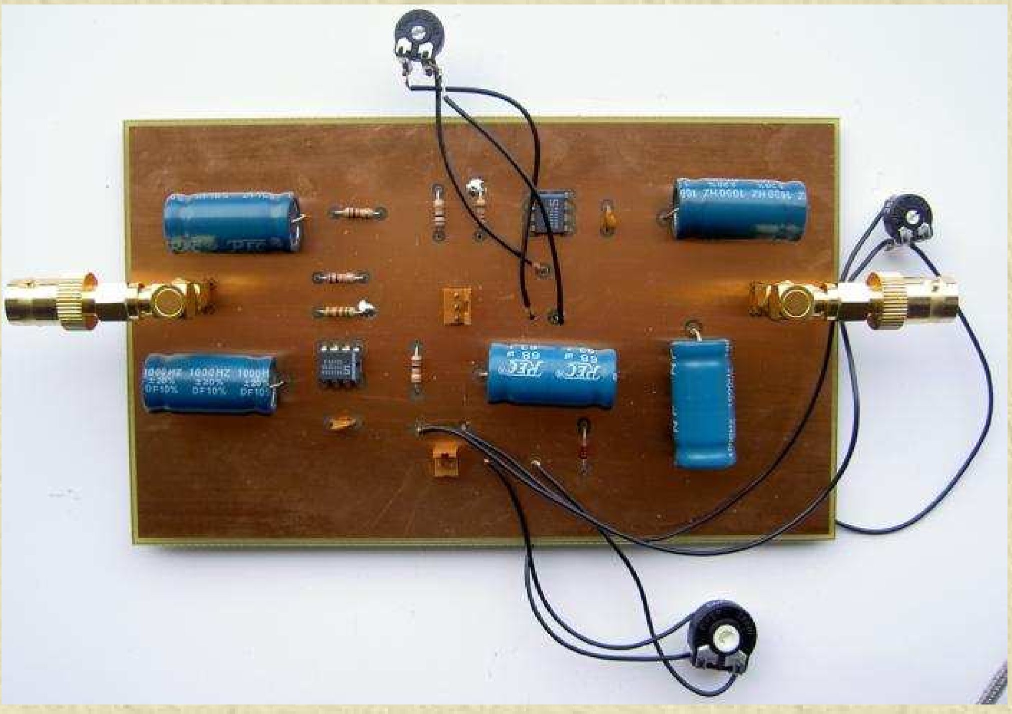


Simulation result

- The second-order product is reduced by about 25dB, on the meantime the third-order product is increased about 3dB.
- Finally the dynamic range is increased about 17dB.
- The circuit cancels the second-order products under the condition that the input level is small.

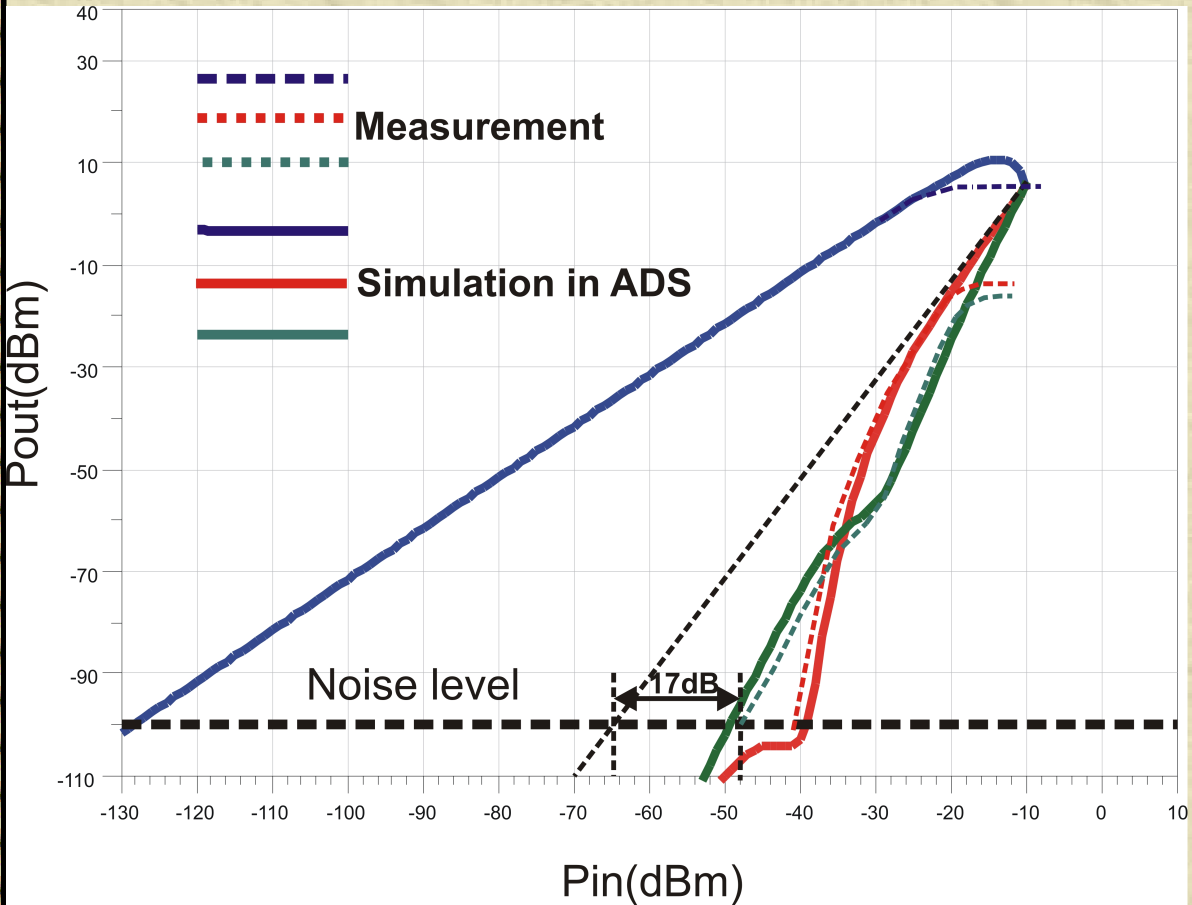
Postdistortion Technique

- Circuit Realization
  - The potentiometers are used to replace the corresponding resistors (R_3 , R_5, R_{10}).
  - Diode is the small signal type 1N4148 which is biased through R_5 and the bias-voltage of +12V.



Postdistortion Technique

- Measurement



The measurement results

- The measurement results are very similar to the simulated ones by setting the values of the potentiometers properly.
- Other small signal diode types beside 1N4148 are used in the measurement (small signal schottky diode), where the measurement results are almost the same.

Conclusion

- Contrary to the superheterodyne receiver, the nonlinearity of direct conversion reciever is characterized by the second-order intermodulation distortion.
- The mixer in direct conversion receiver is the most critical component since many challenges(second-order distortion, frequency drift, flicker noise, DC offset) are closely related with the mixer.
- The desgined postdistortion circuit, which is composed of both linear components(op amp) and the nonlinear component(diode), provide us much flexibility through the linear control of resistors, and finally it increased the spurious-free dynamic range about 17dB.

The end of this presentation

Thanks for your appreciation